Session 1aID

Interdisciplinary: Plenary Lecture: Studying the Sea With Sound

N. Ross Chapman, Chair
School Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada

Chair's Introduction—7:55

Invited Paper

8:00

1aID1. Studying the sea with sound. Stan E. Dosso and Jan Dettmer (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

Because electromagnetic radiation is strongly attenuated in seawater while sound propagates efficiently to long (even global) ranges, scientists and engineers have devised many ingenious methods to use acoustics in the ocean in place of light, radio, and microwaves. Myriad underwater acoustic applications include remote sensing, remote control, communications, navigation, and source detection/localization. This talk will present a semi-historical overview of the use of sound to study the sea (including the seabed), from philosophical musings of Aristotle, through the Renaissance, two world wars, and into the modern era of advanced measurement technologies and computer analysis. A final emphasis involves on-going research to estimate seabed geophysical properties and quantify their uncertainty and variability using a variety of ocean acoustic measurements and probabilistic inversion theory.

MONDAY MORNING, 3 JUNE 2013 513ABC, 8:55 A.M. TO 12:00 NOON

Session 1aAAa

Architectural Acoustics and Signal Processing in Acoustics: Advanced Analysis of Room Acoustics: Looking Beyond ISO 3382 I

Boaz Rafaely, Cochair
Dept. of Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva 84105, Israel

Samuel Clapp, Cochair
Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180

Michael Vorländer, Cochair
ITA, RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany

Chair's Introduction—8:55

Invited Papers

9:00

1aAAa1. An objective measure for the sensitivity of the room impulse response. Rok Prislan (Faculty of Mathematics and Phys., Univ. of Ljubljana, Jadranska 19, Ljubljana 1000, Slovenia, rok.prislan@gmail.com), Jonas Brunskog, Finn Jacobsen, and Cheol-Ho Jeong (Acoust. Technol., Dept. of Elec. Eng., Tech. Univ. of Denmark, Lyngby, Denmark)

This study is relevant for a number of important acoustic measurements in reverberation rooms such as measurement of sound transmission and measurement of sound power levels of noise sources. From a pair of impulse responses measured in a room differing only in the position of the sound source, it might be possible to quantify the sensitivity of the room due to changes in initial conditions. Such changes are linked to mixing. The proposed measure is the maximum of the absolute value of the cross-correlation between the time windowed sections of the two impulse responses. By integrating this quantity normalized by the energy of the impulse response of the room, a single number rating is obtained. The proposed measure is examined experimentally, and the results are discussed. The results indicate that the number of absorbers and diffusers in the room influences the proposed measures systematically.

9:20

1aAAa2. Room acoustic transition time based on reflection overlap. Cheol Ho Jeong, Jonas Brunskog, and Finn Jacobsen (Acoust. Technol., Tech. Univ. of Denmark, Oersteds Plads, Bldg. 352, Lyngby 2800, Denmark, chj@elektro.dtu.dk)

A transition time is defined based on the temporal overlap of reflected pulses in room impulse responses. Assuming specular reflections only, the temporal distance between adjacent reflections, which is proportional to the volume of a room, is compared with the characteristic width of a pulse at time t, which is mainly controlled by the absorption characteristics of the boundary surfaces of the room. Scattering, diffuse reflections, and diffraction, which facilitate the overlapping process, have not been taken into account. Measured impulse responses show that the transition occurs earlier in a room with nonuniform absorption and furniture than in a room that satisfies the underlying assumptions.

9:40

1aAAa3. What is "clarity," and how it can be measured? David H. Griesinger (Research, David Griesinger Acoust., 221 Mt Auburn St #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Human hearing did not evolve to detect reflections and reverberation. It is the ability to detect the direct component of a sound field that allows us to separate simultaneous signals, determine their direction, their timbre, their distance, and their importance for our attention. "Clarity" is perceived when we can separately detect direct sound. But ISO3382 measures concentrate on reflections, and are either blind to the direct component of a sound field or misinterpret its significance. They fail to predict sound quality in individual seats. We will demonstrate the vital importance of clarity through demonstrations of "clear" and "muddy" in speech and music. We will then present three physiologically based methods that measure the degree of clarity in a particular acoustic environment. The first, LOC, uses a simple nerve firing model to analyze an impulse response for the build-up of reflected energy at the onsets of sounds. The second method measures the degree of phase randomization above 1000 Hz caused by a particular impulse response. The third measure—based on a computer model of human hearing—measures clarity directly from binaurally recorded speech. All three measures predict perceived clarity with useful accuracy.

10:00

1aAAa4. Perceptual limits for detecting interaural-cue manipulations measured in reverberant settings. Stefan Klockgether (Acoust. Group, Univ. of Oldenburg, Carl-von-Ossietzky-strasse 9-11, Oldenburg 26129, Germany, steven.van.de.par@uni-oldenburg.de), Jasper van Dorp Schuitman (Philips Res. Europe, Eindhoven, Netherlands), and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Oldenburg, Germany)

In this study, binaural room impulse responses (BRIRs) were manipulated to determine the just noticeable differences in the interaural time delay (ITD), interaural level difference (ILD), and interaural cross-correlation (ICC) in reverberant settings. The BRIR were split in two sections, the first 75–150 ms of the BRIR was found to be direction dependent, and for this first part either an extra ITD or ILD was applied. These manipulations were expected to change the perceived direction of the sound source. Changes in the ICC were applied to the remaining part of the BRIR, which was expected to change the overall spatial impression, but not the perceived location. Each of these three differently manipulated BRIRs was convolved with an anechoic musical instrument, and the just noticeable change in ITD, ILD, or ICC was determined in a listening experiment. Due to the convolution with a temporally varying musical instrument stimulus, a complex spectrotemporal pattern of binaural cues is created. An analysis of these cues will be presented and it will be compared to the listening test results. This analysis will be based on a model of human auditory processing, which predicts perceptual cues related to room acoustic perception.

10:20–10:40 Break

10:40

1aAAa5. Accuracy in the acoustical parameters evaluation according to ISO-3382. Miguel Arana, Ricardo San Martin, Abel Arregui, and Jorge Machin (Physics, Public Univ. of Navarre, Campus de Arrosadia, Pamplona, Navarre 31006, Spain, marana@unavarra.es)

An exhaustive characterization of the new auditorium of the Navarre Conservatory of Music (Pamplona, Spain) has been carried out. All monaural acoustic parameters in all seats (375) have been measured for three source positions on the stage. For acoustic characterization, countless results can be obtained in accordance (in all cases) with the views of the ISO-3382 for the presentation of the results. The spatial dispersion for each source position and combinations thereof will be shown. Finally, the accuracy on the acoustic evaluation of the room will be discussed from a statistical point of view.

11:00

1aAAa6. On the effects of pre-processing of impulse responses in the evaluation of acoustic parameters on room acoustics. Andrea Venturi (Univ. of Bologna, Via delle Scienze, Parma 43100, Italy, andrea.venturi78@gmail.com), Angelo Farina, and Andrea Venturi (Univ. of Parma, Parma, Italy)

The evaluation of room acoustics characteristics in rooms has been thoroughly described in several papers since 1960s. Moreover, the ISO 3382 standard describes several acoustic parameters and their measurements. However, there are only a few information about the methods of pre-processing the impulse responses that are required before calculating those acoustic parameters. In this paper, the main processing methods (based on Luneby, Chu, and Hirata methods) are analyzed. Moreover, they are compared with the Schroeder (background integrated) methods. In a further step, these methods are applied in some acoustic measurements employed in some opera houses in Italy. Finally, after a full discussion about the uncertainties that is beyond these methods, the acoustic parameters are compared with the JND that is actually accepted in the evaluation of the mono-aural, binaural, and spatial acoustic parameters.

11:20

1aAAa7. The influence of noise on monaural room acoustic parameters utilizing different evaluation methods. Martin Guski and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustraße 50, Aachen 52066, Germany, mgu@akustik.rwth-aachen.de)

Besides clarity and definition, the reverberation time is the most common room acoustic parameter. The latter is also an essential quantity for acoustic measuring techniques (i.e., sound insulation, scattering, or diffuse absorption). The unavoidable occurring noise in every measurement is the most significant factor that causes incorrect evaluated parameters. Therefore, it is important to respond to the effects caused by noise. In this study, different noise compensation methods are compared theoretically and based on measurements. At first all methods are investigated theoretically utilizing a simple parametric model impulse response. As a second step, long-term measurements have been conducted in an auditorium to analyze the performance of the different techniques under realistic conditions. Therefore, the excitation signal has been varied in volume to obtain measurements with different noise levels, and the evaluated room acoustic parameters are examined as a function of peak-signal to noise ratio. Theoretical and measured results coincide with each other for each analyzed method. The performances of the examined evaluation methods differ clearly. In particular, the three methods defined by ISO 3382 show different behaviors. The advantages and the limitations of each noise compensation method are presented.

11:40

1aAAa8. Including directivity patterns in room acoustical measurements. Martin Pollow, Johannes Klein, Pascal Dietrich, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustraße 50, Aachen 52056, Germany, mpo@akustik.rwth-aachen.de)

Room acoustical measurements according to ISO 3382 require source and receiver to be of omnidirectional sensitivity. Therefore, radiation patterns of natural sources and receivers (although audible) are not accounted for when using the obtained room impulse responses (RIRs) for room acoustic analysis or even auralization. In order to include this spatial information in the RIR, it is necessary to measure the RIR for each pair of radiation patterns of source and receiver. This could be done by electronic beamforming during the measurement using array systems or by mechanical modification of the transducer (as, e.g., a dummy-head with its corpus). In this contribution, an alternative approach is shown, using the superposition of a set of sequential measurements done with a spherical sound source. At the cost of longer measurement times, the obtained data can be used universally to synthesize RIRs of arbitrary directivity up to a certain maximal spatial resolution, as long as the room is considered as a linear and time-invariant system during the measurement. The measurement device, obtained results, and a study of the validity of the superposition approach are presented in this talk. Based on this representation of the RIR, more advanced spatial room acoustic analysis accounting for arbitrary sets of source and receiver directivity becomes possible.

MONDAY MORNING, 3 JUNE 2013 — 513DEF, 9:00 A.M. TO 12:00 NOON

Session 1aAAb

Architectural Acoustics and Noise: Cultivating the Sustainable in Architectural Acoustics

Jesse Ehnert, Chair
Arpeggio, LLC, 1060 Mercer St., SE, Atlanta, GA 30316

Invited Papers

9:00

1aAAb1. Achieving the acoustical credit within the Leadership in Energy and Environmental Design (LEED®) for healthcare green building rating system and changes within the forthcoming 2014 Guidelines. Daniel M. Horan (Cavanaugh Tocci Assoc., Inc., 327 F Boston Post Rd., Sudbury, MA 01776, dhoran@cavtocci.com) and Jean-François Latour (Acoust. and Vibr., SNC-Lavalin Inc., Longueuil, QC, Canada)

The 2009 LEED® for Healthcare green building rating system includes a total of two possible points to be earned by satisfying the requirements of the Indoor Environmental Quality Credit 2 (IEQ Credit 2: Acoustic Environment). This credit references criteria that are defined by the Facility Guidelines Institute's 2010 *Guidelines for Design and Construction of Health Care Facilities* (2010 FGI Guidelines). The acoustical design requirements of the 2010 FGI Guidelines will be summarized, as well as a brief history of the FGI document itself. The FGI Guidelines document is currently being revised for the 2014 edition. Proposed changes in the 2014 edition will also be briefly discussed, as we anticipate that future versions of the LEED® rating system will reference the 2014 FGI Guidelines. The presentation will be made in English by Mr. Horan, secretary of the FGI's Acoustical Working Group (AWG). Mr. Latour, a French-Canadian member of the AWG, will be available to help respond to any French-speaking audience members during Q&A.

9:20

1aAAb2. Achieving Leadership in Energy and Environmental Design acoustical requirements in a commercial office project. Ethan Salter (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, ethan.salter@cmsalter.com)

This paper will discuss a recently completed a Leadership in Energy and Environmental Design Commercial Interiors (LEED-CI) office renovation project in San Francisco, California. The project had multiple aims: office space, test laboratory for new materials and technologies, and teaching tool for clients and students. The owner, an acoustical consulting firm, decided on LEED-CI certification

because of the belief that acoustic comfort could be achieved while simultaneously meeting other LEED requirements (e.g., design, products and materials, construction methods, and operations). Recently, acoustical requirements have been adopted into various LEED rating systems because occupant acoustic comfort in many LEED-certified buildings has been poor. The organization responsible for LEED, the US Green Building Council, is taking steps to more comprehensively adopt acoustical standards throughout their portfolio. By using the LEED Innovation in Design (ID) Pilot Credit Library, projects can attempt to achieve a wider range of potential credits. One of those credits, Pilot Credit 24, addresses acoustic comfort, including sound isolation, speech privacy, background noise, and reverberation time. The project is one of the first to achieve Pilot Credit 24 requirements. This paper will discuss the project design objectives, Pilot Credit 24 requirements, and how the project achieved those requirements.

Contributed Papers

9:40

1aAAb3. Straw bale sound insulation: Blowing away the chaff. Stephen Dance and Paul Herwin (Urban Eng., London South Bank Univ., Borough Rd., London SE1 0AA, United Kingdom, dances@lsbu.ac.uk)

Popular opinion states that straw bale walls are good at isolating sound. Cheap load bearing straw bale houses could contribute substantially to low carbon sustainable construction. However, literature on the subject was found to be highly anecdotal. The paper presents a summary of nine laboratory and field sound insulation test reports and two especially commissioned tests. Data were compared to European party wall sound insulation criteria, and it was found that straw walls could perform as well as, but sometimes worse than, conventional constructions, due to poor performance at low frequencies. Better performance could help to promote the use of straw bales in multi-unit housing. It was found that by adding a plasterboard layer on studs to just one side of a plastered straw bale wall would allow the construction to pass all of the criteria reviewed.

10:00–10:20 Break

10:20

1aAAb4. From felt to fungus: New materials and applications—Focus on sustainability. Dawn Schuette and Scott Pfeiffer (Threshold Acoust. LLC, 53 W Jackson Blvd, Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

A two-part presentation of new materials for use in architectural acoustics. This presentation emphasizes new materials or new applications of standard products that provide acoustic benefit in a highly sustainable context. The companion session is presented in "New Materials for Architectural Acoustics." Current trends in architecture are bringing more organic approaches to the use of natural materials. Exploiting these trends with approaches that have definable acoustic behavior leads to more flexibility in architectural design and yields acoustical application of materials that are not traditionally part of the acoustical treatment vocabulary. Case studies will be presented featuring new materials and/or methods being employed for sustainable acoustic solutions.

10:40

1aAAb5. Development of an ecological, smooth, unperforated sound absorptive material. Seda Karabulut (R&D, MEZZO Studyo LTD., METU,R&D, MEZZO Studyo LTD, Ankara, Turkey, sedakarabulutt@gmail.com) and Mehmet Çalışkan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

Material selection for acoustically comfortable environments is a very important issue especially for rooms for speech and music as well as for large volumes like shopping centers and foyers. Energy efficient and sustainable materials are devised in construction industry for healthy environments; hence, ecological sound absorbing materials for acoustically sensitive environments are being preferred to get credits for international certification procedures like LEED and BREAM. Nevertheless, most of the acoustic materials in construction industry are perforated with mineral wool based absorption materials behind and have great effects on design of the building environments. Architects usually prefer seamless unperforated materials to avoid changes in the appearance of design environments for acoustical requirements. This article is about development of an ecological unperforated acoustical material, which is made of reed and pumice stone. Different layers of pumice stone and reed glued with an ecological binder are evaluated according to frequency range they are effective, and an optimization is done to create an acoustic material that is effective especially in medium frequencies. Acoustical performance of the material is justified with measurements of sound absorption coefficient in Kundt Tube.

11:00

1aAAb6. Design, optimization, and testing of door-grille silencers. Vivek V. Shankar and Murray Hodgson (Acoust. & Noise Res. Group, Univ. of British Columbia, 2054-6250 Appl. Sci. L., Vancouver, BC V6T1Z4, Canada, vivekvs1@gmail.com)

It is a common practice to install doors that have openings in them to improve cross airflow through horizontal ventilation. However, excessive outdoor noise and poor noise privacy are known associated issues. Grilles are often installed in these door openings to address this issue. While they may reduce the noise level slightly, they have proven not to be very effective. Effective silencers would be too thick to be installed in doors. This work investigates the design and development of a novel door silencer that reduces the sound transmission to acceptable limits without compromising the airflow. A model of the silencer has been designed and tested—using the Acoustics module of the COMSOL Finite Element software—in a diffuse field environment, and validated with STC ratings. The airflow was modeled using the COMSOL CFD module. The dimensions of the ventilation opening and its position in the door have also been optimized. A real prototype of the model has then been built and its performance tested. Various design guidelines have then been proposed for the design of these doors.

11:20

1aAAb7. The furniture industry needs a new evaluation standard for evaluation of sound absorption. Klas G. Hagberg (Acoust., WSP, Box 13033, Goteborg 41526, Sweden, klas.hagberg@wspgroup.se) and Delphine Bard (Acoust., dBA R&D, Neuchatel, Switzerland)

Since decades the standard ISO 11654 are prevailing for evaluating sound absorption of products. The standard is developed and fully adapted to ceiling manufacturer, in particular mineral wool ceiling manufacturer. However, the standard is used independently of which interior product it is applied to, casing a lot of "misuse" and confusion amongst many manufacturer. In particular, when it comes to evaluation of various types of office screens, the ISO 11654 becomes a problem. There is no need, and probably not even possible, to calculate absorption factor for an office screen correctly. Therefore, Sweden decided to develop a new standard, SS 25269—"Acoustics—Evaluation of sound absorption of single objects." Hence office screens should be treated as single objects to cover a wide range of variety. The standard specifies an evaluation method of the sound absorption using only sound absorption area for each object tested. It will simplify the evaluation and minimize risk for errors since there is no need to state the product area in order calculate absorption coefficient. Furthermore, the area does not have to be stated yet again when performing calculation of room characteristics when using the same products in the finished room.

11:40

1aAAb8. Audio and acoustic design of the University of Sydney's Indoor Environmental Quality Laboratory. Densil Cabrera, Robert Crow, Luis Miranda, and Richard de Dear (Faculty of Architecture, Design & Planning, The Univ. of Sydney, G04, Sydney, NSW 2006, Australia, densil.cabrera@sydney.edu.au)

The quality of indoor environments such as commercial offices is affected by many factors, including temperature, humidity, air movement,

illumination, ambient sound, and room acoustics. In 2012, a new laboratory was established at the University of Sydney to examine how such factors affect human occupants. In terms of sound, the design of the laboratory has three components: the acoustic design of the testing rooms; the audio system design (for introducing artificial soundscapes); and the design of generic soundscapes to support experimental work in the laboratory. Acoustic design considerations of the laboratory allow for the testing rooms to be configured as high grade office environments. The laboratory has a 24-channel audio system for introducing realistic and potentially complex sound fields in to the testing rooms, both from within and outside the rooms. Parametrically controlled soundscapes have been developed for interior sources (such as building services noise) and exterior sources (such as transport noise). This paper describes how the combination of the laboratory's acoustics, audio systems, and soundscapes can be used for scientific studies of indoor environmental quality.

MONDAY MORNING, 3 JUNE 2013 — 510D, 8:55 A.M. TO 12:00 NOON

Session 1aAO

Acoustical Oceanography: Estuarine Acoustics

Andone C. Lavery, Cochair
Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536

David R. Barclay, Cochair
Memorial Univ. of Newfoundland, P.O. Box 4200, St John's, NF A1C 5S7, Canada

Chair's Introduction—8:55

Invited Paper

9:00

1aAO1. The impact of acoustic oceanographic methods on estuarine dynamics research. W. Rockwell Geyer, Peter Traykovski, and Andone Lavery (Woods Hole Oceanogr. Inst., 98 Water St., MS-12, Woods Hole, MA 02543, rgeyer@whoi.edu)

Estuaries present unique challenges for observational oceanographers, due to their intense spatial gradients and unrelenting temporal variability. The influence of spatial and temporal variation of estuarine structure and flow on the time-averaged regime is the most important research problem in estuarine physical oceanography. Acoustic methods have played an essential role in revealing this spatial and temporal variability, and new advances in acoustic methods are continuing to provide the most important advances in observations of estuarine processes. The measurement of acoustic backscatter has been a mainstay of estuarine physical oceanography, first for providing qualitative images of the density structure, then for quantifying suspended sediment distributions, and most recently for quantifying the intensity of stratified turbulence. Improved resolution of new systems is revealing the internal structure of shear instability and the mechanics of the transition to turbulence. Acoustic Doppler techniques are so routine now as to be taken for granted, but their impact on the field cannot be overstated, and the new advances in pulse-coherent velocity profiling are continuing this revolution in acoustical oceanography. Acoustic propagation in estuaries has not yet received much attention, but its importance to the operation of autonomous vehicles and long-term monitoring should bring this challenging acoustics problem to the forefront.

Contributed Papers

9:20

1aAO2. The spatial properties of breaking wave generated and bedload transport generated noise in the sediment layer of a shallow water wave guide. David R. Barclay, Len Zedel (Phys. and Physical Oceanogr., Memorial Univ. of Newfoundland, P.O. Box 4200, St John's, NF A1C 5S7, Canada, dbarcl@gmail.com), Alex E. Hay, and Matthew G. Hatcher (Oceanogr., Dalhousie Univ., Halifax, NB, Canada)

In May of 2012, three weeks of ambient noise measurements from a hydrophone buried 30 cm deep in the sediment were recorded at Advocate Beach, a 1:10 sloped beach at the head of the Bay of Fundy, Nova Scotia. While tides varied the mean water depth between 0 and 4 m, 0.8 m surface waves passed overhead, driving sediment bedload transport and creating an ambient noise field in the sediment consisting of two primary components: noise generated by bubbles formed in breaking waves at the surface and noise generated by the collisions of sand, gravel, and cobble in the bedload transport along the seabed. Both of these noise sources are stochastic and can be described by their second order statistics: power spectral density, spatial coherence, and directional density. In an effort to distinguish these two noise sources, the spatial properties of three full wave models of the noise field in the sediment are compared, using an infinite sheet of sources placed near the surface of a Pekeris waveguide to model breaking wave noise, near the fluid–fluid interface of a Pekeris waveguide to model bedload transport noise, or near the fluid–fluid interface of two infinite half-spaces to model bedload transport noise. Using integral transforms to solve the wave equation, each noise model is shown to be spatially inhomogeneous with a unique depth dependent intensity and coherence.

9:40

1aAO3. Acoustic measurements of the spatial distribution of suspended sediment at three sites on the Lower Mekong River. Stephanie A. Moore (Civil Eng., Univ. of Ottawa, 161 Louis Pasteur St, Ottawa, ON K1N6N5, Canada, moore@uottawa.ca), Guillaume Dramais (UR HHLY Hydrology Hydraulics, Irstea Lyon, Lyon, France), Philippe Dussouillez (Ctr. Européen de Recherche et d'Enseignement des Géosciences de l'Environnement, Aix-en-Provence, France), Jerome Le Coz (UR HHLY Hydrology Hydraulics, Irstea Lyon, Lyon, France), Colin Rennie (Civil Eng., Univ. of Ottawa, Ottawa, ON, Canada), and Benoit Camenen (UR HHLY Hydrology Hydraulics, Irstea Lyon, Lyon, France)

The Mekong River spans thousands of kilometers, flows through six countries, and its basin is one of the world's richest in terms of biodiversity.

However, land-use changes, dredging of the river bed, and the construction of dams are changing its sediment dynamics and morphology. The resultant increases in bank erosion and reduction in sediment supply to floodplains may have adverse effects on the economical and biological productivity of the region. In order to monitor these changes, the current conditions must be well understood. Comprehensive measurements of the spatial distribution of sediment (both suspended and bed load) were made at three locations in different physiographic regions of the Lower Mekong at the end of the 2012 rainy season. Acoustic Doppler Current Profilers and a multifrequency acoustic backscatter system, the AQUAscat, were used in combination with water sampling to provide high resolution measurements of concentration and grain size. The AQUAscat consisted of four monostatic transducers operating at 0.5, 1, 2.5, and 4 MHz. At each site, it was deployed horizontally at five across-stream positions and 5–10 depths per vertical; a 10 m profile was recorded at each point. The spatial distribution of particle size and concentration were determined using multifrequency inversions of (1) backscattered intensity and (2) attenuation calculated from the intensity profiles. This data set provides a baseline to which to compare future measurements at these sites.

10:00–10:20 Break

10:20

1aAO4. Acoustic propagation characteristics of the estuarine salt wedge. D. Benjamin Reeder (Oceanography, Naval Postgrad. School, 73 Hanapepe Loop, Honolulu, Hawaii 96825, dbreeder@nps.edu)

The estuarine environment often hosts a salt wedge—a layer of denser seawater advected by the rising tide under fresh water discharged by the river. The nature of the stratification is a function of the tide's range and speed of advance, river discharge volumetric flow rate, and river mouth morphology. The competing effects of temperature and salinity on sound speed present the question: Is the salt wedge acoustically observable? Using temperature and salinity profiles collected *in situ*, numerical results show that the salt wedge can impact acoustic propagation. Acoustically, this environment consists of two isospeed layers separated by a thin gradient. While this three-layer very shallow water acoustic waveguide is typically dominated by high angle multipath propagation, refraction occurring in the gradient layer allows some low-angle energy from near-surface sources to be trapped in the upper layer. Acoustic fluctuations observed at an upstream or downstream receiver depend upon the interaction between the advancing and receding tide and the river discharge, which can include the presence of internal waves propagating along the top of the salt wedge.

10:40

1aAO5. Quantification of stratified turbulence using acoustic propagation and broadband scattering techniques. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., MS 11 Bigelow 211, 98 Water St., Woods Hole, MA 02543, alavery@whoi.edu), Wayne R. Geyer (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), and Malcolm E. Scully (Physical Oceanogr., Old Dominion Univ., Norfolk, VA)

Narrowband acoustical backscattering techniques have been used for decades as a tool for remote imaging of small-scale physical processes in energetic coastal environments, such as oceanic internal waves and microstructure, on spatial and temporal scales difficult to probe with in situ measurements. However, to date, it has been challenging to infer quantitative information about turbulent intensity from the measured backscatter, in part due to uncertainty in the sources of scattering. In contrast to narrowband techniques, emerging broadband techniques result in increased spectral classification and quantification capabilities. Broadband backscattering collected in the Connecticut River Estuary in 2009 in concert with in situ measurements of turbulence have illustrated the potential of these techniques for quantitative remote-sensing of microstructure intensity over relevant spatial and temporal scales. These measurements have resulted in remote quantification of finescale variability of turbulent mixing as well as examination of the mechanisms and structure of shear instability across a broad range of stratification and shear conditions. Upcoming acoustic propagation measurements directed at measuring acoustic scintillation in the Connecticut River Estuary in the presence of strongly stratified turbulence and shear instabilities, which allow remote-sensing of the path-averaged statistical structure and motion of the intervening flow, will also be discussed.

Invited Paper

11:00

1aAO6. Acoustics and estuarine ecology: Using active and passive methods to survey the physical environment, vegetation, and animals in North Carolina's coastal estuaries. Joseph J. Luczkovich (Inst. for Coastal Sci. and Policy, East Carolina Univ., M. S. 169, Greenville, NC 27858, luczkovichj@ecu.edu), Mark W. Sprague (Physics, East Carolina Univ., Greenville, NC), Cecilia S. Krahforst (Coastal Resources Management, East Carolina Univ., Greenville, NC), John P. Walsh (Geological Sci., East Carolina Univ., kitty Hawk, NC), Audrey J. Pleva (Biology, East Carolina Univ., Greenville, NC), and Dean E. Carpenter (Albemarle-Pamlico National Estuary Partnership, Raleigh, NC)

Estuarine systems have complex interactions of physical and biological processes. Regular observations are needed in order to understand their dynamics. Acoustic observation systems (echosounders, acoustic Doppler current profilers (ADCPs), and passive acoustic dataloggers) can provide observations on a wide spectrum of processes in estuaries. We have used echosounders to monitor changes in bathymetry, submerged aquatic vegetation, fishes, and invertebrates over time. In addition, sediment changes, resuspension events, turbidity, and waves are monitored using ADCPs. The higher trophic level species of fishes and marine mammals that are soniferous have been monitored by passive acoustic methods. We provide examples of each acoustic method used to study the dynamics of seagrasses, fishes, and the physical environment of the Albemarle, Pamlico, Currituck, and Core Sounds in North Carolina. While it is possible to combine methods to use acoustics to measure the dynamics of estuarine systems (estuarine observing systems), the challenge we face is to ground-truth these acoustic metrics using traditional sampling methods (e.g., quadrats for plants, trawls for fishes, and water samples for sediments) and integrate each of these measures. We could then examine the effect of storms, waves, and resuspension events on estuarine plant and animal distributions and abundances using acoustics metrics.

Contributed Paper

11:20

1aAO7. Investigation of low-frequency acoustic tissue properties of seagrass. Gregory R. Enenstein, Craig N. Dolder, Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 4700 W Guadalupe St #A-437, Austin, TX 78751, gregenenstein@gmail.com), and Jean-Pierre Hermand (Acoust. & Env. Hydroacoustics Lab, Université Libre de Bruxelles (ULB), Brussels, Belgium)

Understanding the acoustic properties of seagrass is important for applications in mine hunting, shallow water sonar performance, and acoustic remote sensing for ecological surveys. Previous laboratory and field investigations have shown that the plant biomass and tissue structure of seagrass, rather than just the overall gas content, play a determinant role in its acoustic behavior. Hence, effective medium models of propagation through seagrass meadows have been ruled out, and a complete description of both tissue structure and tissue elastic properties is required to describe the acoustic response of seagrass meadows. To begin to address these deficiencies, a resonance tube experiment was set up to determine the low-frequency acoustic response of multiple species of seagrass in relation to leaf biomass and tissue acoustic compliance independent of tissue structure. Responses to frequency-modulated signals in the range from 0.5 to 10 kHz were obtained for *Thalassia testudinum* (turtle grass) and *Halodule wrightii* (shoal grass), two species with well-differentiated morphological features. An elastic waveguide model was used to account for the minor effect of the tube walls on the resonance characteristics. Initial measurements of tissue compliance will be presented. [Work supported by ONR and ARL:UT.]

11:40–12:00 Panel Discussion

MONDAY MORNING, 3 JUNE 2013 — 519A, 9:00 A.M. TO 11:40 A.M.

Session 1aBA

Biomedical Acoustics: Ultrasound Tomography

Yun Jing, Chair

Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695

Invited Papers

9:00

1aBA1. Three-dimensional nonlinear inverse scattering: Quantitative transmission algorithms, refraction corrected reflection, scanner design, and clinical results. James Wiskin (Bioengineering, Univ. of Utah, 3216 Highland Dr., Ste. 100, Salt Lake City, UT 84106, jwiskin.cvus@gmail.com), David Borup (CVUS, LLC., Salt Lake City, UT), Michael Andre (Radiology, Univ. of California VA Medical Ctr., San Diego, CA), Steven Johnson (CVUS, LLC., Salt Lake City, UT), James Greenleaf (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN), Yuri Parisky (Radiology, Mammoth Hospital, Mammoth Lakes, CA), and John Klock (CVUS, LLC., Salt Lake City, UT)

Research in quantitative whole breast ultrasound imaging has been developing rapidly. Recently, we published results from 2D transmission inverse scattering algorithms, based on optimization, incorporating diffraction, refraction, and limited multiple scattering effects, using data collected from an early prototype, which showed the feasibility of high resolution quantitative imaging of the breast tissue speed and attenuation, and concomitant refraction corrected reflection imaging. However, artifact problems in speed and attenuation result from the 2D algorithms and the data characteristics. The reflection algorithm uses the speed map to model refractive effects of rays, so these artifacts are unacceptable. The 3D inverse scattering algorithm presented here, using data from a new prototype, overcomes most of these artifacts. We then use a 3D refraction corrected 360° compounded reflection algorithm for high resolution speckle free reflection images. We discuss the transmission and reflection algorithms and the advanced scanner used to collect the data, as well as initial clinical results from the Mayo Clinic, Breast Cancer Imaging Center, Orange County, and the University California, San Diego. We show examples of cysts, fibroadenomas, calcifications, cancers, and DCIS, in dense, fatty, and average breast tissue, and compare these with hand-held ultrasound, MRI, and mammography, where available.

9:20

1aBA2. Quantitative ultrasound tomography. Koen W.A. v. Dongen and Neslihan Ozmen-Eryilmaz (Lab. of Acoust. Wavefield Imaging, Delft Univ. of Technol., P.O. Box 5046, Delft 2600 GA, Netherlands, k.w.a.vandongen@tudelft.nl)

Whole breast ultrasound is gaining interest as a possible alternative to mammography, as it is cost effective, patient friendly, and avoids the usage of ionizing radiation. Due to the similarity in both measurement setups, scientists have investigated if tomographic reconstruction algorithms originally developed for x-ray tomography, such as inverse radon transforms, are also applicable to ultrasound tomography. However, the multiple scattering of an acoustic wave field inside the breast as well as diffraction and refraction effects results in a severe blurring of the obtained images. In order to overcome these limitations, people are developing imaging algorithms, which are based on the acoustic wave equations. To show the limitations and potentials of the various imaging algorithms, we computed synthetic data for a cancerous breast model using a full wave solution method. Next, we tested and compared different imaging algorithms varying from a ray based inverse radon transform up to a full-wave nonlinear inversion technique. The latter one has the advantage that, as we will show, it allows for accurate speed of sound profile reconstructions at the cost of a severe computational load.

Contributed Papers

9:40

1aBA3. A contrast source inversion method for breast cancer detection. N. Ozmen-Eryilmaz and K. van Dongen (Lab. of Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CJ, Netherlands, n.ozmen-eryilmaz@tudelft.nl)

Tomographic ultrasound imaging is gaining popularity in breast cancer detection. Reconstructing the acoustic properties of a breast from the ultrasound measurements is stated as a nonlinear inverse problem, which is usually solved by linearized methods because of computational efficiency. However, linearization of the problem reduces the quality of the reconstruction. To improve the accuracy, we developed and tested a three-dimensional nonlinear inversion method that allows for three-dimensional reconstruction of the breast in terms of speed of sound. The method, referred to as contrast source inversion (CSI), uses an integral equation formulation to describe the inverse acoustic scattering problem. The resulting integral equation is solved to reconstruct the unknown contrast (speed-of-sound profile of the breast). The contrast and contrast sources (the product of the contrast with the total field) are iteratively updated by minimizing a cost functional using conjugate gradient directions. In this study, we tested the CSI method on synthetic data retrieved form full-wave simulations for a realistic three-dimensional cancerous breast model. Results show that the CSI method outer performs other conventional methods as it yields speed-of-sound reconstructions that are akin to the model. This shows that the approach offers a contribution to the detection of breast cancer.

10:00

1aBA4. Approximation error method for full-wave tomography. Janne Koponen, Tomi Huttunen, Tanja Tarvainen (Appl. Phy., Univ. of Eastern Finland, Yliopistonranta 1, PL 1627, Kuopio 70211, Finland, janne.koponen@uef.fi), and Jari Kaipio (Mathematics, Univ. of Auckland, Auckland, Finland)

In ultrasound tomography (UT), the speed of sound (SOS) is reconstructed based on ultrasound measurements made on the surface of the object. As a part of the reconstruction process, which includes the solution of the inverse problem, propagation of acoustic signals in the medium is simulated using a forward model. Consequently, modeling errors can generate artifacts into reconstructed SOS. Accurate full-wave models can be computationally heavy and thus impractical in many real applications. On the other hand, approximate models typically lead to less accurate reconstructions. In this study, measurement noise and modeling errors of UT are modeled in Bayesian framework, and a numerical method that takes both errors into account is developed. The performance of the method is investigated by numerical simulations in which artifacts generated by a fast but less accurate forward model and approximate boundary conditions are compensated.

10:20

1aBA5. Flawed transducer detection using random sample consensus for ultrasound tomography. Tianren Wang and Yun Jing (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Eng. Bldg. 3, 3141, Raleigh, NC 27695, twang10@ncsu.edu)

In this paper, we present a random sample consensus (RANSAC) based ultrasound travel-time tomography method. Conventionally, all the time-of-flight (TOF) data between each two transducers are used to estimate the sound speed distribution. However, failing to identify the inaccurate TOF data (outliers) due to flawed transducers would reduce the accuracy of the estimated sound speed distribution. In our proposed approach, a small subset of TOF data were first randomly selected from the original TOF data, and then applied to the tomography algorithm to estimate a rough sound speed distribution. The rest of the TOFs data was applied to the rough distribution, and the goodness of fit was calculated. If most of the data fitted well in the estimated distribution, then all the well-fitted data (including the subset) was used to estimate a final sound speed distribution. Otherwise, there were outliers expected in the subset, and a new subset of the TOFs data would be randomly selected again. This repeated until most of the data fitted well in the estimated distribution. Simulation results showed that our method could effectively detect and eliminate outliers and increase the accuracy of estimating sound speed distribution.

10:40

1aBA6. Expressiveness of temperature-induced changes in backscattered energy in conventional B-mode images. Cesar A. Teixeira (CISUC, Univ. of Coimbra, Polo II, Pinhal de Marrocos, Coimbra 3030-290, Portugal, cteixei@dei.uc.pt), Marco von Kruguer (Biomedical Eng. Program, COPPE—Federal Univ. of Rio de Janeiro, Rio de Janeiro, Brazil), André V. Alvarenga (Lab. of Ultrasound-Directory of Scientific and Industrial Metrol. (Dimci), INMETRO, Rio de Janeiro, Brazil), and Wagner C. Pereira (Biomedical Eng. Program, COPPE—Federal Univ. of Rio de Janeiro, Rio de Janeiro, Brazil)

Changes on conventional B-mode images have been correlated with temperature, aiming to develop a reliable method for noninvasive temperature estimation. The assumption is that temperature variations induce wave propagation changes that modify the backscattered ultrasound signal and these changes have an expression in ultrasonographic images. One of the main effects is the change on the image intensity that is mainly caused by temperature-related changes in backscattered energy (CBE) from tissue inhomogeneities. It is reported that CBE is dependent on medium speed-of-sound and density, behaving in different ways for lipid or aqueous scatterers. In this paper, we demonstrate that CBE has an expression on B-mode images recorded from conventional ultrasound scanners. We observed that different regions have positive, negative, or undefined correlations with temperature, and that this behavior is due to the dependence of CBE with scatterers type. This differentiated behavior enables the segmentation of different structures inside the same tissue. Our experimental setup consisted in the temperature elevation from 36 to 44 °C (hyperthermia range) of *ex-vivo* tissue samples. We considered bovine muscle and porcine muscle and fat. For both samples, we observed coherent segmentations of the different structures, pointing for a potential clinical application of the proposed analysis.

11:00

1aBA7. Electromagnetic hydrophone for high-intensity focused ultrasound measurement. Pol Grasland-Mongrain (Université de Lyon, 151 Cours Albert Thomas, Lyon 69424, France, pol.grasland-mongrain@inserm.fr), Jean-Martial Mari, Bruno Gilles, and Cyril Lafon (LabTAU, INSERM U1032, Lyon, France)

An ultrasonic hydrophone based on the Lorentz force is introduced. When a metallic wire is moved by ultrasound while submitted to a magnetic field, the Lorentz force induces an electrical current proportional to the integral of pressure along the wire. 2D pressure field mapping is achieved by performing a tomography through wire translations and rotations in the imaging plane. Performances of this hydrophone are assessed in this study. Signal is linear over pressure from 10 kPa to at least 10 MPa with a determination coefficient R^2 above 0.997. Excellent resistance to cavitation has been observed. Frequency bandwidth was measured against three different wire diameters: 70 μm, 100 μm, and 210 μm. Results showed that upper cut-off frequency decreases with increasing wire diameter. Additional measurements showed that wire tension has no visible effect on the signal. Such characteristics are potentially of great interest for high-intensity focused ultrasound and shockwave transducers calibration.

11:20

1aBA8. Ultrasonic projection imaging of biological media. Krzysztof J. Opielinski and Tadeusz Gudra (Electron., Wroclaw Univ. of Technol., Wybrzeze Wyspianskiego 27, Wroclaw, Low Silesia 50-370, Poland, krzysztof.opielinski@pwr.wroc.pl)

The study presents the method of ultrasonic projection imaging of biological media, using single ultrasonic probes and 2-D piezoelectric transducer arrays. Dedicated research stands were set up and used to perform ultrasonic projection scanning of various biological media (and phantoms of the media) that were submerged in water. Based on such measurements, images of the heterogeneous internal structure of the studied objects were

obtained, which show two-dimensional distributions of the projection values of acoustic parameters. Those parameters were derived from recorded pulses of ultrasonic wave transmitted sequentially through a fixed projection surface. The obtained projection images were analyzed with respect to the method and quality of representation of the studied structures. Additionally, contrast resolution of ultrasonic projection images of the heterogeneous structure of a biological medium was estimated in relation to the size of the heterogeneity and with respect to scanning resolution and longitudinal resolution. Ultrasonic projection imaging can be applied in medicine for diagnostic examination of women's breast.

MONDAY MORNING, 3 JUNE 2013 512AE, 9:00 A.M. TO 12:00 NOON

Session 1aEA

Engineering Acoustics: Thermoacoustics I

Roger T. Richards, Chair
NUWC, Newport, RI 02841

Contributed Papers

9:00

1aEA1. Heat transfer enhancement through thermoacoustically driven streaming. Randall A. Ali and Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Grad. Prog. in Acoust., P.O. Box 30, State College, PA 16804-0030, randallali@gmail.com)

We recently reported on a simple standing-wave thermoacoustic engine that was intended for use as a self-powered monitor of temperature within a resonator that was similar to a nuclear fuel rod [J. Acoust. Soc. Am. **132**(3), Pt. 2, 1993–1994 (2012)]. An additional potential benefit of such a device is the enhanced heat transfer between the heat source and the surrounding coolant produced by the acoustic streaming generated by the high-amplitude acoustic standing wave within the resonator. By adding a remotely operated linear actuator that can depress a valve at the ambient-temperature end of the resonator, we are able to squelch the acoustic resonance by modification of that boundary condition without changing any other operating parameters (e.g., heater power). We will report heat transfer measurements made in a calorimeter at several input thermal power levels, with and without the presence of the thermoacoustic oscillations. [Work supported by the U.S. Department of Energy.]

9:20

1aEA2. Finite element simulation of a two-dimensional standing wave thermoacoustic engine. Jan A. de Jong, Ysbrand H. Wijnant, and André de Boer (Eng. Technol., Univ. of Twente, P.O. Box 217, Enschede, Overijssel 7500AE, Netherlands, j.a.dejong@utwente.nl)

A finite element analysis has been performed on a theoretical two-dimensional standing wave thermoacoustic engine using the linearized thermoacoustic equations in the frequency domain. This analysis is used to obtain the stability curve of the thermoacoustic engine, which in turn is used to calculate the oscillation onset temperature difference across the stack. The results are compared with existing theory including the long-pore approximation, as originally derived by Rott *et al*. In addition, the time-averaged effects of the acoustic wave are obtained using weakly nonlinear thermoacoustic theory. This includes the second order time-averaged equations for energy, momentum, and continuity. The saturation amplitude of the acoustic pressure oscillation and the required heat input to sustain the oscillation is obtained. The theory allows for calculation of acoustic streaming patterns. The particle path calculations provide insight to the minor loss mechanism occurring at the interface between the stack and the free tube for low acoustic velocity amplitudes (laminar flow).

9:40

1aEA3. Calculation of thermoacoustic functions with computational fluid dynamics. Simon Bühler (Thermal Eng., Univ. of Twente, P.O. Box 217, Enschede 7500AE, Netherlands, s.buhler@utwente.nl), Douglas Wilcox (Chart Inc., Troy, NY), Joris P. Oosterhuis, and Theo H. van der Meer (Thermal Eng., Univ. of Twente, Enschede, Netherlands)

Thermoacoustic functions are important parameters of one-dimensional codes used for the design of thermoacoustic engines. The thermal and viscous thermoacoustic functions allow the inclusion of three dimensional effects in one-dimensional codes. These functions are especially important in the regenerator of a thermoacoustic engine, where the thermoacoustic heat pumping occurs. Even though analytical solutions were derived for uniform pores, the thermoacoustic functions for complex geometries such as stacked screen or random fiber regenerators cannot be calculated analytically. In order to gain more insight into the geometry induced complex flow fields, the procedure of Udea *et al*. (2009) to estimate the thermoacoustic functions was applied in computational fluid-dynamic simulations. By using two measurement locations outside of the regenerator and modeling the regenerator as an array of uniform pores, it is possible to estimate the thermoacoustic functions for complex geometries. Furthermore, a correction method is proposed to quantify the entrance effects at the beginning and end of a regular pore. The simulations are first validated for a uniform cylindrical pore with the help of the analytical solution. Then the correction method is successfully applied to a cylindrical pore with the results closely matching the analytical solution. Finally, the method is applied to the model of a staggered, stacked screen regenerator.

10:00

1aEA4. Acoustic characteristics of a flexible sound generator based on thermoacoustic effect. Takehiro Sugimoto and Yoshiki Nakajima (NHK Sci. & Technol. Res. Lab., 1-10-11 Kinuta, Setagaya-ku, Tokyo 1578510, Japan, sugimoto.t-fg@nhk.or.jp)

A flexible sound generator based on the thermoacoustic effect was proposed for use with thin and flexible devices. The sound generator was composed of three thin films made of aluminum, polyimide, and graphite. The aluminum functions as an electrode for heat radiation, the polyimide as a heat insulator, and the graphite as a heat sink. Thickness of each layer is 50 nm, 75 μm, and 40 μm, respectively. The area of the electrode is 100 mm $\times$ 4 mm. The proposed sound generator was modeled considering several boundary conditions and using the heat conduction equation. Then, radiated sound was analytically described as a function of the input signal's frequency. Experimental measurement was carried out and the frequency response calculated by the model agreed with the measurement result. An experimental study was conducted on the relationship between the fundamental response and the harmonic distortion. Surface vibration was observed with the laser Doppler velocimeter. The observation revealed that the proposed device is a vibration-free sound generator. Detailed comparison between the calculation and the measurement will be discussed in the presentation.

10:20–10:40 Break

10:40

1aEA5. Study on thermoacoustic system to drive by low temperature—Effects of loop-tube thermoacoustic system connected with parallel double stacks on the onset temperature ratio. Yosuke Nakano (Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, dum0331@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Univ. of Shiga Professor, Hikone, Shiga, Japan), and Yoshiaki Watanabe (Doshisha Univ., Kyotanabe, Kyoto, Japan)

As the temperature ratio of the both ends of the stack increases gradually and reaches a critical value, sound waves begin to oscillate in the thermoacoustic system. This temperature ratio is called the onset temperature ratio. It is necessary to decrease the onset temperature ratio for practical use of the thermoacoustic system: the use of the factory exhaust heat and the solar heat. In a previous study, thermoacoustic system with series connected a number of prime movers was designed. This system can drive by lower onset temperature ratio than thermoacoustic system with a prime mover. However, it is considered that the heat loss increase when the heat is carried to a number of high heat exchangers. Therefore, loop-tube thermoacoustic system connected with parallel double stacks (parallel loop system) was proposed. This system can drive two prime movers by a heat input part because it is connected prime movers in parallel. In this report, the onset temperature ratio of this system was compared with that of normal loop-tube system with a prime mover. As a result, we confirmed that parallel loop system can drive by lower onset temperature ratio than normal loop-tube system.

11:00

1aEA6. The effect of resonance mode control by expanding of cross-section area on cooling capacity in a loop-tube type thermoacoustic cooling system. Manabu Inoue (Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe 610-0321, Japan, dmm1011@mail4.doshisha.ac.jp), Shin-ichi Sakamoto (Univ. of Shiga Professor, Hikone, Japan), Yosuke Nakano, and Yoshiaki Watanabe (Doshisha Univ., Kyotanabe, Japan)

One of assignments for practical realization of a loop-tube type thermoacoustic system is improvement of cooling capacity. It is known that cooling capacity of the system is improved by setting phase adjuster (PA), which reduces cross-section of the system locally. This is considered that because setting PA enables to control resonance mode and setting PA controls to one wavelength. However, there is much dissipation of energy in PA because of reducing cross-section. More cooling capacity of the system is expected by control resonance mode at less dissipation of energy. Therefore, expanding phase adjuster (EPA) as a devise, which expands cross-section locally, is proposed to reduce dissipation of energy. If setting EPA enables to control resonance mode, cooling capacity is expected to improve more. At first, we make sure that setting EPA enables to control resonance mode. Next, we make a comparison between the system set EPA and PA. As a result, cooling capacity of the system with PA is higher than with EPA. However, more cooling capacity is expected by shift the condition of EPA because energy used on cooling in the system with EPA is more than with PA.

11:20

1aEA7. On discontinuity waves and vibrations in thermo-piezoelectric bodies. Adriano Montanaro (Dept. of Mathematics, Univ. of Padua, via Trieste, 63, Padova 35121, Italy, montanar@math.unipd.it)

With regard to a body composed of a linear thermo-piezoelectric medium, referred to a natural configuration, we consider processes for it constituted by small displacements, thermal deviations, and small electric fields superposed to the natural state. We show that any discontinuity surface of order r greater than 1 for the above processes is characteristic for the linear thermo-piezoelectric partial differential equations. We show that discontinuity surfaces of order 0 generally are not characteristic; hence, the conditions are written, which characterize the discontinuity surfaces of order 0 that are characteristic. We find the ordinary differential equations of propagation for plane progressive waves and standing waves. Then we characterize the ones whose wavefronts are characteristic.

11:40

1aEA8. Computational fluid dynamics simulation of Rayleigh streaming in a vibrating resonator. Joris P. Oosterhuis, Simon Bühler (Thermal Eng., Univ. of Twente, P.O. Box 217, Enschede 7500AE, Netherlands, j.p.oosterhuis@utwente.nl), Douglas Wilcox (Chart Inc., Troy, NY), and Theo H. van der Meer (Thermal Eng., Univ. of Twente, Enschede, Netherlands)

Rayleigh streaming is a time-averaged flow that can exist in the thermal buffer tubes of thermoacoustic prime movers and refrigerators and is driven by the viscous stresses close to the solid boundaries. This mean flow leads to mean convective heat transport, which can have large impact on the performance of thermoacoustic devices. Rayleigh streaming in a standing wave resonator is simulated using a commercially available computational fluid dynamics (CFD) code and is compared to existing analytical models of Hamilton *et al*. (2003). A test case is developed, and a standing wave is generated by applying a harmonic volume force to the domain. Both the inner and outer streaming vortices are well described for a range of radii from $R/\delta\nu = 3\ldots20$ and the magnitude of the streaming velocity matches analytical values. This paper shows the possibility of using available as-is CFD software for the simulation of streaming in a standing wave resonator. The presented results pave the way for the simulation of more complex geometries and studies to reduce the negative effects Rayleigh streaming can have on thermo-acoustic prime mover and refrigerator efficiency.

Session 1aMU

Musical Acoustics: String Instrument Measurements

Agnieszka Roginska, Cochair
New York Univ., 35 West 4th St., Rm. 1077, New York, NY 10012

Chris Waltham, Cochair
Phys. & Astronomy, UBC, 6224 Agricultural Rd., Vancouver, BC V6T 1Z1, Canada

Contributed Papers

9:00

1aMU1. Eigenvalue shapes compared to forced oscillation patterns of guitars. Malte Muenster, Jan Richter, and Rolf Bader (Systematic Musicologie, Univ. of Hamburg, Pilatuspool, 19, Hamburg, Hamburg 20355, Germany, m.muenster@arcor.de)

Thirty-two guitars are measured geometrically and acoustically. The geometries of the top plate with its bracing as well as its thickness, the back plate with all bracing, the ribs, and rims are transferred to a CAD model. The top plate and the back plate of these guitars are measured using a 121-microphone array, back-propagating the sound field onto the top and back plates. Therefore, the guitars are once driven by impulses at the guitar bridge, once by plucking all notes on all strings up to the 12th fret to reconstruct the forced-oscillation patterns. Large differences are found with respect to the basic modes between the different guitars in terms of frequency and shape of their eigenmodes. Comparing the measured and calculated eigenvalues with the forced-oscillation modes driven by the strings, it appears that the eigenmode shapes often differ from the forced-oscillation patterns considerably.

9:20

1aMU2. Calculating guitar sound radiation by forward-propagating measured forced-oscillation patterns. Jan Richter, Malte Münster, and Rolf Bader (Univ. of Hamburg, Gefionstrasse 11, Hamburg 22769, Germany, janrichter81@gmx.de)

The radiation patterns of 32 guitars are investigated. Therefore, the top and back plates are measured using a 121-microphone array, back-propagating the recorded sound field onto the guitar top and back plates. Both, the eigenvalues and the forced oscillation patterns are measured, the latter by plucking the guitar strings for all possible notes. For each note, the forced-oscillation radiation pattern is calculated for 20 partials up to 4 kHz. These radiation patterns are then forward-propagated into the surrounding space around the guitar. Considerable differences appear between the different guitars within the same frequency region in terms of shape and intensity. Also, for similar frequencies, different patterns may appear, depending on the string and note played.

9:40

1aMU3. Measuring the haptic behavior of an acoustic guitar as perceived by the player by means of a vibrating actuator. Marcello Giordano and Marcelo M. Wanderley (Input Devices and Music Interaction Lab., CIRMMT, McGill Univ., 555, Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, marcello.giordano@mail.mcgill.ca)

Two sets of recordings of the vibration produced by plucking the fifth and the second string of an acoustic guitar were acquired using an accelerometer secured to the neck of the instrument. Vibrations from both sets could be reproduced using a recoil-type vibrating actuator attached at the neck of the guitar. In one of the sets, salient spectral features of the original recordings were altered. We performed a preliminary study involving nine volunteer participants, blindfolded and artificially deafened using earplugs and loud white noise played through headphones. They were asked to discriminate, by holding the neck of the instrument with their left hand, between "fake" (i.e., actuator-produced) or real vibrations, produced by the experimenter plucking either string two or five. Our aim was to assess if any of the spectral features altered in the second set of recordings increased the recognition rate of actuator-produced vibrations as being "fake." These features, if present, would then be likely to carry crucial information and should be therefore modeled with extreme care in the simulation of the haptic behavior of the instrument. Results show that, at least for string five, we were capable of identifying one feature (a peak in the vibration spectrum located at 548 Hz), which, if altered, made the recognition rate as "fake" rise, statistically significantly, from 55% to 89%.

10:00

1aMU4. Acoustic imaging of string instrument soundboxes. Chris Waltham, Evert Koster, Nils Smit-Anseeuw, and Aaron Zimmer (Phys. & Astronomy, UBC, 6224 Agricultural Rd., Vancouver, BC V6T 1Z1, Canada, cew@phas.ubc.ca)

A circular 30-microphone array of 90 cm radius has been used to produce acoustic images of several string instruments. The soundboxes were excited by an automated impact hammer, and the instruments were suspended inside the array in such a manner that both the soundbox and hammer mechanism could be rotated in the horizontal plane. By normalizing all microphone signals to the hammer signal, data could be assembled as if from an array of many times 30 microphones. Images were formed using the inverse frequency response function method. The array and data analysis code were tested with a rectangular plate set in a large plywood baffle, a system that was straightforward to simulate numerically. Due to the limited spatial resolution set by the array geometry and frequency range—typically 200–1000 Hz—instruments with long soundboxes were chosen for initial testing: a gothic harp, a guzheng, and a guqin.

10:20–10:40 Break

10:40

1aMU5. Characterization of bridge motions on the violin using polymer sensor technology. Gunnar Gidion and Reimund Gerhard (Phys. and Astronomy, Univ. of Potsdam, Geschwister-Scholl-Str. 75, Potsdam, Brandenburg 14471, Germany, gunnargidion@web.de)

Recent developments in minimally invasive polymer-film sensors permit the *in-situ* detection of mechanical vibrations in musical instruments without significantly disturbing the acoustics of the instrument. As an example, we present measurements of vibrations of a violin between the feet of the bridge and the top plate. To this end, calibrated fluoropolymer-film sensors were matched to the geometry of the bridge feet. The forces exerted on the top plate by either bridge foot can be measured separately during excitation of a

string with the bow. The differences in amplitude and phase between bass and treble foot vibrations exhibit the distinctly asymmetrical nature of bridge motions, which of course also depend on the string and the note that are being played, respectively. In comparison with the simultaneously detected string vibrations and the radiated sound, the filter characteristics of the bridge are clearly identified in the spectral representation. As the bridge is also the main agent for the coupling from the body to the string, it is suggested that the observed variations in bridge motion are closely connected to the fact that the playability of a violin changes sometimes quite drastically from note to note.

11:00

1aMU6. High resolution radiation pattern measurements of a grand piano—The effect of attack velocity. Agnieszka Roginska, Justin Mathew, Jim Anderson (New York Univ., 35 West 4th St., Rm. 1077, New York, NY 10012, roginska@nyu.edu), and Alex U. Case (Univ. of Massachusetts Lowell, Lowell, MA)

The sound radiation pattern of a grand piano is highly complex and depends on the shape of the soundboard, construction of the frame, reflections from the lid, and other parts of the instrument's structure. The spectral energy generated by and emitted from the instrument is further complicated by the sound production mechanism (hammers, strings), the attack velocity, and results in independently complex behaviors depending on the register of the piano. This paper presents the acoustic measurements of the radiation pattern of a grand piano using a high spatial resolution measurement technique. Measurements of a Yamaha Diskclavier were taken using a 32-channel microphone array with a 2-in. spacing between capsules. The complex radiation patterns and overtone structure is analyzed for middle-C at three attack velocities—pianissimo, mezzo forte, and forte. Comparisons of the effect of attack strength on frequency response and radiation pattern are presented.

11:20

1aMU7. Properties of violin glides in the performance of cadential and noncadential sequences in solo works by Bach. Jiaxi Liu (Faculty of Music, Univ. of Cambridge, Darwin College, Silver St., Cambridge CB3 9EU, United Kingdom, jiaxi.liu1@gmail.com)

This study examines the articulatory changes ("glides") between the leading tone and tonic of cadential vs noncadential semitone sequences in solo violin performance. It was predicted that though these glides would have similar slopes, they would differ in duration and in semitone intonation, and that these latter properties could characterize the expression of cadential finality and the structural insignificance of noncadential sequences. Cadential (46) and noncadential (58) targets from 17 recordings by 13 professional violinists were analyzed using narrow-band spectrograms. Glide durations comprised 16% of the overall duration of semitone sequence irrespective of structure function. However, cadential glides comprised 28% of the duration of the leading tone compared with 11% for noncadential glides. As predicted, the leading tone tended to be sharp in both contexts, but the mean cadential interval was nonsignificantly larger by 18 cents, mainly because the tonic tended to be tuned more accurately in cadential sequences. Finally, the glide direction was linear and followed the natural vibrato trajectory in both contexts as expected. These data confirm that articulatory modifications play a prominent role in the performance of intended musical structure and suggest that such distinctions will influence structural expectancies.

MONDAY MORNING, 3 JUNE 2013 — 511BE, 8:55 A.M. TO 12:00 NOON

Session 1aNS

Noise: Advanced Hearing Protection and Methods of Measurement I

Christian Giguere, Cochair
Audiol./SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H8M5, Canada

Elliott H. Berger, Cochair
Occupational Health & Env. Safety Div., 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

Chair's Introduction—8:55

Invited Papers

9:00

1aNS1. Attenuation as a function of the canal length of custom-molded earplugs. Jennifer B. Tufts, Siyuan Chen (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269, jennifer.tufts@uconn.edu), and Lynne Marshall (Naval Submarine Medical Res. Lab., Groton, CT)

The fit of a custom-molded earplug (CMEP) and the amount of attenuation it provides can be affected by variables related to the original earmold impression and the subsequent manufacturing process. One variable thought to affect the amount of attenuation is the length of the canal portion of the CMEP. In this pilot study, we systematically examined the relationship between CMEP canal length and attenuation in four human subjects. Two men and two women were fitted with CMEPs extending past the second bend of the ear canal. The attenuation provided by the CMEPs was measured over four visits to the laboratory. Prior to each visit, the canal portion of the subject's test CMEP was shortened by 2 mm. As expected, attenuation decreased as canal length decreased for all subjects. However, the rate and pattern of decrease varied markedly. Anecdotal reports of comfort as a function of canal length also varied markedly. Results suggest that the critical region/s in the ear canal for maintaining a good acoustic seal may vary from person to person. Implications for future study will be discussed.

9:20

1aNS2. "Calibrating" the insertion depth of roll-down foam earplugs. Elliott H. Berger (Occupational Health & Env. Safety Div., 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650, elliott.berger@mmm.com)

Since introduction in 1972, roll-down slow-recovery foam earplugs have become nearly ubiquitous. They are used widely in industries and by consumers around the world. Their performance has been reported in numerous journal articles and they are often part of laboratory experiments, either as the object of the study or as a reference device that is used as a control or to assure exclusion of noise from the ear to facilitate data acquisition. As such it is important to be able to describe their performance since although they generally provide high levels of protection, the amount of protection and its spectral dependence is a function of insertion depth. Real-ear attenuation results will be presented for a range of insertions from that which caps the earcanal to full earcanal insertion past the second bend. The results will be compared to published data to demonstrate how they can be used to estimate the quality of fit that was likely achieved vs that which was reported. These data will be useful to researchers who wish to "calibrate" the quality of insertion they are achieving in their own studies going forward. [The author is an employee of 3M and the research was funded by 3M.]

9:40

1aNS3. Attenuation characteristics of fit-compromised earmuffs and various non-standard hearing protectors. Laurie Wells, Elliott H. Berger, and Ron W. Keiper (Occupational Health & Env. Safety Div., 3M, 3M Ctr., 235-2W-75, St. Paul, MN 55144-1000, Laurie.Wells@mmm.com)

Excessive noise exposure can be successfully mitigated by proper use of legitimate hearing protection devices. However, real-life circumstances sometimes drive people to use compromised or alternative means of protection. This paper reports attenuation data measured in the 3M E•A•RCAL facility over several years, in conformance with ANSI real-ear attenuation at threshold test standards (S3.19-1974, S12.6-1984, and S12.6-2008 Method A) and also provides, for comparison, one dataset from the open literature (fingers/palms). The loss of attenuation was measured for various earmuffs worn in less than ideal conditions, including earmuffs worn in conjunction with various safety glasses, hairnets, head covers, hoods, earmuff cushion covers, and ball caps. Data were also obtained for non-standard means of blocking sound, including long hair, cotton balls, and even use of palms and/or fingers to block the ears. Results demonstrated that the effects on earmuff attenuation varied from none at all (suitable cushion cover) to as much as 12 dB (hooded sweatshirt). Realizing that people adapt hearing protectors to meet their needs is one step toward optimizing hearing protection selection and use; knowing the significance of these adaptations is the next step. [The authors are employees of 3M and the research was funded by 3M.]

10:00

1aNS4. Comparison of subjective and objective methods for the measurements of hearing protector devices attenuation and occlusion effect. Hugues Nélisse (Service de la recherche, IRSST, 505 Blvd De Maisonneuve Ouest, Montreal, QC H3A 3C2, Canada, hugues.nelisse@irsst.qc.ca), Cécile Le Cocq (Département de génie mécanique, École de Technologie Supérieure, Montréal, QC, Canada), Jérôme Boutin (Service de la recherche, IRSST, Montréal, QC, Canada), Jérémie Voix, and Frédéric Laville (Département de génie mécanique, École de Technologie Supérieure, Montréal, QC, Canada)

With the increase popularity of individual fit testing and miniaturization of electronic components, the field-microphone-in-real-ear approach (F-MIRE) is becoming more appealing and well suited for estimating hearing protection devices (HPD) attenuation both in laboratory and in "real world" occupational conditions. The approach utilizes two miniature microphones to simultaneously measure the sound pressure levels in the ear canal under the hearing protector, as well as outside of the protector. In this study, experiments on several human subjects were carried out in order to examine the various factors relating the subjective and objective attenuation values. The subjects were first instrumented on both ears with miniature microphones outside and underneath the protector. They were then asked to go through a series of subjective hearing threshold measurements followed by objective microphone recordings using high level diffuse field broadband noises. Earmuffs, earplugs, and double-protection were tested for each subject, and attenuation values were compared. Additionally, an objective scheme to measure the occlusion effect was developed and tested using subjects' voice as the excitation and the same microphone setup. Results obtained for the attenuation values as well as the occlusion effect levels are presented and discussed.

10:20–10:40 Break

Contributed Papers

10:40

1aNS5. Implementation of a simplified, artificial external ear test fixture for measurement of the earplug induced auditory occlusion effect. Martin Brummund (Dept. of Mech. Eng., École de technologie supérieure, 1100, rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, martin.brummund.1@ens.etsmtl.ca), Franck Sgard (Service de la recherche, IRSST, Montreal, QC, Canada), Yvan Petit, Frédéric Laville (Dept. of Mech. Eng., École de technologie supérieure, Montreal, QC, Canada), and Jérôme Boutin (Service de la recherche, IRSST, Montreal, QC, Canada)

Earplugs remain a frequently used short-term solution for occupational hearing conservation. Due to comfort limitations, as induced by, e.g., the occlusion effect, workers often only wear earplugs for limited amounts of time and are likely to develop professional hearing loss. The occlusion effect expresses itself in the low frequencies through an altered perception of the wearer's own voice and the amplification of physiological noises that occur upon earplug insertion. While many studies examined the occlusion effect experimentally, no study was found that attempted to implement an artificial external ear model dedicated to the measurement of the objective occlusion effect. A simplified external ear test fixture can help to better assess and design earplugs, because it allows standardized experimental testing. This work describes the implementation of a cylindrical artificial test fixture of the human outer ear that comprises the auditory canal as well as the bony, cartilaginous, and skin tissues that are made up of rigid polyurethane foam and two different types of silicone, respectively. Obtained measurement results are compared to literature findings.

11:00

1aNS6. Impulse noise attenuation by earplugs measured with the use of an acoustical test fixture and with the participation of subjects. Rafal Mlynski and Pawel Gorski (Wibroacoustic Hazards, Central Inst. for Labour Protection – National Res. Inst., Warszawa, Poland, pawel@ciop.pl)

The effectiveness of impulse noise attenuation by hearing protector devices is most often determined collecting the data from measurements. In impulse noise conditions with high peak sound pressure level, it is necessary to replace a subject in a measurement with an acoustical test fixture. The use of the acoustical test fixture is important because of the potential risk of hearing damage occurring during impulse noise tests, performed with the participation of subjects. The impulse noise attenuation by earplugs determined from measurements carried out using acoustical test fixture was compared with attenuation determined with the participation of subjects (MIRE technique). The acoustical test fixture complied with the acoustic and mechanical requirements described in Standard No. ISO 4869-3 and was equipped with a chamber representing the external ear canal and a 2 cm^3 chamber reflecting the acoustic properties of the middle ear. The results of measurements carried out with two different methods were comparable.

11:20

1aNS7. Influence of the external ear tissue domains on the sound attenuation of an earplug predicted by a finite element model. Guilhem Viallet (Mech. Eng., École de technologie supérieure, 1100, rue Notre-Dame ouest, Montreal, QC H3C 1K3, Canada, guilhem.viallet.1@ens.etsmtl.ca), Franck Sgard (Noise and Vib., Institut de Recherche de Robert Sauvé en santé et en sécurité du travail, Montreal, QC, Canada), and Frédéric Laville (Mech. Eng., École de technologie supérieure, Montreal, QC, Canada)

Earplugs are a widespread solution to prevent the problem of hearing loss in the workplace environment, but they do not always perform as desired. Using a model of the ear canal occluded by an earplug could be helpful to perform sensitivity analyses (geometry and materials of the earplug) and to better understand the role of the earplug. The human external ear is a complex system made up of different tissues with a 3D geometry. In practice, it is reduced to a 2D cylindrical geometry for the acoustical tests fixtures. The purpose of this study is to compare the insertion loss predicted by a 3D complex finite element model of the ear canal surrounded by different tissue domains (skin, soft tissue, and bone) and occluded by a silicon earplug versus a 2D axisymmetric model of the same system. In both models, some investigations are made in order to verify if the models could be simplified by replacing the tissue domains by mechanical impedances. These investigations are made to reduce the complexity of the models and to discuss the relevance of whether or not including external ear tissue domains in a sound attenuation model of an earplug.

11:40

1aNS8. Sound transfer path analysis to model the vibroacoustic behavior of a commercial earmuff. Sylvain W. Boyer (Département de Génie Mécanique, École de Technologie Supérieure, 1100 rue Notre-Dame, Ouest, Montréal, QC H3C 1K3, Canada, sylvain.boyer.1@ens.etsmtl.ca), Olivier Doutres (Groupe d'Acoustique de l'Universite de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC, Canada), Franck Sgard (Institut de Recherche Robert-Sauvé en Santé et en Sécurité du Travail, Montréal, QC, Canada), Frédéric Laville (Département de Génie Mécanique, École de Technologie Supérieure, Montréal, QC, Canada), and Jérôme Boutin (Groupe d'Acoustique de l'Universite de Sherbrooke, Université de Sherbrooke, Montréal, QC, Canada)

Hearing protection devices (HPD), such as earmuffs, are widely used to protect workers from noisy environments. Numerical predictive tools can be used to simulate the vibroacoustic behavior of earmuffs and thus assess their sound attenuation and improve their acoustical design. The present work describes the implementation of a vibroacoustic finite element numerical model of an earmuff coupled to a rigid baffle in the frequency range from 20 to 5000 Hz. An experimental assessment of the sound transfer paths through each element of the earmuff (cup, cushion, and foam lining) using a specific acoustical test bench is first proposed. This analysis is then used to target the right level of model complexity for each component. An experimental validation of the FEM model is then carried out.

MONDAY MORNING, 3 JUNE 2013 — 519B, 8:55 A.M. TO 11:20 A.M.

Session 1aPAa

Physical Acoustics and Biomedical Acoustics: Acoustics in Microfluidics and for Particle Separation I: Standing Waves, Streaming, and Radiation Forces

Lawrence A. Crum, Cochair
Appl. Phys. Lab., Univ. of Washington, Ctr. for Industrial and Medical Ultrasound, Seattle, WA 98105

Michel Versluis, Cochair
Phys. of Fluids Group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands

Chair's Introduction—8:55

Invited Papers

9:00

1aPAa1. First-principle simulation of the acoustic radiation force on microparticles in ultrasonic standing waves. Mads Jakob Herring Jensen (COMSOL A/S, Diplomvej 373, Lyngby 2800, Denmark, mads@comsol.dk) and Henrik Bruus (Dept. of Phys., Techn. Univ. of Denmark, Lyngby, Denmark)

The recent development in the field of microparticle acoustophoresis in microsystems has led to an increased need for more accurate theoretical predictions for the acoustic radiation force on a single microparticle in an ultrasonic standing wave. Increasingly detailed analytical solutions of this specific problem can be found in the literature [Settnes and Bruus, Phys. Rev. E **85**, 016327 (2012), and references therein], but none have included the complete contribution from thermoviscous effects. Here, we solve this problem numerically by applying a finite-element method to solve directly the mass (continuity), momentum (Navier-Stokes), and energy conservation equations using perturbation theory to second order in the imposed time-harmonic ultrasound field. In a two-stage calculation, we first solve the first-order equations resolving the thermoviscous boundary layer surrounding the microparticle and with a perfectly matched layer as a non-reflecting boundary condition for the scattered waves. These first-order solutions are then used as source-terms for solving the time-averaged second-order equations [Muller *et al*., Lab Chip **12**, 4617 (2012)] and in particular to determine the second-order time-averaged hydrodynamic stress on the particle surface. From this, we deduce the radiation force and compare it as a function of the physical parameters to existing analytical results.

9:20

1aPAa2. Acoustic standing wave based microsystem for low-concentration oil detection and separation. Han Wang (Dept. of Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), Zhongzheng Liu (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX), Chiwan Koo (Dept. of Biomed. Eng., Texas A&M Univ., College Station, TX), Sungman Kim (Dept. of Elec.and Comput. Eng., Texas A&M Univ., College Station, TX), Younghak Cho (Dept. of Mech. Syst. Design Eng., Seoul National Univ. of Sci. and Technol., Seoul, Republic of Korea), Yong-Joe Kim (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX), and Arum Han (Dept. of Elec. and Comput. Eng., Texas A&M Univ., 309C WERC, TAMU 3128, College Station, TX 77843-3128, arum.han@ece.tamu.edu)

Detection and quantification of extremely small amount of oil on site and at low cost has broad applications in environmental monitoring, both in oil spills as well as in routine marine/coastal ecosystem monitoring. For example, dispersed oil, generated through the use of chemical dispersants in oil spills to break up oil slick into small droplets so that they can be rapidly diluted in 3D space, are the greatest concern and poses the most challenges in detection. Fluorometry is the current standard method, however is bulky and expensive, limiting its wide deployment in the field. Here we demonstrate for the first time the development of an acoustic standing wave based microfluidic platform capable of processing large amount of liquid samples from which dispersed oil can be concentrated and separated to a detectable level by acoustophoretic force. The microfluidic platform consists of a recirculation channel structure into which dispersed oil droplets can be continuously separated from the main sample flow stream. A piezoelectric transducer attached at the bottom of the silicon-glass microfluidic channel creates the acoustic standing wave that exerts acoustophoretic force to oil droplets. An optical detector measures the presence of concentrated oil droplets by their distinct fluorescent signatures.

Contributed Papers

9:40

1aPAa3. Large volume flow rate acoustophoretic phase separator for oil water emulsion splitting. Jason P. Dionne (FloDesign Sonics Inc., 380 Main St., Wilbraham, MA 01095, j.dionne@fdsonics.com), Brian McCarthy, Ben Ross-Johnsrud (Mech. Eng., Western New England Univ., Springfield, MA), Louis Masi (FloDesign Sonics Inc., Wilbraham, MA), and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Efficient separation technologies for multi-component liquid streams that eliminate waste and reduce energy consumption are needed. Current technologies suffer from high cost of energy, use of consumables, fouling, and limited separation efficiency of micron-sized particles. We propose a novel platform technology consisting of a large volume flow rate acoustophoretic phase separator based on ultrasonic standing waves. The acoustic resonator is designed to create a high intensity three dimensional ultrasonic standing wave resulting in an acoustic radiation force that is larger than the combined effects of fluid drag and buoyancy, and is therefore able to trap, i.e., hold stationary, the suspended phase. The action of the acoustic forces on the trapped particles results in concentration, agglomeration, and/or coalescence of particles and droplets. Heavier than water particles are separated through enhanced gravitational settling, and lighter particles through enhanced buoyancy. A first prototype consists of a 2 in. by 1 in. flow chamber driven by a single 1 in. by 1 in. transducer at 2 MHz, with flow rates of 30 L/h, and measured oil separation efficiencies in excess of 95%. A second prototype is designed to further scale the system to flow rates of 150 L/h. [Work supported by NSF SBIR 1215021 and NSF PFI:BIC 1237723.]

10:00

1aPAa4. Acoustic radiation force on a sphere without restriction to axisymmetric fields. Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Lab., The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, hamilton@mail.utexas.edu)

The analysis presented at the previous ASA meeting related to investigation of the acoustic radiation force on a sphere embedded in a soft elastic medium with shear modulus that is several orders of magnitude smaller than its bulk modulus. The acoustic field was assumed to be axisymmetric and the spherical scatterer to be located on the axis of the acoustic beam. When one of these conditions is violated, the problem loses its symmetry. In this talk, the acoustic radiation force is considered in the more general case of nonaxisymmetric fields. The calculation is performed in Lagrangian coordinates. All acoustic fields, incident as well as scattered, depend on all three spherical coordinates. The incident and scattered waves, which include both potential and solenoidal parts, are expanded with respect to spherical harmonics. An analytical expression for the acoustic radiation force derived in this investigation may contain as many spherical harmonics as needed. In limiting cases when the scatterer is in liquid and only two modes, monopole and dipole, remain in the scattered fields, the solution for the acoustic radiation force recovers the results reported by Gor'kov [Sov. Phys. Doklady **6**, 773 (1962)]. [Work supported by NIH DK070618 and EB011603.]

10:20

1aPAa5. Viscous contributions to low-frequency scattering, power absorption, radiation force, and radiation torque for spheres in acoustic beams. Philip L. Marston (Phys. and Astron. Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

An analysis of the dipole response of solid spheres illuminated by plane acoustic traveling waves [Settnes and Bruus, Phys. Rev. E **85**, 016327 (2012)] has implications for estimating the magnitude of viscous corrections to quantities of broader interest. Their results may be recast to give the viscous correction to the dipole scattering s-function for solid spheres. For the present discussion, it may be assumed that the Stokes layer is thin relative to the sphere's radius, giving a simple reduction in magnitude of the dipole s-function (which is unimodular in the lossless case). The power absorption for plane waves and Bessel beams follow immediately from a prior formulation [Zhang and Marston, Phys. Rev. E **84**, 035601 (2011)] as does the axial radiation force. The plane-wave force agrees after correcting a minor error in Settnes and Bruus. A condition $G = 0$ previously noted for low-ka negative radiation forces on spheres in Bessel beams [Marston, J. Acoust. Soc. Am. **120**, 3518–3524 (2006)] still gives negative forces for sufficiently large spheres. The torque caused by first-order vortex beams may also be estimated [Zhang and Marston, Phys. Rev. E **84**, 065601 (2011)]. [Work supported by ONR.]

10:40

1aPAa6. Direct numerical simulations of acoustic streaming in standing wave tubes using the Lattice Boltzmann Method. Yasser Rafat, Kaveh Habibi, and Luc Mongeau (Mech. Eng., Mcgill Univ., 817 Sherbrooke St. West, Montreal, QC, Canada, yasser.rafat@mail.mcgill.ca)

One important factor in the efficiency of thermoacoustic engines is acoustic streaming, which causes convective heat transfer between high and low temperature reservoirs. Most experimental and numerical studies performed so far have focused on Rayleigh streaming. Less work has been done on acoustic streaming due to the stack. Most numerical studies of Rayleigh streaming were performed using Navier-Stokes based numerical methods. In this study, large eddy numerical simulations were performed using schemes based on the lattice Boltzmann method (LBM). The model considered a simplified thermoacoustic refrigerator made of a rectangular standing wave resonator with a flat plate spoiler. Low-amplitude results obtained for Rayleigh streaming velocity magnitudes were compared with linear acoustic theory for verification. High amplitude recirculated streaming flow structures around the edges of the flat plate spoiler were identified. These are likely to contribute to heat transfer much more than Rayleigh streaming. Parametric studies were performed to investigate the effects of Strouhal number and spoiler edge shape. The results confirm that vertical edge streaming flows play a significant role in thermoacoustic heat transport.

11:00

1aPAa7. Three-dimensional analysis of the acoustic radiation pressure: Application to single-beam acoustical tweezers. Diego Baresch, Régis Marchiano (Institut Jean le Rond d'Alembert, UMR CNRS 7190, UPMC-CNRS, 4, Place Jussieu, Paris 75005, France, diego.baresch@upmc.fr), and Jean-Louis Thomas (Institut des NanoSciences de Paris, UMR CNRS 7588, UPMC-CNRS, Paris, France)

Recent studies on the acoustic radiation forces exerted by sound impinging spherical objects suggest the use of structured wavefronts for particle entrapment and controlled manipulation. In the scope of understanding why it is made possible to trap and manipulate small particles with sound, we present a general model for the acoustic radiation forces in three dimensions. A first generalization comes from the extension of well known results for the radiation pressure of plane waves to incident wavefields having arbitrary wavefronts. Second, the elastic spherical target of any dimension is allowed to be arbitrarily located within the wavefield. Introducing a new class of "single-beam" acoustical tweezers, we discuss the capabilities of different acoustical beams to achieve particle trapping and manipulation tasks. In addition, using an efficient experimental setup, we report the propagation of a peculiar beam carrying orbital angular momentum, namely an acoustical vortex, which is our selected candidate to achieve the first three-dimensional acoustic trap for elastic particles.

MONDAY MORNING, 3 JUNE 2013 — 519B, 11:20 A.M. TO 12:00 NOON

Session 1aPAb

Physical Acoustics and Biomedical Acoustics: Acoustics in Microfluidics and for Particle Separation IIa: Bubbles and Drops

Yong-Jae Kim, Cochair
Mech Eng., Texas A&M Univ., College Station, Texas 77843

Martin Wiklund, Cochair
Appl. Phys., Royal Inst. of Technol., KTH-Albanova, Stockholm 10691, Sweden

Invited Papers

11:20

1aPAb1. Perturbation analysis of flow about spherically pulsating bubble at the velocity node of a standing wave. Mohammad AlHamli (Aerosp. & Mech. Eng., Univ. of Southern California, Olin Hall OHE 430 (MC-1453), Los Angeles, CA 90089-1453, moalhamli@gmail.com), Alexey Y. Rednikov (TIPs-Fluid Phys., Université Libre de Bruxelles, Brussels, Belgium), and Satwindar S. Sadhal (Aerosp. & Mech. Eng., Univ. of Southern California, Los Angeles, CA)

An analysis using the singular perturbation method for a radially pulsating gas bubble at the velocity node of a standing wave was conducted with $\varepsilon = U_0/(a\omega) \ll 1$ as a small parameter and $\omega a^2/\nu \gg 1$ as a large parameter. Here,a, U_0, ω, and ν are length scale, velocity scale, frequency, and kinematic viscosity, respectively. While the mean oscillatory flow around the gas bubble has no net time-averaged flow component, viscous steady streaming arises due to the nonlinearity of the flow dynamics. However, with bubble surface being considered shear-free, the vorticity generation in the system is quite weak as compared with what would result from a solid boundary. Not surprisingly, the steady streaming is also weak. As already known, the steady streaming would not arise with purely radial pulsations of a bubble in an otherwise quiescent liquid. For the case of a non-pulsating bubble at the velocity node, streaming is seen at $O(\varepsilon^2)$. However, as seen with the case of a radially pulsating bubble at the velocity antinode, interaction of two oscillatory fields creates streaming at lower order. The phase difference between radial and lateral oscillations was found to play a significant role in both the streaming direction and intensity.

11:40

1aPAb2. Acoustic bubble sorting of ultrasound contrast agents. Michel Versluis (Phys. of Fluids Group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl)

Ultrasound contrast agents are coated microbubbles with radii ranging from 1 to 10 μm. Medical transducers typically operate at a single frequency; therefore, only a small selection of bubbles will contribute to the nonlinear contrast through resonance. Thus, the sensitivity of contrast-enhanced ultrasound can be improved by narrowing down the size distribution. Monodisperse bubble can be formed in a flow-focusing geometry. However, it requires extensive skills in microfluidics technology and in surface chemistry. Here, we present a simple lab-on-a-

chip technique to sort microbubbles on-line in a traveling ultrasound wave. A broad range of the parameter space of bubble size and frequency has been characterized to provide physical input parameters for a simple force balance model. We find good agreement for the modeled displacement as a function of the bubble radius for a range of sizes in the unbounded fluid. Within the confinement of the sorting chip, we find good agreement for the resonance behavior and overall with a smaller displacement than predicted as a result of bubble–wall interactions.This novel sorting strategy may lead to an overall improvement of the sensitivity of contrast echo of at least one order of magnitude.

MONDAY MORNING, 3 JUNE 2013 514ABC, 8:55 A.M. TO 12:00 NOON

Session 1aPP

Psychological and Physiological Acoustics: In Memory of Bertram Scharf: Five Decades of Contributions to Auditory Perception

Mary Florentine, Cochair
SLPA & ECE, Northeastern Univ., 106-A FR, 360 Hutntington Ave., Boston, MA 02115

Huanping Dai, Cochair
Speech Lang. and Hearing Sci., Univ. of Arizona, P.O. Box 21071, 1131 E. 22nd St., Tucson, AZ 85721-0071

Chair's Introduction—8:55

Invited Papers

9:00

1aPP1. Bertram Scharf and his critical contributions to the field. Harry Levitt (Adv. Hearing Concepts, P.O. Box 610, Bodega Bay, CA 94923, harrylevitt@earthlink.net)

Jerry Tobias introduced me to Bert Scharf via his book Foundations of Modern Auditory Theory. Our research in the 1970s was on the prediction of speech intelligibility from acoustic measurements; the similarity between Fletcher's 20 frequency bands of equal contributions to intelligibility and psychoacoustic measurements of critical bands was of great interest. Bert's chapter on critical bands provided an insightful, comprehensive, concise, and critical review of the state of the art in critical-band research. Our mutual research interests brought us together when he asked me to review a draft of a paper he was preparing. My review was highly critical and after submitting it to him I felt I had been too unforgiving in my review (reviewers can be wrong). I called him to explain that I had been overly critical of a fine paper. His response was the opposite of what I expected. He said it was the best review he had received and that he was extremely grateful for my input. I realized then that he—not only set high standards for others—but for himself as well. By adhering consistently to his high standards, his many contributions to the field have been long-lasting and critical.

9:20

1aPP2. An overview of Bertram Scharf's research in France on loudness adaptation. Sabine Meunier (LMA-CNRS-UPR 7051, Aix-Marseille Univ, Centrale Marseille, 31 chemin Joseph-Aiguier, Marseille 13402, France, meunier@lma.cnrs-mrs.fr)

Since 1978, Professor Bertram Scharf divided his time between the United States and France. He was a Visiting Scientist at the Laboratoire de Mécanique et d'Acoustique in Marseille until the mid-1990s and collaborated with the University of Marseille (Faculté de Médecine) until his death. One of Bertram Scharf's major contributions to the field of psychoacoustics is in the area of loudness. He first studied spectral loudness summation, when he started working at Harvard University. In France, his work on loudness focused mainly on loudness adaptation. He wrote, "Loudness resembles pain in that it decreases as a function of time only under special stimulus conditions." Bertram Scharf's work with his French colleagues defined aspects of loudness adaptation in its direct (simple loudness adaptation) and indirect (induced loudness adaptation) forms. They studied how the auditory system recovers from loudness adaptation and examined a possible physiological basis for loudness adaptation.

9:40

1aPP3. Spectral loudness summation: From the 1960s to the present. Jesko L. Verhey and Jan Hots (Dept. of Exp. Audiol., Otto von Guericke Univ. Madgeburg, Leipziger Str. 44, Magdeburg 39120, Germany, jesko.verhey@med.ovgu.de)

In general, the loudness level of a broader sound is higher than that of a narrower sound centered at the same frequency, an effect commonly referred to as spectral loudness summation. From the late 1950s onwards, Scharf published several articles on this topic investigating how stimulus parameters such as the level, number, and spectral separation of components of a complex tone and the spectrum shape affects the magnitude of spectral loudness summation, and how spectral loudness summation is altered in hearing-impaired listeners and under masking. In a contribution to the proceedings of the first international symposium on hearing, Scharf also provided important information of the effect of duration on spectral loudness summation, stimulating our own research in this field. This talk will provide an overview of Scharf's work on spectral loudness summation and how the view on this topic has changed over time. It will be shown that even today there are aspects of this effect that are not completely understood in the light of current loudness models.

10:00–10:20 Break

10:20

1aPP4. Context effects in loudness. Michael Epstein (Auditory Modeling and Processing Lab., Speech-Lang. Pathol. and Audiol., Northeastern Univ., 360 Huntington Ave., 106A FR, Boston, MA 02115, m.epstein@neu.edu) and Mary Florentine (Commun. and Digital Signal Processing Ctr., Speech-Lang. Pathol. and Audiol., Northeastern Univ., Boston, MA)

Bertram Scharf made contributions to numerous topics in the loudness literature. In particular, he brought a great deal of insight into the current understanding of contextual effects in loudness. Some of the contextual effects that he studied include: (1) loudness adaptation, the decline in loudness of the latter portion of a continuous sound, (2) induced loudness reduction, the phenomenon by which a preceding stronger tone reduces the loudness of a weaker tone, (3) temporary loudness shift, a decline in the loudness of weaker sounds due to a physical fatigue of the cochlear amplifier, and (4) loudness enhancement, in which a brief sound is made louder when it follows a stronger sound within a short duration. Context effects serve as complex reminders of the necessity of careful design of any psychoacoustical experiment in which level varies. These effects also result in the breakdown of all loudness models, as virtually all calculations of loudness are performed for sounds without regard for previous stimuli. [Work supported by NIH-NIDCD.]

10:40

1aPP5. Connecting cues to signals in auditory attention. Ervin Hafter (Psychology, Univ. of California, 1854 San Lorenzo Ave., Berkeley, CA 94707, hafter@berkeley.edu)

Among his many fields of study, Bert Scharf played a major role in our understanding the role of auditory attention, especially at the level of basic psychophysics. Of special importance to this talk is the profound influence that he had on research in our lab (myself, Bert Schlauch, Joyce Tang, Kourosh Saberi, and Poppy Crum) through his work on signal uncertainty in masking and its alleviation by specific informational cues. Scharf's resurrection of the probe-signal method led us to examine the effects of uncertainty on both the means and variances of effective bandwidths used by listeners in a detection task. Also shown will be cases where we used different kinds of cues to study detection at various levels of processing including judgments based on: specific spectral components, complex pitches derived from sets of harmonics and locations in frequency reliant on mentally tracking an FM stimulus through a period of occlusion.

11:00

1aPP6. Neural correlates of auditory attention. Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

Bert Scharf's seminal studies on selective auditory attention were, in many ways, ahead of the times. Twenty years ago, many psychoacousticians viewed any consideration of "cognitive" factors or any effects driven by the intent of the listener, rather than the acoustics of the input, as outside of their realm of interest. However, today, a plethora of laboratories are exploring questions about what acoustic features enable listeners to focus attention, how bottom-up stimulus attributes interact with top-down control signals to determine what source a listener attends in a mixture of sources, and what neural mechanisms realize such selective auditory attention. This talk reviews recent work exploring selective auditory attention using a combination of behavioral studies and neuro-imaging techniques, all of which suggest that (1) listeners can focus attention on one, and only one, auditory object or stream at a time, and (2) executive control regions of the brain are engaged during attention to reduce across-object interference in the representation of whatever object is in the attentional foreground. These studies underscore the importance of auditory attention in allowing us to communicate in everyday settings containing multiple sound sources, and thus the foresight of Bert in tackling this problem when most others did not.

11:20

1aPP7. Tuning in the time domain revealed through detection of auditory signals of unexpected duration or presentation time. Beverly Wright (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, b-wright@northwestern.edu) and Huanping Dai (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Bert Scharf was interested in how expectation affects auditory performance. He explored this question using the probe-signal method of Greenberg and Larkin, in which the listener is led to expect a particular stimulus but is occasionally presented with an unexpected but equally detectable one. The detectability of unexpected stimuli provides insight into the listener's template for the expected stimulus. Bert's expectation research, which focused on the frequency domain, inspired us to extend the inquiry to the time domain. We have seen that signal detection can be quite poor for signals of unexpected duration as well as for signals presented at unexpected times, indicating that listeners attend selectively to these two temporal aspects of sound. However, this temporal tuning is much broader for starting time (hundreds of milliseconds) than for signal duration (can be $<$25 ms). Thus, it appears that listeners can select the template for signal detection with considerable accuracy, but do not apply the selected template strictly to the expected starting time of the signal. We are grateful to Bert for his mentorship and keen interest in this topic. [Work supported by NIH/NIDCD.]

11:40

1aPP8. Inspiration from Bertram Scharf's work. Fan-Gang Zeng (Ctr. for Hearing Res., Depts. of Anatomy and Neurobiology, Biomedical Eng., Cognit. Sci. and Otolaryngol.– Head and Neck Surgery, Univ. of California, 110 Med Sci E, Irvine, CA 92697, fzeng@uci.edu)

In 1990, Bertram Scharf and I discussed about me doing a post doc in his laboratory. The opportunity to work with Bertram did not materialize, but his work in loudness, efferents, and attention has been a continuing inspiration not only for my own research but also for auditory perception, physiology, and audio engineering in general. For example, a Google Scholar search of "Bertram Scharf" on November 8, 2012, produced 1517 citations for his top 10 papers, with 6 on loudness and critical bands, 2 on efferents, and 2 on attention. Here I highlight two recent projects that have been inspired by Scharf's work. The first project showed significant loudness adaptation in patients with auditory neuropathy, particularly those with otoferlin deficits. This result directly supports Scharf's proposition that simple loudness adaptation is due to a sensory process, which in this case can be pinned down to transmitter release and replenishment in the hair cell and nerve synapse. The second project extended Scharf's theoretical work in efferents and attention to improving feedback control in cochlear implant users and tinnitus sufferers. This line of work could improve cochlear implant speech perception in noise and reduce internal gain to alleviate tinnitus.

Session 1aSA

Structural Acoustics and Vibration: Measurement and Modeling of Structures with Attached Noise Control Materials I

Franck C. Sgard, Cochair
IRSST, 505 Boulvd de Maisonneuve O, Montreal, QC H3A3C2, Canada

Noureddine Atalla, Cochair
GAUS Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC J1K 2R1, Canada

Chair's Introduction—8:55

Invited Papers

9:00

1aSA1. Tunneling effect on the sound transmission loss of a flat structure coupled with a porous material. Franck C. Sgard (Direction Scientifique, IRSST, 505 Boulvd de Maisonneuve O, Montreal, QC H3A3C2, Canada, frasga@irsst.qc.ca), Noureddine Atalla, Mohammad Gholami (GAUS, Univ. of Sherbrooke, Montreal, QC, Canada), and Hugues Nelisse (Direction Scientifique, IRSST, Montreal, QC, Canada)

It is well known that when measuring sound transmission loss (STL) in a laboratory, among all test conditions, the location of a specimen in an aperture affects the results, due to the tunneling effect. Previous studies have considered this effect for flat single panels and double walls but the case of a panel with attached sound package seems to have received very little attention. This paper deals with the application of a modal approach to study the STL of a rectangular plate coupled with a porous material located inside a tunnel. The sound absorbing material is supposed to be described by a transfer matrix calculated using a Transfer Matrix Method, which relates interstitial pressure and total normal stress on both sides of the material. The model is validated by comparison with finite element/boundary element computations. Numerical results are shown to illustrate the validity of the proposed hybrid modal-TMM methodology and its use to investigate the niche effect in the presence of a sound absorbing material.

9:20

1aSA2. A hybrid modeling approach for vibroacoustic systems with attached sound packages. Luca Alimonti, Noureddine Atalla, Alain Berry (Mech. Eng., Universitè de Sherbrooke, 1747 Rue Marcil, Sherbrooke, QC J1J 2H7, Canada, luca.alimonti@usherbrooke.ca), and Franck Sgard (IRSST, Montreal, QC, Canada)

Modeling complex vibroacoustic systems including poroelastic materials using finite element (FE) based methods can be computationally expensive. Several attempts have been made to alleviate this drawback, such as high order hierarchical basis and substructuring approaches. Still, these methods remain computationally expensive or limited to simple configurations. On the other hand, analytical approaches, such as the Transfer Matrix Method (TMM), are often used, thanks to the lower computational burden. However, since the geometrical flexibility of the FE method is always needed in the low/mid-frequency range, attempts have been made to couple the FE model of the master system with a TM model of the sound package. Although these hybrid approaches seem promising, the open literature is not comprehensive. The aim of this work is to present a hybrid FE-TMM approach based on a Green's function formulation. The idea is to account for the sound package by approximating the effects over the treated surface using fundamental solutions (i.e., Green's functions) obtained by the TMM. A benchmark representative of typical applications is used to illustrate the capabilities of the presented methodology in terms of efficiency and accuracy in comparison to other classical methods.

9:40

1aSA3. A finite element solution strategy based on Padé approximants for fast multiple frequency sweeps of coupled elastic, poroelastic, and internal acoustic problems. Romain Rumpler and Peter Göransson (Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., Teknikringen 8, Stockholm 10044, Sweden, rumpler@kth.se)

Analyses involving structural-acoustic finite element models including three-dimensional modeling of porous media are, in general, computationally costly. While being the most commonly used predictive tool in the context of noise and vibrations reduction, efficient solution strategies enabling the handling of large-size multiphysics industrial problems are still lacking, particularly in the context where multiple frequency response estimations are required, e.g., for topology optimization, multiple load cases analysis, etc. In this work, an original solution strategy is presented for the solution of multi-frequency structural-acoustic problems including poroelastic damping. Based on the use of Padé approximants, very accurate interpolations of multiple frequency sweeps are performed, allowing for substantial improvements in terms of computational resources, i.e., time and memory allocation. The method is validated and demonstrated for its potential on 3D applications involving coupled elastic, poroelastic, and internal acoustic domains.

10:00

1aSA4. Discontinuous Galerkin Methods for poroelastic materials. Olivier Dazel (LAUM UMR CNRS 6613, University Le Mans, Avenue Olivier Messiaen, Le Mans Cedex F-72085, France, olivier.dazel@univ-lemans.fr) and Gwenael Gabard (ISVR, Univ. of Southampton, Southampton, United Kingdom)

In this work, we are interested in the development of a Discontinuous Galerkin Method (DGM) for sound absorbing materials. These materials are commonly used for noise and vibration control. The objective of this method is to discretize the structure and to represent in each element the field as a superposition of local solutions such as plane waves. This type of methods have shown their efficiency by requiring much smaller numbers of degrees of freedom compared to standard polynomial interpolations (i.e., FEM) especially when the frequency is increased. For poroelastic materials, the solutions are expressed in terms of Biot waves (two of them associated to compression waves and one corresponding to shear waves). The poroelastic problem is expressed as a first order model and the formulation of numerical flux at interfaces between elements is derived and implemented. Compared to classical DG methods for standard acoustics, this method is applied here to a dissipative medium and to a two-displacement field involving shear and compression waves. Several two-dimensional cases will be presented in order to validate the method and to compare against analytical and finite-element solutions (in displacement and mixed formulation). Results will be discussed in terms of accuracy of the method, errors, and conditioning of the linear systems.

10:20

1aSA5. Numerical simulation of acoustic waves in air and poroelastic media using the partition of unity finite element method. Jean-Daniel Chazot (Roberval UMR7337, Universite de Technologie de Compiegne, Rue Personne de Roberval, Compiegne 60200, France, jean-daniel.chazot@utc.fr), Benoit Nennig (LISMMA, SUPMECA Paris, Saint-Ouen, France), and Emmanuel Perrey-Debain (Roberval UMR7337, Universite de Technologie de Compiegne, Compiegne, France)

Foams and fibrous materials are used in a large range of applications such as automotive or building acoustics. Their properties can be described with either a poroelastic model or an equivalent fluid model. These models, also used in geophysics, are now widely spread and are also available in some commercial finite element software applications. However, the discretization level required to achieve reasonable accuracy is not always acceptable in the mid-frequency range. In such case, the Partition of Unity Finite Element Method (PUFEM) using plane wave functions seems appropriate to avoid this limitation. The PUFEM has been recently applied to rigid frame materials with and without coupling with an acoustic cavity. It has also been applied efficiently to poroelastic materials. The present work focuses on the coupling between a poroelastic material, i.e., described with Biot's equations, and an air cavity. Some practical examples are tested to demonstrate the efficiency of the PUFEM for solving noise control problems at medium frequency, but also to underline the precautions that must be taken when dealing with an air-porous interface.

10:40

1aSA6. On internal mean flow in porous absorbers and its effect on attenuation properties. Susann Boij (Marcus Wallenberg Lab. for Sound and Vib. Res., Dept. of Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., 2004 Yolo Ave., Berkeley, California 94707, sboij@kth.se), Anna Färm (Scania CV AB, Stockholm, Sweden), Peter Göransson (Marcus Wallenberg Lab. for Sound and Vib. Res., Dept. of Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., Stockholm, Sweden), and Ragnar Glav (Scania CV AB, Stockholm, Sweden)

In vehicle applications, absorbing materials are often used to attenuate sound. In, for example, exhaust systems and on noise encapsulations, the absorber is exposed to flow. This creates a boundary layer above the absorber, which affects the impedance of the surface, and hence alters the absorption properties. In addition to this effect, the flow itself may enter the absorbent material due to high pressure and forced flow paths. An investigation of the effects that internal flow in the absorber imposes on the acoustic properties is presented. One way to describe the effect is by a change in flow resistivity. The effect is investigated for typical absorbers used in noise encapsulation for trucks. The Transfer Matrix Method is applied to calculate the resulting absorption and reflection coefficient for absorbers with changed flow resistivity in layers at the surface. The possibility to model the changed properties of the absorber with internal mean flow by means of Biot theory is also explored, together with a discussion on suitable experimental methods to verify and further investigate the effects.

11:00

1aSA7. Investigating the transmission loss effect via optimizing the insulator package on vehicle's firewall. Sajjad Beigmoradi (Automotive Eng. Dept., Iran Univ. of Sci. & Technol., No. 13, Emmami Alley, Golzarand Alley, Safdari St. Navab Safavi St, Tehran, Iran, s.beigmorady@gmail.com), Kambiz Jahani, and Hassan Hajabdollahi (Mech. Eng. Dept., Iran Univ.of Sci. & Technol., Tehran, Islamic Republic of Iran)

Nowadays, noise and vibration of attributes of motor vehicles have a dominant effect on customers' judgment about the cars, and hence, car manufacturers and OEM suppliers have dedicated remarkable time, budget, and concern to the investigations in this field. From NVH perspective, firewall is the foremost structural member in vehicle body design, since it is the main pass for transferring the engine induced noise to the passenger cabin. In this research, effect of the insulator package is studied through different configurations considering the radiated noise level and the optimized design is proposed using an optimization procedure. Indeed, it is concluded that adding the optimized insulator package can significantly refine the noise transmissibility while avoiding structural modifications in the firewall.

11:20

1aSA8. A method for measuring the acoustic properties of a porous sample mounted in a rigid ring in acoustic tubes. Thomas Dupont, Philippe Leclaire (Drive, ISAT, Université de Bourgogne, BP 31 - 49 rue Mlle Bourgeois, Nevers 58027, France, thomas.dupont@u-bourgogne.fr), Raymond Panneton, Kévin Verdière (Gaus, Université de Sherbrooke, Sherbrooke, QC, Canada), and Saïd Elkoun (Gaus, Université de Sherbrooke, Sherbrooke, Alberta, Canada)

This study presents a method to measure the acoustic properties of a homogeneous porous material with a support or a reduction element in an acoustic tube. Some materials tested have a lateral size much smaller than the tube's diameter, as they cannot be produced in the correct dimensions without corrupting the material; this also permits the testing of the same samples in a large frequency bandwidth by using different section tubes. Moreover, the acoustic leaks on the material boundaries can significantly change the transmission loss measured in tubes. To rectify these problems, rings can be placed on each material surface. The presence of these rings can influence the acoustic indicator measurement; while this effect is negligible for tubes with a large cross section, it is not for tubes with a small cross section. To correct, or remove, the influence of the rings, we propose to use an application of the parallel assembly process of the transfer matrix method, which has recently been proposed by Panneton *et al*. [*Proceeding Internoise*, New York (2012)]. Measurements on classical porous materials with and without reductions are proposed and compared to simulated results. The ring's effects and the proposed corrections are discussed for different materials.

11:40

1aSA9. Acoustic characterization of graded porous materials under the rigid frame approximation. Jean-Philippe Groby, Olivier Dazel (Laboratoire d'Acoustique de l'Universit, Le Mans, France), Laurent De Ryck (LMS Int., Leuven, Belgium), Amir Khan, and Kirill Horoshenkov (School of Eng., Univ. of Bradford, Great Horton Rd., Bradford BD7 1DP, United Kingdom, a.khan117@bradford.ac.uk)

Graded porous materials are of growing interest because of their ability to improve the impedance matching between air and material itself. Theoretical models have been developed to predict the acoustical properties of these media. Traditionally, graded materials have been manufactured by stacking a discrete number of homogeneous porous layers with different pore microstructure. More recently, a novel foaming process for the manufacturing of porous materials with continuous pore stratification has been developed. This paper reports on the application of the numerical procedure proposed by De Ryck to invert the parameters of the pore size distribution from the impedance tube measurements for materials with continuously stratified pore microstructure. Specifically, this reconstruction procedure has been successfully applied to retrieve the flow resistivity and tortuosity profiles of graded porous materials manufactured with the method proposed by Mahasaranon *et al*. In this work, the porosity and standard deviation in pore size are assumed constant and measured using methods, which are applied routinely for homogeneous materials characterization. The numerical method is based on the wave splitting together with the transmission Green's functions approach, yielding an analytical expression of the objective function in the least-square sense. The objective function is constructed to minimize the discrepancy between the predicted and measured reflection coefficient spectra.

MONDAY MORNING, 3 JUNE 2013 — 515ABC, 8:55 A.M. TO 11:40 A.M.

Session 1aSCa

Speech Communication: Distinguishing Between Science and Pseudoscience in Forensic Acoustics I

Geoffrey Stewart Morrison, Cochair
Forensic Voice Comparison Lab., School of Elec. Eng. & Telecommunications, Univ. of New South Wales, NSW 2052, Australia

James Harnsberger, Cochair
Univ. of Florida, 402 NW 24th St., Gainesville, FL 32607

Chair's Introduction—8:55

Invited Papers

9:00

1aSCa1. Distinguishing between science and pseudoscience in forensic acoustics. Geoffrey Stewart Morrison (Forensic Voice Comparison Lab., School of Elec. Eng. & Telecommunications, Univ. of New South Wales,NSW 2052, Australia, geoff-morrison@forensic-voice-comparison.net)

In this presentation, I argue that one should not attempt to directly assess whether a forensic analysis technique is scientifically acceptable. Rather one should first specify what one considers to be appropriate scientific principles governing acceptable practice, then consider any particular approach in light of those principles. I focus on one principle: The validity and reliability of an approach should be empirically tested under conditions reflecting those of the case under investigation using test data taken from the relevant population. Versions of this principle have been key elements in several reports on forensic science, including forensic voice comparison, published over the last four-and-a-half decades. I consider the aural-spectrographic approach to forensic voice comparison (also known as "voiceprint" or "voicegram" examination) in light of this principle, and also the currently widely practiced auditory-acoustic-phonetic approach (these two approaches do not appear to be mutually exclusive). Finally, I challenge the audience members to consider what each of them thinks constitutes the relevant scientific principles regarding acceptable practice, and then consider their own approach to forensic-acoustic analysis in light of those principles.

9:20

1aSCa2. A Canadian perspective on forensic science versus pseudoscience. Brent Ostrum (Sci. and Eng. Directorate, Canada Border Services Agency, 14 Colonnade Rd., Ste. 280, Ottawa, ON K2E 7M6, Canada, brent.ostrum@cbsa-asfc.gc.ca)

This presentation will provide my personal observations regarding forensic science versus pseudoscience in the Canadian legal system. I am neither a lawyer nor a judge; rather, I am a forensic scientist with over 25 years of experience in the Canadian system. My presentation focuses on relevant criteria for expert evidence considered in Canadian courts. The key ruling in R. v. Mohan (1994) provides

the start of the discussion with subsequent court rulings adding various elements. In Canada, we have had several judicial inquiries, such as the Kaufmann Commission, that can serve to guide experts. Select aspects of the 2009 NAS report "Strengthening Forensic Science in the United States: A Path Forward" will also be referenced. There are some common "criteria" often used by courts in different jurisdictions to assess expert evidence, including forensic acoustics. In other words, some basic expectations for all forms of expert evidence can be identified. I will attempt to show how select "sciences" have tried to fulfill those expectations. This will involve some commentary on issues of individual examiner competency, oversight at a system level (e.g., accreditation), and the need for proper and adequate method validation.

9:40–10:00 Break

10:00

1aSCa3. Voice stress analyses: Science and pseudoscience. Francisco Lacerda (Dept. of Linguist., Stockholm Univ., Universitetsvagen 10 C, Stockholm SE-106 91, Sweden, frasse@ling.su.se)

Voice stress analyses could be relevant tools to detect deception in many forensic and security contexts. However, today's commercial voice-based lie-detectors are not supported by convincing scientific evidence. In addition to the scientific implausibility of their working principles, the experimental evidence invoked by the sellers is either anecdotal or drawn from methodologically flawed experiments. Nevertheless, criminal investigators, authorities, and even some academics appear to be persuaded by the ungrounded claims of the aggressive propaganda from sellers of voice stress analysis gadgets, perhaps further enhanced by the portrays of "cutting-edge voice-analysis technology" in the entertainment industry. Clearly, because there is a serious threat to public justice and security if authorities adopt a naïve "open-minded" attitude toward sham lie-detection devices, this presentation will attempt to draw attention to plausibility and validity issues in connection with the claimed working principles of two commercial voice stress analyzers. The working principles will be discussed from a phonetics and speech analysis perspective and the processes that may lead naïve observers into interpreting as meaningful the spurious results generated by such commercial devices will be examined. Finally, the scope and limitations of using scientific phonetic analyses of voice to detect deception for forensic purposes will be discussed.

10:20

1aSCa4. Assessing acoustic features in the speech of asylum seekers. Judith K. Rosenhouse (Linguist. Unit, SWANTECH Ltd., 89 Hagalil St., Haifa 32684, Israel, swantech@013.net)

One of the areas of forensic linguistics concerns asylum seekers who speak languages which are foreign to the official language of the country where they apply for asylum. Identifying and verifying their real national background may be difficult if their speech manner reveals non-typical properties of their (real or alleged) native languages. Governments submit such asylum seekers' speech samples for linguistic analysis on various levels, including phonetic acoustics. This aspect of forensic linguistics raises questions about the scientific merit of such an analysis. Our aim is to examine some of the questions which relate to segmental and supra-segmental features that are analyzed acoustically based on recorded samples of asylum seekers' (alleged) native language and compared with the same features as known from the literature. We demonstrate such issues by examples from the speech of Arabic-speaking asylum seekers whose native tongue is (supposed to be) some local dialect but the recording includes various foreign features reflecting different dialects or languages. These questions involve sociolinguistic factors that affect individual speakers' speech production due to a complex and unstable life-history. We suggest that the acoustic methods currently used in speech analysis in this context could be considered pseudo-science in many cases.

10:40

1aSCa5. Analysis criteria for forensic musicology. Durand R. Begault, Heather D. Heise, and Christopher A. Peltier (Audio Forensic Ctr., Charles M. Salter Associates, Inc., 130 Sutter St., Floor 5, San Francisco, CA 94104, durand.begault@cmsalter.com)

Expert testimony for forensic musicology addresses a broad spectrum of legal issues, including the authentication and differentiation of published compositions and musical recordings, performance rights, and legal determinations regarding copyright infringement. While legal cases involving music and performance infringement date back as far as the 19th century, the field of forensic musicology has no stated methodology by which an objective forensic determination can be made. Expert opinions based merely on subjective impression or resulting from the "golden ear" syndrome are pseudo-scientific and not objectively based. This paper proposes scientific methods and recommendations for analysis based on stated criteria, with the goal of controlling examiner bias. Considerations include analyses of composition, performance, and acoustical features, and factors such as melody, harmony, rhythm, and orchestration; pitch, tone, vibrato, and embellishment; metadata analysis; recording technologies; and digital signal processing, including "effects." By engaging in a series of structured categorizations, the forensic expert can establish a consistent, replicable, and objectively verifiable means of determining whether or not a recorded piece of music has been misappropriated.

11:00–11:40 Panel Discussion

Session 1aSCb

Speech Communication: Digital Speech Processing (Poster Session)

Mark VanDam, Chair
Washington State Univ., P.O. Box 1495, Spokane, WA 99202

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

1aSCb1. Precision and error of automatic speech recognition. Mark VanDam (Speech and Hearing Sci., Washington State Univ., P.O. BOX 1495, Spokane, WA 99202, mark.vandam@wsu.edu) and Noah H. Silbert (Ctr. for the Adv. Study of Lang., Univ. of Maryland, College Park, MD)

Automatic speech recognition (ASR) software developed by the LENA Research Foundation (Boulder, CO) is an increasingly important tool in psycholinguistics. Naturalistic day-long recordings are segmented and assigned talker labels including those for *KEY CHILD, ADULT MALE*, and *ADULT FEMALE*. Performance of the system is a serious concern for ASR in general, not just the LENA system. Additional evidence of the software's performance is necessary to better interpret and understand accumulating research using this tool. Here we analyze the correspondence between computer and human segment labels corresponding to children, mothers, and fathers. Segments machine-labeled as *ADULT MALE, ADULT FEMALE*, and *KEY CHILD* were played to judges who identified each segment as *Mother, Father, Child*, or *Other*. Judges' responses were analyzed in terms of agreement, precision, and error. Overall agreement between machine and human coding was in the 70% range with Cohen's $\kappa > .55$. Machine coding appears to be better at coding *KEY CHILD* segments than the *ADULT* segments, and agreements for *ADULT MALE* labels were better than for *ADULT FEMALE* labels. The ASR system performed similarly when assigning segment labels for children and fathers, but less well for mothers. Overall error rates were generally very low. [Work support from NIH-NIDCD R01DC009560.]

1aSCb2. Unsupervised machine learning for the accurate classification of the discourse maker *like* in code-switching utterances. Page E. Piccinini (Linguistics, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0108, ppiccinini@ucsd.edu) and Eric R. Kramer (Medical Scientist Training Program, Univ. of California, San Diego, La Jolla, CA)

Spanish-English bilinguals use the discourse marker *like* in English, Spanish, and code-switching utterances. An acoustic analysis found that the [l] and diphthong in *like* is produced differently depending on the type of utterance in which it occurs. To investigate the possible perceptual relevance of these differences, we built a logistic-polynomial regression model to classify *like* tokens based on acoustic data. The model first projects F1 and F2 values onto a space of time-dependent polynomials. We then apply multinomial logistic regression to classify these polynomials as English, Spanish, or code-switching. The area under the curve was 0.75, showing classification was significantly greater than random. This model outperforms models that rely on static values for F1 and F2, either at the midpoint of the token (DeLong's test, $p < 0.004$), or two data points: the midpoint of the [a] of the diphthong, and the midpoint of the [ɪ] of the diphthong (DeLong's test, $p < 0.04$). The superiority of the polynomial model suggests that the time-dependent progression of F1 and F2 values, rather than absolute formant values, is important for predicting an imminent code-switch. We hypothesize that listeners leverage these time-dependent changes in F1 and F2 to anticipate code-switches.

1aSCb3. Using a computational model for the auditory midbrain to explore the neural representation of vowels. Laurel H. Carney, Jiashu Li, Tianhao Li (Biomedical Eng. and Neurobiol. & Anatomy, Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu), and Joyce M. McDonough (Linguistics, Univ. of Rochester, Rochester, NY)

A formant-based approach to representing vowel quality is anchored in acoustic theory and is well documented in perception studies and in auditory modeling. This ongoing study investigates the representation of vowels in the responses of auditory models at the level of the midbrain (inferior colliculus). Previous modeling and physiological results have shown that formant structure is conveyed by changes in neural rates of midbrain cells that are tuned to amplitude modulations near voice pitch frequency. The current study examined model population responses to 20 speakers (10 males, 10 females) reciting 12 English vowel contrasts from the Hillenbrand *et al*. database [*J. Acoust. Sci. Am.* 97, 3099 (1995)]. Pairwise correlations across model population responses for each vowel were used to evaluate variability in the neural representations. Results show that the acoustical variability associated with the vowel contrasts is maintained in these neural representations. Thus, variability in the acoustic vowel space is maintained after the nonlinear responses of realistic auditory-nerve models and midbrain models for amplitude modulation tuning. Our goal is to extend our knowledge of the neural representation of the vowel space using a computational model for the responses of auditory neurons to ensembles of speech tokens.

1aSCb4. Generative approach for robust acoustic model training for blindly separated speech recognition. Norihide Kitaoka, Yuto Dekiura, and Kazuya Takeda (Dept. of Media Sci., Nagoya Univ., Furo-cho Chikusa-ku, Nagoya 464-8603, Japan, kitaoka@nagoya-u.jp)

We propose a generative acoustic model training method for robust speech recognition with blind sound source separation as a front-end. Multiple microphone systems are often used for the separation. In such situation, separated speech is severely distorted and thus the recognition rate significantly drops. If we can measure transmission characteristics from the sound sources with various directions to the microphones, we can simulate to receive various mixed speech made by multiple speakers speaking with overlaps to each other. Then we separate the simulated overlapped speech using a blind source separation method such as frequency domain independent component analysis (FDICA) and use the separated speech to train HMM acoustic models to recognize such separated speech. Our method can generate such distorted speech enormously without recording the real speech spoken to the microphone system. We evaluate the models in the continuous Japanese speech recognition and show the effectiveness.

1aSCb5. Speaker identification in reverberant environments. Noha Korany (Elec. Eng., E. E. Dept., Faculty of Eng., Alexandria Univ., Alexandria 21544, Egypt, nokorany@hotmail.com)

The performance of speaker identification process degrades in reverberant environments, as reverberation leads to clear physical effects on the perceived signals. This paper investigates the effect of room reverberation on

the identification rate. However, various reverberant environments are simulated, and the impulse response is convolved with dry speech signals. The reverberant speech database is used by the identification engine within the train and the test phases. Then, statistical identification technique using the Gaussian Mixture Model (GMM) is implemented. Three types of features, Mel-Frequency Cepstrum Coefficients (MFCC), Perceptual Linear Predictive Cepstrum Coefficients (PLPCC), and Relative Spectral Perceptual Linear Predictive Cepstrum Coefficients (RASTA-PLPCC) are extracted. Various types of features are integrated and used for the classification problem. Finally, the performance of the recognition process is evaluated while varying the duration of the train and the test signals, the features used for the classification problem, and the room reverberation. A series of physical measures that correlate with various attributes of the sound perceived in rooms, such as the reverberation time T60, the clarity index C80, the definition D, are calculated. Then, their effect on the identification rate is investigated.

1aSCb6. Experimental study of shout detection with the rahmonic structure. Naoto Kakino, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is012085@ed.ritsumei.ac.jp)

The surveillance systems with microphones have been developed to achieve a secure society. These systems can detect hazardous situations with observed speech but are generally very expensive. This is because the conventional systems manually detect hazardous situations by security officers. Thus, we focus on the automatic shout detection method, which can estimate hazardous situations. The acoustic model based on the Gaussian mixture model has been proposed as the conventional method to identify shouted and natural speeches. However, these methods have a problem that it is necessary to prepare huge training samples to accurately detect shouted speech. In the present paper, we focus on the rahmonic structure, which shows a subharmonic of fundamental frequency in the cepstrum domain because the rahmonic structure tends to arise in the shouted speech. In the present paper, we therefore propose the detection method of the shouted speech based on rahmonic structure. More specifically, we investigate rahmonic structure in the shouted speeches, and detect the shouted speech by utilizing the rahmonic structure model. We conducted evaluation experiments to confirm the effectiveness of the proposed method. As a result of evaluation experiments, we confirmed the effectiveness of the proposed method.

1aSCb7. Suppression of clipping noise in observed speech based on spectral compensation with Gaussian mixture models and reference of clean speech. Makoto Hayakawa, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is033080@ed.ritsumei.ac.jp)

In recent years, the development of communication system allows people to easily record and distribute their speech. However, in the speech recording, clipping noise degrades sound quality when the level of input signal is excessive for the maximum range of amplifier. In this case, it is necessary to suppress clipping noise in the observed speech for improving its sound quality. Although a linear prediction method has been conventionally proposed for suppressing clipping noise, it has a problem with degradation of the restoration performance by cumulating error when the speech includes a large amount of clipping noise. This paper describes a method for suppression of clipping noise in observed speech based on spectral compensation. In this method, the power spectral envelope of speech on each frame in the lower frequency band is noise suppressed to by using Gaussian Mixture Models (GMM), and the one in the higher frequency band is restored by referring to the clean speech. We carried out evaluation experiments with a speech quality, and confirmed the effectiveness of the proposed method toward the speech, which includes a large amount of clipping noise.

1aSCb8. Detection for Lombard speech with second-order mel-frequency cepstral coefficient and spectral envelope in beginning of talking-speech. Takayuki Furoh, Takahiro Fukumori, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0038sv@ed.ritsumei.ac.jp)

In noisy environments, the recorded speech is distorted by the additional noise and the Lombard effect. Thus, the automatic speech recognition (ASR) performance is degraded in noisy environments. To solve this problem, noise reduction methods have been proposed as the conventional study. However, in the conventional study, the improvement of ASR performance for the Lombard effect was not discussed well enough. In the present paper, we focus on the robustly detection for Lombard effect speech (Lombard speech). This is because the ASR system can employ a suitable acoustic model by detecting the Lombard speech. We previously proposed the detection for Lombard speech based on second-order MFCC and fundamental frequency. The previously proposed method however requires longer utterances to detect Lombard speech. We therefore newly propose the detection method for Lombard speech with second-order MFCC and spectral envelope in beginning of talking-speech. To detect the Lombard speech at a short time, the proposed method employs variable weights corresponding to elapsed time for second-order MFCC and spectral envelope. As a result of evaluation experiments, we confirmed that the detection time was reduced from the conventional method.

1aSCb9. The detection of the sleepiness from the sounds obtained inside of the body. Masanoti Akita, Hiroyuki Kamitabira, Tomohiko Yoshida, Syogo Kanemitsu, and Yoichi Midorikawa (Faculty of Eng., Dept. of Elec. and Electron. Eng., Oita Univ., 700 Dannoharu, Oita 8701192, Japan, makita@oita-u.ac.jp)

This paper shows that the detecting method of the sleep-in sleep state or sleepiness from the sound signals in the human body. In former report, we showed that the detection of the sleepiness concerns with the piezoelectric sensors attached on car seat. Our preliminary examination showed that the spectrum of signals from the piezoelectric sensor have the tendency that the shapes of the spectral envelopes are flattened. And the sounds in human body are considered to have similar features. In this experiment, the signals of the piezoelectric sensor on the seat and the sounds in the human body are measured at the same time and the relation between the sounds and sleepiness are examined. The sounds inside the body are measured using NAM microphone system. The spectral envelopes of the signals from the left side and the right side of breath are calculated. The spectral envelopes from the seat are calculated at the same time. Twenty-two measurements by four examinees are done, and 8 sleeping data are measured. Flatness of the envelopes is defined using the lower order of cepstral coefficients, and the increase of the flattened spectrum is observed by the sounds from the sleeping data of three quarters.

1aSCb10. Modeling occurrence tendency of adventitious sounds and noises for detection of abnormal lung sounds. Takanori Okubo, Masaru Yamashita, Katsuya Yamauchi, and Shoichi Matsunaga (Engineering, Nagasaki Univ., 1-14, Bukyo-machi, Nagasaki 852-8521, Japan, mat@cis.nagasaki-u.ac.jp)

Diagnosis of pulmonary emphysema by using a stethoscope is based on the common knowledge that abnormal respiratory (adventitious) sounds usually appear in patients with pulmonary emphysema. However, the spectral similarity between adventitious sounds and noises at auscultation makes highly accurate automatic detection of adventitious sounds difficult. In this paper, we have proposed a novel method for distinguishing between normal lung sounds in healthy subjects and abnormal sounds, including adventitious sounds in patients, taking into account the occurrence tendency of adventitious sounds and noises. According to our investigation results, adventitious sounds occur repeatedly in successive inspiratory/expiratory phases of patients. On the other hand, noise sounds mix at random in lung sounds of both patients and healthy subjects. In our method, the occurrence tendency of these sounds is described using Gaussian distribution of a random variable obtained by subtracting the acoustic likelihood for abnormal respiration from the likelihood for normal respiration. The spectral likelihood calculated using hidden Markov models and the validity score of the occurrence tendency of the adventitious/noise sounds are combined to derive the classification result. Our method achieved a higher classification rate of 94.1% between normal and abnormal lung sounds than that achieved using the conventional method (87.4%).

1aSCb11. Audio quality evaluation by experienced and inexperienced listeners. Nadja Schinkel-Bielefeld (Audio Dept., Fraunhofer Inst. IIS, Am Wolfsmantel 33, Erlangen 91058, Germany, nadja.schinkel-bielefeld@iis.fhg.de), Netaya Lotze (Deutsches Seminar, Leibniz Universität Hannover, Hannover, Germany), and Frederik Nagel (Audio Dept., Fraunhofer Inst. IIS, Erlangen, Germany)

Basic audio quality of coded audio material is commonly evaluated using ITU-R BS-1534 Multi Stimulus with Hidden Reference and Anchors (MUSHRA) listening test. MUSHRA guidelines call for experienced

listeners. However, the majority of consumers using the final product are no expert-listeners. Also the degree of expertise in a listening test may vary among listeners in the same laboratory. It would be useful to know how the audio quality evaluation differs between trained and untrained listeners and how training and actual tests should be designed in order to be as reliable as possible. To investigate the rating differences between experts and non-experts, we performed MUSHRA listening tests with 13 experienced and 11 inexperienced listeners using 5 speech and audio codecs delivering a wide range of basic audio quality. Except for the hidden reference, absolute ratings of non-experts were consistently at least 10% higher than those of experts. However, they could be mapped to each other by a z-transform. For lower quality values, confidence intervals were significantly larger for non-experts than for experts. Experienced listeners set more than twice as many loops as non-experts, compared more often between codecs and listened to high quality codecs for a longer duration than non-experts.

1aSCb12. Influence of amplification scheme and number of channels on aided speech-intelligibility performance. Amyn M. Amlani (Dept. of Speech and Hearing Sci., Univ. of North Texas, 907 W Sycamore St., P.O. Box 305010, Denton, TX 76203, amlaniam@unt.edu), Sneha V. Bharadwaj (Dept. of Commun. Sci. and Disord., Texas Woman's Univ., Denton, TX), and Shirin J. Jivani (Dept. of Speech and Hearing Sci., Univ. of North Texas, Denton, TX)

Modern hearing aids offer a wide range of channels (i.e., filters) and amplification schemes. Our previous work revealed that increasing the number of channels, in conjunction with a fast-fast amplification scheme, results in (a) the spectral flattening of the vowels /i, u, ^/ (Amlani *et al*., 2011), and (b) reduced consonant- and vowel-identification accuracy in impaired listeners (Amlani *et al*., 2012). In the present study, we assess the performance of impaired listeners and their normal-hearing controls on the perception of everyday speech using the Connected Speech Test (Cox *et al*., 1987, 1988). The stimuli were processed through a simulated hearing aid with varying amplification schemes (linear, compression [fast-fast, slow-slow, fast-slow]) and number of channels (2, 8, 16). Findings revealed that while speech-intelligibility performance improved markedly with everyday speech compared to /CVC/ words for both groups, normal-hearing listeners identified the target words significantly better than impaired listeners did. Speech-intelligibility performance was similar across number of channels for normal-hearing listeners, but decreased significantly with a fast-fast amplification scheme. For impaired listeners, performance declined for channels greater than 2 and with the inclusion of the fast-fast amplification scheme. We discuss the implication of these findings relative to clinical application and hearing aid design.

1aSCb13. Relationship between subjective and objective evaluation of noise-reduced speech with various widths of temporal windows. Mitsunori Mizumachi (Dept. of Elec. Eng. and Electron., Kyushu Inst. of Technol., 1-1 Sensui-cho, Tobata-ku, Kitakyushu, 805-8440, Japan, mizumach@ecs.kyutech.ac.jp)

It is necessary to enhance adverse speech signals for building useful speech interfaces. Speech enhancement is essential under noisy and reverberant acoustic environments. Therefore, quality assessment of the enhanced speech signals should be also an important issue in noise reduction and dereverberation. Subjective evaluation is given by carrying out listening tests, and objective evaluation is provided by speech distortion measures. However, there is the discrepancy between subjective and objective evaluation of speech distortion. The author has investigated the relationship between subjective and objective evaluation of noise-reduced speech signals. The objective speech distortion was calculated in each short-term frame, of which length was fixed, and the statistical characteristics of the short-term speech distortion were investigated using higher-order statistics such as skewness and kurtosis. The preliminary result suggested that skewness of the short-term speech distortion could give an explanation for the discrepancy between subjective and objective evaluation. Further investigation of the relationship between subjective and objective evaluation of noise-reduced speech signals is carried out with a variety of temporal window widths. [Work supported by NEDO, Japan.]

1aSCb14. A Hidden Markov Model based speaker identification system using mobile phone database of North Atlantic Treaty Organization words. Shyam S. Agrawal, Shweta Bansal, and Dipti Pandey (KIIT College of Eng., Sohna Rd.,Near Bhondsi, Gurgaon, Haryana 122102, India, dr.shyamsagrawal@gmail.com)

This paper describes results of an experiment to conduct text independent speaker identification of large number of speakers (about 100) using a standard vocabulary of about 23 NATO words—such as Alfa, Bravo, etc. These words in isolation were spoken in a sound treated room by Hindi natives having very good education in English (both males and females) and recorded by a three channel data recording system—the cardioid microphone, electret condenser microphone, and a NOKIA mobile telephone. The pre-processed digitized database of isolated words was further processed to determine 39 MFCC's and their derivatives and used to build an HMM model for each speaker based on all the words. The HMM model was trained using an HTK tool kit to generate the model parameters and tested using Viterbi algorithm. The identification of speakers was done in a closed set manner, based on comparison of each NATO word in the model. In addition to correct identification, false acceptance and false rejection scores were also found. The results show varying performance due to variations in channels, male/female speakers. The overall identification scores vary between 60% and 70%. The paper gives detailed analysis of results.

1aSCb15. A time-synchronous histogram equalization for noise robust speech recognition. Fumiya Takahashi, Masaharu Kato, and Tetsuo Kosaka (Grad. School of Sci. and Eng., Yamagata Univ., 4-3-16 Jonan, Yonezawa, Yamagata 992-8510, Japan, tar11506@st.yamagata-u.ac.jp)

The histogram equation (HEQ) technique is commonly adopted for feature space normalization in speech recognition systems. In this technique, a transform function is calculated directly from the histograms of both training and test data, and the nonlinear effects of additive noise are compensated. In order to estimate the transform function accurately, a certain amount of data are required. However, this is not suitable for real-time application because at least several seconds of evaluation data need to be accumulated before the transform function can be calculated. This means that the system cannot start the recognition process until the end of utterance. In this research, we aim to develop a new speech recognition method based on the HEQ technique for real-time processing. This method is called "time-synchronous frame-weighted HEQ (ts-FHEQ)." In the time-synchronous decoding, lack of data for estimating the histogram becomes a major problem. To resolve this problem, we introduce a frame weighting approach, where the degree of transform is controlled according to the number of data frames. Our speech recognition experiments verified that the proposed technique shows good performance and achieves substantial reduction of calculation time.

1aSCb16. An investigation of vowel substitution rules in the automatic evaluation system of English pronunciation. Kei Sato, Masaharu Kato, and Tetsuo Kosaka (Grad. School of Sci. and Eng., Yamagata Univ., 4-3-16 Jonan, Yonezawa, Yamagata 992-8510, Japan, tna01014@st.yamagata-u.ac.jp)

We investigate the performance improvement of an automatic evaluation system of the English pronunciation of Japanese learners. In this system, Japanese and English acoustic models are used to detect mispronunciation at a phoneme level. Hidden Markov models (HMMs) are used as acoustic models. Mispronunciation is detected by comparing the output likelihoods of the two models. In order to improve the performance of this system, we investigate certain mispronunciation rules, which represent common mispronunciations among Japanese learners. We use four mispronunciation rules: vowel insertion (at the end of a word), vowel substitution, vowel insertion (between consonants), and consonant substitution. In this system, the accuracy of the mispronunciation rules is particularly important. The rules are determined on the basis of the knowledge of phonetics in our previous system. However, the effectiveness of the rules has not been analyzed quantitatively, and we do so in this work. A knockout procedure is used to select effective rules. By selecting effective rules, we found that the correlation coefficient between the subjective evaluation value and the system performance improved from 0.757 to 0.858.

1aSCb17. A method to estimate a temporally stable spectral envelope for periodic signals. Masanori Morise and Yoichi Yamashita (Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu, Shiga 525-8577, Japan, masanori.morise@gmail.com)

Vocoder-based speech synthesis system requires fundamental frequency (F0) and spectral envelope. Since the sound quality of synthesized speech depends on the estimation performance, methods that can accurately estimate two parameters are crucial to synthesize natural speech. In particular, spectral envelope estimation is more difficult than F0 estimation. One of the problems in spectral envelope estimation is that the result depends on the windowing time, type of window function, and its length. Conventional methods such as LPC or Cepstrum cannot remove the temporal variation. TANDEM-STRAIGHT can estimate a temporally stable spectral envelope, whereas the processing that consists of averaging two power spectra, smoothing and post-processing is complex. To simplify the processing, we propose a method based on pitch synchronous analysis and spectral smoothing. The proposed method can estimate a temporally stable spectral envelope from only one power spectrum processed by a specialized window function. The window function and its length are determined to remove the temporal variation. The objective evaluation was conducted to verify the temporal variance and the estimation performance. The result suggested that the proposed method can estimate the temporally stable power spectrum as well as TANDEM-STRAIGHT.

1aSCb18. Processing time improvement for speech enhancement based local projection using dynamic parameters. Phongphan Phienphanich, Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., 99 Moo 18 Phaholyothin Rd., Ampher Khlongluang, Pathumthani 12120, Thailand, 5410030067@student.tu.ac.th), and Chutamanee Onsuwan (Linguistics, Thammasat Univ., Khlongluang, Pathumthani, Thailand)

Local projection (LP) has been widely used to enhance speech by transforming noisy speech into two orthogonal subspaces: noise (S1) and signal plus small amount of noise (S2) subspace. S1 is removed and S2 is transformed into time domain resulting in the enhanced speech. Satisfactory results with significantly improved speech quality have been reported by several works although the processing time was not taken into account. Four parameters to be considered are embedding dimension (d), time delay (τ), numbers of iteration, and minimal embedding dimension. Speech quality is increased by the increase of d-parameter, resulting in decrease of τ-parameter value ($d\times\tau$ kept constant) and the dramatic increase of the processing time. The goal is to come up with the best $d\times\tau$ parameter for each iteration, while speech quality remains almost unaffected. Rather than using a fixed $d\times\tau$ parameter, a dynamic approach is taken. Specifically, τ is initially set to 1 and incremented by 1 for next iteration. The experimental results tested on Thai initial rhyming words corrupted by three noise types (white, car, and street) each at SNR levels of 10, 5, 0, -5 dB showed that the proposed method significantly reduced the processing time for white noise by 35% ($p < 0.01$).

1aSCb19. Acoustic-to-articulatory inversion by analysis-by-synthesis using cepstral coefficients. Julie Busset and Yves Laprie (LORIA/CNRS, 615 rue du jardin botanique, Villers-lès-Nancy 54600, France, yves.laprie@loria.fr)

This paper deals with acoustic to articulatory inversion of speech by using an analysis by synthesis approach. We used old X-ray films of one speaker to (i) the develop a linear articulatory model presenting a small geometric mismatch with the subject's vocal tract mid sagittal images (ii) and design an adaptation procedure of cepstral vectors used as input data. The adaptation exploits the bilinear transform to warp the frequency scale in order to compensate for deviation between synthetic and natural speech. This enables the comparison of natural speech against synthetic speech without using cepstral liftering. A codebook is used to represent the forward articulatory to acoustic mapping, and we designed a loose matching algorithm using spectral peaks to access it. This algorithm, based on dynamic programming, allows some peaks in either synthetic spectra (stored in the codebook) or natural spectra (to be inverted) to be omitted. Quadratic programming is used to improve the acoustic proximity near each good candidate found during codebook exploration. The inversion has been tested on speech signals corresponding to the X-ray films. It achieves a very good geometric precision of 1.5 mm over the whole tongue shape unlike similar works evaluating the error at 3 or 4 points corresponding to sensors located at the front of the tongue.

1aSCb20. An overview of the development of resources, techniques, and, systems for Indian spoken languages. Shyam S. Agrawal (KIIT, College of Eng., Sohna Rd.,Near Bhondsi, Gurgaon, Haryana 122102, India, dr.shyamsagrawal@gmail.com)

India possesses a large variety of languages and dialects spoken in different parts of the country. These languages possess some unique linguistic, phonological, and phonetic properties different from European languages. Research is being done in several of Indian languages—such as Hindi, Bangla, etc. to study the articulatory, acoustic—phonetic and prosodic nature for the purpose of creating standards of phonetic representation of phonemes and Pronunciation Lexicon in Indian Languages. Comprehensive and task specific language corpora, speech databases in laboratory as well as in mobile communication situation and the tools/technics required for processing of speech signals are being developed. The emphasis is on developing multilingual human-machine interaction systems. Some of the recently developed systems include multi-lingual speech recognition system for voice enabled services, multilingual text to speech synthesis system, speaker and language identification system for general purpose and forensic applications. Recognition of emotions in spoken speech, spoken language translation system, etc. The paper presents an overview of such studies conducted in various laboratories, academic institutions, and industries in India pertaining to these areas. The technologies used for data collection, processing, and recognition/ synthesis, etc., utilized and status of the development have been mentioned.

1aSCb21. Performance estimation of speech recognition based on Perceptual Evaluation of Speech Quality and acoustic parameters under noisy and reverberant environments with Corpus and Environment for Noisy Speech RECognition 4. Takahiro Fukumori, Masato Nakayama, Takanobu Nishiura, and Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, cm013061@ed.ritsumei.ac.jp)

CENSREC-4 evaluation framework has been distributed for evaluating distant-talking speech under various noisy and reverberant environments. It however has not been evaluated how variable noisy and reverberant features in this contains. We thus try to evaluate CENSREC-4 with our designed noisy and reverberant criteria based on PESQ and acoustic parameters. We specifically focus on criteria to represent the difficulty of noisy and reverberant speech recognition, and also confirm why it is difficult to easily evaluate the recognition accuracy in a part of CENSREC-4 corpus with our proposed noisy and reverberant criteria. We first designed the noisy and reverberant criteria using the relationship among the D value, the PESQ, and the ASR performance. We then tried to estimate the recognition accuracy in various noisy and reverberant environments with CENSREC-4. We carried out evaluation experiments to confirm the difficulty to easily evaluate the recognition accuracy in a part of CENSREC-4 courpus0. As a result of evaluation experiments, we confirmed that it was difficult to estimate the accuracy of noisy and reverberant speech recognition in heavy noisy and reverberant environment with CENSREC-4. We therefore confirmed that CENSREC-4 contained very challenging and variable noisy and reverberant data.

1aSCb22. On instantaneous vocal tract length estimation from formant frequencies. Adam Lammert and Shrikanth Narayanan (Signal Anal. and Interpretation Lab., Univ. of Southern California, 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089, lammert@usc.edu)

The length of the vocal tract and its relationship with formant frequencies is examined at fine temporal scales with the goal of providing accurate estimates of vocal tract length from acoustics on a spectrum-by-spectrum basis despite unknown articulatory information. Accurate vocal tract length estimation is motivated by applications to speaker normalization and biometrics. Analyses presented are both theoretical and empirical. Various theoretical models are used to predict the behavior of vocal tract resonances in the presence of different vocal tract lengths and constrictions. Real-time MRI with synchronized audio is also utilized for detailed measurements of vocal tract length and formant frequencies during running speech, facilitating the examination of short-time changes in vocal tract length and corresponding changes in formant frequencies, both within and across speakers. Previously proposed methods for estimating vocal tract length are placed within a coherent framework, and their effectiveness is evaluated and compared. A data-driven method for VTL estimation emerges as a natural extension of this framework, which is then developed and shown to empirically outperform previous methods on both synthetic and real speech data. A theoretical justification for the effectiveness of this new method is also explained. [Work supported by NIH.]

MONDAY MORNING, 3 JUNE 2013 510A, 9:00 A.M. TO 12:00 NOON

Session 1aSP

Signal Processing in Acoustics, Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Array Signal Processing for Three-Dimensional Audio Applications I

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Invited Papers

9:00

1aSP1. Control of frame loudspeaker array by minimizing fluctuations of frequency response and synthesized wave front. Akio Ando (Sci. and Technol. Res. Lab., NHK, 1-10-11 Kinuta Setagaya, Tokyo 157-8510, Japan, andio@a.memail.jp) and Aya Tokioka (Dept. of Commun. and Integrated Syst., Grad. School of Sci. and Eng., Tokyo Inst. of Technol., Tokyo, Japan)

In sound reproduction with accompanying pictures, the localization of sound on the image display is problematic because a loudspeaker cannot be placed on the display. To gain a stable localization, the use of loudspeaker array set on the frame of the display may be a solution. In general, the loudspeaker array enables to control the perceptual depth of sound image by generating the appropriate curvature of the wave front corresponding to the source position. However, the frequency response and the shape of the wave front reproduced by such a frame loudspeaker array sometimes deteriorate, particularly when the virtual sound source has a certain distance back from the display. In this study, new parameters are introduced to scale the deterioration of the frequency response and the shape of the wave front. A new method is also introduced to generate the input signals to the loudspeakers on the basis of minimization of these parameters. The experimental result showed that the method generates the sound with small deterioration of the frequency response and the wave front regardless of the depth of the virtual source position, meaning that it can be used for the sound reproduction for 3D television.

9:20

1aSP2. Optimal beamformer designed for robustness against channel mismatch based on Monte Carlo Simulation. Mingsian R. Bai and Ching-Cheng Chen (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

Design of beamformers that withstand mismatch in channel characteristics of gain, phase, and position has been a key issue in array signal processing. These mismatch factors are random in nature and generally intractable by deterministic approaches. This paper examines the effects of channel mismatch on beamformer performance from a statistical perspective. The aim of this work is twofold: analysis and synthesis. In the analysis phase, the mismatch factors of microphone characteristics are assumed to be random variables following either uniform or Gaussian distribution. In the light of the Monte-Carlo Simulation (MCS), statistics including the mean, maximum, minimum, and the probability density function (pdf) of Directivity Factor (DF) can be efficiently obtained with random sampling. This provides useful information for choosing performance measures in the next synthesis phase. Optimal parameters of superdirective array designed using least squares (LS) and convex optimization (CVX) are determined based on the preceding performance measures. Simulation results have shown that the proposed statistical approach with different performance measures provided various performance-robustness tradeoffs in terms of parameter range for optimal beamformers.

9:40

1aSP3. Hybrid immersive three-dimensional sound reproduction system with steerable parametric loudspeakers. Chuang Shi, Ee-Leng Tan, and Woon-Seng Gan (School of Elec. & Electron. Eng., Nanyang Technol. Univ., 50 Nanyang Ave., S2-B4a-03, DSP Lab, Singapore 639798, Singapore, shichuang@ntu.edu.sg)

A loudspeaker must be both dispersive and directive to accurately reproduce spatial audio from digital media. To address this problem, an audio system that has a unique combination of conventional and parametric loudspeakers has previously been proposed and proved to be effective to reproduce an immersive 3D soundscape. However, this system has two drawbacks: (1) There is only one fixed "sweet spot," and (2) only one listener within the "sweet spot" can enjoy the complete experience. Therefore, a hybrid 3D sound

reproduction system combining conventional loudspeakers with a pair of steerable parametric loudspeakers is proposed in this paper. By using this new combination of conventional and steerable parametric loudspeakers, the "sweet spot" can be steered toward the listener's head position. Thus, the listener no longer needs to keep his head stationary while watching movies or playing games, which resulting in a more relaxing and pleasant experience. Furthermore, a dual-beamsteering method is proposed for the parametric loudspeaker, which provides a flexible software-control solution to allow the 3D sound experience to be enjoyed by two listeners simultaneously. This paper provides the system overview and highlights the key processing techniques in rendering a "steerable" immersive 3D soundscape.

10:00

1aSP4. Virtual sound source generation: Its various methods and performances. Dong-Soo Kang (Ctr. for Noise and Vib. Control, Dept. of Mech. Eng., KAIST, Daejeon, South Korea), Jung-Min Lee (Grad. School of Culture Technol., KAIST, Daejeon, South Korea), Jung-Woo Choi, Min-Ho Song, and Yang-Hann Kim (Ctr. for Noise and Vib. Control, Dept. of Mech. Eng., KAIST, YuseongGu GuseongDong 373-1, Daejeon, South Korea, yanghannkim@kaist.ac.kr)

There are many means to generate a virtual sound source, or sources in the region of interest. For example, Wave Field Synthesis (WFS) or Higher Order Ambisonics (HOA) are good examples. These methods normally assume that loudspeakers are spatially distributed in the space. The region of interest where the desired sound is generated can have arbitrary shape; enclosed by surrounding loudspeakers or partially enclosed. Therefore, the performance of the method would be affected by the boundary conditions as well as the wave length of desired wave field. In other words, how the waves are distributed in the selected space. In recent work [Choi and Kim, IEEE Trans. Speech Audio Process. **20**(7), 1976–1989 (2012)], a new approach was proposed to generate virtual sources in the space that is enclosed by an array of loudspeakers, which have been believed to be problematic with well-known methods. It is proved to be mathematically exact solution. However, "exact solution" does not necessarily mean that it is better than the others. In this paper, performances of these three methods are compared. Theoretical and experimental comparisons have been attempted and observed in this paper.

10:20

1aSP5. Evaluation of system configuration to check the suitability for the sound field rendering using the inverse approach. Jeong-Guon Ih (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 373-1 Guseong-Dong, Yuseong-Gu, Daejeon 305-701, South Korea, J.G.Ih@kaist.ac.kr), Wan-Ho Cho (Div. of Physical Metrol., Korea Res. Inst. of Standards and Sci., Daejeon, South Korea), and Seung-Wan Hong (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

Sound field control by the inverse approach based on the acoustical holography is useful to render an arbitrary target sound field within a selected control zone if the target condition is given in a detailed format. This method needs information on the various factors constituting the total system: source array configuration, relative position of source and control region, assigning method of target field condition, conditioning method, etc. Because these factors heavily affect the accuracy of the generated sound field, a proper definition of the problem including all factors related to the system configuration is important. In this work, we have studied on the condition of major factors of various configurations to generate the target sound field efficiently with high accuracy. Because the difference between target and generated sound field strongly depends on the noise and the information error existing in the actual situation, the expected accuracy should be calculated in relation to the characteristics of system transfer matrix. To this end, variances of uncorrelated noise, condition number, and linear independency of the transfer matrix are evaluated to check the suitability of transfer matrix for accurately rendering the sound field in both free-field and enclosed space.

10:40

1aSP6. Extension of perceived source width using loudspeaker array. Jung-Woo Choi and Yang-Hann Kim (Ctr. for Noise and Vib. Control, Dept. of Mech. Eng., KAIST, YuseongGu GuseongDong 373-1, Daejeon 373-1, South Korea, khepera@kaist.ac.kr)

In this paper, a method to extend the perceived spatial width of a virtual sound source using multiple loudspeakers is proposed. Control of perceived source width or apparent source width (ASW) has been attempted by decreasing the inter-aural correlation. For this purpose, numerous decorrelators were proposed for stereo loudspeakers or headphones. However, these techniques are inadequate for sound field reproduction system incorporating multiple loudspeakers. For sound field reproduction, extension of source width has to be realized with three requirements. First, extension should not deteriorate the localization cue, provided by the reproduction system. Second, the coloration artifact, which induces by extra wavefronts other than the direct wave, should be minimized. Most importantly, the effect of source width extension has to be maintained over a large listening area. To design a spatial decorrelator that can meet these requirements, we design a proper target sound field with reduced inter-aural correlation over a zone of interest. The target sound field is reproduced by loudspeakers driven from a multipole expansion technique. The performance of the proposed method is verified by examining the inter-aural correlation coefficient (IACC) of the reproduced sound field over a wide area, as well as the ITD and ILD distributions.

Contributed Papers

11:00

1aSP7. Sound-field reconstruction performance of a mixed-order Ambisonics microphone array. Márton Marschall and Jiho Chang (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, mm@elektro.dtu.dk)

Recently, there has been increasing interest in using spherical microphone arrays for spatial audio recordings. Accurate recordings are important for a range of applications, from virtual sound environments for hearing research through to the evaluation of communication devices, such as hearing instruments and mobile phones. Previously, a mixed-order Ambisonics (MOA) approach was proposed to improve the horizontal spatial resolution of spherical arrays. This was achieved by increasing the number of microphones near the horizontal plane while keeping the total number of transducers fixed. The approach is motivated by the fact that human spatial hearing is most acute in the horizontal plane. This study presents simulations of the performance of an MOA rigid-sphere microphone array, and its robustness to variations in microphone characteristics. Specifications of a commercially available microphone were used to simulate self-noise, sensitivity, and phase response variations between the microphones. To quantify

the reconstruction error and the "sweet area" as a function of source elevation, the reconstructed sound field based on a simulated array measurement was compared to the reference sound field for both horizontal and elevated sources. It is expected that the MOA approach results in a larger sweet area for mid to high frequencies for horizontal sources.

11:20

1aSP8. Aircraft sound environment reproduction: Sound field reproduction inside a cabin mock-up using microphone and actuator arrays. Philippe-Aubert Gauthier, Cédric Camier, Olivier Gauthier, Yann Pasco, and Alain Berry (Mech. Eng., Université de Sherbrooke, 51, 8e Ave. Sud, Sherbrooke, QC J1G 2P6, Canada, philippe_aubert_gauthier@hotmail.com)

Sound environment reproduction of various flight conditions in aircraft cabin mock-ups is useful for the design, demonstration, and jury testing of interior aircraft sound quality. To provide a faithfully perceived sound environment, time, frequency, and spatial characteristics should be preserved. Physical sound field reproduction approaches for spatial sound reproduction are mandatory to immerse the listener in the proper sound field so that localization cues are recreated. A 80-channel microphone array was built and used to capture a 2-h recording of in-flight sound environments within an actual Bombardier CRJ aircraft. An instrumented cabin mock-up was used to reproduce, in the least-mean-square sense, the recorded sound field using a 41-channel trim-panel actuator array. In this paper, experiments with multichannel equalization are reported. One of the practical difficulties was related to the use of the trim panels as sound sources. Windows and trim panels introduce audible squeaks and rattles if driven at low frequencies. Bass management was therefore implemented. Floor shakers and a subwoofer were used to recreate the low frequency content while the trim panels were only used for the high frequency range. The paper presents objective evaluations of reproduced sound fields. Results and practical compromises are reported.

11:40

1aSP9. Design and implementation of a personal audio system in a car cabin. Jordan Cheer and Stephen J. Elliott (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Highfield, Southampton, Hampshire SO17 2LG, United Kingdom, j.cheer@soton.ac.uk)

The generation of personal listening zones in a car cabin would allow the different occupants to listen to different audio programs without the use of headphones. This would allow, for example, the driver to listen to a navigation system while the rear passengers watched a film. Personal audio systems have previously been implemented in mobile devices and monitors, for example, however, the investigation of the effects of an enclosure on the generation of personal listening zones has been limited. This paper presents an investigation of the effects of a car cabin sized enclosure on the generation of independent listening zones in the front and rear seats. The standard car audio loudspeaker array is used to produce independent listening zones at low frequencies, while a second array of small loudspeakers positioned at the four headrest positions is used to provide control over the rest of the audio bandwidth. The performance of the arrays is first simulated in a car cabin sized rectangular enclosure to understand the physical limits on the achievable control. The proposed arrays are then implemented in a real car to validate the simulation results and the results of a real-time implementation are presented.

MONDAY MORNING, 3 JUNE 2013 511AD, 8:55 A.M. TO 12:00 NOON

Session 1aUW

Underwater Acoustics: Seabed Scattering: Measurements and Mechanisms I

Charles W. Holland, Cochair
Appl. Res. Lab., The Penn. State Univ., P.O. Box 30, State College, PA 16801

Dale D. Ellis, Cochair
DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada

Gavin Steininger, Cochair
School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. (Ring Rd.), Victoria, BC V8P 5C2, Canada

Chair's Introduction—8:55

Invited Papers

9:00

1aUW1. The small-slope approximation for layered seabeds. Darrell Jackson (Appl. Phys., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, drj@apl.washington.edu)

The small-slope approximation has found application to unlayered seabeds and is generally regarded as an improvement over methods that employ either small-roughness perturbation theory or the Kirchhoff approximation. Unfortunately, the usual small-slope ansatz fails when applied to layered seabeds, as it is inconsistent with perturbation theory. This ansatz is replaced by an alternative, which is found to satisfy the criteria of reciprocity and consistency with the perturbation and Kirchhoff approximations. This approach will be illustrated by computation of the coherent reflection coefficient and scattering strength for a seabed consisting of a single rough fluid layer over a semi-infinite, elastic basement with flat upper boundary. Computation time is significantly longer than for the unlayered case, increasing as the desired accuracy increases. The results will be contrasted with those obtained using a variety of existing approximations.

9:20

1aUW2. Seafloor measurements using synthetic aperture sonar. Anthony Lyons, Daniel Brown (Appl. Res. Lab., Penn State Univ., University Park, State College, PA 16803, apl2@psu.edu), Derek Olson (Grad. Program in Acoust., Penn State Univ., State College, PA), and Shawn Johnson (Appl. Phys. Lab., Johns Hopkins Univ., Laurel, MD)

The past decade has seen considerable growth in the use of synthetic aperture sonar (SAS) imaging systems in both the civilian and military domains. Although SAS systems are almost always uncalibrated, they can still yield information about the seafloor given an understanding of the mechanisms affecting the statistical properties of the images produced by these systems. This talk will describe our recent efforts to link SAS image statistics to seafloor properties through the use of seafloor scattering models. Sample results from several SAS systems encompassing frequencies ranging from 6 to 300 kHz will be shown.

Contributed Papers

9:40

1aUW3. Seabed roughness parameters for the Malta Plateau from joint backscatter and reflection inversion. Gavin Steininger (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. (Ring Rd.), Victoria, BC V8P 5C2, Canada, gavin.amw.steininger@gmail.com), Charles W. Holland (Appl. Res. Lab., The PennState Univ., State College, PA), Stan E. Dosso, and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper presents seabed interface-scattering and geoacoustic parameters estimated on the Malta Plateau, Mediterranean Sea, by joint Bayesian inversion of monostatic backscatter and spherical-wave reflection-coefficient data. The data are modeled assuming a stack of homogeneous fluid sediment layers overlying an elastic basement. The scattering model also assumes a randomly rough water-sediment interface with a von Karman roughness power spectrum. Scattering and reflection data are inverted simultaneously using a population of interacting Markov-chains to sample roughness and geoacoustic parameters as well as residual error parameters. Trans-dimensional sampling is applied to treat the unknown number of sediment layers and unknown autoregressive order of the errors (to represent residual correlation). Results are considered in terms of marginal posterior probability profiles and distributions, which quantify the effective data information content to resolve scattering/geoacoustic parameters and structure. Results indicate well-defined scattering (roughness) parameters in good agreement with existing measurements, and a multi-layer sediment profile over a high-speed (elastic) basement, consistent with independent knowledge of sand layers over limestone. [Work supported by ONR.]

10:00

1aUW4. Energy exchange and scattering loss within a two-way coupled-mode formulation. Steven A. Stotts and Robert A. Koch (Env. Sci. Lab., Appl. Res. Labs/The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78759, stotts@arlut.utexas.edu)

The loss of energy due to propagation in an ocean environment arises from several possible processes. In a two-way coupled-mode description scattering effects from a rough bottom Pekeris waveguide can be isolated by excluding bulk attenuation, and intermodal energy exchange can be examined. The range- and frequency-dependent interaction of a single trapped mode exchanging energy with multiple continuum modes is identified within this framework. The goal of the analysis is two-fold. First, it provides insight into additional loss mechanisms that can be incorporated into current local mode models (c.f. Koch and Stotts, A Derivation of Energy Loss via Coupled Modes, *165th Meeting of the Acoustical Society of America*, June 2–7, 2013). Second, a comparison can be made to previous descriptions of additional attenuation, such as Kirchhoff scattering loss. Varying the bottom roughness permits tests of the applicability of the Born approximation.

10:20

1aUW5. Derivation of energy loss via coupled modes. Robert A. Koch and Steven A. Stotts (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, koch@arlut.utexas.edu)

A coupled mode formalism can describe energy loss within an ocean waveguide due to all possible mechanisms, including bulk attenuation, scattering from rough surfaces, and volume inhomogeneities. First order corrections produced from mode-couplings can be incorporated into the modal loss. Accounting for these losses within an adiabatic mode approach would provide improvements over current standard modal propagation models. Highlights of the formalism for scattering from a rough surface will be provided.

10:40

1aUW6. Influence of rough seabed surface on statistics of modal energy flux. David P. Knobles and Jason D. Sagers (ARL, UT at Austin, 10000 Burnet Rd., Austin, TX 78758, knobles@arlut.utexas.edu)

When sound propagates through a random media, the wave properties of the acoustic field may be viewed as stochastic variables. It is thus natural to investigate the relationship between the statistical properties of the acoustic field and the random fluctuations of the waveguide. The interest here is a 2-D ocean waveguide with a rough seabed. Numerical solutions to the 2-way integral coupled mode equations (ICME), for random realizations of the roughness from a wavenumber power spectrum, provide the statistics of the modal intensity and cross-mode coherence with range. The roughness induces mode coupling within the trapped spectrum, between the trapped and the continuum spectrum, and to the back propagating modal spectrum. Instead of a master equation for the modal intensities to study the connection between the statistics of the acoustics and the fluctuations in the waveguide as has been advocated in previous studies, the conservation law for acoustic energy flux is used to develop an expression for the individual modal Poynting vectors. In addition to exact numerical computations of the range derivatives of the modal energy flux vectors for both forward and backward propagation, a Poynting vector master equation is derived for the case where the Born approximation is valid.

11:00

1aUW7. An initial model-data comparison of reverberation and clutter from a near-shore site in the Gulf of Mexico. Dale D. Ellis (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, daledellis@gmail.com) and John R. Preston (Appl. Res. Lab., The Penn State Univ., State College, PA)

Reverberation measurements were made in the Gulf of Mexico off Panama City, Florida, USA, in April 2012 in preparation for the main Transmission Reverberation Experiment (TREX) in May 2013. The data were gathered using the triplet section of the ONR Five Octave Research Array (FORA), deployed as a fixed receiver. By steering cardioid beams to the right or left, the array can reduce ambiguity. Beamformed data from the 2012 trial show background noise with high directionality and variability due to nearby shipping. Model predictions of reverberation and target are compared with data using a range-dependent Clutter Model, which uses adiabatic normal modes as the computational engine. The initial predictions use isovelocity water, over a sandy bottom halfspace with Lambert scattering, and bathymetry from the $GEBCO_{08}$ database. These initial results will be presented, hopefully supplemented by improved predictions with better environmental inputs and additional clutter data obtained during the May 2013 experiment. [Work supported by ONR Code 322 OA.]

11:20

1aUW8. Evidence for a common scale O(0.1) m that controls seabed scattering and reverberation in shallow water regions. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16801, holland-cw@psu.edu)

Analysis of the spectral content of long-range reverberation yields two observations. First, there is a remarkably similar scale, O(0.1)m, between three diverse continental shelf regions. This is surprising given general understanding of the complexity and diversity of geologic processes. Second, there is strong evidence that the scale is associated with heterogeneities within the sediment. Thus, sediment volume scattering, not interface scattering, controls long-range reverberation from a few hundred Hertz to several kilohertz. This is also unexpected given that at long-ranges the vertical grazing angles are less than the critical angle, and hence, the penetration of the acoustic field into the sub-bottom is expected to be modest. The consistency of the scale, O(0.1)m, suggests an underlying feature or mechanism that is consistent across many ostensibly diverse geological settings. Neither the feature nor mechanism is known at this time. Several hypotheses will be presented. [Work supported by ONR Ocean Acoustics.]

11:40

1aUW9. Rayleigh scattering of sound by spherically symmetric bodies. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

An obstacle's shape is often approximated by a sphere in analyses of sound scattering by air bubbles, objects on or near the seafloor, marine organisms, clouds of suspended particles, etc. Here, an asymptotic technique is developed to study low-frequency sound scattering from spherically symmetric inhomogeneous obstacles. The obstacle can be fluid, solid, or a fluid-filled solid shell. Physical properties of the obstacle are arbitrary piece-wise continuous functions of the distance to its center. The radius of the obstacle is assumed to be small compared to the wavelengths of sound in the surrounding fluid as well as of compressional and shear waves inside the obstacle. General properties of the sound scattering by spherically symmetric bodies are established. Resonant Rayleigh scattering is studied in detail. For plane and spherical incident waves, it is discussed which physical and geometrical parameters of the obstacle can be retrieved from the scattered acoustic field.

MONDAY AFTERNOON, 3 JUNE 2013 513ABC, 12:55 P.M. TO 4:20 P.M.

Session 1pAAa

Architectural Acoustics and Signal Processing in Acoustics: Advanced Analysis of Room Acoustics: Looking Beyond ISO 3382 II

Boaz Rafaely, Cochair
Dept. of Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva 84105, Israel

Samuel Clapp, Cochair
Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180

Michael Vorländer, Cochair
ITA, RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany

Chair's Introduction—12:55

Invited Papers

1:00

1pAAa1. Theoretic considerations on how the directivity of a sound source influences the measured impulse response. Ingo B. Witew, Tobias Knüttel, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, ingo.witew@akustik.rwth-aachen.de)

In previous investigations, it has been shown that the directivity of a measurement sound source has a significant influence on the measured room impulse response (RIR). Using a specialized method of analysis, the sources influence can be identified even in the very late part of the RIR even in very reverberant environments. These results seem to be surprising at first and contradict intuitive expectations. In this contribution, the findings are briefly discussed, and the congruence with general room acoustic theory is revised and discussed.

1:20

1pAAa2. Enhanced spatial analysis of room acoustics using acoustic multiple-input multiple-output systems. Hai Morgenstern and Boaz Rafaely (Ben-Gurion Univ. of the Negev, Ben-Gurion Univ. of the Negev, Beer-Sheva 84105, Israel, hai.morgenstern@gmail.com)

Standard acoustic measurements in enclosures typically employ single-input single-output (SISO) acoustic systems. The parameters obtained from these measurements describe features of energy decay and do not characterize spatial attributes of the enclosure. Directional analysis of enclosures became popular with the introduction of microphone and loudspeaker arrays. In particular, spherical arrays have been shown to be highly beneficial for spatial analysis. Spherical microphone arrays facilitate the estimation of the arrival direction of the direct and reflected sound, while the use of both loudspeaker and microphones arrays can support the estimation of both radiation and arrival directions, with the application of conventional beamforming methods. However, when several reflections are attributed to the same time bin in a discrete impulse response, reflection paths may not be uniquely determined by existing beamforming techniques. We present a new method to uniquely determine source and receiver directions for multiple reflections when time separation is unfeasible. The paper presents the formulation of the proposed method, also showing a simulation study to demonstrate the performance of the proposed method.

1:40

1pAAa3. Spatio-temporal energy measurements in renowned concert halls with a loudspeaker orchestra. Sakari Tervo, Jukka Pätynen, and Tapio Lokki (Media Technology, Aalto Univ., P.O. Box 15500, Aalto FI-00076, Finland, Tapio.Lokki@aalto.fi)

Room acoustical parameters are commonly measured as spatial averages from a few source positions to several listening positions. In addition, ISO3382-1:2009 suggests that subjective listener aspect could be predicted with a few parameters at mid frequencies. It is obvious that such recommendation is inadequate to describe all perceptual differences between concert halls or seating positions. Explaining multidimensional subjective impressions is not possibly with a few averaged numbers. In our opinion, to compare accurately between concert halls, the measurement positions should be exactly at the same distance from the sources in each hall. And sources should be numerous to represent the typical real source on a wide area—a symphony orchestra. In this paper, we compare renowned European concert halls with spatio-temporal visualization of sound energy. Measurements at identical distances in halls enable an accurate comparison of sound energy distributions. The analysis of measured spatial impulse responses is performed with spatial decomposition method (SDM). The spatial sound energy distribution is presented at different frequency bands as a function of time to visualize the cumulative energy before and after 100 ms.

2:00

1pAAa4. Three-dimensional spatial analysis of concert and recital halls with a spherical microphone array. Samuel Clapp (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, clapps@rpi.edu), Anne Guthrie (Arup Acoust., New York, NY), Jonas Braasch, and Ning Xiang (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

The most well-known acoustical parameters—including reverberation time, early decay time, clarity, and lateral fraction—are measured using data obtained from omnidirectional or figure-of-eight microphones, as specified in ISO 3382. Employing a multi-channel receiver in place of these conventional receivers can yield new spatial information about the acoustical qualities of rooms, as well as the potential for new parameters that could have greater predictive power in terms of listeners' subjective preferences. In this research, a spherical microphone array was used to measure the room impulse responses of a number of different concert and recital halls. The data were analyzed using spherical harmonic beamforming techniques, along with other direction of arrival estimation algorithms, to understand how the soundfield evolves spatially over time at different points in the room. The results were compared to geometrical acoustic simulations and used to differentiate between listener positions which exhibited similar values for the standard parameters. In addition, new parameters were examined, including soundfield homogeneity and other spatial ratios.

2:20

1pAAa5. Listening space measurements resolved by direction and spectro-temporally. Adam O'Donovan, Dmitry N. Zotkin (Comput. Sci. and UMIACS, Univ. of Maryland, College Park, MD), and Ramani Duraiswami (VisiSonics Corp., A.V. WIlliams Bldg., #115, College Park, MD 20742, ramani@umiacs.umd.edu)

Measurement of a listening space might be done to characterize it in a gross way, or to identify some deficiencies in the space, which can then be corrected. Alternately, these measurements might be performed to create inputs to an auralization software, which might seek to recreate a virtual listening experience. We propose that it might be possible to perform audio-visual measurements of a listening space that allow the entire response of the listening space to be understood and visualized. The goal is to understand the response completely in terms of its components: along direction, frequency, early and late characteristics, and finally at the level of the structural elements of the scattering surfaces. Measurements with spherical arrays of cameras and microphones provide measurements that allow the response to be decomposed in the desired fashion. Post-processing software that allows the measurements to be analyzed instantly following the measurement will also be described and demonstrated.

2:40–3:00 Break

3:00

1pAAa6. Large-scale multiple input/multiple output system identification in room acoustics. Martin Schneider and Walter Kellermann (Multimedia and Signal Processing, Univ. Erlangen Nürnberg, Cauerstr.7, Erlangen, Germany, schneider@LNT.de)

In audio reproduction scenarios, room acoustics may be described as a MIMO system response from multiple loudspeaker to multiple microphones in the listening space. This system response may, e.g., be used for an equalization of the listening room and must be identified from observing the available loudspeaker and microphone signals in real-world systems. For few transducers this task is mostly solved, but massive multichannel reproduction with dozens to hundreds of loudspeakers left many research questions open. This contribution points out the fundamental challenges, previous solutions and recent advances. As a key issue, the so-called nonuniqueness problem for MIMO system identification by adaptive filtering will be discussed along with decorrelation schemes for the loudspeaker signals to alleviate this problem. Successful adaptation algorithms suitable for these scenarios imply considerable computational demands and require additional measures to ensure robustness. Recently emerging system models in spatial transform domains allow for approximative models and seem to be promising for robust real-time implementations.

3:20

1pAAa7. Robustness analysis of room equalization for soundfield reproduction within a region. Dumidu S. Talagala, Wen Zhang, and Thushara D. Abhayapala (Res. School of Eng., College of Eng. and Comput. Sci., Australian National Univ., Canberra, ACT 0200, Australia, thushara.abhayapala@anu.edu.au)

Recent works on soundfield reproduction have presented several methods of recreating a desired soundfield within a region. Estimation or prior knowledge of the inverse reverberant channels now becomes an essential element of equalizing the room effects. However, it has been shown that designing point-to-point equalizers by sampling the reverberant soundfield is only practical within a few tenths of

a wavelength of the sampled locations. This work investigates the robustness of the equalization process applied to a region, with respect to changing of actual microphone positions from their expected locations. We use a modal description of the equalized soundfield to obtain theoretical results for region equalization error due to positioning errors. Simulation results suggest that equalizing the reverberant soundfield recorded at multiple positions around the edge of the reproduction region is more immune to the positioning errors.

Contributed Papers

3:40

1pAAa8. A numerical and experimental validation of the room acoustics diffusion theory inside long rooms. Chiara Visentin, Nicola Prodi (Dipartimento di Ingegneria, Università di Ferrara, via Saragat 1, Ferrara 44122, Italy, chiara.visentin@unife.it), Vincent Valeau (Institut PPRIME, CNRS-Université de Poitiers-ENSMA, Poitiers, France), and Judicäel Picaut (IFSTTAR, LUNAM Université, Bouguenais, France)

The paper focuses on the validation of the recently proposed room-acoustics diffusion theory by means of numerical simulations and experimental measurements. The analysis aims to verify the equation underlying the theory (Fick's law of diffusion) which relates the energy density gradient and the sound intensity inside a room through a constant diffusion coefficient. In this work, the acoustic quantities are numerically/experimentally derived under stationary conditions, and their ratio is employed to estimate the effective value of the diffusion coefficient inside long rooms. The numerical study was carried out with particle-tracing simulations. The measurements were performed with a Microflown® three-dimensional sound intensity probe inside a 1:16 scale model of a long room, varying the absorption and the scattering properties at the boundaries. A comparison between numerical and experimental results is carried out with a least-square algorithm, showing a fair agreement between the diffusion coefficients estimated with the two methods. The results lead to the conclusion that the reverberant sound field inside long rooms can be described by a non-homogeneous diffusion process: the local diffusion coefficient is not a constant inside the room but increases with the distance from the source and depends on the acoustical properties of the room boundaries.

4:00

1pAAa9. Energy evolution in enclosure geometries as exhibited by a finite difference time domain method. Zackery Belanger (53 3rd St, Troy, New York 12180, zb@archgeometer.com)

Measurements and simulations conducted for the purpose of extracting or constructing impulse responses are inextricably dependent on a limited number of receiver locations. Wave-based simulations offer the opportunity to discard this dependency and assess an entire evolving sound field. In this work, a finite difference time domain method is implemented to simulate an evolving sound field in a range of enclosure geometries with reflective boundaries. The distribution of energy is monitored statistically as mixing and ergodic states are approached, and evidence is presented for the predictability of this evolution based on room geometry alone. A system for importing geometries from the Rhinoceros 3D CAD software, using the Grasshopper parametric environment, is also presented. A form known as the Barnett Billiard Table in mathematics is included in this study, which exhibits exceptional behavior even without the presence of diffusive treatment.

MONDAY AFTERNOON, 3 JUNE 2013 — 513DEF, 12:55 P.M. TO 3:20 P.M.

Session 1pAAb

Architectural Acoustics and Musical Acoustics: Vibration in Music Performance

Clemeth Abercrombie, Cochair
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M. Ercan Altinsoy, Cochair
Chair of Commun. Acoust., Dresden Univ. of Technol., Helmholtzstr. 18, Dresden 01062, Germany

Chair's Introduction—12:55

Invited Papers

1:00

1pAAb1. Perceptual evaluation of violin vibrations and audio-tactile interaction. M. Ercan Altinsoy, Sebastian Merchel, and Sebastian Tilsch (Chair of Commun. Acoust., Dresden Univ. of Technol., Helmholtzstr. 18, Dresden 01062, Germany, ercan.altinsoy@tu-dresden.de)

When playing a violin, the musician communicates with his instrument not only through his ears but also his fingers, chin, shoulder, and eyes. He uses different sensory inputs, which are provided by different sensory channels, such as auditory, tactile, kinesthetic, and visual, to play his musical instrument. The perceived vibrations are useful for the player to feel and to control the instrument. The interaction between sound and vibration plays also a role on the overall instrument perception. In this study, violin vibrations and their interaction with violin sounds were evaluated. Therefore, the vibration amplitudes of the neck and the violin sounds were recorded simultaneously during normal playing. The vibration recordings were analyzed, and then additional stimuli were generated by filtering or modifying frequency components. In the first experimental session, the vibration stimuli, which were presented to the subjects via a mini electrodynamic shaker, were evaluated. In the second experimental session, an investigation with multimodal (auditory-haptic) stimuli was conducted. The results show the importance of vibrations on the overall perception of the instrument and provide information on useful vibration features for the player-instrument interaction.

1:20

1pAAb2. Telehaptic interfaces for interpersonal communication within a music ensemble. Jonas Braasch, Pauline Oliveros, and Doug Van Nort (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Visual communication is an important aspect of music performance, for example, to pick up temporal cues and find the right entries. Visual cues can also be instrumental to negotiate the solo order in improvized music or enable social exchange, for example, by signaling someone that her solo was well received. The problem with visual communication is that one has to catch someone else's attention, and visual cues outside someone's visual field cannot be detected, even more so if the addressee is busy reading a music score or closing his eyes in a Free Music session. Acoustic communication does not encounter these challenges, but of course someone does not want to disturb the music with other acoustic signals. The haptic modality has the advantage that it does not necessarily interfere with the acoustic signal and does not require attention. However, it allows interpersonal communication if both parties are within close proximity. Using telematic interfaces solves the problem of proximity by allowing participants to communicate over any physical distance. In the project presented here, haptic interfaces were explored in connection with an intelligent music system, CAIRA, to examine both the effect of human/machine and inter-human communication. [Work supported by the National Science Foundation, No. 1002851.]

1:40

1pAAb3. Stage floor vibrations and bass sound in concert halls. Anders Askenfelt (Dept. of Speech, Music and Hearing, KTH Royal Inst. of Technol., Lindstedtsvägen 24, Stockholm 10044, Sweden, andersa@csc.kth.se) and Knut Guettler (Norwegian State Acad. of Music, Jar, Norway)

The double bass and cello sections in the orchestra transmit vibrations to the stage floor through the end pins. Whether or not these vibrations may contribute to the perceived sound in the hall has been investigated since the 1930s. In this study, the conditions for an efficient transfer of instrument vibrations to the floor, as well as the radiation from the floor to the audience area, are investigated. The study includes measurements of the impedance matching between bass and stage floor, the vibration velocity transfer to the floor via the endpin, and radiation from point-driven bending waves in the stage floor well below the coincidence frequency. The impedance conditions and radiation properties for the stage floors of five concert halls were investigated. In the most promising hall, a full-scale experiment was run with an artificially excited double bass supported via the end pin on the stage floor, and on a concrete support below, respectively. The contribution from the stage floor radiation to the sound level in the audience area was 5 dB or more between 30 and 60 Hz. This range covers the fundamental frequencies over one octave starting from the lowest note (B0) of a five-string bass.

2:00

1pAAb4. Recent experiences with vibration of stage and audience floors in concert halls. Thomas Wulfrank (Kahle Acoust., Ave. Moliere 188, Brussels 1050, Belgium, thomas.wulfrank@kahle.be), Igor Lyon-Caen (Alstom Transport TGS, Saint-Ouen, France), Yann Jurkiewicz, Johan Brulez, and Eckhard Kahle (Kahle Acoust., Brussels, Belgium)

Vibration of (wood) surfaces plays a significant role in concert hall acoustics, as confirmed by musicians and music lovers. Many acoustic engineers, on the other hand, tend to have strong reservations against vibrating surfaces, and usually try to minimize surface vibration in order to maximize RT and airborne strength (G) at bass frequencies. This has led to a generally accepted preference for massive and stiff surface constructions in new halls. Problems have been known to occur when this general guideline was also applied to the design of wooden floors, in particular stage floors. Despite some good scientific research in this field, a big gap still remains between the vibro-acoustic behavior of wooden floors and subjective preferences of musicians and audiences. This paper further explores the role of vibrations in concert hall design, and the need for balancing surface reflectivity versus vibration transmission. Recent experiences, including the new Konserthus in Stavanger and the renovation of the Bolshoi Hall of the Moscow Conservatory, will be described as well as vibration measurements carried out on a number of existing stage floors. Some implications for the design of wooden floor constructions will be discussed.

Contributed Paper

2:20

1pAAb5. Auditory-tactile music perception. Sebastian Merchel and M. Ercan Altinsoy (Chair of Commun. Acoust., TU Dresden, Helmholtzstr. 18, Dresden 01069, Germany, sebastian.merchel@tu-dresden.de)

The coupled perception of sound and vibration is a well-known phenomenon during live pop or organ concerts. However, even during a symphonic concert in a classical hall, sound can excite perceivable vibrations at the body surface. However, the concert visitor might not be aware of those vibrations, because the tactile percept is integrated with the other senses into one multi-modal percept. This article discusses the influence of whole-body vibrations on the listener experience during the reproduction of concerts recordings. Four sequences were selected from classical and modern music, which include low frequency content (e.g., organ, kettledrum, contrabass). A stimulus length of 1.5 min was chosen in order to provide enough time for habituation. The audio signal was reproduced using a surround setup. Additional seat vibrations have been generated from the audio signal. Test participants were asked to rate the overall quality of the concert experience. The results show that vibrations have a significant influence on our perception of music. This finding is interesting in the context of audio reproduction, but also for the construction of concert venues.

2:40–3:20 Panel Discussion

Session 1pAB

Animal Bioacoustics: Sound Generation and Perception in Animals

Nancy Allen, Chair
Defence R&D Canada - Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada

Contributed Papers

1:00

1pAB1. Acoustic communication in crocodiles: How do juvenile calls code information? Nicolas Mathevon, Amélie Vergne (ENES Lab CNRS UMR8195, Univ. of Lyon/Saint-Etienne, 23 rue Michelon, Saint-Etienne 42023 cedex 2, France, mathevon@univ-st-etienne.fr), and Thierry Aubin (Equipe Communications Animales CNPS CNRS URM 8195, Univ. Paris XI-Orsay, Orsay, France)

In spite of its importance for the understanding of the evolution of sound communication, information concerning the vocal world of crocodilians is limited. Experimental works have brought evidence of the biological roles of juvenile sound signals, with "hatching calls" eliciting care by the mother and synchronizing clutch hatching, "contact calls" gathering groups of juveniles, and "distress calls" inducing maternal protection. Recently, we investigated the question of species-specific information coding within juvenile calls. The analysis of signal acoustic structure shows inter-specific differences between calls. However, using playback experiments, we bring the evidence that these differences are not relevant to animals, either juveniles or adults. By using calls modified in the temporal and the frequency domains, we isolate the acoustic cues necessary to elicit a behavioral response from receivers, underlying the importance of the frequency modulation slope. Considering previous results underlying the absence of information about individual identity in juvenile calls, we make the hypothesis that these signals basically support a "crocodilian" identity.

1:20

1pAB2. Design and field test of a low-cost-portable linear array for marine mammal localization. Omar A. Bustamante (ESIME, Austral #77 Col. Atlanta, Cuautitlan Izcalli 54740, Mexico, omarb.p@hotmail.com), Eduardo Romero Vivas (CIBNOR, La Paz, BCS, Mexico), and Sergio Beristain (ESIME, IPN,IMA, Mexico, D. F., Mexico)

Marine mammals are reliable bioindicators of aquatic systems health. Within this group, cetaceans are well known by their high dependence on sound for many of their vital activities such as socialization and mating, prey catching, and navigation. Due to its high dependence on sound, bioacoustic methods become very important for the study of these species. Acoustic monitoring in the field is usually performed using omnidirectional hydrophones to assess the presence of mammals, but for some behavioral studies, it is also important to locate the animals, something which is not possible with that arrangement. Although there are very well known techniques to detect the direction of arrival of the sound, the equipment required is highly specialized and expensive. In this paper, the design and field test of digital and analog versions of a portable linear array of hydrophones capable of locating animal sounds by beamforming, using low cost and easily available equipment is presented. The array was tested in La Paz bay, Mexico, by experts of the Marine Mammals Research Program of the University of Baja California Sur, which were able to locate dolphins (*Tursiops truncatus*) only by their sound, despite strong sources of noise in the area.

1:40

1pAB3. Auditory reaction time measurements and equal-latency curves in the California sea lion (*Zalophus californianus*) and bottlenose dolphin (*Tursiops truncatus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmpfoundation.org) and James J. Finneran (U.S. Navy Marine Mammal Program, SSC Pacific, San Diego, CA)

Subjective equal-loudness contours are used to create weighting functions for human noise-mitigation criteria. Comparable direct measurements of subjective loudness with animal subjects are, however, difficult to conduct. Using methods similar to those used in previous mammalian studies, this study estimated subjective loudness through the measurement of response time (RT) in an auditory signal-detection task. Measurements were conducted in a sound-attenuating hut with a California sea lion and under water in a quiet pool with a bottlenose dolphin. Tonal stimuli were presented at supra- and near-threshold sound pressure levels (SPLs) using a method of constants. Median RT increased with decreasing SPL for both species across all tested frequencies. A Piéron function, which models RT as a function of SPL, was fitted to the RT-SPL curves in a nonlinear fashion. Equal-latency curves were based on the Piéron functions at each frequency. Preliminary results for the sea lion suggest that the equal-latency curves are similar to the audiogram at longer median RTs (~300 ms), with increasing deviation from the audiogram at the fastest median RTs (~200 ms). Continued testing with additional subjects will provide further data for designing marine mammal auditory weighting functions. [Funded by U.S. Navy Living Marine Resources Program.]

2:00

1pAB4. Methods for determining free-swimming positioning and echolocation beam patterns. Danielle Greenhow (College of Marine Sci., Univ. of South Florida, 140 7th Ave. S, St. Petersburg, FL 33701, dgreenho@mail.usf.edu), Heidi Harley (Div. of Social Sci., New College, Sarasota, FL), Wendi Fellner (The Seas, Epcot, Animal Programs, Walt Disney World Resort, Lake Buena Vista, FL), Adrienne Cardwell (Mote Marine Lab., Sarasota, FL), and David Mann (College of Marine Sci., Univ. of South Florida, Sarasota, FL)

Echolocation beam patterns were recorded using a 25-element autonomous, self-contained hydrophone array during free-swimming echoic match tasks. Two GoPro HERO2 cameras mounted at 45-degree angles to the plane of the 4.5×4.5 ft PVC array recorded the position of the dolphin during approach and investigation of the sample object presented in front of the array. Using a pre-session calibration file and MaxTRAQ3D software, analysis of video determined exact positioning of the head region during echolocation, including distance from the array and angle to the plane of the array. An LED and ITC-1042 transducer, operated by an Arduino Uno processor board and a Hafler amplifier, were mounted on the array. Simultaneously emitted visual pulses (LED) and acoustic pulse trains were used for synchronization of the acoustic recorders and video cameras. The distinct pulse train was used during post-recording analysis to synchronize the hydrophones using a custom-designed MATLAB routine. The hydrophones are sampled at 400 kHz and analyzed in real-time. The maximum and minimum values and time stamps within a 0.32 ms window are stored continuously to reduce the amount of data that needs to be stored. Using the 3-D positioning of the animal, dynamic beam patterns can be reconstructed.

2:20

1pAB5. Threshold of hearing for swimming Bluefin tuna (*Thunnus orientalis*). Arthur N. Popper (Dept. of Biol., Univ. of Maryland, College Park, MD, apopper@umd.edu), Jonathan Dale (Tuna Res. and Conservation Ctr., Hopkins Marine Station, Stanford Univ., Pacific Grove, CA), Michael D. Gray (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), William Keith (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI), Barbara A. Block (Tuna Res. and Conservation Ctr., Hopkins Marine Station, Stanford Univ., Pacific Grove, CA), and Peter H. Rogers (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Hearing thresholds for three pairs of 1 m long Pacific bluefin tuna (*Thunnus orientalis*) were measured utilizing operant conditioning procedure with a food reward and a staircase psychophysical technique. Fish, swimming at 1–4 m/s, quickly learned to approach the feed when they heard a sound. Measurements were made at the Tuna Research and Conservation Center (Stanford University) in a 9.14 m diameter, 1.65 m deep indoor cylindrical tank. The acoustic stimulus was produced by radially oriented piezoelectric line sources centered at the bottom of the tank, which produced a circumferentially uniform sound field. The acoustics of the tank was thoroughly characterized for both acoustic pressure and particle motion using hydrophones and two neutrally buoyant accelerometers with response axes oriented in the radial and vertical directions. Thresholds, expressed in terms of pressure and particle acceleration, were obtained at six sinusoidal frequencies between 325 Hz and 800 Hz, a range that was limited by source and tank acoustics. The lowest mean threshold for the three fish pairs, expressed in terms of acoustic pressure, was 83 dB re 1 μPa at 500 Hz. [Work supported in part by ONR/CNR Challenge Grant: "Mitigation of flow noise effects by fish."]

2:40

1pAB6. Prey pursuit strategy of Japanese horseshoe bats, *Rhinolophus ferrumequinum Nippon*, during target selection task. Yuki Kinoshita, Daiki Ogata (Faculty of Life and Med. Sci., Doshisha Univ., 1-3 Miyakotani Tatara, Kyotanabe 610-0321, Japan, dmm1014@mail4.doshisha.ac.jp), Ikkyu Aihara (Brain Sci. Inst., RIKEN, Wako, Japan), Yoshiaki Watanabe, Hiroshi Riquimaroux, Tetsuo Ohta, and Shizuko Hiryu (Faculty of Life and Med. Sci., Doshisha Univ., Kyotanabe, Japan)

We investigated prey pursuit behavior of Japanese horseshoe bats, while they were tasked to make a choice between two tethered fluttering moths during flight. Echolocation pulses were recorded by a telemetry microphone mounted on the bat, combined with a 17-ch horizontal microphone array to measure pulse directions. Flight paths of the bat and moths were monitored by using two high-speed video cameras. Acoustical measurements of CF echoes from fluttering moths (67 kHz : CF2 frequency) was conducted using an ultrasonic loudspeaker, turning the head direction of the moth to the loudspeaker from 0° to 180° in the horizontal plane. Amount of acoustical glints caused by moth fluttering varied with the sound direction, showing the maximum between 70° and 100°. In the flight experiment, moths chosen by the bat fluttered within or moved across these angles to the bat's pulse direction, which would cause dynamic changes in frequency and amplitude of acoustical glints during flight. This result suggests that dynamic changes in acoustical glints appear to attract the bats for prey selection. Furthermore, mathematical modeling implied that the bats possibly took the optimum flight path for capturing a target, which the bat selected based on the acoustical cues in the echoes.

3:00

1pAB7. The alignment problem for bat biosonar beampatterns. Philip Caspers (Dept. of Mech. Eng., Virginia Tech, 1110 Washington St., Blacksburg, VA 24061, pcaspers@vt.edu), Rongjiang Pan (School of Comput. Sci. & Technol., Shandong Univ., Jinan, China), Alexander Leonessa, and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA)

Unlike the beampatterns of technical acoustical systems, the biosonar beampatterns of bats are highly variable in the shapes of their main- and sidelobes over frequency. Some of this variability could represent adaptations to different sensing tasks. In order to understand such possible adaptations, a quantitative method for the analysis of variability (e.g., principal component analysis) is needed. Since the orientation of biosonar beampatterns is highly variable *in-vivo*, e.g., due to ear/head movements, and not preserved in isolated noseleaf/ear samples, orientation is left out of the initial analysis. Instead, beampatterns should be aligned to characterize their orientation-independent features. For this purpose, a framework to characterize the beampattern alignment problem and perform the alignment has been drawn up. For each frequency, beampatterns are compared using a distance metric (e.g., a p-norm). By investigating the value of this distance metric over the space of all possible beampattern rotations, it is possible to gain insights into the alignment problem, e.g., with regard to the existence of multiple minima in the metric. This space can also be used to test alignment strategies across multiple frequencies, e.g., through a weighted sum of the respective distances.

3:20

1pAB8. Effects of the source location on numerical biosonar beampattern predictions for bat noseleaves. Anupam Kumar Gupta and Rolf Müller (Dept. of Mech. Eng., Virginia Tech, 1110 Washington St., Blacksburg, VA 24061, anupamkg@vt.edu)

Approximately 300 bat species are known to emit their ultrasonic biosonar pulses through the nostrils. In these animals, ultrasound is produced by the larynx, propagates along the vocal tract, exits through the nostrils, and is finally diffracted by intricately shaped baffle structures known as "noseleaves." Noseleaf geometry determines diffraction and hence the spatial distribution of the emitted ultrasound. As a consequence, numerical predictions of the noseleaves' acoustic function can be made based on the digital models of noseleaf shape. To limit model size and computational effort associated with numerical beampattern predictions, the vocal tract is often only partially included in these models or left out completely. In order to investigate the effect of source placement within a complete or partial vocal tract attached to a noseleaf shape on the numerical beampattern prediction, the noseleaf of the Great Roundleaf Bat (*Hipposideros armiger*) was studied as a model. Numerical beampattern estimates were obtained for a single monopole source positioned near the vocal folds or closer to the nostrils. Two monopoles sources placed in each nostril were also investigated. It was found that source positioning could impact the beampattern whenever they broke the symmetry in the near-fields of the two nostrils.

3:40

1pAB9. Investigation of acoustic gaze strategy by *Pipistrellus abramus* and *Rhinolophus ferrumequinum Nippon* during obstacle avoidance flight. Yasufumi Yamada, Arie Oka, Shizuko Hiryu, Tetsuo Ohta, Hiroshi Riquimaroux, and Yoshiaki Watanabe (Faculty of life and Med. Sci., Doshisha Univ., 1-3, 1-3, Miyako-tani, Tatara, Kyotanabe,Kyoto 610-0394, Japan, dml0142@mail4.doshisha.ac.jp)

We investigated the acoustic gaze (angle between pulse and flight directions) and the beam width of echolocation pulses emitted by bats during obstacle avoidance flight in a chamber (7×3×2 m). Echolocation pulses were recorded by a telemetry microphone mounted on the bat's back and a horizontal 20-ch microphone array set up in the chamber. Flight path measurements were conducted using two high-speed video cameras. While the bat showed a circular flight avoiding the surrounding walls, the acoustic gaze showed significant linear correlation to the angular velocity of the flight. This means that the bat adjusted the pulse direction to precede its own flight direction. The correlation coefficient increased with complexity of obstacle conditions. We compared changes in acoustic gaze between FM bats (mean beam width: ±50°) and CF-FM bats (mean beam width: ±22°). We found that FM bats smoothly shifted the acoustic gaze within 10° during the flight whereas the CF-FM bats frequently shifted the acoustic gaze within 25°. These results indicate that the shifting acoustic gaze by CF-FM bats compensates their own narrow beam width. Both bat species may keep approximately ±50° of their echolocation sights in order to sense the space for moving forward safely.

4:00

1pAB10. Buckling as a source of sound, with application to the modeling of cicada sound generation. Allan D. Pierce (Woods Hole Oceanogr. Inst., P.O. Box 339, East Sandwich, MA 02537, adp@bu.edu), Derke Hughes (Naval Undersea Warfare Ctr., Newport, RI), Kossi Edoh (North Carolina A&T Univ., Greensboro, NC), Richard A. Katz, and Robert M. Koch (Naval Undersea Warfare Ctr., Newport, RI)

A basic model of a ribbed finite plate is first considered, with the plate connected to a parallel surface by a nonlinear spring. When individual ribs are placed under compression, the linearized version of the model predicts eventual exponential growth of the transverse displacement when the compressional load exceeds the buckling load. The nonlinear spring, however, stops this growth and a subsequent oscillation ensues. The anatomy of the cicada is, of course, much more complicated, and the basic model is extended to give a mathematical formulation of the model proposed by Bennet-Clark and Daws (J. Exper. Biol. 1999) for the *Cyclochila australasiae* (a relatively large species of cicada), with explicit elasto-mechanical parameters, a principal objective being the computational prediction of the observed far-field waveforms. Possible explanations are advanced for the anatomical cause of the nonlinear springs. Further extensions of the theory are applied to the development of mathematical models for species of cicadas (considerably smaller) commonly found in the United States.

4:20

1pAB11. Nature of nonlinear mechanisms in the generation and propagation of sound in the cicada mating call. Derke Hughes (NUWC-DIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@verizon.net), Kossi Edoh (North Carolina A&T State Univ., Greensboro, NC), and Allan Pierce (Woods Hole Oceanogr. Inst., Thalmouth, MA)

Experiments and analyses (Hughes *et al*., J. Acoust. Soc. Am. 2009) on the relationship of body surface displacements with external acoustic pressure measurements while cicadas are generating mating calls resulted in the conclusion that the relationship is substantially nonlinear. The present analysis assesses whether the propagation through the air is nonlinear propagation. Suspicion that such might be the case is suggested by the fact that the sound levels at distances of the order of several meters from a sounding cicada are as high as 90 dB at distances of the order of several meters. Computational results are reported in the present paper that indicate that nonlinear effects are not important for propagation of typical cicada frequencies of 10 kHz over a radial distance interval beginning 2 cm and ending at 10 m. A suggested explanation is based on the observation that the displacement of the body surface varies with position over the surface. The time history of displacements at different points is not as repetitive during successive tone burst generations as is the total volume displacement of the surface. This is consistent with the observation that buckling of the long ribs in the tymbals is the ultimate cause of the surface displacement vibrations.

4:40

1pAB12. Giraffe Helmholtz resonance. Elizabeth vonMuggenthaler and Meredith Bashaw (Fauna Commun. Res. Inst., 818 Lake Orange Rd., Hillsborough, NC 27278, L@animalvoice.com)

Many animal species, including elephants and okapi, use sounds above and below the range of human hearing to communicate. A longitudinal study presented here suggests giraffe produce infrasonic vocalizations using Helmholtz resonance. Recordings were made of giraffe (*Giraffe camelopardalis*) in controlled indoor conditions and naturalistic outdoor conditions. The portable recording and analysis system consisted of a trigger oscilloscope; DAT recorders; Nagra IV-SJ; and computers. Each signal was low-pass/high-pass filtered, and FFT and STFT were performed using PolynesiaTM real-time scrolling analysis. In controlled recordings and in naturalistic situations, two types of signals were identified: audible bursts [(11 Hz (75 dB +/− 3) to 10,500 Hz (80 dB +/− 3)] dominant frequencies between 150 and 200 Hz and covert vocalizations [(14 Hz (60 dB +/− 3) to 250–275 Hz (30 dB +/− 3)] dominant frequencies between 20 and 40 Hz. Both audible and covert signals coincided with neck throw or head toss behaviors. The shape of the giraffe's respiratory apparatus during this behavior and the frequencies produced implicate Helmholtz resonance as a production mechanism. In naturalistic recordings, two of the five infrasonic vocalizations identified were produced during close range social interactions, suggesting that giraffe use these vocalizations to communicate with con-specifics. The social functions, air and seismic transmission mechanisms of these vocalizations should be further assessed.

MONDAY AFTERNOON, 3 JUNE 2013 — 518C, 1:35 P.M. TO 3:40 P.M.

Session 1pBAa

Biomedical Acoustics: Acoustic Microscopy: Biomedical Applications

John S. Allen, Chair
Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822

Chair's Introduction—1:35

Invited Papers

1:40

1pBAa1. An all-optical photoacoustic microscopy system for remote, noncontact characterization of biological tissues. Ashwinkumar Sampathkumar, Parag V. Chitnis (Biomedical Eng., Riverside Res., 156 William St., #9, New York, NY 10038, asampathkumar@riversideresearch.org), and Ronald H. Silverman (Dept. of Ophthalmol., Columbia Univ. Medical Ctr., New York, NY)

Conventional photoacoustic microscopy (PAM) employs light pulses to produce a photoacoustic (PA) effect and detects the resulting acoustic waves using an ultrasound transducer acoustically coupled to the target tissue. The resolution of conventional PAM is limited by the sensitivity and bandwidth of the ultrasound transducer. We have investigated an all-optical, "pump-probe" method employing interferometric detection of the acoustic signals that overcomes limitations of conventional PAM. This method does not require contact with the specimen and provides superior resolution. A 532-nm "pump" laser with a pulse duration of 5 ns excited the PA effect in tissue. Resulting acoustic waves produced surface displacements that were sensed interferometrically with a GHz bandwidth using a 532-nm CW "probe" laser using a Michelson interferometer. The pump and probe beams were coaxially focused using a 50× objective giving a diffraction-limited spot size of 0.5 μm. The phase-encoded probe beam was demodulated using a homodyne interferometer. The detected time-domain signal was time reversed using k-space wave-propagation methods to produce a spatial distribution of photoacoustic sources in the target tissue. Performance was assessed using 3D images of fixed, *ex vivo*, retina specimens. Apparatus design and imaging results for the all-optical PA system and possible applications will be discussed.

2:00

1pBAa2. Multimodal ultrasound microscopy for biomedical imaging. Yoshifumi Saijo (Grad. School of Biomedical Eng., Tohoku Univ., 4-1 Seiryomachi, Aoba-ku, Sendai 980-8575, Japan, saijo@idac.tohoku.ac.jp)

Acoustic microscopy provides not only high resolution imaging but also basic data for interpreting clinical ultrasound images and information on biomechanics of the tissues. Multimodal ultrasound microscope is developed for quantitative measurement of sound speed of the tissue. The frequency dependent characteristics of the amplitude and phase of a single pulse deduce the tissue thickness and sound speed. Specific acoustic impedance and elastic bulk modulus are derived by the sound speed and density of the tissue. Ultrasound impedance microscope visualizes microscopic image of the tissue surface by just touching the probe to the tissue. The reflection from the interface between the tissue and plastic plate is obtained to visualize two-dimensional distribution of specific acoustic impedance of the tissue. The multimodal ultrasound microscope realized conventional C-mode, surface impedance mode, B-mode, 3D mode, and combination of photoacoustic imaging. The series of the ultrasound measurements of gastric cancer, renal cancer, prostatic cancer, myocardial infarction, atherosclerosis, cartilage-bone complex, and brain have provided important information for clinical ultrasound imaging and pathophysiology from the point of view of biomechanics. Development of higher frequency transducer or arrayed transducer with newest technologies would realize higher resolution imaging and easier handling.

2:20

1pBAa3. Speed of sound of fatty and fibrosis liver measured by 80-MHz and 250-MHz scanning acoustic microscopy. Tadashi Yamaguchi (Res. Ctr. for Frontier Med. Eng., Chiba Univ., 1-33 Yayoicho, Inage, Chiba 2638522, Japan, yamaguchi@faculty.chiba-u.jp), Kenta Inoue (Grad. School of Eng., Chiba Univ., Chiba, Japan), Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), Kazuto Kobayashi (Honda Electron. Co., Ltd., Toyohashi, Aichi, Japan), and Yoshifumi Saijo (Grad. School of Biomedical Eng., Tohoku Univ., Sendai, Miyagi, Japan)

Early detection of hepatitis is critical for proper patient management and improving disease prognosis. Ultrasound imaging is ideally suited for early-stage assessments, but conventional ultrasound images based on backscatter do not display quantitative tissue information because conventional ultrasound lacks essential modeling of the complex interaction between ultrasound and liver tissue in normal and diseased states. Therefore, speed-of-sound (SOS) measurements were obtained from three types of rat livers (normal, fatty, and fibrosis). Livers were harvested, fixed, and embedded in paraffin; a single 10-μm thin section was obtained using a microtome and placed on a microscope slide. A scanning acoustic microscope incorporating transducers operating at 80- and 250-MHz center frequencies was used to scan the 10-μm section. An adjacent 4-μm thin section was stained with H&E (normal and fatty livers) or Azan (fibrosis livers). The SOS measured with both transducers displayed the same trend: SOS in fatty liver was lower than in normal liver and SOS in fibrosis liver was higher than in normal liver. SOS differences were greater at 250 MHz because of the improved spatial resolution, which allowed choosing region-of-interests containing only fat or fibrosis tissue. These initial results also were used to correlate the pathologic state with the SOS.

2:40

1pBAa4. Sound speed estimation in single cells using the ultrasound backscatter power spectrum. Eric M. Strohm and Michael C. Kolios (Physics, Ryerson Univ., 350 Victoria St, Toronto, ON M5B2K3, Canada, estrohm@ryerson.ca)

The acoustic properties of single cells such as their size, sound speed and attenuation are known to change depending on the type, state, or disease progression of a cell. Typically, ultrasonic pulse echo methods on adherent cells are used. We propose using the ultrasound backscatter power spectrum on cells in suspension to extract the acoustic parameters. When the ultrasound wavelength is on the same order as the dimensions of the cell, periodically varying minima and maxima occur throughout the power spectrum that depend on the sound speed and density of the object and the surrounding fluid, respectively. The ultrasound parameters can be determined by comparing the measured spectrum to a theoretical scattering model. We measured the backscattered ultrasound signals from single MCF7 breast cancer cells in suspension using a 200 MHz transducer. The cell diameter was determined through simultaneous optical imaging. The sound speed was calculated by adjusting the parameters in the scattering model until a good fit of the spectral features between the model and measured agreed. The sound speed from single cells found to vary between 1540 to 1580 m/s when the density was fixed at 1050 kg/m^3.

3:00

1pBAa5. Ultrasonic biomicroscopy and micro-Raman spectroscopy for the mechanochemical characterization of atheromatous lesions. Pavlos Anastasiadis (Molecular Biosciences and Bioengineering, Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, pavlos@hawaii.edu), Shiv Sharma (Hawaii Inst. of Geophys. & Planetology, Univ. of Hawaii, Honolulu, HI), Michelle L. Matter (John A. Burns School of Med., Univ. of Hawaii, Honolulu, HI), and John S. Allen (Mech. Eng., Univ. of Hawaii, Honolulu, HI)

Atherosclerosis is defined as a focal, inflammatory, and fibro-proliferative response to endothelial injury. The development of atheromatous lesions in the coronary tree is predominantly a quiescent asymptomatic process without any clinical manifestations. The unpredictable and acute nature of cardiovascular complications such as vulnerable plaque rupture makes diagnosis and treatment of this disease an outstanding medical challenge. We investigate non-invasive techniques that facilitate mechanical measurements at the microscopic level, which can then be directly correlated to biomarker localization within lesion sites. To characterize these sites *in vitro*, we used time-resolved scanning acoustic microscopy (TRSAM). This technique allows for non-invasive interrogation of tissue samples with optical resolution at the micrometer scale. Furthermore, we combined TRSAM with micro-Raman (micro-RS) spectroscopy to investigate plaque morphology with regard to specific biomarkers. We characterized mechanoelastic and biochemical regions containing high cholesterol, phosphate, and carbonate apatite that are characteristic of atherosclerotic lesions. The mechanoelastic evaluation of these regions was determined using TRSAM. Calcified lesions, for example, exhibit ultrasonic velocities of 1810 m/s $\pm$ 25 m/s and are more rigid and stiffer than normal blood vessel tissues.

Contributed Paper

3:20

1pBAa6. An analysis of the acoustic properties of the cell cycle and apoptosis in MCF-7 cells. Maurice M. Pasternak, Eric M. Strohm, and Michael C. Kolios (Physics, Ryerson Univ., 10 Torresdale Ave. B-1, Toronto, ON M2R3V8, Canada, emilku@hotmail.com)

Through the use of high frequency acoustic microscopy, the acoustic properties of cells through various stages of interphase (G1/G2), mitosis (metaphase, M-phase), and apoptosis were ascertained. The cell thickness, sound velocity, acoustic impedance, density, bulk modulus, and attenuation were determined through a quantitative analysis of the pulse echoes from the cell membrane and substrate using a 375 MHz transducer. Hoechst 33342, Annexin-V, propidium iodide, and FITC-448 Anti-cyclin B1 and D1 mouse antibodies were used to identify cell cycle stage. ANOVA and Tukey *post hoc* statistical tests were used to quantify differences between cell stages. A total of 174 cells, 58 within each category, were measured. A statistically significant increase in thickness (9.4–11.4 μm), and decrease in attenuation (1.20–1.05 dB/cm/MHz) was observed between G1 and G2 cells, respectively. A statistically significant increase in thickness, and decrease in acoustic impedance, density, bulk modulus, and attenuation was observed between M-phase and G1 or G2. During apoptosis, minor differences were observed between interphase and early apoptosis; however, significant differences in nearly all properties were observed as the cells progressed to late stage apoptosis. The differences found indicate considerable structural and/or organizational alterations occurring as the cell progresses through these phases.

MONDAY AFTERNOON, 3 JUNE 2013 — 519A, 1:00 P.M. TO 5:20 P.M.

Session 1pBAb

Biomedical Acoustics: Ultrasound Contrast Agents and Passive Cavitation Mapping of High Intensity Focused Ultrasound Lesion Formation

Eleanor Stride, Chair
Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom

Contributed Papers

1:00

1pBAb1. Acoustic characterization and modeling of poly-lactic acid-encapsulated contrast microbubbles. Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu) and Shirshendu Paul (Mech. Eng., Univ. of Delaware, Newark, DE)

Biodegradable polymers like polylactic acid hold potential for better stability and control over encapsulation properties of ultrasound contrast microbubbles. We report here several interesting acoustic properties of air-filled PLA shelled microbubbles through both *in vitro* experiments and mathematical modeling. Attenuation measurements with PLA microbubbles (average diameter 1.9 micrometer), indicated a resonance frequency of 2.5–3 MHz, which, in contrary to other encapsulated microbubbles, is lower than the resonance frequency of a free bubble of similar size. Pressure dependent scattering measurements at two different excitation frequencies (2.25 and 3 MHz) show strongly non-linear behavior with distinct second and subharmonic responses. Subharmonic responses are registered above a relatively low generation threshold of 100–150 kPa. To investigate the underlying mechanisms, we utilized several preexisting interfacial models describing encapsulated bubble dynamics. The attenuation data were utilized to determine the interfacial rheological properties of the encapsulation for each of these models. The model predictions are then compared with scattered nonlinear—sub- and second harmonic—responses. Our studies indicate that the extremely low surface elasticity (around 0.01 N/m) and reduced surface tension (0.01–0.03 N/m) along with the polydispersity of the bubble suspension play a critical role in determining the acoustic properties of PLA microbubbles.

1:20

1pBAb2. Acoustic and optical characterization of targeted ultrasound contrast agents. Camilo Perez, Jarred Swalwell (Ctr. for Industrial and Med. Ultrasound (CIMU), Univ. of Washington Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105-6698, camipiri@uw.edu), Juan Tu (Key Lab. of Modern Acoust., Nanjing Univ., Nanjing, China), Hong Chen, Andrew Brayman, and Thomas J. Matula (Ctr. for Industrial and Med. Ultrasound (CIMU), Univ. of Washington Appl. Phys. Lab., Seattle, WA)

We previously developed a flow cytometer system that incorporates the action of ultrasound to characterize shell properties of ultrasound contrast agents (UCA's). The most recent manifestation involves a flow cytometer modified with a custom square quartz flow cell in place of the standard nozzle and fluid jet. Acoustic coupling to the carrier sheath fluid and UCA samples occurs through a PZT bonded to one side of the flow cell. The PZT-driven UCA oscillations were processed and fitted to the Marmottant UCA model. Shell properties for UCAs (including Definity, Optison, SonoVue, and even homemade bubbles) were determined. A major limitation of the previous work involved a lack of knowledge of the actual acoustic pressure incident on the UCA. The focus of this talk will be on optimization of the pressure inside the flow cell using finite element methods, and the comparison with additional measurements of unpublished data from targeted UCA's. [Work funded in part by the Life Sciences Discovery Fund #3292512.]

1:40

1pBAb3. Investigating the effect of fabrication method on the stability and acoustic response of microbubble agents. Graciela Mohamedi (Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Rd. Campus, Headington, Oxford OX3 7DQ, United Kingdom, graciela.mohamedi@eng.ox.ac.uk), Neveen A. Hosny (Dept. of Chem., Imperial College London, London, United Kingdom), Paul Rademeyer (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Yoonjee Park (Dept. of Biomedical Eng., Boston Univ., Boston, MA), Joshua Owen (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Tuan Pham, Joyce Y. Wong (Dept. of Biomedical Eng., Boston Univ., Boston, MA), Marina Kuimova (Dept. of Chem., Imperial College London, London, United Kingdom), and Eleanor Stride (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Microbubbles stabilized by a surfactant or polymer coating are already in clinical use as ultrasound imaging contrast agents. They have also been widely investigated as vehicles for drug delivery and gene therapy that can be tracked and triggered using ultrasound. Extensive studies have been made of the effects of the coating material and gas core on microbubble characteristics, but the influence of the fabrication method has received less attention. The aim of this study was to compare the behavior of microbubbles prepared using different techniques. Phospholipid-coated microbubbles were produced using sonication, electrospraying, or in a specially designed microfluidic device. The microbubbles were observed using optical, electron, and fluorescence lifetime imaging microscopy (FLIM) to interrogate their surface microstructure and stability over time. Their acoustic response was then determined in a flow chamber by detecting the pressure scattered from individual microbubbles as they passed through the focal region of a transducer (center frequencies 1, 2.25, and 3.5 MHz; peak negative pressures 50–300 kPa). The method of bubble generation was found to significantly affect the bubble surface characteristics, stability, and acoustic response. The results demonstrate that the processing method affects not only the bubble size distribution but other characteristics important for biomedical applications.

2:00

1pBAb4. Radiation for bubble contrast agents in inhomogeneous media. Chrisna Nguon, Max Denis, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, 63 Hemlock St., Dracut, MA 01826, chrisna_Nguon@student.uml.edu)

The acoustic field generated by a distribution of micro-bubbles serving as contrast agents for a three-dimensional scattering volume is evaluated. A dual-frequency incident field generated by a confocal transducer insonifies the target and creates a scattered field that includes the difference frequency component that is of interest for improving the resolution in imaging of biological tissue. The scattered pressure is computed for a range of compressibility contrast parameters and wavenumbers using Born series and Padé approximants to ensure convergence as the medium contrast is increased. The effect of this pressure field on the resonant radiation of bubbles is examined, and bubble parameters that influence the amplification of the field measured exterior to the scattering volume are identified. A baseline comparison of the scattered pressure field with and without the presence of bubble contrast agents is presented.

2:20

1pBAb5. Temporal evolution of subharmonic emissions from a lipid-encapsulated contrast agent. Himanshu Shekhar (Elec. and Comput. Eng., Univ. of Rochester, 212 Conant Rd. Apt. C, Rochester, NY 14623, himanshushekhar@rochester.edu), Joshua J. Rychak (Targeson Inc., San Diego, CA), and Marvin M. Doyley (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

A large body of work has investigated the influence of the excitation pulse, agent size distribution, and the ambient pressure on the subharmonic response of microbubble contrast agents (MCA). The purpose of this study was to investigate the temporal evolution of the subharmonic emissions, i.e., whether the subharmonic response is influenced by the time elapsed since agent constitution. We measured subharmonic emissions from a commercial lipid-encapsulated contrast agent (Targestar-P®, Targeson Inc., San Diego) over the time span of 60 min. The excitation parameters were as follows: 10-MHz frequency, 30–290 kPa pressures, 60 cycles, and 1-kHz pulse repetition frequency. The subharmonic emissions were observed to increase by 11 dB over 60 min relative to those measured immediately after reconstitution. The most striking increase (>8 dB) was observed in the first 15 min. Although we did not observe a change in the agent size distribution, the pressure threshold for subharmonic emissions reduced by nearly two-fold within the time span of our measurements. This work demonstrates that time evolution of subharmonic emissions could bias quantitative estimates obtained from techniques such as subharmonic imaging and subharmonic-aided pressure estimation. Additionally, these findings suggest the possibility for improving subharmonic emission by careful agent design.

2:40–3:00 Break

3:00

1pBAb6. Simulations of transcranial passive acoustic mapping with hemispherical sparse arrays using computed tomography-based aberration corrections. Ryan Jones (Med. Biophysics, Univ. of Toronto, 2075 Bayview Ave., Focused Ultrasound Lab (C713), Toronto, ON M4N 3M5, Canada, ryanjones017@gmail.com), Meaghan O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), and Kullervo Hynynen (Med. Biophysics, Univ. of Toronto, Toronto, ON, Canada)

Passive acoustic mapping (PAM) is receiving increasing interest as a method for monitoring focused ultrasound (FUS) therapy. PAM would be beneficial during transcranial cavitation-enhanced FUS treatments, particularly non-thermal, cavitation-mediated applications such as FUS-induced blood–brain barrier disruption or sonothrombolysis, for which no real-time monitoring technique currently exists. However, the use of PAM in the brain is complicated by the presence of the skull bone. If not properly accounted for, skull-induced aberrations of propagating cavitation emissions will lead to image distortion and artifacts upon reconstruction. Through the use of numerical simulations, this study investigated the feasibility of transcranial PAM via hemispherical sparse hydrophone arrays. A multi-layered ray acoustic transcranial ultrasound propagation model based on computed tomography-derived skull morphology was developed. By incorporating skull-specific aberration corrections into a conventional passive beamforming algorithm [Norton and Won, IEEE Trans. Geosci. Remote Sens. **38**, 1337–1343 (2000)], simulated acoustic source fields were spatially mapped through digitized human skulls. The effects of array sparsity and receiver element configuration on the formation of passive acoustic maps were examined. Multiple source locations were simulated to determine the imageable volume within the skull cavity. Finally, the reconstruction algorithm's sensitivity to noise was explored.

3:20

1pBAb7. Transcranial spatial and temporal assessment of microbubble dynamics for brain therapies. Costas Arvanitis and Nathan McDannold (Radiology, Brigham and Women's Hospital, Harvard Med. School, 221 Longwood Ave., Rm. 514a, Boston, MA MA 02115, cda@bwh.harvard.edu)

Harnessing ultrasound/microbubble interactions in the brain may make possible a number of therapeutic ultrasound applications, such as targeted drug delivery, sonothrombolysis, and cavitation-enhanced ablation. However, methods to guide these emerging therapies are presently lacking. Here, we integrated a linear US imaging transducer with a clinical transcranial MRI-guided focused ultrasound (MRgFUS) system and evaluated passive cavitation imaging to monitor microbubble-enhanced sonications. A nonhuman primate skull filled with brain-mimicking phantom was used for the experiments. First, we sonicated the phantom over a range of powers (20–60 W) to induce cavitation-enhanced heating. Using transcranial passive cavitation mapping and MR thermometry, we assessed the ability of the integrated system to simultaneously visualize temperature changes and microbubble activity. In another experiment, we traversed the phantom with a 2 mm channel through which microbubbles could flow and applied burst sonications (5 W) to generate stable and inertial cavitation. In the first experiment, cavitation activity and heating were colocalized. In the second, the location of the cavitation activity was coincident with the targeted location in the channel within the expected resolution of the passive imaging. We conclude that combined MR/ultrasound imaging can provide comprehensive guidance to simultaneously localize and quantify both acoustic cavitation activity and heating.

3:40

1pBAb8. Spatial specificity and sensitivity of passive cavitation imaging for monitoring high-intensity focused ultrasound thermal ablation in ex vivo bovine liver. Kevin J. Haworth (Internal Med., Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209, kevin.haworth@uc.edu), Vasant A. Salgaonkar (Biomedical Eng. Program, Univ. of Cincinnati, San Francisco, California), Nicholas M. Corregan (Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Christy K. Holland (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH)

Passive cavitation images (PCIs) generated from scattered acoustic waves are a potential technique for monitoring lesion formation during high-intensity focused ultrasound (HIFU) thermal ablation. HIFU lesion prediction by PCIs was assessed in *ex vivo* bovine liver samples ($N = 14$) during 30-s sonications with 1.1-MHz continuous-wave ultrasound (1989 W/cm^2 estimated spatial-peak intensity). Treated samples were sectioned, optically scanned, and the HIFU lesions segmented based on tissue discoloration. During each insonation, a 192-element, 7-MHz linear array (L7/Iris 2, Ardent Sound) passively recorded emissions from a plane containing the HIFU propagation axis oriented parallel to the image azimuth direction. PCIs were formed from beamformed A-lines filtered into fundamental, harmonic, ultraharmonic, and inharmonic frequency bands. Lesion prediction was tested using binary classification of local tissue ablation based on thresholded PCIs, with spatial specificity and sensitivity of lesion prediction quantified by the area under receiver operating characteristic curves (AUROC). Tadpole-shaped lesions were best predicted by harmonic emissions ($AUROC = 0.76$), prefocal lesions were best predicted by harmonic or ultraharmonic emissions ($AUROC = 0.86$), and cigar-type focal lesions were best predicted by fundamental and harmonic emissions ($AUROC = 0.65$). These results demonstrate spatial specificity and sensitivity when predicting HIFU lesions with PCIs. [Work supported in part by NIH grants F32HL104916 and R21EB008483.]

4:00

1pBAb9. Real-time three-dimensional passive cavitation detection for clinical high intensity focused ultrasound systems. Jamie Collin, Christian Coviello, Erasmia Lyka (Inst. of Biomedical Eng., Univ. of Oxford, ORCRB, Headington, Oxford OX3 7DQ, United Kingdom, jamie.collin@eng.ox.ac.uk), Tom Leslie (Dept. of Urology, Oxford Univ. Hospitals, Oxford, United Kingdom), and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Bubble activity during high intensity focused ultrasound (HIFU) surgery has been linked with desirable effects, such as an enhanced heat deposition caused by inertial cavitation, and undesirable effects, such as lesion migration caused by boiling bubbles. There is presently no reliable way of achieving spatiotemporal monitoring of cavitation activity during clinical HIFU treatments. In the present work, a near-acoustically transparent two-dimensional 32-element PVDF array was designed and mounted on the therapy transducer of a clinical HIFU device (Model JC200, Chongqing Haifu) to enable detection of acoustic emissions arising from cavitation during therapy. The signal detected by each of the elements was digitized and processed in real time on a graphical processing unit (GPU), and beamformed using our previously described passive acoustic mapping (PAM) algorithm to produce real-time three-dimensional (3D) maps of cavitation activity with a frame rate in excess of 5 Hz. The system was initially validated in agar-based tissue-mimicking materials, demonstrating that the displayed volume of cavitation activity agreed with predictions based on in situ pressure calibrations. The system was further validated during clinical HIFU treatments of kidney tumor, liver tumor, and uterine fibroid ablation, and was found to enable accurate localization of the HIFU focus at sub-lesioning intensities.

4:20

1pBAb10. Passive acoustic mapping using optimal beamforming for real-time monitoring of ultrasound therapy. Christian Coviello (Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., ORCRB, Oxford OX3 7DQ, United Kingdom, christian.coviello@eng.ox.ac.uk), Richard Kozick (Dept. of Elec. Eng., Bucknell Univ., Lewisburg, PA), James J. Choi (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Miklós Gyöngy (Faculty of Information Technol., Pázmány Péter Catholic Univ., Budapest, Hungary), Jamie Collin, Carl R. Jensen, Penny Probert Smith, and Constantin C. Coussios (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

In ultrasound therapy, passive acoustic mapping (PAM) has been shown to be an effective method for imaging the acoustic emissions generated during treatment providing the potential for real-time therapy monitoring. In both high intensity ultrasound (HIFU) ablative surgery and targeted drug delivery, imaging artifacts at higher amplitude exposure conditions have been observed, which make the localization and dosimetry of therapeutically relevant cavitation activity a challenge. Due to these artifacts, correlating drug release or lesion volumes to the PAMs is hindered for many exposures. It is proposed that incorporating optimal beamforming techniques into the PAM framework can reduce and remove these artifacts, allowing determination of the extent of cavitation activity during ultrasound therapy. Additionally, optimal beaforming is found to yield improved resolution, good interference suppression, and robustness against steering vector errors. A description of the origin of the artifacts as well as reduction of them by implementing optimal beamforming within PAM will be demonstrated in the context of targeted drug delivery.

4:40

1pBAb11. Passive acoustic mapping of magnetic microbubbles in an *in vitro* flow model. Calum Crake, Marie de Saint Victor, Christian Coviello, Joshua Owen, Constantin-C Coussios, and Eleanor Stride (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, calum.crake@eng.ox.ac.uk)

Magnetic microbubbles can be successfully retained near a vascular target and simultaneously imaged using conventional B-mode ultrasound. When further modified to carry a drug, they could enable significant enhancements in targeted drug delivery for applications such as sonothrombolysis, where stable cavitation has been shown to play a key role. However, the effect of the increased proximity of the microbubbles under the effect of the magnetic field on their acoustic response remains unknown. Passive acoustic mapping is a method that enables real-time spatiotemporal monitoring of cavitation dynamics in an arbitrary plane or volume within the field of view of the ultrasound probe, and classification of the type of cavitation activity on the basis of the spatial distribution of frequency-domain emissions. In the present work, PAM is used to investigate the effect of bubble proximity and flow rate on the type, sustainability, intensity, and spatial distribution of cavitation activity observed for both magnetic and non-magnetic microbubbles excited by 0.5 MHz therapeutic ultrasound in an *in vitro* flow model. It is hoped that this study will not only yield a new method for real-time monitoring of drug delivery using magnetically trapped microbubbles, but will also help elucidate complex bubble–bubble interactions in therapeutic ultrasound fields.

5:00

1pBAb12. A two-component speckle model for detection of microbubble signals in linear contrast-enhanced ultrasonography. Matthew R. Lowerison (Dept. of Medical Biophys. and Robarts Res. Inst., The Univ. of Western Ontario, 95 Salem Place, London, ON N6K 1T8, Canada, mloweri@imaging.robarts.ca), M Nicole Hague, Ann F. Chambers (London Regional Cancer Program, London, ON, Canada), and James C. Lacefield (Dept. of Medical Biophys. and Robarts Res. Inst., The Univ. of Western Ontario, London, ON, Canada)

Contrast-enhanced ultrasound (CEUS) serves oncology by imaging tumor blood supply to enable quantification of longitudinal vascular changes and monitoring of treatment responses. Unfortunately, the linear subtraction methods commonly used for preclinical imaging are susceptible to registration errors and motion artifacts that lead to reduced contrast-to-tissue ratios. In this presentation, an alternative approach is proposed to improve discrimination between the contrast and tissue signals by comparing the

first-order speckle statistics of images acquired before and after injection of microbubbles. The microbubble signal component is modeled as a temporally varying random process superimposed on a Rayleigh-distributed speckle signal representing backscatter from tissue. Images were acquired at 18 MHz from a murine orthotopic (mammary fat pad) xenograft breast cancer model following a bolus injection of microbubbles. Images were processed using gold-standard pulse inversion nonlinear CEUS, conventional linear subtraction, and the proposed statistical method. In comparison to conventional linear CEUS, the statistical method produced a wash-in curve that showed closer agreement to the gold-standard nonlinear CEUS data. The statistical method eliminates baseline image subtraction from linear CEUS processing, which should streamline the imaging workflow, improve the robustness of image quantification, and enable real-time perfusion imaging with linear CEUS.

MONDAY AFTERNOON, 3 JUNE 2013 512AE, 1:00 P.M. TO 4:40 P.M.

Session 1pEAa

Engineering Acoustics: Active and Passive Control of Fan Noise

Alain Berry, Cochair
Dept. Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC J1K 2R1, Canada

Anthony Gerard, Cochair
Soft dB, 1240 Ave. Beaumont, Bureau 206, Mont Royal, QC H3P3E5, Canada

Invited Paper

1:00

1pEAa1. Active control of axial and centrifugal fan noise. Scott D. Sommerfeldt and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

Both axial and centrifugal fans are used to cool information technology (IT) equipment. These fans generate noise that can be annoying to their users, particularly the tonal noise that can be radiated. Work has focused on developing a method to attenuate the tonal noise associated with both of these types of fans. A compact system is used, whereby control sources are placed in close proximity to the fan. A genetic algorithm has been implemented to determine optimal source configurations. The attenuation associated with some configurations is found to be much more sensitive to error than others. For a given configuration, by using a relatively simple point source model it becomes possible to identify optimal near-field error sensor locations, which results in a compact noise control solution that provides significant global attenuation of the radiated tonal noise. This paper will review progress that has been made to apply this method to both axial and centrifugal fans. Experimental results confirm that it is feasible to achieve significant global control using this method.

Contributed Papers

1:20

1pEAa2. Industrial fan noise control using flow obstructions. Remy Oddo, Anthony Gérard (GAUS, 2500 Boul. de l'Universite, Sherbrooke, QC J1K2R1, Canada, Remy.Oddo@USherbrooke.ca), Michel Pearson (Soft dB, Québec, QC, Canada), Adrien Amyotte, Patrice Masson (GAUS, Sherbrooke, QC, Canada), Franck Sgard (IRSST, Montréal, QC, Canada), and Alain Berry (GAUS, Sherbrooke, QC, Canada)

Fans are used in a lot of industrial processes and are sometimes a source of important noise for workers. The first aim of this study was to identify some problematic fans in Québec industries, for which the noise exposure exceeds the CSST (Comission de la santé et sécurité au travail) limit of 90 dBA for 8 h. We have focused on fans having a high tonal content, for which the Simple Silence technology can be applied, i.e., tonal fan noise control using obstructions in the flow. Two analytical models of the tonal noise radiated by these fans have been proposed: a free field model based on the Lowson model and an in-duct model based on the Goldstein model. The first free-field model has been applied to the control of the noise from a series of eight evaporator fans, having a strong blade passage frequency (BPF) tone at 90 Hz. These fans have been controlled *in-situ*, in a cold storage room, using trapezoidal obstructions in the downstream flow field of the fans. The second in-duct model has been applied to control the tonal noise from an in-duct air-extractor fan used in the underground gold mining galleries. Several obstructions have been tested in the upstream flow field.

1:40

1pEAa3. Analyzing the impact of the inlet temperature on the acoustic noise production form a supersonic jet using large eddy simulations. Bernhard Semlitsch, Mihai Mihaescu, Laszlo Fuchs (Mechanics - Linne Flow Ctr., KTH - Royal Inst. of Technol., Osquars Backe 18, Stockholm 10044, Sweden, bernhard@mech.kth.se), and Ephraim Gutmark (School of Aerosp. Syst., UC - Univ. of Cincinnati, Cincinnati, OH)

Non-ideal expanded supersonic jets emerging from a convergent-divergent nozzle produce three different types of noise, i.e., the shock-associated broadband noise, the screech noise, and the turbulent mixing noise. Interesting to note that the screech tone outside of the nozzle was exposed mostly in downscaled laboratory experiments, while under realistic conditions the exhaust jet of a gas turbine engine does not show this phenomena. Apart from a geometric scaling difference, usually a lower temperature is employed in experimental studies. It is believed that the screech tone occurs due to self-excitation of the shear-layer in a feedback-loop. Acoustic waves generated by vortical structures interacting with a shock are propagating upstream within the subsonic region of the shear-layer or outside of it. They eventually hit the nozzle's lip and excite instabilities at a certain frequency. The compressible Navier-Stokes equations are simulated numerically by using Large Eddy Simulation approach. The effect of supersonic jet operation temperature onto the associated noise is investigated. The jet-exit Mach-number is 1.56, while the total temperature ratios considered at the inlet plane of the nozzle are 1.27, 2.05, and 3.65. The differences in the near-field acoustics will be presented in each of the cases and the flow-acoustic interaction will be analyzed and quantified.

2:00

1pEAa4. Low frequency sound absorption of resonators with flexible tubes. Frank Simon (DMAE, Onera, 2 Ave. Edouard Belin, Toulouse 31055, France, frank.simon@onera.fr)

Classically, passive acoustic liners, used in aeronautic engine nacelles to reduce radiated fan noise, have a quarter-wavelength behavior, thanks to perforated sheets backed to honeycombs (SDOF, DDOF). So, their acoustic absorption ability is naturally limited to medium and high frequencies because of constraints in thickness. To drastically improve their capabilities to the lowest frequencies, the combination with active control systems or the using of foam architecture have shown an interest, but the industrial application is tricky (i.e., problems of fouling, robustness). A possible approach is to carry out a perforated panel resonator with flexible tube bundles to shift the resonance frequency to a lower frequency by a prolongation of air column length (Yadong Lu *et al.*, Internoise 2007). This paper describes theoretically this concept that allows a significant change in the acoustic impedance due to the large thickness of the resistive and reactive material and the coupling with the surrounding cavity. Applied to aeronautical configurations, the resonance frequency decreases considerably compared to a conventional resonator (factor of about 1/5) but with a reduction of the maximum absorption when the tubes fill the cavity. Experiments in impedance tube validate the theoretical approach.

Invited Paper

2:20

1pEAa5. Volumetric resistance blower. Mark MacDonald and Douglas Heymann (Intel Corp., HF2-40, 5200 Elam Young Pkwy, Hillsboro, OR 97124, mark.macdonald@intel.com)

This paper reports on a new low-noise blower rotor technology developed by Intel Corporation (patents pending). The new approach replaces the traditional centrifugal blower rotor with a block of continuous porous media. The porous media can be as simple as a low-cost, block of open-cell foam and has no blades or macroscale structure. As the porous media rotor rotates, viscous and inertial forces from the volumetric resistance of the porous media cause the air within the rotor to rotate with it, creating centrifugal forces that overwhelm the flow resistance in the radial direction and create a flow pattern similar to that achieved in a traditional blower. However, because of the lack of distinct blades, the porous-media generates nearly zero aerodynamic tonal noise and significantly reduced broadband noise. This allows the rotor to be operated at significantly higher RPM and reduced clearances relative to the traditional rotor design for further improved performance. This paper will discuss numerical modeling and experimental development of the new blower type. An iso-flow comparison of porous-media and traditional rotors with the same motor and housing demonstrate a 5 dBA reduction in broadband noise and a factor of two reduction in tonality while maintaining comparable overall efficiency. Impact of porosity and different rotor support structures are also discussed.

Contributed Papers

2:40

1pEAa6. Effect of standoff distance on the reconstruction of in-duct velocity field and regeneration of pressure field. Yong-Ho Heo and Jeong-Guon Ih (Mech. Eng., KAIST, Mech. Blg. Rm. 5121, KAIST, Daejeon, Chungcheongnam-do 305-701, South Korea, yonghoheo@kaist.ac.kr)

Identification of in-duct acoustic source characteristics is essential in the design of fluid machinery system for reducing and predicting the flow-generated noise. To this end, the inverse estimation method can be employed by using the measured sound field and matrix formulation for wave propagation within a duct. In this paper, the effect of the distance between source and measurement plane is investigated. At each standoff distance, pressures are measured at three planes with two different spacings to widen the estimation frequency range, and measurements are conducted with three different standoff distances. Modal decomposition is applied to estimate modal amplitudes, and the result is used to reconstruct the velocity field at the source plane and to obtain the regenerated pressure field at the measurement planes. It is shown that the modal amplitude identified by measured pressure field at the short standoff distance, i.e., at nearfield, can yield an accurate reconstructed velocity field of the source and regenerate the pressure field with smaller error, which is similar to the other inverse techniques such as equivalent source method and nearfield acoustical holography. A field reduction example by suppressing some parts of source velocity field is shown for demonstrating the effectiveness of the method.

3:00

1pEAa7. Numerical investigation of acoustically excited flow through an orifice using lattice Boltzmann method. Chenzhen Ji and Dan Zhao (Aerosp. Eng., Nanyang Technol. Univ., 50 Nanyang Ave., Singapore 639798, Singapore, cji1@e.ntu.edu.sg)

Two-dimensional time-domain numerical investigation of sound-induced flow through an orifice with a diameter 6 mm is conducted by using lattice Boltzmann method. Emphasis is placed on characterizing its acoustic damping behaviors. The main damping mechanism is identified as incident waves interact with the shear layers formed at the orifices rims and the acoustic oscillations destabilize the shear layers to form vortex rings. And acoustic energy is converted into vortical energy. To quantify the orifice damping effect, power absorption coefficient is used. It is related to Rayleigh conductivity and describes the fraction of incident acoustical energy being absorbed. Numerical simulations are conducted in time domain by forcing a fluctuating flow with multiple tones through the orifice. This is different from frequency-domain simulations, of which the damping is characterized one frequency at a time. Comparing our results with those from Howe's theoretical model, good agreement is observed. In addition, orifice thickness effect on its damping is discussed.

3:20

1pEAa8. Scattering of sound waves at an area expansion in a cylindrical flow duct. Susann Boij (Marcus Wallenberg Lab. for Sound and Vib. Res., Dept. of Aeronautical and Vehicle Eng., KTH Royal Inst. of Technol., 2004 Yolo Ave., Berkeley, California 94707, sboij@kth.se), Özge Yanac Cinar, Gökhan Cinar (Gebze Inst. of Technol., Kocaeli, Turkey), and Börje Nilsson (School of Comput. Sci., Phys. and Mathematics, Linneaus Univ., Växjö, Sweden)

Sound propagation in pipes and ducts with flow, like ventilation ducts and exhaust pipes, is influenced by flow separation and vortex production at sharp edges along the ducts, such as at bends and area expansions. Shear layers form at the separation points, and such layers are unstable to low frequency acoustic disturbances. An analytical model, aiming at physical insight into this interaction, is presented. Results in the plane wave region for the so called scattering matrix for a sudden area expansion with flow in cylindrical pipes are compared with experimental values. Both the magnitude and the phase, in the form of an end correction, is presented. The model is also compared to a 2 dimensional model, in order to evaluate the anticipated increased accuracy of the 3 dimensional modeling. The scattering coefficients are strongly dependent on the flow speed, which is up to a Mach

number of 0.5. It is observed that for low frequencies, the interaction is dominated by the dynamics of an unstable shear layer downstream of the edges. For higher frequencies, the wave propagation is mainly affected by convective effects. Differences in properties for the 2D and the 3D cases are also explored.

3:40

1pEAa9. Design of a built-in electroacoustic resonator for active noise reduction. Romain Boulandet, Etienne Rivet, and Hervé Lissek (Laboratoire d'ElectroMagnétisme et Acoustique, Ecole Polytechnique Fédérale de Lausanne, ELB Station 11, Lausanne CH-1015, Switzerland, romain.boulandet@gmail.com)

The paper focuses on the design of a built-in electroacoustic resonator for active noise reduction purposes. This concept basically encompasses a loudspeaker connected to a synthetic electrical load that enhances the ability of the transducer to dissipate a certain part of the incoming acoustic energy. The strategy is therefore to control the dynamics of boundaries in closed sound spaces (such as room, cavity, etc.) rather than targeting a global control that requires significant input of additional acoustic energy. The main attraction of the proposed methodology is its ability to achieve broadband sound absorption while bypassing the use of sensors, the sensing of sound field information being incorporated within the synthetic electrical load admittance (current/voltage transfer function). Computational and experimental results are provided to illustrate the benefits and potential of a built-in electroacoustic resonator compared to other options. Concluding remarks and discussions on foreseen future developments are then provided.

4:00

1pEAa10. The threshold of the difference between a mathematical model applied to active noise control and data recorded. Ricardo A. Quintana and Adriana P. Gallego (Universidad Distrital Francisco José de Caldas, Calle 7A # 73 - 98 Apto 503 Int 4, Bogotá 11001000, Colombia, rquintana@raqsacoustic.com)

Nowadays, there are many methods used to obtain mathematical models applied to active noise control, especially when the transfer function is required. Inside rooms, the global active sound control has bad results due to the reflections and the diffuse field. Then, authors have applied system identification to find more complex mathematical models based on measured data. Also, the number of system identification methodologies is increasing and it carries to having many unexplored models. In order to know which models are useful for global active noise control inside rooms, a relationship between the sound pressure level decreased and the error of the mathematical model is presented. First, the concept of "a useful mathematical model" is defined under any context based on an analysis of the error (FIT). In addition, this concept is delimited to the active noise control context. Finally, an example is presented.

Invited Paper

4:20

1pEAa11. Upgrade of a multi-channel active noise control system for an industrial stack. Louis-Alexis Boudreault, André L'Espérance, and Alex Boudreau (Soft dB Inc., 1040, Ave. Belvedere, Ste. 215, Quebec, QC G1S 3G3, Canada, la.boudreault@softdb.com)

Active noise control has been studied in the 1990s as an innovative way to reduce the noise in specific situations. Some applications are well known today and found commercial success like noise-canceling headphones. However, the use of active noise control in industrial applications is more complex, thus being an uncommon solution in this field. The use of active noise control for industrial stack noise is one of these applications. One of the first large-scale implementation has been set up at the end of the 1990s. This system was a 10-channel active noise control system installed on a 1.8 m wide chimney to attenuate a 320 Hz pure tone. At that time, an 8 dB noise reduction was achieved at error microphones. Fifteen years later, it has been decided to upgrade the system with the latest generation of digital signal processor (DSP) allowing a real-time optimization and better tracking speed. This paper describes the overall system and the updated multi-channel active noise controller developed for this application. It also presents the improvements, the achieved noise reduction, and the associated environmental benefits.

MONDAY AFTERNOON, 3 JUNE 2013 — 512BF, 1:00 P.M. TO 5:00 P.M.

Session 1pEAb

Engineering Acoustics: Transduction, Transducers, and Energy Harvesting

Stephen Butler, Chair
NUWC, Newport, RI 02841

Contributed Papers

1:00

1pEAb1. Ultra-low frequency underwater acoustic projectors: Present status and future trends. Bertrand Dubus, Pascal Mosbah (ISEN, IEMN UMR CNRS 8520, 41 Boulevard Vauban, Lille cedex 59046, France, bertrand.dubus@isen.fr), Jean-Rémi Hartmann, and Jacky Garcin (Techniques Navales /SDT/SCN/LSM/DSM, DGA, Toulon, France)

Ultra-low frequency (ULF) underwater transducers, used in the 10–400 Hz frequency range, have usually radiating surfaces the dimensions of which are small with respect to the acoustic wavelength. To radiate a high acoustic power with a monopolar ULF transducer, a large volume velocity is required to counterbalance the low radiation resistance. Three transduction technologies are available to realize compact high power ULF transducers: hydroacoustic, electromagnetic, and active material-based. In the latter case, piezoelectric ceramics and magnetostrictive rare-earth alloys are often associated to flexural vibration such as found in flextensional transducers. Compared to these materials, piezoelectric single crystals, which exhibit lower stiffnesses and produce higher strains together with higher energy densities, are potential active materials for future ULF underwater transducers. In this work, ULF transducers are analyzed in terms of their working frequencies, acoustic powers, and masses. Thirty-two ULF underwater projectors build during the last 25 years are considered. For single crystal transducers, prototypes working at higher frequencies as well as transducers modeled with finite element method are taken into account. Using these data and classical scaling laws, abacuses displaying acoustic power-frequency curves for given masses are constructed for each technology. They show that single crystals transducers could provide more compact and powerful solutions for frequencies above 40 Hz.

1:20

1pEAb2. Performance of transducers with segmented piezoelectric stacks using materials with high electromechanical coupling coefficient. Stephen C. Thompson, Richard J. Meyer, and Douglas C. Markley (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16803, sct12@psu.edu)

Abstract underwater acoustic transducers often include a stack of thickness polarized piezoelectric material pieces of alternating polarity interspersed with electrodes and bonded together. The stack is normally much shorter than a quarter wavelength at the fundamental resonance frequency, so that the mechanical behavior of the transducer is not affected by the segmentation. When the transducer bandwidth is less than a half octave, as has conventionally been the case, stack segmentation has no significant effect on the mechanical behavior of the device. However, when a high coupling coefficient material such as PMN-PT is used to achieve a wider bandwidth, the difference between a segmented stack and a single piezoelectric piece with the same overall dimensions can be significant. This paper investigates the effects of stack segmentation on the performance of wideband underwater acoustic transducers, particularly tonpilz transducer elements. Included is discussion of transducer designs using single crystal piezoelectric material with high coupling coefficient compared with more traditional PZT ceramics.

1:40

1pEAb3. On the spatial distributions for randomly spaced arrays. Jenny Au and Charles Thompson (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, jenny_au@student.uml.edu)

In this work, we examine the statistical characteristics of the randomly spaced transducer arrays. Approaches to realizing the linear and two-dimensional arrays are considered. The array elements are equally weighted in amplitude and their contribution to received signal is by virtue of adjustment by their spatial location. The cumulative distribution for the number of transducers as a function of transducer spacing and its relationship to the spatial objective function is given. The case for Dolph–Chebychev objective function is shown in detail and in closed-form. The statistical effect spatial binning of transducer elements is examined.

2:00

1pEAb4. Electroacoustic metamaterials: Achieving negative acoustic properties with shunt loudspeakers. Herve Lissek (LEMA, Ecole Polytechnique Federale de Lausanne, STI IEL LEMA, Station 11, Lausanne 1015, Switzerland, herve.lissek@epfl.ch)

Acoustic metamaterials constitute a new class of structures that exhibit acoustic properties not readily available in nature. These properties can be a negative mass density, expressing the opposition of the acceleration of a particle to the application of pressure, or a negative bulk modulus, signifying the rarefaction of the particle in reaction to a compression (resp. a condensation in reaction to a depression). However, these artificial behaviors result from a periodic arrangement of passive unit-cells (such as membranes and side holes), and not from individual "meta-properties" of each unit-cell. It is however possible to observe such intrinsic metamaterial properties out of a passive electroacoustic resonator. This concept encompasses a loudspeaker, connected to a specific electric load, thus altering the acoustic dynamics of the loudspeaker diaphragm when subject to an exogenous sound source. It is especially possible to achieve negative acoustic impedance at its diaphragm, thanks to the connection of passive electric shunt circuits, such as simple RLC series resonators. This paper aims at highlighting the metamaterial nature of such electroacoustic resonators through computational and experimental results, followed by discussions on ongoing developments.

2:20

1pEAb5. Loudspeaker for low frequency signal driven by four piezoelectric ultrasonic motors. Juro Ohga (Ohga Acoust. Lab., 2-24-3 Tamanawa, Kamakura 247-0071, Japan, johga@nifty.com), Hirokazu Negishi (MIX Corp., Yokosuka, Japan), Ikuo Oohira (I. Oohira and Assoc., Yokohama, Japan), Hiroya Saito, Kunio Oishi (Tokyo Univ. of Technol., Hachioji, Japan), and Kazuaki Maeda (TOA Corp., Takarazuka, Japan)

The authors are developing a completely new direct-radiator loudspeaker as an alternative of the conventional electrodynamic loudspeaker. It is driven by continuous revolution of piezoelectric ultrasonic motors, It is useful for radiation of very low frequency signal because it shows almost flat phase frequency characteristics in low frequency region. A preliminary model, named dual-motor, de-spin (DMDS) model, included co-axial two ultrasonic motors. Stator of one motor is fixed to the base and of the other is connected to the cone radiator. Velocity modulation for any motor induces driving force for the cone radiator. Output sound at low frequency range (for example, 30–120 Hz) by this model was excellent because it has no significant resonance in this frequency region. However, its operation was occasionally instable. At this Congress, a highly improved model named quad-motor, de-spin (QMDS) model is presented. It uses two co-axial DMDS mechanisms. The experimental model has a cone radiator of 46 cm in diameter and an enclosure of 400 L. Its working frequency range is same as DMDS model. Harmonic distortions included in the output signal are improved to be less than 10%. Its sound quality is excellent.

2:40

1pEAb6. Influence of nonlinear parameters in Mirror filter to compensation performance of nonlinear distortions. Natsuki Uesako and Yoshinobu Kajikawa (Faculty of Eng. Sci., Kansai Univ., 3-3-35 Yamate-cho, Suita-shi, Osaka 564-8680 Japan, natsuki.uesako@gmail.com)

Mirror filter is used for the compensation of nonlinear distortions for electro-dynamic loudspeaker systems and is based on the nonlinear differential equations. The design of Mirror filter requires the estimated parameters of a target loudspeaker system. If you obtain the corresponding parameters of a target loudspeaker system and arrange Mirror filter designed using those parameters in front of the loudspeaker, then the nonlinear distortions can be compensated. Hence, the estimated parameters are very important to achieve high compensation performance. In this paper, we therefore examine the effects of the estimated parameters to the compensation performance. Concretely, we clarify the effects by varying each nonlinear parameter in Mirror filter. Simulation and experimental results demonstrate that the compensation performance for the second order nonlinear distortions depends on a nonlinear parameter of the force factor in loudspeaker systems.

3:00–3:20 Break

3:20

1pEAb7. A new loudspeaker for low frequency radiation by linear motion type piezoelectric ultrasonic actuators. Hiroya Saito (School of Comput. Sci., Tokyo Univ. of Technol., 1401-1,Katakura, Hachioji 192-0982, Japan, hirosaito12@gmail.com), Hirokazu Negishi (MIX Corp., Yokosuka, Japan), Juro Ohga (Shibaura Inst. of Technol./MIX Corp., Kamakura, Japan), Ikuo Oohira (Self-Employee, Yokohama, Japan), Kazuaki Maeda (TOA Corp., Takarazuka, Japan), and Kunio Oishi (School of Comput. Sci., Tokyo Univ. of Technol., Hachioji, Japan)

The authors had proposed new direct-radiator loudspeaker constructions with a conventional paper cone radiator driven by ultrasonic motors (USM), as a substitution for voice-coil motor. However, those models needed a revolution to linear motion conversion mechanism, and avoiding zero region non-linearity, like class A amplifier. These complications came from the conventional USM, since it is a rotational and having zero region non-linearity inherently. Here, the authors would propose a new mechanism by using new ultrasonic linear actuators, called longitudinal-bending multilayered transducers with independent electrodes (LBMTIE). The beauty of LBMTIE is linear and to control vertical motion and horizontal motion independently, hence zero region non-linearity avoided. Therefore, it is possible to substitute the voice-coil motor directly, which avoids the complicated mechanisms mentioned above. In this LBMTIE driven loudspeaker, vertical

movement voltage be fixed and horizontal voltage is driven by audio signal, like voice-coil motor. In addition, there is a big contrast against conventional voice-coil motor, which is a typical transducer, as its electrical input and sound pressure output are direct proportion each other. This is because LBMTIE driven loudspeaker may behave a sort of modulator, which is not direct proportion in between input electric power and output sound pressure level.

3:40

1pEAb8. Velocity control with class D amplifiers. Robert C. Randall (Ship and Torpedo Electron., Raytheon, 188 Hanover St. Apt 3, Fall River, Massachusetts 02720, bobrandall81@gmail.com) and David A. Brown (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Fall River, MA)

A SONAR array's radiation pattern is affected by the acoustic interactions, which may limit the effectiveness of beamforming algorithms when transmitting. A negative feedback system with a velocity sense signal fed back to the power amplifier can mitigate the array interactions proportional to the loop gain, and be effective across a broad frequency range without requiring *a priori* knowledge of the input signals. The tradeoffs between motional current velocity control and accelerometer based velocity control are summarized. Class D switching amplifiers can achieve greater than 90% efficiency and are increasingly being used to drive SONAR arrays. When a velocity control system is used with a Class D amplifier, feedback stability becomes a significant concern due to obtaining the feedback signal after the amplifier's LC output filter. The array equations are still obtained by converting the amplifier into a Thevenin equivalent force and output impedance, which now includes the amplifier's output filter and the synthesized impedance due to the velocity control loop. Sample beampatterns both with and without velocity control are shown, concluding that velocity control is well suited for use with complex dynamic transmit beamforming.

4:00

1pEAb9. A system for ultrasonic transmission of power and signal to an implanted hearing aid. Hugo Vihvelin, Jeffrey Leadbetter, Jeremy A. Brown, and Robert Adamson (School of Biomedical Eng., Dalhousie Univ., P.O. Box 15000, Halifax, NS B3H4R2, Canada, hugo.vihvelin@gmail.com)

We will report on development of a system for efficiently powering implanted hearing aids by transmitting ultrasonic acoustic energy across the skin. As compared to traditional magnetic induction coil power delivery systems, ultrasound-based systems offer a more compact form factor for the same power handling capability and lower electrical loss. Part of the challenge of building such a system for implanted hearing aids is developing efficient modulation and demodulation electronics that can deliver both electrical power and an acoustic frequency signal to the implanted device. We present the design and implementation of an amplitude modulated system in which the power is delivered on the carrier and signal in the modulation sidebands. The transmitter consists of an efficient PWM encoder driving an LC resonator tuned to the ultrasound transducer resonance frequency. The receiver consists of an efficient rectifying demodulator that provides supply voltages to internal electronics as well as the acoustic signal. Power loss mechanisms, form factor considerations, linearity, and overall system performance will be discussed.

4:20

1pEAb10. The design of ultrasonic lead magnesium niobate-lead titanate composite transducers for power and signal delivery to implanted hearing aids. Jeff Leadbetter, Jeremy Brown, and Rob Adamson (Dalhousie Univ., 1276 South Park St., Rm. 3189, Dickson Bldg., VG Site, Halifax, NS B3H 2Y9, Canada, jeff.leadbetter@dal.ca)

We present a system for efficiently powering implanted hearing aids by transmitting an ultrasonic signal across the skin. The use of ultrasound as method for power and signal transfer is known for embedded systems in industrial applications and has more recently been investigated for use with other medical implants. In our application, ultrasonic transducers are investigated as they offer substantially reduced size relative to traditional magnetic induction coil power delivery. The developed transducers use lead magnesium niobate-lead titanate (PMN-PT) piezoelectric material in a 1–3 composite formulation. PMN-PT offers an electromechanical coupling factor (kt, an indicator of maximum efficiency) that is up to 60% greater than traditional piezoceramics, while the use of composite transducers removes geometric constraints that can limit the achieved efficiency. The fabrication methods for the transducers are detailed. Experimental results are presented to show the composite transducers achieve a kt of 0.86 (out of 1.00), and a power transmission efficiency that is improved by 38% relative to a similar non-composite transducers. It is also demonstrated that these transducers offer sufficient bandwidth for amplitude or frequency modulation schemes to transmit data signals along the power carrier beam.

4:40

1pEAb11. Biocompatible wireless power transferring and charging based on ultrasonic resonance devices. Sung Q Lee, Woosub Youm, and Gunn Hwang (Nano Convergence Sensor, Electron. and Telecommunications Res. Inst., 161 gajungro yousung, Daejon 305-350, South Korea, hermann@etri.re.kr)

To increase application area of implantable devices for medical treatment including implantable cardiac defibrillator or deep brain stimulator, the rechargeable battery module is highly requested. The previous Li-type battery has limited current sources, so that the patient is forced to have surgery just for changing battery. Previous technologies such as magnetic resonance and induction coupling have limited applications because of its short transfer distance compared to device size and magnetic field intensity limitation for the safety of body exposure. As an alternative, the biocompatible wireless power transferring and charging technology is proposed using ultrasonic resonance devices. For the high efficient power transferring, optimal transfer frequency is calculated based on the acoustic radiation and damping effect. Then, the optimal load resistance is selected for matching power condition in receiver. And, transmitter is designed to match the optimal transfer frequency. The ultrasonic resonance transmitter and receiver are manufactured with the size of 20 mm diameter, 6.0 mm height. The energy conversion efficiency from input electrical power of transmitter and output power of receiver is about 25.6% at 10 cm distance, experimentally. The maximum transferring power is up to 15 mW. This result is quite high considered with the device size and the power transfer distance.

Session 1pMU

Musical Acoustics and Psychological and Physiological Acoustics: Player/Instrument Coupling

Gary Scavone, Cochair
McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada

Tamara Smyth, Cochair
Music, Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093

Invited Papers

1:00

1pMU1. A morphological and acoustic study on the effect of a trumpet player's vocal tract. Tokihiko Kaburagi (Grad. School of Design, Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8540, Japan, kabu@design.kyushu-u.ac.jp), Naoyuki Yamada (Nagareyama City Hall, Nagareyama, Japan), Takashi Fukui (Shikumi Design Co., Ltd., Fukuoka, Japan), and Eriko Minamiya (Yamaha Corp., Hamamatsu, Japan)

A morphological and acoustic study is presented to examine the role of the vocal tract in playing the trumpet. Preliminary results obtained from one professional player are shown, and the effectiveness of the method is demonstrated. Images of the vocal tract with a resolution of 0.5 mm (2 mm in thickness) were recorded with magnetic resonance imaging to observe tongue posture and to estimate the vocal-tract area function during actual trumpet performance. The input impedance was then calculated for the player's air column, including both the supra- and sub-glottal tracts, using an acoustic tube model that also considers wall losses. Finally, a time-domain blowing simulation was performed with a lip vibration model (Adachi and Sato, J. Acoust. Soc. Am. **99**, 1200–1209, 1996). In this simulation, the oscillating frequency of the lips was slightly affected by using different shapes of the vocal tract measured for the player. In particular, when the natural frequency of the lips was gradually increased, the transition to higher modes occurred at different frequencies for different vocal-tract shapes. Furthermore, simulation results showed that the minimum blowing pressure required to attain lip oscillation can be reduced by properly adjusting the vocal-tract shape.

1:20

1pMU2. Simulating different upstream coupling conditions on an artificial trombone player system using an active sound control approach. Vincent Fréour (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Music Technol. Area, Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, vincent.freour@mail.mcgill.ca), Thomas Hélie, Nicolas Lopes, René Caussé (IRCAM - CNRS UMR 9912, UPMC, Paris, France), and Gary P. Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Recent research suggests that the ability to finely tune vocal-tract resonances during trombone playing may constitute an important aspect of performance expertise. Artificial player systems, designed to reproduce the behavior of a real player, often neglect this component by not providing any control of upstream resonances. However, they offer great experimental platforms for quantitative studies on sound production mechanisms, allowing independent adjustment of certain control parameters. An active sound control method was designed to improve high tone support and investigate different conditions of coupling between the artificial lips, the downstream air-column, and the upstream cavity during sustained tones played by an artificial valve-trombone player system. Upstream input impedance at the fundamental frequency was controlled through real-time adjustment of the phase and amplitude ratio between the acoustic pressure generated on both sides of the lips. The phase difference between the upstream and downstream pressures was swept linearly while maintaining different conditions of upstream energy and fixed trombone fingering. Observations during this procedure included: (1) significant fundamental frequency variations in the neighborhood of a downstream impedance peak; and (2) variation of the downstream energy and optimal phase tuning with regard to the mechanical efficiency of the lip-valve system suggested at the energy maximum.

1:40

1pMU3. Saxophone modeling and system identification. Tamara Smyth (Music, Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, trsmyth@ucsd.edu) and Marjan Rouhipour (Computing Sci., Simon Fraser Univ., Vancouver, BC, Canada)

In this work, saxophone instrument frequency responses are estimated at both the mouthpiece corresponding to the input impedance and outside the bell, using acoustic measurement and post signal processing. The measurement technique is based on one previously developed for measuring the acoustic properties of instrument bells but is adapted to account for the fact that the saxophone bell does not easily separate from the instrument bore and must be measured as a single unit. Furthermore, measurements are taken of the instrument configured with all possible fingerings covering the playable range of the B-flat tenor saxophone, and instrument reflection and transmission functions are estimated for, and applied to a waveguide model of, each tone-hole configuration. Having the instrument frequency responses at both the mouthpiece and the bell for every possible fingering allows for an improved parametric synthesis, but also allows for saxophone system identification, inverse modeling, and estimation of player-instrument control parameters during real-time performance.

2:00

1pMU4. Measuring lips control on flute-like instruments using active vision. Benjamín Carriquiry, Patricio de la Cuadra (Centro de Investigación en tecnologías de audio, Music department, Pontificia Universidad Católica de Chile, Jaime Guzman Errazuriz 3300, Providencia, Santiago 07866, Chile, pcuadra@uc.cl), and Benoit Fabre (LAM-IJLRA, Université Pierre et Marie Curie (Paris 6), Paris, France)

Flute-like instruments operate under a feedback system between an air jet and a resonator. The characteristics of the air jet and its interaction with the resonator are defined by construction in some flute-like instruments such as the organ pipes, or completely determined by the musician's control like in the Shakuhachi, panpipe, or transverse flute. In this article, a 3D lips detection system based on active vision using laser grid triangulation has been designed and implemented. Simple musical gestures from transverse flute players have been measured and analyzed, outlying the strategies used to coordinate the parameters available, such as lips to labium distance, jet speed, and jet shape, to orchestrate an adequate sound. The analysis is presented in a non-dimensional representation capable of generalizing to other members of the flute family. The measurement system developed is compared with previous strategies and reveals a promising tool to further understand the complexity of the human control in this popular family of instruments.

2:20

1pMU5. Analysis of bow-change strategies. Knut Guettler (Retired, Eilins vei 20, Jar 1358, Norway, kg@knutsacoustics.com)

One of the most important skills of the accomplished bowed-string player is the smooth bow change. Smooth changes are often necessary in order to keep a phrase flowing, and equally important in situations where the bow is too short for the duration of the given note, the latter requiring a bow change of least possible audibility. The problem arises from the fact that a change of bowing direction requires the rotation of the Helmholtz corner to be reversed, and the phases of the string-velocity frequency components thus to be shifted 180°. In between the two states, there exists no transition that can fully maintain the sound flow without introducing undesirable noises. However, by choosing the right bowing strategy and gesture, the tradeoff between transition time and noise content can be optimized for the purpose. In practice, different players solve this problem in a number of ways. The present study, which is mainly based on numeric simulations, analyzes the sounding outcome of a variety of possible bowing parameters.

2:40

1pMU6. Perception and production of complex bowing movements in violin performance. Erwin Schoonderwaldt (Inst. of Music Physiol. and Musicians' Medicine, Hanover Univ. of Music, Drama and Media, Emmichplatz 1, Hannover, NDS 30175, Germany, schoondw@kth.se), Matthias Demoucron (Inst. for Psychoacoustics and Electron. Music, Ghent Univ., Ghent, Belgium), Eckart Altenmüller (Inst. of Music Physiol. and Musicians' Medicine, Hanover Univ. of Music, Drama and Media, Hanover, Germany), and Marc Leman (Inst. for Psychoacoustics and Electron. Music, Ghent Univ., Ghent, Belgium)

In bowed-string instruments, the primary function of bowing movements is to control the parameters that govern the stick-slip interaction between the bow and the string, giving the performer control of the sound. Not less importantly, bowing movements have to be planned ahead in order to anticipate future events. In fast, repetitive bowing movements involving string crossings and bow changes the primary and anticipatory movements become integrated, forming an overall, in the simplest case circular movement pattern. The relative timing of string crossings and bow changes is an inherent property of the shape of these patterns, which therefore has an important influence on the quality of the note transitions. We will present two complementary studies that provide insight in this coordination phenomenon. A perceptual study has been conducted using a virtual violin, in which the participants could influence the relative timing between string crossings and bow changes by a simple slider, giving insight in the perception of such transitions and typical temporal constraints. Analyses of bowing movements show in detail how the coordination is realized in performance, and how the performer adapts her/his movement patterns to performance constraints, such as tempo and dynamic level.

3:00–3:20 Break

3:20

1pMU7. Time-domain simulation of the bowed cello string: Dual-polarization effect. Hossein Mansour (Music, McGill Univ., Ste. 500- 550 Sherbrooke o, Montreal, QC H3A 1B9, Canada, hossein.mansour@mail.mcgill.ca), Jim Woodhouse (Engineering, Cambridge Univ., Cambridge, United Kingdom), and Gary P. Scavone (Music, McGill Univ., Montreal, QC, Canada)

A detailed time-domain simulation is implemented to model the bowed cello string. Building on earlier simulation models, several new features have been added to make the model more realistic: in particular, both polarizations of the string motion are included, as well as the longitudinal vibrations of the bow hair. These additional features can be turned on and off in the model to evaluate their relative importance. In all previous simulations, the bow-hair was assumed stiff enough to suppress any motion of the string perpendicular to the bowing direction. High-speed video recordings, on the contrary, have suggested that the amplitude of this motion is not negligible compared to the motion of the string in the bowing direction. The major source of this motion is tracked down to the X-Y coupling through the bridge. Although this extra dimension of vibration may not necessarily contribute much to the radiated sound by itself, it can modulate the effective bow force and hence affect the stick-slip motion of the string. The longitudinal vibration of the bowhair is also included in our model. The compliance of the bowhair was accounted for in previous studies in a crude way, but without enough detail to capture the difference between different bows.

3:40

1pMU8. Characterization of bowing strokes in violin playing in terms of controls and sound: Differences between bouncing and on-string bow strokes. Alfonso Perez Carrillo (Schulich School of Music, McGill Univ., 555 Sherbrooke West, Montreal, QC H3A, Canada, alfonso.perezcarrillo@mail.mcgill.ca)

Bowing is the main element in sound production during a violin performance and one of the most basic and important expressive resources for the musician. In the lowest level, control parameters such as force, velocity, or bow-bridge distance are directly determining the characteristics of the sound. In a higher level, bowing strokes constitute one of the main mechanisms for structuring the performance. There are many different kinds of bowing strokes, and they are commonly classified into on-string, if the attack happens with the bow on the string and off-string, if the bow is bouncing. From a database of violin performances containing multimodal data including sound and gestures, a set of spectral features and instrumental controls is extracted and the database is segmented into intra-note segments (attack, sustain, and release). A characterization of bowing strokes and a comparison between bouncing and on-string strokes in terms of bowing controls and sound at the intra-note segments is presented.

4:00

1pMU9. Analysis/synthesis of bowing control applied to violin sound rendering via physical models. Esteban Maestre (Music Technol. Group, Roc Boronat 138, Barcelona 08018, Spain, esteban.maestre@upf.edu)

A prominent challenge in instrumental sound synthesis is to reproduce the expressive nuances naturally conveyed by a musician when controlling a musical instrument. Despite the flexibility offered by physical modeling synthesis, appropriately mapping score annotations to sound synthesis controls still remains an interesting research problem, especially for the case of excitation-continuous instruments. Here we present our work on modeling bowing control in violin performance, and its application to sound synthesis via physical models. Minimally invasive sensing techniques allow for accurate acquisition of relevant timber-related bowing control parameter signals. The temporal contours of bowing control parameters (bow velocity, bow force, and bow–bridge distance) are represented as sequences of low-order polynomial curves. A database of parametric representations of real performance data is used to construct a generative model able to synthesize bowing controls from an annotated score. Synthetic bowing controls are then used to render realistic performances by driving a violin physical model.

Contributed Papers

4:20

1pMU10. On the relation between gesture, tone production, and perception in classical cello performance. Magdalena Chudy (Ctr. for Digital Music, School of Electron. Eng. and Comput. Sci., Queen Mary Univ. of London, Mile End Road, London E1 4NS, United Kingdom, magdalena.chudy@eecs.qmul.ac.uk), Alfonso Pérez Carrillo (Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montreal, QC, Canada), and Simon Dixon (Ctr. for Digital Music, School of Electron. Eng. and Comput. Sci., Queen Mary Univ. of London, London, United Kingdom)

On bowed string instruments such as violin or cello, the quality of sound depends mostly on the performer's bowing technique, which determines the interaction between the bow hair and the string. An accomplished string player has numerous ways of shaping the spectrum of a desired sound. This research investigates the combination of bowing gestures necessary for production of a rich tone. In particular, bowing control parameters such as bow force, bow velocity, and bow–bridge distance captured by a dedicated sensing system are analyzed and compared against audio features. Using audio and gesture measurements of six advanced cellists recorded on two different instruments of a luthier class, we characterize a sound palette and respective bowing control patterns of each player in performed music excerpts and scales. We especially focus on how performers adjust their bowing technique to control the timber of an instrument on which they have never practiced before. Observed differences between the players on the measured audio features show consistency with the bowing parameters adapted for balancing the timbral changes due to instrument, string, and fingering position. To perceptually evaluate the recorded samples, expert musicians were asked to rank the players in terms of sound quality and tone richness.

4:40

1pMU11. Differences in technique and sound in beginner and expert cello performances and use of acoustic information to provide support for performance techniques. Taichi Sato, Shoichi Miyagawa, and Hiromi Yamatani (Tokyo Denki Univ., Adachi-ku, 5 Senju-Asahi-cho, Tokyo 120-8551, Japan, taichi@mail.dendai.ac.jp)

We studied the differences in performance between beginners and expert musicians by taking cello performance as the object of our research. By principal component analysis of the beginning part of a performance—the part when a performer just begins to play—we were able to distinguish between performances by beginners and those by experts. We showed that the process of mastery of an instrument, in which the sound of a beginning performer becomes better, can be evaluated using principal component analysis. We studied the characteristics of the ability to skillfully draw the bow across the cello strings, and we created acoustic information based on these characteristics. By giving this acoustic information to beginners, the beginners learned to become almost as skillful as expert cellists in their ability to draw the bow skillfully across the strings, even though they were only beginners.

Session 1pNSa

Noise: Advanced Hearing Protection and Methods of Measurement II

Jeremie Voix, Cochair
École de technologie supérieure, Université du Québec, 1100 Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada

Elliott H. Berger, Cochair
Occupational Health & Env. Safety Div., 3M, 7911, Zionsville Rd., Indianapolis, IN 46268-1650

Invited Papers

1:00

1pNSa1. Improving speech intelligibility in active hearing protectors and communication headsets with subband processing. Anthony J. Brammer, Gongqiang Yu, Eric R. Bernstein, Martin G. Cherniack, and Donald R. Peterson (Ergonomics Technol. Ctr., Univ. of Connecticut Health Ctr., 263 Farmington Ave., Farmington, CT 06030-2017, brammer@uchc.edu)

Parallel subband processing, in which the full bandwidth of environmental noise and a communication channel are processed separately in contiguous, restricted frequency bands, has been proposed as a means to improve speech intelligibility in noise for active hearing protectors and communication headsets (Bernstein *et al*., Int. J. Ind. Ergonom., in press). An active, adaptive feed-forward control structure has been employed to improve the audibility of sounds in the communication channel of a circumaural hearing protector / headset while active noise reduction (ANR) is used to complement the passive attenuation of the device from 50 to 800 Hz. The communication channel subbands have been implemented as octave bands from 125 Hz to 8 kHz. The performance of the device has been evaluated in a diffuse field when worn by human subjects. Word intelligibility in industrial noise was evaluated when the active system was not operating, when the device was operating as a fullband ANR system with fixed communication channel gain, and as a subband ANR system with adaptive gain of the communication channel signal to improve the speech signal-to-noise ratio. A significant improvement in speech intelligibility can be obtained with the subband system. [Work supported by NIOSH 5R01OH 008669.]

1:20

1pNSa2. Advanced hearing protection and auditory awareness in individuals with hearing loss. Christian Giguere, Chantal Laroche, and Véronique Vaillancourt (Audiol./SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H8M5, Canada, cgiguere@uottawa.ca)

In-ear and earmuff-type electronic protection devices are rapidly being introduced into the marketplace and deployed in noisy industrial workplaces and military settings. In these environments, workers must be sufficiently protected from noise while being able to maintain good communication abilities and situational awareness. Features such as level-dependent attenuation or amplification, user-adjustable talk-through circuitry, noise reduction, and remote communication capabilities are commonly found in advanced devices. The benefits of these features depend on their complex interaction with the signal and noise characteristics, the hearing status and language proficiency of the workers, and the nature of the auditory task. Yet, detailed electro-acoustical specifications are rarely reported by manufacturers. Measurement standards are also lacking, though the advent of ANSI/ASA S12.42 should in part address this situation. In this paper, the characteristics of two advanced devices are reported over a test battery including measurements of the passive sound attenuation, level-dependent talk-through sound transmission, and speech recognition in noise for listeners with a wide range of hearing profiles. Electronic level-dependent attenuation provided superior speech recognition performance than passive attenuation for all groups of listeners, and often exceeded unprotected speech recognition performance.

1:40

1pNSa3. Supplemental text messaging for the resolution of auditory overload. Sharon M. Abel (Individual Behaviour and Performance Section, Defence R&D Canada - Toronto, 1133 Sheppard Ave. W., Toronto, ON M3K 2C9, Canada, sharon.abel@drdc-rddc.gc.ca), Geoffrey Ho (Human Systems Integration Section, Defence R&D Canada - Toronto, Toronto, ON, Canada), Ann Nakashima, and Ingrid Smith (Individual Behaviour and Performance Section, Defence R&D Canada - Toronto, Toronto, ON, Canada)

Military signal operators listen, transcribe, and respond to audio traffic over multiple audio channels, in high-level noise from vehicles and weapons. The messages typically overlap in time and may be difficult to disentangle. Two studies were carried out to determine the benefit of supplemental texting. Normal-hearing participants were tested in a mock up of a military command post. Brief messages were played simultaneously over a communications headset (dichotic) and a loudspeaker array, either in quiet or in a background of vehicle noise. The at-ear speech-to-noise ratio was 5 dB. Only those messages beginning with a pre-assigned call sign were encoded. Mean scores of 84% or better were observed for messages presented over the headset, although there was a clear right ear advantage in noise. Messages coming over the loudspeakers were more difficult to understand but a visual cue directing attention to the source of an incoming targeted message resulted in a significant improvement of 7%. Replacing audio messages over the loudspeakers in noise with visual or audiovisual presentations resulted in an improvement from 71% to 96% that did not negatively affect headset performance. The data suggest that texting is a viable option for communication in cases of degraded audio.

Contributed Paper

2:00

1pNSa4. Investigation of the role for noise canceling headphones to assist speech recall in noise. Marion Burgess and Brett Molesworth (School of Eng. and Information Technol., Univ. NSW, Canberra, Northcott Dr., Canberra, ACT 2612, Australia, m.burgess@adfa.edu.au)

There are many situations where it is necessary to hear, understand and be able to recall spoken information in less than ideal listening conditions. For example within an aircraft cabin where, despite improvements in modern passenger aircraft, noise generated from aircraft engines and aerodynamic airflow make it difficult to hear important on-board safety announcements such as the preflight safety brief. The benefits of headphones that incorporate active noise control in such environments are the focus of a series of research studies. In this paper, we discuss the techniques developed to investigate the use of active noise control headphones on the intelligibility and recall of speech generated outside the headsets in noise typical of that in a commercial aircraft cabin. The initial studies were directed towards assessing the effects on the recall for safety announcements. These studies have been extended to investigate if there are any benefits for those for which English is a second language. The results suggest that the use of active noise control headphones can minimize communication errors in a range of situations and this paper will discuss the methodology adopted and summarize the outcomes.

Invited Papers

2:20

1pNSa5. Integration of a distance sensitive wireless communication protocol to hearing protectors equipped with in-ear microphones. Rachel E. Bou Serhal, Tiago Falk, and Jérémie Voix (Universite de Quebec, Ecole de Technologie Superieure, 1280 Rue Saint Marc Apt. PH3, Montreal, QC H3H 2G1, Canada, rachel.bou.serhal@etsmtl.ca)

Using radio communication in noisy environments is a practical and affordable solution allowing communication between workers wearing hearing protection devices (HPD). However, typical radio communication systems have two main limitations when used in noisy environments: first, the background noise is disturbing the voice signal picked-up and transmitted, and second, that voice signal goes to all listeners on the same radio channel regardless of their physical proximity. A new concept of a so-called "Radio-Acoustical Virtual Environment" (RAVE) addressing these two issues is presented. Using an intra-aural instantly custom molded HPD equipped with both an in-ear microphone and miniature loudspeaker, undisturbed speech is captured from inside the ear canal and transmitted over the wireless radio to the remote listener. The transmitted signal will only be received by listeners within a given spatial range, such range depending on the user's vocal effort and background noise level. This paper demonstrates the technological challenges to overcome and the methodology involved in the implementation of RAVE.

2:40

1pNSa6. Sensorial substitution system from vision to audition using transparent digital earplugs. Damien Lescal, Jean Rouat (GEGI, Université de Sherbrooke, Sherbrooke, QC, Canada), and Jérémie Voix (Génie mécanique, École de Technologie Supérieure, 1100, rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, jeremie.voix@etsmtl.ca)

Since the Tactile Vision Substitution System (TVSS) developed by Bach-Y-Rita in 1960's, several sensorial substitution systems have been developed. In general, the so-called "sensorial substitution" system transform stimuli characteristic of one sensory modality (for example, vision) into stimuli of another sensory modality (for example, audition). These systems are developed to help handicapped persons. We developed a sensorial substitution system from vision to audition. An artificial neural network is used to identify the important parts in the image. The virtual acoustic space technic is used to generate localizable sounds. A sound is associated to each important parts of the image. The entire real-time system has been implemented on iOS platforms (IPhone/IPad/IPod Touch). We associated our system with transparent digital earplugs. This way the user is aware of the audio scene happening around him. The system has been tested on non-blind persons and the results are presented.

3:00–3:20 Break

3:20

1pNSa7. An active hearing protection device for musicians. Antoine Bernier and Jérémie Voix (École de Technologie Supérieure, 1100 Rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, antoine.bernier@ens.etsmtl.ca)

Professional musicians have to deal with two great problems when wearing hearing protection devices (HPDs): the occlusion effect and the isolation effect. The occlusion effect is an unnatural and annoying perception of one's own voice when wearing HPDs. It will affect all musicians whose instrument induces vibrations to the skull, including a singer's vocal tracts and instruments mechanically coupled to the head, such as a trumpet or a violin. The isolation effect is the unnatural sensation of being isolated from a given sound environment. It is caused by a non-uniform attenuation of the HPD over the frequency spectrum and the absence of compensation for psychoacoustic factors, such as uneven loudness perception. These two effects cause a shift of perception between the musician's perception and the audience's perception and therefore compromise the musician's ability to offer a good performance to his audience. This paper presents the design and implementation of an active musician's HPD featuring a feedback active noise control system for occlusion effect reduction as well as psychoacoustic compensations for isolation effect reduction. The proposed test procedure and preliminary performance assessments are presented to validate both the test procedure and the system for future subjects trials on a larger scale.

3:40

1pNSa8. A case-study on the continuous use of an in-ear dosimetric device. Kuba Mazur and Jeremie Voix (Universite du Quebec, Ecole de Technologie Superieure, 1100, rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, kuba.mazur@etsmtl.ca)

In order to further understand the combined effects of occupational and recreational noise exposure with regards to noise induced hearing loss (NIHL), an in-ear dosimeter prototype meant for continuous use was developed. The device acts as a hearing protection device (HPD) and can measure and log effective in-ear sound pressure level as well as unprotected levels. To enable its continuous use, this HPD is also equipped with a bypass feature for "transparent" hearing, input for music or communication devices and interfaces with Android smartphones. The proposed device allows for the implementation of an algorithm accounting for the auditory fatigue recovery rate, providing a true representation of the current accumulated noise dose. This allows for 24 h dosimetry and avoids having the user manually reset the dose back to 0% on the next day and thus assuming complete fatigue recovery has occurred. This paper details the proposed recovery algorithm, presents collected field data, and discusses the benefits as well as real-world challenges of using such a device.

4:00

1pNSa9. Estimation of noise exposure level for subjects wearing hearing protector devices. Cécile Le Cocq (Génie mécanique, École de technologie supérieure, 1100 Notre Dame Ouest, Montréal, QC H3C1K3, Canada, cecile.lecocq@etsmtl.ca), Hugues Nélisse, Jérôme Boutin (Prévention des risques mécaniques et physiques, Institut de recherche Robert-Sauvé en santé et en sécurité du travail, Montréal, QC, Canada), Jérémie Voix, and Frédéric Laville (Génie mécanique, École de technologie supérieure, Montréal, QC, Canada)

Industrial workers are exposed to noise levels that could damage their hearing. The Field-MIRE (F-MIRE) method has been developed to quantify earplug and earmuff attenuations with two microphones located under and outside of the HPD. This technique has been designed to be used in the field, but does not give a direct access to the noise exposure level, that is, the noise level at the head location without the subject. In this article, we present a combination of the F-MIRE method with a proposed technique to estimate the sound pressure level without subject, in order to quantify both the ambient and protected noise exposure levels and deduce the effects on worker hearing. Several experiments have been conducted on four subjects with three types of earplugs and with five types of earmuffs. First, the best location for the microphone outside of the HPD has been determined. Second, correction factors that need to be applied on the outside microphone measurement to estimate the sound pressure level without subject have been quantified. Finally, the proposed technique has been validated with measurements taken in a simulated workplace.

4:20

1pNSa10. Improved hearing conservation in industry: More efficient implementation of distortion product otoacoustic emissions for accurate hearing status monitoring. Annelies Bockstael, Hannah Keppler, and Dick Botteldooren (Ghent Univ., Sint-Pietersnieuwstraat 41, Gent 9000, Belgium, annelies.bockstael@intec.ugent.be)

Preventing occupational hearing damage requires close monitoring of workers' hearing. Implementing Distortion Product Otoacoustic Emissions (DPOAEs) in-field is a sensitive and feasible approach provided that a combination of minimal measuring time and infrequent false-positives—i.e., cases where elevated background compromises DPOAEs—is found. This paper investigates how measurement time can be reduced by carefully selecting the tested frequency span and resolution, and how false-positives are minimized by comparing DPOAEs acquired in noise with DPOAEs previously obtained in optimal test conditions. To test this, DPOAEs have been registered with a 1/8-octave band resolution from 841 Hz to 8 kHz for 60 subjects, in quiet conditions and in white noise levels ranging from 54 dB(A) to 90 dB(A). Measurement accuracy is confronted to decrease the measurement time as a function of frequency resolution and range. Diagnostic importance and sensitivity to background noise is addressed for different frequency regions. Within-subject variation of DPOAEs in noisy conditions is assessed both between different noise conditions and between subsequent probe placement. Obtained test–retest statistics quantify normal variability and allow within normal working routines to select for further investigation persons with DPOAEs falling outside this range.

4:40

1pNSa11. Use of passive hearing protectors and adaptive noise reduction for field recording of otoacoustic emissions in industrial noise. Vincent Nadon (École de technologie supérieure, 6080 rue Laurendeau, appartement 2, MONTREAL, QC H4E3X5, Canada, vincent.nadon.1@ens.etsmtl.ca), Annelies Bockstael (INTEC, Ghent Univ., Ghent, Belgium), Hannah Keppler (Dept. of Oto-Rhino-Laryngology and Logopaedic-Audiol. Sci., Ghent Univ., Ghent, Belgium), Dick Botteldooren (INTEC, Ghent Univ., Ghent, Belgium), Jean-Marc Lina, and Jérémie Voix (École de technologie supérieure, Montreal, QC, Canada)

Distortion product otoacoustic emissions (DPOAEs) can detect noise-induced hearing loss in-field, but their data extraction is very sensitive to background noise. This paper investigates how passive and active noise reduction enhance DPOAE recording based on data collected in white noise from 54 dB(A) to 90 dB(A). Despite considerable high-frequency attenuation from a proper placed DPOAE probe, 54 dB(A) background noise deteriorates the test outcome substantially. More low-frequency attenuation by an extra passive earmuff enables measurements in white noise levels of 70 dB(A). The relationship between external sound level and noise recorded by the DPOAE system has been statistically modeled. Additionally, the upper limits of attenuation improvement are analyzed by quantifying residual physiological noise. Furthermore, for an earplug integrating microphone and speakers of the DPOAE measurement probe, adaptive noise reduction processing on the DPOAE signal is used to improve the signal-to-noise ratio. The adaptive noise reduction (ANR) is implemented using the NLMS algorithm to filter out the ambient noise, measured by the first microphone measuring the DPOAE signal, with a second miniature microphone mounted flush with the external faceplate of the isolating DPOAE probe. Simulated data show that DPOAE response extraction is possible in an environment with noise levels exceeding 70 dB(A).

Session 1pNSb

Noise: Community Noise

Eric L. Reuter, Chair
Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801

Invited Papers

1:00

1pNSb1. Quantifying the ambient community noise environment for optimal industry siting. Tim C. Wiens, Gordon L. Reusing, Slavi Grozev, and Zachary Zehr (Noise and Vib. Services, Conestoga-Rovers & Assoc., 651 Colby Dr., Waterloo, ON N2V 1C2, Canada, twiens@craworld.com)

A road traffic noise model was developed to approximate the ambient noise levels present within a 200 km^2 urban center. Road corridors that included highways, city streets, and country side-roads with varying traffic volumes were modeled to evaluate the existing ambient conditions within the project area. To calculate noise levels, an acoustical model and Traffic Noise Model (TNM) calculation standard was used to account for a variety of real-world variables such as daily average traffic counts, turning counts, speed limits, road composition, elevation, road width, and traffic composition. The model generated noise contours that were used to identify areas of elevated ambient noise levels within the project area that may prove suitable for a medium-sized industrial facility. This modeling technique and ambient community noise analysis allowed for the identification of an optimal site within the project area and also proved to be an approach that can be used to industry's advantage. Urban noise is an emerging issue for growing communities. Locating new facilities within urbanized areas with elevated ambient conditions may minimize community noise impacts, reduce post-construction noise abatement costs, and ultimately promote complementary adjacent land use and sustainable urban densification.

1:20

1pNSb2. Proposed standard—Guidance for developing state noise regulations and local noise ordinances. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

The American National Standards Institute (ANSI) Accredited Standards Committee S12 (Noise) Working Group (WG) 41 has been developing a draft standards document for over 12 years. The current document is in Draft #9, now under development. This document represents the consensus of many stakeholders in the community noise arena, including industry, government, consulting, and the public. The purpose of the document is to provide guidance to government officials, acoustical consultants, and other interested persons on how to develop a community noise ordinance or regulation, which is appropriate for the existing local circumstances. The document addresses issues such as public and government priorities and values, and available resources, and also provides the technical basis to manage the local sound environment. The keys to the effectiveness of the document are that it provides a menu of options for the user, discusses the trade-offs involved for decisions that must be made by government officials, and emphasizes that enforcement of a community noise ordinance is crucial to its success. A description of the current draft is presented.

Contributed Papers

1:40

1pNSb3. Do recent findings on jet noise answer aspects of the Schultz curve? Micah Downing (Blue Ridge Res. and Consulting, 15 W. Walnut St., Ste. C, Asheville, NC 28801, micah.downing@blueridgeresearch.com), Kent Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Sally Anne McInerny (Dept. of Mech. Eng., Univ. of Louisiana at Lafayette, Lafayette, LA), Tracianne Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael James (Blue Ridge Res. and Consulting, Asheville, NC)

Recent research efforts on nonlinear propagation from high performance jet aircraft have revealed an interesting challenge to predicting community response. This challenge focuses on receiver perception of these unique acoustical signals, which contain acoustical shocks that appear to increase their relative loudness and/or noisiness. This current finding suggests a need for an improved description of a receiver perception of the loudness of these signals in order to improve the assessment of noise impacts from these aircraft. Looking backwards, an interesting question emerges: did the earlier low bypass jet engines on commercial and transport aircraft also include these acoustical shocks? If they did contain these features, then the perceptual differences observed between aircraft and other transportation noise sources may be partially explained.

2:00

1pNSb4. Acoustical indicator of noise annoyance due to tramway in in-curve operating configurations. Arnaud Trollé, Catherine Marquis-Favre, and Achim Klein (Lab. of Bldg. and Civil Eng., Univ. of Lyon, Labex CELYA, National School of State Public Works, Rue Maurice Audin, Vaulx-en-Velin 69518 CEDEX, France, arnaud.trolle@entpe.fr)

Tramway gives rise to annoyance. Particularly, tramway in an in-curve operating configuration often emits squeal noises leading to inhabitants' complaints. Noise exposure levels were not sufficient to account for annoyance. Other acoustical factors could explain noise annoyance. A laboratory experiment is carried out in order to identify and characterize the influential acoustical factors of in-curve tramway noises. Subjects are asked to rate the short-term annoyance caused by 14 tramway pass-by noises, recorded *in situ* for various in-curve operating configurations. A psychoacoustical analysis shows that the overall perceived noise level, the irregular character and the treble character of tramway noises influence noise annoyance. These

acoustical features are taken into account through the following indices: the mean loudness, the variance of time-varying A-weighted pressure normalized by RMS A-weighted pressure, and a psychoacoustical index, constructed to account for squeal noise, expressed by the total energy of the tonal components within critical bands from 12 to 24 Barks. A multilevel regression analysis reveals that a combination of these indices proves to be satisfactory for predicting short-term annoyance due to in-curve tramway noises. These results are consistent with those from a previous experiment that implied 61 tramway pass-by noises corresponding to different operating situations.

2:20

1pNSb5. Rail noise and vibration in Australia—A case study. Vincent Chavand (Air & Noise Service Line, GHD, Level 3, GHD Tower, 24 Honeysuckle Dr., Newcastle, NSW 2300, Australia, vincent.chavand@ghd.com)

This paper reviews the various stages of a major rail project undertaken in New South Wales (NSW), Australia, between 2010 and 2012. This case study involves 40 km of new track adjacent to existing railway lines in the Hunter Valley, NSW. The project is located within a mixture of rural and urban settings and had the potential to impact on a large number of sensitive receivers during both the construction and operational phases. The project approvals required compliance with a number of relatively new noise and vibration guidelines and policies, which provides an opportunity for the author to reflect on the recent evolutions in noise and vibration control practices and policing in Australia. This paper reviews the project from the approvals process to its commissioning phase from a noise and vibration point of view. It explores the construction and operational noise modeling methodologies, reviews the design process and adopted mitigation measures and, in doing so, it discusses the practical challenges met through the course of the works.

2:40–3:00 Break

3:00

1pNSb6. Noise pollution in urban settings of the Western Amazonia and an approach to cope with. Stephan Paul (Undergrad. Program Acoust. Eng., Fed. Univ. of Santa Maria, Av. Roriama 1000, Camobi, Santa Maria, RS 97105-900, Brazil, stephan.paul@eac.ufsm.br), Isabel C. Kuniyoshi (Dept. of Speech and Hearing Sci., Faculdade Sao Lucas, Porto Velho, Brazil), Flávio André M. de Araújo (Office of the Federal Public Prosecutor of Rondonia State, Porto Velho, Brazil), and Lucinara Camargo (Dept. of Environmental Protection SEMA, Municipality of Porto Velho, Porto Velho, Brazil)

The Amazon Region is usually well known for its nature and not for the noise pollution that recently took over in many cities, both major ones such as Porto Velho, Manaus, or Belém, as well as minor cities. Transparent facades with large openings of places like restaurants and bars, heavy traffic, and young adults competing with high power car sound systems are the most common sources of noise pollution and annoyance. Results from noise measurements done at different moments at several "hot spots" in Porto Velho, capital of the state of Rondônia, will be presented. A-weighted equivalent sound pressure levels at all measurement points were found to be in exceedance of 70 dBA during several hours at night, causing a lot of annoyance to the population. Besides the results of measurements some results of the state authorities approach to combine a campaign to raise noise awareness and noise policy enforcement will be presented.

3:20

1pNSb7. Sound power level of speaking people. Marco Caniato (Univ. of Trieste, via valerio 6/a, Trieste 34100, Italy, mcaniato@units.it), Federica Bettarello (Acusticamente - Designing Team, Conegliano, Italy), and Michele Taffarel (M&T Eng., Conegliano, Italy)

In restaurants and cafés many sound sources are present: music, refrigerant, and cooling equipment and people speaking. The smoking prohibition law did move out people creating a lot of aggregation areas outdoor, both in summer and in winter time. As a matter of facts many cafés open on the outer part a stallage in order to provide beverages to outside costumers. In this way, the "people speaking" source became a common problem to deal with and solve. In order to characterize this particular sound source in terms of sound power level of a typical stallage situation full of speaking people, sound pressure power measurements according to ISO 3446 standard were carried out. The results confirm the first investigation achievements provided by Sepulcri *et al*. with a non-direct method.

3:40

1pNSb8. New athletic fields for Saint Joseph's University: A community noise battle. Felicia Doggett (Metropolitan Acoust., LLC, 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, felicia@metropolitanacoustics.com)

Saint Joseph's University, located in Philadelphia, purchased a vacated private school campus adjacent to theirs. This new campus, which resided in neighboring Lower Merion Township, included 15 acres of open fields that the university desired to turn into NCAA (National Collegiate Athletic Association) baseball, softball, and field hockey fields. Proposed improvements to the fields included permanent bleacher seating, press boxes, dugouts, batting cages, artificial turf playing surfaces, re-grading of the fields, and sound reinforcement systems among other changes. The surrounding community vehemently opposed the proposed changes largely due to the potential noise generated from cheering crowds and sound systems. Through a year and a half of township hearings with testimony provided by a plethora of expert witnesses, Saint Joseph's University finally received approval with various restrictions to build their fields. This presentation explores the research, measurements, and modeling methods undertaken to quantify the acoustical implications on the surrounding community.

4:00

1pNSb9. Noise control for rooftop chiller units: An application in Istanbul. David Meredith (Kinetics Noise Control, 6300 Irelan Place, Dublin, OH 43065, dmeredith@kineticsnoise.com), Hakan Dilmen, Merve Çay (Pro-Plan Proje Mühendislik San. ve Tic. Ltd, İstanbul, Turkey), H. Temel Belek, and Ahmet Arisoy (İstanbul Teknik Üniversitesi Makina Fakültesi, İstanbul, Turkey)

A telecommunication corporation located in Gayrettepe, Istanbul, installed 20 rooftop chillers on their six-story building to air condition their newly introduced client Internet server hall in spring 2010, causing annoyance in the mostly residential neighborhood during the summer months. After a series of night-time acoustic measurements were performed to characterize the noise emitted by the chillers, a three-dimensional noise model of the area was created. Using this model, a ventilated noise control barrier was designed to bring down the contribution of the chiller units to the overall environmental noise level. Following the manufacturing and installation of the barrier, a series of night-time measurements were performed anew, which demonstrated that the application has mitigated the contribution of the noise emitted by the rooftop units to within regulation limits.

Session 1pPAa

Physical Acoustics and Biomedical Acoustics: Acoustics in Microfluidics and for Particle Separation IIb: Bubbles and Drops

Martin Wiklund, Cochair
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Yong-Jae Kim, Cochair
Mech. Eng., Texas A&M Univ., College Station, TX 77843

Contributed Papers

1:00

1pPAa1. Acoustic bubble behavior in a standing wave field. Cyril Desjouy, Pauline Labelle, Bruno Gilles, Jean-Christophe Bera, and Claude Inserra (U1032, LabTAU, Inserm, 151 Cours Albert Thomas, Lyon cedex 03 69424, France, cyril.desjouy@inserm.fr)

This paper focuses on the experimental and numerical studies of acoustic cavitation induced micro bubbles in a standing waveguide filled with water. It is shown that the cylindrical geometry of the system used in this study allows the micro bubbles to self organize into particular patterns. At high pressure amplitudes, the cavitation bubbles tend to aggregate into well known cluster patterns and at relatively low pressure amplitudes, the cavitation micro bubbles aggregate into ring patterns. This study highlights that the shape of these ring patterns is directly related to the Bjerknes force distribution in the resonator. It is also shown both experimentally and numerically that cavitation bubbles may exhibit spiraling behavior around this ring pattern. This spiraling phenomenon is numerically studied and the conditions for which a single cavitation bubble follows an orbital trajectory in the cylindrical waveguide are established, and the influences of the acoustic pressure amplitude and the initial bubble radius are investigated.

1:20

1pPAa2. The roles of acoustic cavitations in the ultrasonic cleansing of fouled micro-membranes. Yuanxiang Yang (School of Civil and Environmental Eng. and DHI-NTU Ctr., Nanyang Technolog. Univ., Nanyang Ave. 50, N1-B1-3a, Maritime Res. Ctr., Singapore 639798, Singapore, yang0250@e.ntu.edu.sg), Qianxi Wang (School of Mathematics, The Univ. of Birmingham, Birmingham, United Kingdom), and Soon Keat Tan (Nanyang Environment and Water Res. Inst. and Maritime Res. Ctr., Nanyang Technolog. Univ., Singapore, Singapore)

This paper concerned the experimental studies on the cleansing mechanism of acoustic cavitation bubbles near the fouled micro-membranes. The existence of the membrane created asymmetry in the flow field, which forced the cavitation bubble to oscillate non-spherically and finally brought forth the jet impact. The oscillations and micro-jets of the cavitation bubbles enhanced the dynamic features of the fluid nearfield and improved the capability of removing fouling. The study on the acoustic multi-bubble system was quite complicated, so first, we focused on the individual bubble dynamics near the membrane. A succession of individual cavitation bubbles were created by Q-switched Nd: YAG laser pulses and observed using a high-speed camera (up to 100 000 frames per second). The jet flow hit against the membrane surface with velocity above 100 m/s, which was strong enough to remove the adherent fouling. We compared the cleansing effects of the cavitation bubbles with different laser energies and stand-off distances from the membrane surface. And then based on the individual bubble dynamics, we can deduce the influence of cavitations in the ultrasonic cleansing of micro-membranes.

1:40

1pPAa3. Waves of acoustically induced transparency in bubbly liquids: Theory and experiment. Nail A. Gumerov (Inst. for Adv. Comput. Studies, Univ. of Maryland, 115 A.V. Williams Bldg., College Park, MD 20742, gumerov@umiacs.umd.edu), Claus-Dieter Ohl (School of Physical and Mathematical Sci., Div. of Phys. and Appl. Phys., Nanyang Technolog. Univ., Singapore, Singapore), Iskander S. Akhatov (Mech. Eng., North Dakota State Univ., Fargo, ND), Sergei P. Sametov, and Maksim V. Khazimullin (Ctr. for Micro- and Nanoscale Dynam. of Dispersed Systems, Bashkir State Univ., Ufa, Russian Federation)

The theory of self-organization of bubbles in acoustic fields predicts formation and propagation of waves of self-induced acoustic transparency. This is a strongly nonlinear effect, which is a result of a two-way coupling of the sound field with the bubble distribution. We are challenging the theory with an experiment. Here, a homogeneous distribution of gas bubbles is first generated and then an ultrasonic field is switched on. The pressure waves are below the cavitation threshold and in a frequency range from 50 kHz to 200 kHz, mostly above the linear resonance frequency of the bubbles. The ultrasound leads to a rapidly propagating bubble wave away from the transducer. The dynamics is observed with a high-speed camera and analyzed. Interestingly, this transparent region is propagating through the bubbly liquid at velocities substantially higher than the bubble rise velocity due to the gravity. A simplified theoretical model of this acoustically induced transparency is developed. Both, analytical and numerical solutions are obtained. A comparison of the experimental data with the model is presented and the underlying physics of the problem is discussed.

Session 1pPAb

Physical Acoustics and Biomedical Acoustics: Acoustics in Microfluidics and for Particle Separation III: Biological Applications

Michel Versluis, Cochair
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Lawrence A. Crum, Cochair
Appl. Phys. Lab., Univ. of Washington, Ctr. for Industrial and Medical Ultrasound, Seattle, WA 98105

Invited Papers

2:00

1pPAb1. Acoustic deformation of cells. Puja Mishra, Martyn Hill, and Peter Glynne-Jones (Faculty of Eng. and the Environment, Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, pgj98r@gmail.com)

The stretching of cells using optical tweezers has been previously demonstrated, enabling the mechanical properties of individual cells to be assessed with potential applications in, for example, identifying cancer cells and parasite infections such as malaria. We demonstrate here a system that uses acoustic radiation forces to compress levitated cells to an extent comparable to that demonstrated with optical tweezers. While the deformation of levitated droplets has been demonstrated in the past, this paper addresses the challenge of producing significant forces on objects that have low acoustic contrast with their host medium and are small compared to the acoustic wavelength. The acoustic deformation can potentially be applied to many (e.g., thousands) of cells simultaneously, opening the way to higher throughput diagnostic devices. In our system, osmotically swollen red blood cells (RBCs) are used to demonstrate the principle as they are particularly compliant due to the absence of a cytoskeleton. A resonance is formed in a square capillary of inner dimension, 100 μm. Excited by a transducer at a half-wave resonance of 7.9 MHz, cells are both levitated and focused laterally into a single line down the center of the capillary prior to the compression forces being applied. We present finite element models of the acoustic deformation, verifying our code against known results for droplet deformation.

2:20

1pPAb2. Application of acoustic radiation pressure to align cells in a commercial flow cytometer. Gregory Kaduchak and Michael D. Ward (Molecular and Cell Essentials, Life Technologies, 29851 Willow Creek Rd., Eugene, OR 97402, greg.kaduchak@lifetech.com)

Forces derived from acoustic radiation pressure can be used to replace or partly replace hydrodynamic forces to align cells and particles in flow cytometry. The ability to focus cells into a tight line without relying on hydrodynamic forces allows many new possibilities for sample delivery. Dilute samples can be processed quickly. Flow velocities can be varied allowing control of particle delivery parameters such as laser interrogation time and volumetric sample input rates. Recently, a commercial flow cytometer that directs particles into the laser interrogation region using acoustic radiation pressure in a 200 □m channel has been developed. In this talk, the application of acoustic radiation pressure in flow cytometry systems from fundamental principles to implementation details will be presented. Data will be shown for both the operational implementation of the acoustic focusing device as well as demonstrating its ability to perform for complex biological assays.

2:40

1pPAb3. On-chip acoustic sample preparation for cell studies and diagnostics. Martin Wiklund, Ida Sadat Iranmanesh, Athanasia E. Christakou, Mathias Ohlin (Appl. Phys., Royal Inst. of Technol., KTH-Albanova, Stockholm 10691, Sweden, martin@biox.kth.se), Aman Russom (Biotechnology, Royal Inst. of Technol., Stockholm, Sweden), and Björn Önfelt (Appl. Phys., Royal Inst. of Technol., Stockholm, Sweden)

We describe a novel platform for acoustic sample preparation in microchannels and microplates. The utilized method is based on generating a multitude of acoustic resonances at a set of different frequencies in microstructures, in order to accurately control the migration and positioning of particles and cells suspended in fluid channels and chambers. The actuation frequencies range from 30 kHz to 7 MHz, which are applied simultaneously and/or in sweeps. We present two devices: A closed microfluidic chip designed for pre-alignment, size-based separation, isolation, up-concentration, lysis of cells, and an open multi-well microplate designed for parallel aggregation and positioning of cells. Both devices in the platform are compatible with high-resolution live-cell microscopy, which is used for fluorescence-based optical characterization. Two bioapplications are demonstrated for each of the devices: The first device is used for size-selective cell isolation and lysis for DNA-based diagnostics, and the second device is used for quantifying the heterogeneity in cytotoxic response of natural killer cells interacting with cancer cells.

3:00

1pPAb4. Numerical modeling for analyzing microfluidic acoustophoretic motion of cells and particles with application to identification of vibro-acoustic properties. Zhongzheng Liu (Mech. Eng., Texas A&M Univ., College Station, TX), Han Wang, Arum Han (Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), and Yong-Joe Kim (Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77843, joekim@tamu.edu)

Microfluidic, acoustophoretic cell/particle separation has gained significant interest recently. In order to analyze the motion of cells/particles in the acoustophoretic separation, a one-dimensional (1-D) analytical model in a "static" fluid medium has been widely used, while the effects of acoustic streaming, viscous boundary layers, and 2-D and 3-D geometries are usually not considered. Therefore, it is not sufficient to accurately predict the cell/particle motion. Thus, a numerical modeling procedure for analyzing the acoustophoretic cell/particle motion is presented to include the aforementioned effects. Here, the first-order acoustic pressure and the second-order acoustic streaming velocity are first calculated by using a high-order finite difference method. Then, acoustophoretic force is calculated based on the acoustophoretic force equation proposed by Gorkov and is applied to the Newton's equation of motion to calculate the motion of cells/particles. Through various simulations, the effects of acoustic streaming on the motion of cells/particles are studied. Since the acoustophoretic motion depends on the vibro-acoustic properties (e.g., density, compressibility, and size) of particles/cells, the vibro-acoustic properties can be estimated by optimally fitting the experimental and simulated trajectories. The properties obtained from experimental results with polystyrene beads show good agreement with the data reported in literature.

3:20

1pPAb5. Acoustic radiation force to reposition kidney stones. Michael Bailey, Yak-Nam Wang, Julianna C. Simon, Bryan W. Cunitz (Ctr.Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), Jonathan D. Harper, Ryan S. Hsi (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Frank Starr, Marla Paun, Barbrina Dunmire (Ctr.Industrial and Medical Ultrasound, Applied Physics Lab, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Seattle, Washington), Lawrence A. Crum (Ctr.Industrial and Medical Ultrasound, Applied Phys. Lab., Univ. of Washington, Seattle, WA), and Mathew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA)

Our group has introduced transcutaneous ultrasound to move kidney stones in order to expel small stones or relocate an obstructing stone to a nonobstructing location. Human stones and metalized beads (2–8 mm) were implanted ureteroscopically in kidneys of eight domestic swine. Ultrasonic propulsion was performed using a diagnostic imaging transducer and a Verasonics ultrasound platform. Stone propulsion was visualized using fluoroscopy, ultrasound, and the ureteroscope. Successful stone movement was defined as relocating a stone to the renal pelvis, ureteropelvic junction (UPJ), or proximal ureter. Three blinded experts evaluated for histologic injury in control and treatment arms. All stones were moved. 65% (17/26) of stones/beads were moved the entire distance to the renal pelvis (3), UPJ (2), or ureter (12). Average successful procedure per stone required 14 ± 8 min and 23 ± 16 pushes. Each push averaged 0.9 s in duration. Mean interval between pushes was 41 ± 13 s. No gross or histologic kidney damage was identified in six kidneys from exposure to 20 1-s pushes spaced by 33 s. Ultrasonic propulsion is effective with most stones being relocated to the renal pelvis, UPJ, or ureter. The procedure appears safe without evidence of injury. [Work supported by NIH DK43881, DK092197, and NSBRI through NASA NCC 9-58.]

Contributed Papers

3:40

1pPAb6. Macro-scale acoustophoretic separation of lipid particles from red blood cells. Brian P. Dutra (FloDesign Sonics Inc., 380 Main St., Wilbraham, MA 01095, b.dutra@fdsonics.com), Michael Rust (Biomedical Eng., Western New England Univ., Springfield, MA), Daniel Kennedy (Pharmacology, Western New England Univ., Springfield, MA), Louis Masi (FloDesign Sonics Inc., Wilbraham, MA), and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Autologous blood salvage is frequently used in cardiac surgery. However, shed mediastinal blood contains lipid particles ranging in size from 10 to 60 μm. Lipid emboli flow subsequently lodge in the brain capillaries resulting in strokes, leading to neurocognitive dysfunction and death. A novel acoustophoretic filtration system has been developed to separate the lipids from the red blood cells (RBCs). The system works at the macro-scale, supporting flow rates in excess of 2 L/h. The system is designed such that the acoustic radiation force is able to overcome the combined effects of fluid drag and buoyancy forces. Both RBCs and lipid particles are therefore trapped in the ultrasonic standing wave. Due to the opposite contrast factors of lipids and RBCs, the two components separate at opposite nodes within the standing wave, with lipids concentrating at pressure anti-nodes and RBCs at pressure nodes. Subsequent gravitational separation is used to separate the lipids and RBCs. Preliminary results were obtained with a suspension of 10x diluted bovine blood mixed with a 0.75% safflower oil emulsion. Measurements indicate a 15 fold increase in hematocrit of the captured RBCs when compared to the original sample solution, and an excellent separation of the oil droplets.

4:00

1pPAb7. Removal of living cells from biosensing surfaces in droplet-based microfluidics using surface acoustic waves. Adrien Bussonnière, Alan Renaudin (Université de Sherbrooke, Pavillon 3IT Parc innovation, 3000 boulevard de l'université, Sherbrooke, QC J1K 0A5, Canada, adrien.bussonniere@usherbrooke.ca), Yannick Miron, Michel Grandbois (Pharmacologie, Université de Sherbrooke, Sherbrooke, QC, Canada), Michaël Baudoin (FILMS, IEMN, Lille, France), and Paul Charette (Université de Sherbrooke, Sherbrooke, QC, Canada)

Removal of living biological cells from surfaces is a critical process for many applications in the area of biosensing and lab-on-a-chip. Trypsin is one of the most effective biochemical tools used to cleave the cells proteins that are responsible for bonding cells to surfaces [K. A. Walsh, Meth. Enzymol. **19**, 41 (1970)]. We propose a method using Rayleigh-type (20 MHz) surface acoustic wave (SAW)-based mixing [Renaudin *et al*., Lab Chip **10**, 111 (2010)] as an accelerator for trypsin-mediated removal of living cells from surfaces. In the experiments, a 10 μL droplet of Hank's Balanced Salt Solution (HBSS)-trypsin is placed on a piezoelectric substrate covered with human embryonic kidney cells (HEK293). Using phase contrast microscopy, cells removal time for different acoustic power levels and trypsin concentrations is measured. Results from validation experiments show that a minimum of 180 s is necessary to completely release surface-bonded cells covered by the 10 μL droplet without the use of SAW (negative control). By using microstreaming flow in the droplets generated by the SAW, cells are released from the surface in less than 8 s. This work will contribute to improved lab-on-a-chip devices based on living cell biosensing.

4:20

1pPAb8. Acoustophoretic force-based compressibility measurement of cancer cells having different metastatic potential. Han Wang (Dept. of Elec. and Comput. Eng., Texas A&M Univ., College Station, TX), Zhongzheng Liu (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX), Dong Moon Shin, Georgia Chen (Dept. of Hematology and Medical Oncology, Emory Univ. School of Medicine, Atlanta, Texas), Younghak Cho (Dept. of Mech. System Design Eng., Seoul National Univ. of Sci. and Technol., Seoul, South Korea), Yong-Joe Kim (Dept. of Mech. Eng., Texas A&M Univ., College Station, TX), and Arum Han (Dept. of Elec. and Comput. Eng., Texas A&M Univ., 309C WERC, TAMU 3128, College Station, TX 77843-3128, arum.han@ece.tamu.edu)

Mechanical properties of cells such as compressibility are regarded to be different as cancer cells progress into metastatic state. Traditional methods for measuring mechanical properties of single cells such as AFM and micropipette aspiration require labor-intensive procedures and can cause damage to cells due to direct contact, thus unsuitable for high-throughput measurement. Acoustophoretic force exerted on particles under acoustic-standing-waves depends on the particle and medium's vibro-acoustic properties. Thus, cells with different mechanical properties show different mobility under acoustic resonant field, which can be analyzed to decipher the mechanical properties of cells. Here we present a high-throughput, single-cell-resolution, cell compressibility measurement approach based on acoustic-standing-wave-induced force, and the finding that head and neck cancer cells having different metastatic capacities show noticeable differences in compressibility. The acoustophoresis chip has a straight flow channel with a piezoelectric transducer attached at the bottom. Trajectories of moving cells in the channel under acoustic standing wave excitation in the absence of flow are recorded. By using a microfluidic acoustophoretic model, the simulated trajectories of cells are calculated. The mechanical properties of cells are estimated by fitting the experimental and simulated trajectories thereby. Cells with highest metastatic capacity showed highest compressibility, consistent with previously reported clinical observations.

4:40

1pPAb9. Acoustic separation of sub-micron particles in gases. Etienne Robert, Ramin Imani Jajarmi (Mechanics, Kungliga Tekniska Högskolan (KTH), Osquars Backe 18, Stockholm 100 44, Sweden, etienne@mech.kth.se), Jan B. Pettersson (Chemistry and Molecular Biology, Göteborg Univ., Göteborg, Sweden), and Klas Engvall (Chemical Technol., Kungliga Tekniska Högskolan (KTH), Stockholm, Sweden)

In several areas of science and technology, there is a strong need for concentrating, separating, and sorting small particles suspended in gaseous flows. Acoustic fields can be used to accomplish this task, an approach extensively used in liquid phase microfluidics that has great potential for aerosol treatment. This paper presents an experimental investigation of acoustophoresis for very small particles in gases, with sizes ranging from tens to hundreds of nanometers. The phenomenon is studied in a rectangular channel with variable height in which a standing acoustic field is created by a broadband electrostatic transducer operated in the 50–100 kHz range. The flow can either be seeded with particles with a known size distribution or ambient laboratory air can simply be circulated in the channel. Downstream of the separation channel, the flow is separated into enriched and depleted streams with adjustable slits for analysis. The particle number density and size distribution is measured with a scanning mobility particle sizer (SMPS) as a function of position in the standing wave pattern. From these measurement, the separation efficiency is determined as a function of the particle size, excitation frequency, bulk flow velocity, and number of nodes in the channel. Further analysis yields an estimation of the force acting on the particles, which for very small particles yields novel information on the magnitude of acoustophoretic forces in the transition and molecular flow regimes.

MONDAY AFTERNOON, 3 JUNE 2013 — 514ABC, 1:00 P.M. TO 5:00 P.M.

Session 1pPPa

Psychological and Physiological Acoustics: Binaural Hearing and Binaural Techniques I

Janina Fels, Cochair
Inst. of Techn. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52074, Germany

Pablo Hoffmann, Cochair
Aalborg Univ., Fredrik Bajers Vej 7B5, Aalborg 9220, Denmark

Invited Papers

1:00

1pPPa1. Applications of models of binaural hearing. Jens P. Blauert (Inst. of Commun. Acoust., Ruhr-Univ. Bochum, Bochum D-44780, Germany, jens.blauert@rub.de) and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Models of binaural hearing have been proposed for more than 60 decades, but it is only recently that they are ready for technological application. Recognizing this situation, 15 laboratories in Europe and in the United States have founded a research group (AABBA) with the aim of setting in place "Aural Assessment by Means of Binaural Algorithm." Now, after its first 4-years term of activity, the group has published their recent results in a chapter book (Springer)—including a software package for building customized binaural models. In this talk, the group's most relevant results will be described. The results concern the following areas of model application, among others: binaural analysis of aural scenes, binaural de-reverberation, binaural quality assessment of audio channels, loudspeakers and performance spaces, binaural perceptual coding, binaural processing in hearing aids and cochlea implants, binaural systems in robots, binaural/tactile human–machine interfaces, speech-intelligibility prediction in rooms, and/or multi-speaker scenarios. Current members of AABBA are research labs at Universities/Techn. Universities in Helsinki, Boston, Cardiff, Oldenburg, Lyon, Troy, NY (Rensselaer), Bochum, Berlin, Copenhagen, Dresden, Eindhoven, Munich, Paris (Pierre et Marie Curie), Toulouse, and at the Austrian Academy's Acoustics Research Inst. in Vienna.

1:20

1pPPa2. An improved method for head-related transfer function interpolation and grid matching. Aussal Matthieu (DIGITAL MEDIA SOLUTIONS, 45, grande allée du 12 février 1934, Noisiel 77186, France, matthieu.aussal@dms-cinema.com), Alouges François (CMAP, ECOLE POLYTECHNIQUE, Palaiseau, France), and Katz F. Brian (LIMSI-CNRS, Orsay, France)

Today, there exists a growing number of HRTF datasets available with each set often proposing a unique variation on the spatial discretization measurement grid. These differing grids, typically determined by the mechanical system employed, result in datasets, which are not directly comparable or exploitable. To alleviate the limitation of incompatible grids and assist in the adaptation of measurements performed on one grid to another, facilitating the inter-exchange of HRTF sets, a fixed radius HRTF interpolation method is proposed. The approach is based on decomposition of the sound field using spherical harmonics, allowing for a global spatial recomposition. The frequency domain HRTF is separated into its complex components and interpolations are performed independently before reconstitution. Spherical harmonic truncation order is chosen to provide a system which is roughly square, improving matrix inversion with Tikhonov regularization. A high spatial density HRTF was used as a test case for evaluating the interpolation method. A series of measures are employed to quantitatively compare the quality of the interpolation as compared to traditional interpolation methods in both the time and frequency domains.

1:40

1pPPa3. Statistical analysis of head related transfer function data. Yuancheng Luo, Dmitry N. Zotkin (Comput. Sci. and UMIACS, Univ. of Maryland, College Park, MD), and Ramani Duraiswami (VisiSonics Corp., A.V. WIlliams Bldg., #115, College Park, MD 20742, ramani@umiacs.umd.edu)

The head related transfer function (HRTF) is a function that characterizes the response of a given individual to sound from a particular location in an egocentric coordinate system. The range dependence is often neglected, and the HRTF is approximated as a function of frequency and direction, $H(\theta,\varphi,\omega)$. The HRTF displays considerable inter-personal variability, and a major open problem is the development of a generative model for the HRTF from anthropometry. Further, the sampling used in measuring HRTF data varies widely from database to database, and moreover often there are no measurements for elevations below the subject. This raises associated questions of optimal sampling, interpolation, hole-filling, and others. In this work, we model the HRTF via a non-parametric, data-driven, Gaussian Process Regression model. We develop efficient regression techniques to perform inference using this model on measured HRTF data. We then suggest methods for HRTF interpolation, HRTF extrapolation, feature extraction, sampling, and personalization. The methods are tested on the CIPIC database and results presented. [Partial NSF support is gratefully acknowledged.]

2:00

1pPPa4. Toward optimal functional representation of head-related transfer functions in the horizontal plane. Wen Zhang, Thushara D. Abhayapala, Rodney A. Kennedy (Res. School of Eng., Australian National Univ., Canberra, ACT 0200, Australia, wen.zhang@anu.edu.au), and Mengqiu Zhang (School of Elec. Eng., KTH Royal Inst. of Technologyity, Stockholm, Sweden)

Head-related transfer function (HRTF) individualization using principle component analysis (PCA) modeling rely on the empirical data to reduce HRTF dimensionality for an optimal representation and to achieve HRTF personalization by tuning the model weights with the subject anthropometric parameters. However, for these representations, the basis is discrete and data dependent, which can limit its usefulness in universal HRTF representation. This paper studies the optimal functional representation of magnitude HRTF of 45 subjects for sound sources in the horizontal plane. We first use circular harmonics to extract the subject-independent HRTF angular dependence. The remaining spectral components of 45 subjects are then modeled by PCA and two standard functions, i.e., Fourier series and Fourier Bessel series. The metric to evaluate the model efficiency is the expansion weights cumulative variance. We identify that individual magnitude HRTFs over 20 kHz range could be modeled adequately well by a linear combination of only 19 Fourier series; this is a near optimal representation in comparison with the statistical PCA model. Further analysis of the model weights with subjective anthropometric measurements will provide a promising method for HRTF individualization, especially considering the nature of data independent continuous basis functions employed in the proposed functional representation.

2:20

1pPPa5. A three dimensional children head database for acoustical research and development. Stine Harder, Rasmus R. Paulsen (Informatics and Mathematical Modelling, Tech. Univ. of Denmark, Richard Petersens Plads., Bldg. 321, Office 215, Kgs. Lyngby 2800, Denmark, rrp@imm.dtu.dk), Martin Larsen (Oticon A/S, Smørum, Denmark), and Søren Laugesen (Oticon Res. Ctr. Eriksholm, Snekkersten, Denmark)

Most computational–acoustic work within spatial hearing relies on head-related transfer functions from databases of measurements taken on adult humans or dummy heads. We aim to provide a set of 3D digital heads including children, from which head-related transfer functions can be computed instead of measured. However, current volumetric scanning techniques do not have sufficient resolution for accurately scanning the external ear, and computed tomography also involves radiation. In this paper, we propose a framework for scanning, stitching, and meshing complete human heads. The process starts by acquisition of multiple 3D surface scans of the same subject using a high-resolution photogrammetric scanner. Second, the scans are semi-automatically aligned and noise and incoherence is removed. This is followed by an iterative process where a volumetric implicit representation of the head is optimized. The process consists of a regularized surface-reconstruction step followed by an alignment step. Finally, a surface representation of the entire head is extracted using a triangulation of the zero-level iso-surface of the implicit volume. The process has been used to reconstruct the heads of children aged 10 months to 9 years. The data and the associated reconstruction algorithms will be made publicly available for use in acoustical research and development.

2:40

1pPPa6. Consistency among the head-related transfer functions from different measurements. Xiao-li Zhong and Bo-sun Xie (Acoustic Lab., Phys. Dept., School of Sci., South China Univ. of Technol., Bldg. No.18, Wu Shan Rd. No.381, Guangzhou, Guangdong 510641, China, xlzhong@scut.edu.cn)

Empirical measurement is a common approach to obtaining head-related transfer functions (HRTFs). Due to differences in experimental conditions and possible errors, some deviations exist among the data from different measurements even for the same subject. This work aims to evaluate deviations of HRTFs from different measurements. Five sets of KEMAR HRTFs from three groups including MIT-media Lab, CIPIC Interface Lab, and our lab are used. A free-field equalization is applied to the original data so as to eliminate the influence caused by the difference in the response of electro-acoustic measuring chain. The deviations among HRTF magnitudes for different measurements are specified by spectral distortion. Results indicate that the magnitude deviation increases with increasing frequency and reaches more than 10 dB above the frequency of 6 kHz. Salient deviations often occur in contralateral source directions or in ipsilateral source directions near the frequency of pinna-related notch. Nevertheless, most deviations can be effectively reduced after an auditory filter smoothing. This work suggests that the inconsistency in measured HRTF data and its impact on auditory perception should be taken into account when comparing and standardizing HRTFs from different measurements. [Work supported by National Nature Science Fund of China Grant No. 11004064]

3:00–3:20 Break

3:20

1pPPa7. Conventional and spatial principal component analysis on near-field head-related transfer functions. Bo-sun Xie and Cheng-yun Zhang (Acoust. Lab., Phys. Dept., School of Sci., South China Univ. of Technol., Bldg. No.18, Wu Shan Rd. No. 381, Guangzhou 510641, China, phbsxie@scut.edu.cn)

Head-related transfer functions (HRTFs) vary with both frequency and source position. The near-field HRTFs with source distance less than 1.0 m are particularly complicated due to their distance dependence. Principal component analysis (PCA), which is conventionally carried out in the frequency or time domain, has been widely applied to reduce the dimensionality of far-field HRTF data with source distance greater than 1.0 m. The present work first extends the conventional PCA, and then proposes a spatial PCA method in the spatial domain rather than in the frequency or time domain to reduce the dimensionality of near-field HRTF data. An illustrative case indicates that near-field HRTF magnitudes at 9 distances with 493 directions for each distance can be approximately represented by the weighted sum of 15 spectral shape basis functions using the conventional PCA or the weighted sum of 15 spatial basis functions using the spatial PCA. Both representations account for more than 98% energy variation of the original data, and reduce the dimensionality of the original data to about a quarter. The proposed method is also applicable to the head-related impulse responses in the time domain. Furthermore, the spatial PCA scheme is potentially applicable to simplify near-field HRTF measurement.

3:40

1pPPa8. Spatially continuous model of the broadband time-of-arrival in the head-related transfer functions. Piotr Majdak and Harald Ziegelwanger (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Wien 1040, Austria, piotr@majdak.com)

Head-related transfer functions (HRTFs) describe the filtering of the incoming sound by the human anatomy. They contain the so-called broadband time-of-arrivals (TOAs), which interaural differences yield the well-known interaural time differences used to estimate the lateral position of sound sources by the human auditory system. The TOAs are essential for a time-synchronous binaural rendering of multiple virtual sound sources or for interpolation of the timing information in an existing HRTF set. Estimation of the TOA is usually done separately for each spatial direction and is thus prone to errors and directional outliers. A method for a robust estimation of spatially continuous TOA function from a set of listener-specific HRTFs is presented. The method relies on a geometric model of the HRTF-measurement setup represented by parameters like head position, radius, and ear position. The model parameters were fit to HRTFs of a sphere numerically calculated under various conditions, and to measured HRTFs of 160 listeners. The resulting model parameters and TOA functions corresponded well with the measurement geometry and manually derived TOAs, respectively. The model parameters were further compared to those resulting from a simplified model which assumes the listener being in the center of the HRTF-measurement setup, demonstrating the impact of the usually neglected aspect of listener position on the HRTF timing quality.

4:00

1pPPa9. Calculation of listener-specific head-related transfer functions: Effect of mesh quality. Harald Ziegelwanger, Piotr Majdak (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna, Vienna A-1040, Austria, harald.ziegelwanger@oeaw.ac.at), and Andreas Reichinger (Zentrum für Virtual Reality und Visualisierung Forschungs-Gmbh, Wien, Austria)

The geometry of head and ears defines the listener-specific directional filtering of the incoming sound. The filtering is represented by the head-related transfer functions (HRTFs), which provide spectral features relevant for the localization of sound-sources. HRTFs can be acoustically measured or numerically calculated based on a geometric representation of the listener. While the acoustically measured HRTFs usually provide localization performance similar to that obtained in free-field listening, the performance obtained with numerically simulated HRTFs, however, heavily depends on the quality of the geometric and acoustic model of the listener used for the simulation. In this study, we show how to calculate listener-specific HRTFs with spectral features similar to that from acoustically measured HRTFs for the entire audible frequency range. We review the boundary-element method coupled with the fast-multipole method and we present details on the prerequisites like the geometry-capture technique, acoustical parameters, and the numerical algorithms. Further, the effect of the mesh quality on the HRTFs was investigated by systematically varying the average edge length from 1 to 5 mm. The HRTF amplitude spectra were analyzed and evaluated by visual comparison and in a localization model. The optimal average edge length for a fast calculation of HRTFs yielding potentially good localization performance is discussed.

4:20

1pPPa10. Quantitative assessment of spatial sound distortion by the semi-ideal recording point of a hear-through device. Pablo Hoffmann, Flemming Christensen, and Dorte Hammershøi (Acoustics, Aalborg Univ., Fredrik Bajers Vej 7B5, Aalborg 9220, Denmark, pfh@es.aau.dk)

A hear-through device combines a microphone and earphone in an earpiece so that when worn, one per ear, it can work as an acoustically transparent system allowing for simultaneous individual binaural recording and playback of the real sound field at the ears. Recognizing the blocked entrance to the ear canal as the ideal recording point—i.e., all directional properties of the incident sound field are recorded without distortion—it is critical for such device to be sufficiently small so that it can be completely inserted into the ear canal. This is not always feasible and the device may stretch out from the ideal position and thus distort the captured spatial information. Here we present measurements that quantify by how much the directional properties of the sound field are distorted by semi-ideal hear-through prototypes built by mounting miniature microphones on the outer part of selected commercial insert earphones. This includes an analysis of the magnitude by which spatial information is distorted and the extent to which these distortions are direction dependent. Potential strategies for compensating these distortions are also considered.

4:40

1pPPa11. Measuring pressure and particle velocity along the human ear canal. Marko Hiipakka and Ville Pulkki (Aalto Univ., Otakaari 5 A, Espoo 02150, Finland, Marko.Hiipakka@aalto.fi)

A non-invasive method of measuring or estimating accurately the head-related transfer functions (HRTFs) and headphone transfer functions (HpTFs), i.e., the pressure at the eardrum rather than at the blocked ear canal entrance is called for. In this work, a miniature-sized acoustic pressure–velocity sensor is used to measure both pressure and velocity along the ear canals of human test subjects. The measurements are used to study the applicability of a recently proposed method of estimating the pressure at the eardrum from pressure–velocity measurements made at the ear canal entrance. The measurement results are compared to results from computational modeling with human ear canal parameters. In addition, the effect of the PU-sensor itself on the pressure at the eardrum is studied. It is shown that the estimation method is reliable and accurate for most human subjects. The diameter and the shape of the ear canal affect the results in such a way that the best results are obtained with wide and straight ear canals whereas less accurate results are obtained with narrow and curved ear canals. It is concluded that the estimation method facilitates the obtaining of individual HRTFs and HpTFs at the eardrum using non-invasive measurements.

MONDAY AFTERNOON, 3 JUNE 2013 — 516, 1:00 P.M. TO 5:00 P.M.

Session 1pPPb

Psychological and Physiological Acoustics: Psychoacoustics and Perception (Poster Session)

Elin Roverud, Chair
Purdue Univ., West Lafayette, IN 47906

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

1pPPb1. Effects of compression on the use of onset time differences to detect one tone in the presence of another. Sara M. Madsen and Brian Moore (Psychology, Cambridge Univ., Downing St., Desborough, Cambridge CB2 3EB, United Kingdom, smkm2@cam.ac.uk)

It is easier to hear one of two notionally "simultaneous" tone complexes if the onset of the masker complex is delayed relative to that of the signal. However, the ability to use onset asynchrony as a cue may be reduced when using amplitude compression, due to distortion of the onset of sounds (overshoot effects). We assessed how fast- and slow-acting five-channel compression affects the ability to use onset asynchrony to detect one (signal) complex tone when another (masking) complex tone is played almost simultaneously. A 2:1 compression ratio was used with normal-hearing subjects and individual compression ratios and gains recommended by the CAM2 hearing aid fitting method were used for hearing-impaired subjects. For the normal-hearing subjects, performance improved with increasing onset asynchrony in all conditions. The improvement was greatest with fast compression and least with no compression. Preliminary results for the hearing-impaired subjects indicate smaller but similar effects of onset asynchrony and a greater benefit of compression. The benefit of compression probably occurs because compression increases the level of the part of the signal that occurs before the masker relative to the masker. [Work supported by Starkey (U.S.A.) and the MRC (UK).]

1pPPb2. Study on effects of presence of cue-tone on psychophysical tuning curves. Shunsuke Kidani, Ryota Miyauchi, and Masashi Unoki (School of Information Sci., JAIST, 1-1, Asahidai, Nomi, Ishikawa 923-1292, Japan, kidani@jaist.ac.jp)

Our previous study indicated that tunings of the auditory filter were sharpened by the presence of a cue-tone [Kidani *et al*. (2012), ISH2012]. It is unclear, however, whether the variation of the auditory filter due to the cue-tone is caused by excitation or suppression, because tip of the filter is normalized at 0 dB. Psychophysical tuning curves (PTCs) can show that the detection threshold is decreased at the probe frequency or increased around the probe by the presence of cue-tone, indicating excitation and suppression respectively. PTCs, because, are measured as masked threshold of probe by narrow-band noise. This study aims to consider the effect of the presence of cue-tone by measuring of PTCs. In present study, PTCs were measured in

simultaneous masking in the absence and presence of cue-tone for four probe frequencies. The probes were presented at 10 dB above each absolute threshold. The frequency and level of the cue-tone were same as the probe. The result revealed that filter-Q, regarded as the sharpness of tuning, was increased by the presence of cue-tone when the probe frequencies were 1.0 and 2.0 kHz, while the filter-Q was not changed when the probe frequencies were 0.5 and 4.0 kHz.

1pPPb3. Optimizing the simultaneous estimation of frequency selectivity and compression using notched-noise maskers with asymmetric levels. Tomofumi Fukawatase, Toshio Irino, Ryuichi Nisimura, Hideki Kawahara (Faculty of Systems Eng., Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan, tomofuka0522@gmail.com), and Roy D. Patterson (Dept. of Physiol., Development and Neurosci., Cambridge Univ., Cambridge, United Kingdom)

It is important for the development of hearing aids and other audio devices to make accurate estimates of the frequency selectivity and compression of the auditory filter. Previously, we reported a technique for estimating the compression of the auditory filter that combined data from a simultaneous notched-noise experiment and a temporal masking curve (TMC) experiment. Unfortunately, the TMC data derived for individual listeners in forward masking is not stable; the cue to the presence of the signal is not entirely clear in forward masking. In this paper, we report attempts to make the traditional simultaneous notched-noise technique more sensitive to the effects of cochlear compression by varying the relative levels of the noise bands. Asymmetric-level maskers (ALMs) make it possible to estimate the filter shape and compression of the auditory filter simultaneously and reliably; the slope of the input–output function is substantially lower than with symmetric-level maskers. We also describe a procedure for incorporating a sensitivity analysis into the filter-fitting process to determine the minimum number of notched-noise conditions required to produce reliable estimates of selectivity and compression, in hopes of being able to employ the technique with hearing impaired listeners.

1pPPb4. Reliability of procedures used for scaling loudness. Walt Jesteadt and Suyash N. Joshi (Psychoacoust. Lab., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, Walt.Jesteadt@boystown.org)

In this study, 16 normally hearing listeners judged the loudness of 1000-Hz sinusoids using magnitude estimation (ME), magnitude production (MP), and categorical loudness scaling (CLS). Listeners in each of four groups completed the loudness scaling tasks in a different sequence on the first visit (ME, MP, CLS; MP, ME, CLS; CLS, ME, MP; CLS, MP, ME), and the order was reversed on the second visit. This design made it possible to compare the reliability of estimates of the slope of the loudness function across procedures in the same listeners. The ME data were well fitted by an inflected exponential (INEX) function, but a modified power law was used to obtain slope estimates for both ME and MP. ME and CLS were more reliable than MP. CLS results were consistent across groups, but ME and MP results differed across groups in a way that suggested influence of experience with CLS. Although CLS results were the most reproducible, they do not provide direct information about the slope of the loudness function because the numbers assigned to CLS categories are arbitrary. This problem can be corrected by using data from the other procedures to assign numbers that are proportional to loudness. [Work supported by NIH.]

1pPPb5. Sequential dependencies in magnitude scaling of loudness. Suyash N. Joshi and Walt Jesteadt (Psychoacoust. Lab., Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, Suyash.Joshi@boystown.org)

Ten normally hearing listeners used a programmable sone-potentiometer knob to adjust the level of a 1000-Hz sinusoid to match the loudness of numbers presented to them in a magnitude production task. Three different power-law exponents (0.15, 0.30, and 0.60) and a log-law with equal steps in dB were used to program the sone-potentiometer. The knob settings systematically influenced the form of the loudness function. Time series analysis was used to assess the sequential dependencies in the data, which increased with increasing exponent and were greatest for the log-law. It would be possible, therefore, to choose knob properties that minimized these dependencies. When the sequential dependencies were removed from the data, the slope of the loudness functions did not change, but the variability decreased. Sequential dependencies were only present when the level of the tone on the previous trial was higher than on the current trial. According to the attention band hypothesis [Green and Luce, Perception Psychophys., 1974] these dependencies arise from a process similar to selective attention, but observations of rapid adaptation of neurons in the inferior colliculus based on stimulus level statistics [Dean *et al*., Nature Neurosci. (2005)] would also account for the data. [Work supported by NIH.]

1pPPb6. Effect of musical training on static and dynamic measures of spectral-pattern discrimination. Stanley Sheft (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 S. Paulina St., 1012 AAC, Chicago, IL 60612, ssheft@gmail.com), Kirsten Smayda (Univ. of Texas at Austin, Austin, TX), Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), W. Todd Maddox, Bharath Chandrasekaran, (Univ. of Texas at Austin, Austin, TX)

Both behavioral and physiological studies have demonstrated enhanced processing of speech in challenging listening environments attributable to musical training. The relationship, however, of this benefit to auditory abilities as assessed by psychoacoustic measures remains unclear. Using tasks previously shown to relate to speech-in-noise perception, the present study evaluated discrimination ability for static and dynamic spectral patterns by 49 listeners grouped as either musicians or nonmusicians. The two static conditions measured the ability to detect a change in the phase of a logarithmic sinusoidal spectral ripple of wideband noise with ripple densities of 1.5 and 3.0 cycles per octave chosen to emphasize either timbre or pitch distinctions, respectively. The dynamic conditions assessed temporal-pattern discrimination of 1-kHz pure tones frequency modulated by different 5-Hz lowpass noise samples with thresholds estimated in terms of either stimulus duration or signal-to-noise ratio. Musicians performed significantly better than nonmusicians on all four tasks. Discriminant analysis showed that group membership was correctly predicted for 84% of the listeners with the structure coefficient of each measure greater than 0.46. Results suggest that enhanced processing of static and dynamic spectral patterns defined by low-rate modulation may contribute to the relationship between musical training and speech-in-noise perception. [Work supported by NIH.]

1pPPb7. Spectral uncertainty and similarity effects in informational masking identically related to the Simpson-Fitter metric of target-masker separation. An-Chieh Chang, Jacob Stamas (Commun. Sci. and Disord., Dept. Commun. Sci. and Disord., Univ. of Wisconsin, Madison, WI 53705, achang5@wisc.edu), Inseok Heo (Elec. and Comput. Eng., Univ. of Wisconsin - Madison, Madison, WI), Lynn Gilbertson, and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Madison, WI)

Evidence is provided suggesting a primary dependence of informational masking (IM) on the stochastic separation of target and masker given by Simpson-Fitter's da [Lutfi *et al*. J. Acoust. Soc. Am. **132**, EL109-113 (2012)]. The stimuli were synthesized impact sounds of plates played in sequence as masker-target-masker triads. Their spectra varied independently and at random on each presentation as would correspond to changes in plate size. In the 2IFC procedure the listener's task was to choose the larger-sized target. The effect of spectral uncertainty regarding the masker was examined by measuring d' performance for different values of the variance in masker size. The effect of spectral similarity of target and masker was examined by measuring performance for different values of the mean difference between target and masker size. The functions relating d' to da in both cases were identical and of similar slope across listeners. Identical functions were also obtained, though with shallower slopes, when listeners judged the target hit with greater impact force. The results are considered in terms of their implications for the development of a model of IM that emphasizes the statistical properties of signals over loosely defined concepts of target-masker similarity and masker uncertainty.

1pPPb8. Spatial uncertainty and proximity effects in informational masking identically related to the Simpson–Fitter metric of target-masker separation. Jacob Stamas (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Univ. of Wisconsin, Madison, WI 53705, jstamas@wisc.edu), Inseok Heo (Elec. and Comput. Eng., Univ. of Wisconsin - Madison, Madison, WI), Lynn Gilbertson, An-Chieh Chang, and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Madison, WI)

Further evidence is provided suggesting a primary dependence of informational masking (IM) on the stochastic separation of target and masker given by Simpson-Fitter's da [Lutfi *et al*. J. Acoust. Soc. Am. **132**, EL109-113 (2012)]. The stimuli were brief bursts of Gaussian noise or words played

in sequence as masker-target-masker triads. The apparent position of bursts (words), from left to right, was varied independently and at random on each presentation using KEMAR HTRFs. In the 2IFC procedure, the listener's task was to choose the target positioned further to the right. The effect on performance of spatial uncertainty regarding the masker was examined by manipulating the position variance of the masker. The effect on performance of spatial proximity of target to masker was examined by manipulating the position mean difference between target and masker. In both cases, the data were well described by a single linear function relating d' performance to da; intercepts differed across listeners, but slopes were similar. Comparable results presented at this meeting for the effects of spectral uncertainty and similarity of target and masker suggest that the statistical properties of signals may be a more significant determinant of IM than their specific acoustic properties.

1pPPb9. Toward a model of informational masking: The Simpson–Fitter metric of target-masker separation. Lynn Gilbertson, An-Chieh Chang, Jacob Stamas (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Madison, WI), Inseok Heo (Elec. and Comput. Eng., Univ. of Wisconsin - Madison, Madison, WI), and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, 1410 E. Skyline Dr., Madison, WI 53705, ralutfi@wisc.edu)

Informational masking (IM) is the term used to describe masking that appears to have its origin at some central level of the auditory nervous system beyond the cochlea. Supporting a central origin are the two major factors associated with IM: trial-by-trial uncertainty regarding the masker and perceived similarity of target and masker. Here preliminary evidence is provided suggesting these factors exert their influence through a single critical determinant of IM, the stochastic separation of target and masker given by Simpson–Fitter's da [Lutfi *et al*., J. Acoust. Soc. Am. **132**, EL109-113 (2012)]. Target and maskers were alternating sequences of tones or words with frequencies, F0s for words, selected at random on each presentation. The listener's task was to discriminate a frequency-difference in the target tones or identify the target words. Performance in both tasks was found to be constant across conditions in which the mean difference (similarity), variance (uncertainty), or covariance (similarity) of target and masker frequencies were selected to yield the same value of da. The results are discussed in terms of their implications for the development of a model of IM that emphasizes the statistical properties of signals over loosely defined concepts of masker uncertainty and target-masker similarity.

1pPPb10. Extending Schroeder-phase masking: Influence of direction and shape of masker instantaneous frequency. Evelyn M. Hoglund, Yonghee Oh, Joseph F. Hribar, Kelsi J. Wittum, Megan L. Strang, and Lawrence L. Feth (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd, Columbus, OH 43204, hoglund.1@osu.edu)

Schroeder (1970) devised an algorithm to produce low peak factor signals. Schroeder signals with equal amplitude spectra, but reversed phase spectra, reveal large differences in masker effectiveness for listeners with normal hearing [Smith *et al*. (1986)]. Results reported here extend previous work to include detection of multiple bursts of the same frequency, and multiple bursts that increase or decrease in frequency. Signal frequencies were selected to correspond with harmonics in the maskers. Results indicate that changing the frequency of the signal amplifies the difference between the Schroeder-phase maskers, but the direction of the change does not. When a frequency modulated tone is substituted for the Schroeder maskers, masked threshold depends on the shape of the instantaneous frequency (IF) function, as well as the direction of change. For linear FM (with IF similar to the Schroeder maskers), masked thresholds are comparable to the Schroeder-phase maskers. However, for logarithmic FM, IF changes at a constant ERB/s and directional differences are much smaller. A modified channel model [Oh (2012)] shows substantial differences in the basilar membrane response to these different maskers. [Research supported by a grant from the Office of Naval Research #N000140911017.]

1pPPb11. Multicomponent signal detection: Tones in noise and amplitude modulation detection. Eric R. Thompson (Ball Aerosp. & Technol. Corp., 2610 7th St., Bldg. 441, Wright Patterson, OH 45433, eric.thompson.ctr@wpafb.af.mil), Brian D. Simpson, and Nandini Iyer (Air Force Res. Lab., Wright Patterson, OH)

In order to predict the detectability of broadband acoustic signals, a model must include a means of integrating information across frequencies. There have been several previous studies measuring the detectability of multicomponent signals, but it is still not clear what the best model is when signal components are not equally detectable. Some researchers have proposed that thresholds are driven by the most detectable component (max-d' model), while others have found that the best model for their data is a statistical summation model, where component sensitivities are combined using a Pythagorean sum. In the present study, detection thresholds were collected in broadband noise for single tones at three frequencies and three signal-to-noise ratios (SNRs) and for all combinations of SNRs for the three tones presented together. Also, amplitude modulation (AM) detection thresholds were measured for a 16-Hz AM signal imposed on 300-Hz-wide noise bands centered at three frequencies at three modulation depths for each band individually, for combinations of two bands and for all three bands presented together. While both models (max-d' and Pythagorean sum) can predict the general trend of the multicomponent data from the single component data, neither model fits the data very well.

1pPPb12. Investigating the effects of intensity on the bandwidth of peripheral filtering in an amplitude-modulation notch detection task. Matthew L. Richardson, Allison I. Shim, and Bruce G. Berg (Cognit. Sci., UC Irvine, 159 St. Vincent, Irvine, CA 92618, mlrichar@uci.edu)

The effect of intensity on the effective bandwidth of auditory temporal processing is investigated. Thresholds for detecting sinusoidal amplitude-modulation of a 200-Hz wide band of noise centered at 1000 Hz are measured in the presence of a notched noise masker. The masker consists of two, 200-Hz wide, unmodulated bands of noise placed at frequencies above and below the modulated band. Thresholds for a modulation rate of 10 Hz are estimated for different notch bandwidths ranging from 100 Hz to 2740 Hz. The use of a slow modulation frequency aims to avoid possible central limitations of temporal processing at higher modulation frequencies. Intensity is varied across two conditions, with all three bands of noise presented at either 40 dB SPL or 85 dB SPL. Threshold functions for the two intensity levels are essentially identical. The maximum notch width at which an effect of the masker is observed is approximately 500 Hz. The results are consistent with a hypothesis that the filtering characteristics of temporal processing (e.g., envelope model) and spectral processing (e.g., power spectrum model) are different.

1pPPb13. Thresholds of tone pitch contour discrimination for English listeners. Rachael C. Gilbert (Linguistics, The Univ. of Texas at Austin, 4812 Ave. H, Apt B, Austin, TX 78751, rachaelgilbert@gmail.com) and Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

This study aims to provide psychophysical data on English language listeners' ability to discriminate tone pitch contours. Just noticeable differences (JND) of F0 contour changes were measured in six native listeners of American English. Previous work in our lab found English listener thresholds for offsets of falling tones to be significantly lower than those for onsets of rising tones. To what extent this difference is due to the position of the F0 shift versus the direction of the F0 contour is unclear. In this study, we control for four experimental factors: stimulus type (speech, nonspeech), position of F0 shift (onset, offset), direction of shift (upward, downward), and F0 contour direction (falling, rising). Preliminary results reveal that English listeners had significantly lower psychophysical thresholds for F0 shifts at the offset than at the onset. No significant difference was found for F0 shift direction, F0 contour direction, or stimulus type. The current data suggest that the F0 shift position was the primary determinant in our previous study and replicate other findings showing that English listeners perceive tones on a psychophysical base. Future work will examine these results in relation to those of native tone language listeners.

1pPPb14. An automated procedure for detecting human frequency-following responses to voice pitch. Fuh-Cherng Jeng and Jiong Hu (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W224, Athens, OH 45701, jeng@ohio.edu)

The frequency-following response (FFR) to voice pitch has been widely examined in research laboratories and has demonstrated its potential to be transformed into a useful tool for patients with hearing, speech, and language disorders in the clinic. During the past decade, many aspects of the FFR have been reported. The presence of such a response, however, still

relies on subjective interpretation of the observer. Aside from a recent study reporting two algorithms for detecting such a response, there has been limited number of studies reporting the development of an automated procedure for FFR. The purpose of this study is (1) to develop an automated procedure that utilizes the statistical properties of the temporal and spectral energy distributions in the recorded waveforms and (2) to explore the effectiveness, accuracy, and efficiency of the automated procedure and compare them with those obtained from conventional algorithms and human judgments.

1pPPb15. Infants' ability to perceive the pitch of unresolved harmonics. Bonnie K. Lau and Lynne A. Werner (Univ. of Washington, 523 Broadway East Unit 217, Seattle, WA 98102, bonniekwlau@gmail.com)

An important phenomenon for models of pitch perception is that adult listeners' can extract pitch from complexes containing only unresolved harmonics. Although 3-month-olds discriminate resolved harmonics on the basis of missing fundamental (MF) pitch, their ability to discriminate unresolved harmonics is unknown. This study investigated the ability of adults, 7- and 3-month-olds, to perceive the pitch of unresolved harmonics using an observer-based method. Stimuli were MF complexes that were bandpass filtered with a -12 dB/octave slope, combined in random phase, and presented at 70 dB SPL for 650 ms with a 50 ms rise/fall and with a pink noise to mask distortion products. The experiment consisted of two conditions: (1) "low" unresolved harmonics between 2500 and 4500 Hz based on MFs of 160 Hz (H17–H26) and 200 Hz (H13–H22) and (2) "high" unresolved harmonics between 4000 and 6000 Hz based on MFs of 190 Hz (H22–H31) and 200 Hz (H20–H29). To demonstrate MF pitch discrimination, participants were required to ignore spectral changes in complexes with the same fundamental and to respond only when the fundamental changed. Interestingly, variable performance in the "high" condition was observed with adult participants. However, nearly all infants tested categorized complexes by MF pitch in both conditions, suggesting discrimination of unresolved harmonics at 3 months.

1pPPb16. Mistuning detection in a complex of unresolved harmonics: Effects of age. Sara K. Mamo (Div. of Speech and Hearing Sci., Univ. of North Carolina - Chapel Hill, 075 MacNider Hall, CB #7070, Chapel Hill, NC 27599-7070, smamo@med.unc.edu) and John H. Grose (Dept. of Otolaryngol.-Head/Neck Surgery, Univ.of North Carolina - Chapel Hill, Chapel Hill, NC)

Older adults experience speech perception difficulties that are not explained by audiometric thresholds. One hypothesis is that temporal processing deficits contribute to these speech-in-noise difficulties. To test this, threshold for mistuning was measured for a component within a complex of unresolved harmonics—a cue that likely depends on sensitivity to envelope perturbations. The complex comprised harmonics 12–16 of a 100-Hz or 200-Hz fundamental, and the duration was either 170-, 340-, or 680-ms. Presentation level was 70 dB SPL. The starting phases of all components were randomized on each presentation. The 3AFC procedure adaptively varied the mistuning of harmonic 14 to obtain the frequency-shift threshold. Younger and older listeners with audiometrically normal hearing participated. The expectation was that older adults would require greater mistuning to detect changes in envelope periodicity and that this would be more evident at shorter durations. Preliminary results support an overall effect of age as well as effects of fundamental frequency and duration. Results will be considered in the context of parallel speech-evoked ABR studies being undertaken in this population that point to age-related deficits in envelope processing, possibly driven by poor encoding of unresolved harmonics. [Work supported by NIDCD 1-F32-DC012217-01A1 & 5-R01-DC001507.]

1pPPb17. Retention of gap length in normal-hearing listeners. Meghan M. Smith, Dennis Ries, and Audra Woods (Commun. Sci. and Disord., Grover Ctr. W241, 1 Ohio Univ., Athens, OH 45701, ms802311@ohio.edu)

Listeners' ability to retain information about gap length within a noise burst was studied. JND for gap length between the target and comparison stimuli was obtained using the single-interval adjustment-matrix procedure for retention intervals that were silent, included four noise bursts, or included four noise bursts with discrete gaps. JNDs for retention intervals containing either type of noise burst did not differ significantly from those obtained for the silent retention interval. This result differs from that found for retention of pitch and loudness. This might occur as more cortical resources are used for retention of auditory temporal information.

1pPPb18. The influence of feature detection on working memory in complex auditory fields. AnneMarie Chiodi, Aurora Weaver, and Dennis Ries (Commun. Sci. and Disord., Grover Ctr. W241, 1 Ohio Univ., Athens, OH 45701, ac175509@ohio.edu)

Research reveals that feature detectors may enhance listener performance in auditory discrimination and detection tasks involving frequency modulation (FM) [Cusack and Carlyon, J. Exp. Psychol. **29**, 713–725 (2003)]. The influence of these detectors on retention of auditory information is unknown. This study investigated the impact of FM on listener performance in auditory, delayed-comparison tasks for conditions that differed in the number of background stimuli within two perceptual windows separated by various retention intervals. The background stimuli within both windows were either all modulated sinusoids or pure-tones for a given trial. An additional stimulus was presented in each window that could differ in its modulation state (FM or unmodulated sinusoid) across the two windows. The temporal placement and frequencies of all stimuli within the first window were assigned randomly for a given trial and the second window followed these parameters. Listeners were to determine whether the two windows were the same or different. Preliminary results show that same-different judgments of target modulation state was easier in a field of four unmodulated sinusoids than vice versa. This result occurred regardless of retention interval length. The further influence of field complexity, retention interval length, and working memory span will be discussed.

1pPPb19. Boundary effects on the illusory continuity of and interrupted glide through a notched noise. Valter Ciocca (School of Audiol. and Speech Sci., UBC, 2177 Wesbrook Mall, Vancouver, BC V6T 1Z3, Canada, vciocca@audiospeech.ubc.ca) and Nicholas Haywood (MRC Inst. of Hearing Res., Nottingham, United Kingdom)

This study investigated the illusory continuity of an interrupted frequency glide through a notched-noise burst. A 2I-2AFC procedure was used to measure detection of the (target) portion of the frequency glide that overlapped in time with the noise. The portions of the glide preceding and following the noise (flankers) could be present or absent. The center frequency of the notch coincided with either the frequency end-point of the flanker that preceded the noise, or the onset frequency of the flanker that followed the noise. A control condition with a wide-band noise burst (absent notch) was also included. Performance was poorest in the absent notch condition and was significantly poorer with present than with absent flankers. This suggests that listeners perceptually restored the missing target when flankers were present. Performance was also less accurate (indicating stronger illusory continuity) when the notch was centered on the end-point of the flanker that preceded the noise. These results suggest that the masking of the onset of the flanker following the noise provides a stronger cue to the perception of continuity than the masking of the offset of the flanker that precedes the noise.

1pPPb20. Sensory consonance of two simultaneous sine-tones. Reinhart Frosch (ETH and PSI (retired), Sommerhaldenstrasse 5B, Brugg 5200, Switzerland, reinifrosch@bluewin.ch)

In Chapter 4 of my book "Musical Consonance and Cochlear Mechanics" (vdf, Zurich, 2012), four psychoacoustic experiments on the sensory consonance of two simultaneous sine-tones are described. In each of those experiments, the deeper-tone frequency f_d was kept fixed, at $f_d = 132$, 264, 528, or 1056 Hz. Each experiment was done twice, at sound-pressure levels of 50 and 70 dB (SPL). The resulting consonance curves (sensory consonance versus higher-tone frequency f_h) exhibit consonance minima at beat-rates $b_{m.d.} = f_h - f_d$ (where "m.d." = "most dissonant") ranging from 13 Hz (at $f_d = 132$ Hz) to 39 Hz (at $f_d = 1056$ Hz). In Section 15.1 of the above-mentioned book, these most dissonant beat rates are shown to agree well with the following empirical law: $b_{m.d.} = (1.07 s^{-0.5}) * \mathrm{sqrt}(f_{avg})$, where $f_{avg} = f_d + b_{m.d.}/2$. The present study was prompted by the comments of a reader; the just described empirical law is unsatisfactory because in the underlying experiments the deeper-tone frequency f_d [rather than the average frequency $(f_d + f_h)/2$] was kept constant. It was found that the data agree

equally well with the following modified empirical law: $b_{m.d.} = (1.09s^{-0.5}) * sqrt(f_d)$. This modification does not affect the validity of the complex-tone consonance theories described in Chapters 15 and 16 of the mentioned book.

1pPPb21. Dependency of tonality perception on frequency, bandwidth, and duration. Armin Taghipour, Bernd Edler, Masoumeh Amirpour, and Jürgen Herre (Int. Audio Lab. Erlangen, Am Wolfsmantel 33, Erlangen 91058, Germany, armin.taghipour@audiolabs-erlangen.de)

Psychoacoustic studies show that a narrowband noise masker exhibits a stronger simultaneous masking effect than a tonal masker with the same signal power placed at the noise center frequency. Consequently, perceptual audio codecs commonly incorporate some sort of tonality estimation as part of their perceptual model. However, common tonality estimation techniques do not necessarily reflect the perception of tonality by human listeners. As long as the tone and narrowband noise signals are long enough, they are easily distinguishable for normal hearing listeners. However, if the stimulus duration decreases, both signal types approach the shape of impulses, and therefore, at some point become audibly identical. Consequently, at a given frequency and noise bandwidth, there is a duration threshold below which the signals cannot be distinguished. A series of so-called "2-AFC 3-step up-down" psychoacoustic tests are designed and carried out to investigate the frequency and bandwidth dependency of these duration thresholds. The test results, collected from 32 listeners, are statistically evaluated and confirm a decreasing threshold for increasing center frequency and bandwidth. These results can be used to improve psychoacoustic models for audio codecs by using tonality estimators with frequency and bandwidth adapted temporal resolution.

1pPPb22. Reflection orders and auditory distance. Catarina Mendonça (Dept. of Signal Process. and Acoust., Aatlo Univ., Otakaari 5, Espoo FI-02150, Finland, mendonca.catarina@gmail.com), João Lamas (Centro Algoritmi, Univ. of Minho, Guimaraes, Portugal), Tom Barker (Dept. of Signal Process., Tampere Univ. of Technol., Tampere, Finland), Guilherme Campos, Paulo Dias (Departamento de Electrónica, Telecomunicações e Informática, Univ. of Aveiro, Aveiro, Portugal), Ville Pulkki (Dept. of Signal Process. and Acoust., Aatlo Univ., Espoo, Finland), C. Silva, and Jorge Santos (Centro Algoritmi, Univ. of Minho, Guimarães, Portugal)

The perception of sound distance has been sparsely studied so far. It is assumed to depend on familiar loudness, reverberation, sound spectrum, and parallax, but most of these factors have never been carefully addressed. Reverberation has been mostly analyzed in terms of ratio between direct and indirect sound, and total duration. Here we were interested in assessing the impact of each reflection order on distance localization. We compared sound source discrimination at an intermediate and at a distant location with direct sound only, one, two, three, and four reflection orders in a 2AFC task. At the intermediate distances, normalized psychophysical curves reveal no differentiation between direct sound and up to three reflection orders, but sounds with four reflection orders have significantly lower thresholds. For the distant sources, sounds with four reflection orders yielded the best discrimination slopes, but there was also a clear benefit for sounds with three reflection orders. We discuss the results in terms of direct-to-reflected ratio, reflection directionality, and spectral information.

1pPPb23. Ventriloquism effect and aftereffect in the distance dimension. L'uboš Hládek, Christophe C. Le Dantec, Norbert Kopčo (Inst. of Comput. Sci., P. J. Safarik Univ., Jesenná 5, Košice 04001, Slovakia, lubos.hladek@student.upjs.sk), and Aaron R. Seitz (Dept. of Psychol., Univ. of California, Riverside, CA)

When an auditory target is presented simultaneously with a spatially displaced visual target, the perceived auditory target location shifts toward the visual target. This effect, known as the *ventriloquism effect* or *visual capture*, has been extensively studied in the horizontal dimension, but not in distance. Here, we measured distance localization performance in a reverberant room. Stimuli were either audio-visual (AV) 300-ms broadband noise bursts presented synchronously with spatially congruent or incongurent visual stimuli/LEDs, or auditory-only (A-only) noise bursts. One of eight speakers (distance 70 cm to 203 cm directly ahead of the listener) presented a stimulus on each trial. During adaptation runs, the AV stimuli were presented with the V-component closer or further by 30% than the A-component (displacement direction fixed within session). The ventriloquism effect was observed for both V-closer and V-further AV stimuli, with slightly stronger shifts induced by the V-closer stimuli. Ventriloquism aftereffect, assessed by presenting A-only trials interleaved with the adaptation-AV trials, was also observed, but was weaker than the ventriloquism effect. The results suggest that visual targets do capture auditory targets in the distance dimension, but visual modulation might be asymmetrical with respect to distance. [Work supported by EU FP7-247543, VEGA-1/0492/12, NSF (BCS-1057625).]

1pPPb24. An assessment of virtual auditory distance judgments among blind and sighted listeners. Andrew J. Kolarik (Dept. of Psychol., Cambridge Univ., Downing St., Cambridge, CB2 3EB, United Kingdom, ak771@cam.ac.uk), Silvia Cirstea, Shahina Pardhan (Vision and Eye Res. Unit, Anglia Ruskin Univ., Cambridge, United Kingdom), and Brian Moore (Dept. of Psychol., Cambridge Univ., Cambridge, United Kingdom)

Auditory distance perception is a crucial component of blind listeners' spatial awareness. Many studies have reported supra-normal spatial auditory abilities among blind individuals, such as enhanced azimuthal localization [Voss *et al*. (2004)] and distance discrimination [Kolarik *et al*. (in press)]. However, it is not known whether blind listeners are better able to use acoustic information to enhance judgments of distance to single sound sources, or whether lack of visual spatial cues prevents calibration of auditory distance information, leading to worse performance than for sighted listeners. Blind and sighted listeners were presented with single, stationary virtual sound sources between 1.22 and 13.79 m away in a virtual anechoic environment simulated using an image-source model. Stimuli were spoken sentences. Sighted listeners systematically underestimated distance to remote virtual sources, while blind listeners overestimated the distance to nearby virtual sources and underestimated it for remote virtual sources. The findings suggest that blind listeners are less accurate at judging absolute distance, and experience a compression of the auditory world, relative to sighted listeners. The results support a perceptual deficiency hypothesis for absolute distance judgments, suggesting that compensatory processes for audition do not develop among blind listeners when estimating the distance to single, stationary sound sources.

Session 1pSA

Structural Acoustics and Vibration: Measurement and Modeling of Structures with Attached Noise Control Materials II

Franck C. Sgard, Cochair
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Noureddine Atalla, Cochair
GAUS Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC J1K 2R1, Canada

Invited Papers

1:00

1pSA1. Visco-thermal dissipations in heterogeneous porous media. Fabien Chevillotte, Luc Jaouen, and François-Xavier Bécot (Matelys, 1 rue Baumer, Vaulx-En-Velin 69120, France, fabien.chevillotte@matelys.com)

Semi-phenomenological models have been widely used since the 1990's for modeling visco-thermal dissipations of acoustical energy through porous media. These dissipations are taken into account by two complex frequency-dependent functions (the dynamic density ρ_{eq} (ω) and the dynamic bulk modulus K_{eq} (ω)), which are analytically derived from macroscopic parameters. Other models were derived for modeling perforated plates [J. Sound Vib. **303** (2007)], double porosity media [J. Acoust. Soc. Am. **114**(1) (2003)] or, more recently, porous composites made of porous inclusions in a substrate porous media [Acta Acoust. **96** (2010)]. So far, this latter model is not able to consider the shape of the host and the client media. This model can neither be extended to limiting cases of perforated plate model nor double porosity model. Based on a modified equivalent fluid model, this work proposes a unified model which accounts, analytically, for the shape of the inclusions, might they be porous or not. This model enables to describe the acoustic behavior of any kind of composite media from perforated plates to arbitrarily shaped porous composites including configurations of porous inclusions in solid matrix or double porosity media. In addition, possible pressure interactions between the substrate material and the inclusions are accounted for.

1:20

1pSA2. Prediction of acoustic properties of parallel assemblies by means of transfer matrix method. Kévin Verdière, Raymond Panneton, Saïd Elkoun (GAUS, Université de Sherbrooke, 2500 bd de l'université, Sherbrooke, QC J1K2R1, Canada, kevin.verdiere@usherbrooke.ca), Thomas Dupont, and Philippe Leclaire (DRIVE, ISAT, Université de Bourgogne, Nevers, France)

The Transfer Matrix Method (TMM) is used conventionally to predict the acoustic properties of laterally infinite homogeneous layers assembled in series to form a multilayer. In this work, a parallel assembly process of transfer matrices is used to model heterogeneous materials such as patchworks, acoustic mosaics, or a collection of acoustic elements in parallel. In this method, it is assumed that each parallel element can be modeled by a 2x2 transfer matrix, and no diffusion exists between elements. The method is validated by comparison with finite element (FE) simulations and acoustical tube measurements on different configurations at normal and oblique incidence. Then, an overview of the possibilities, such as the combination of series and parallel matrices, the sound absorption coefficient, and the transmission loss of a parallel array of resonators or three-dimensional geometries is presented and discussed.

1:40

1pSA3. Investigations on the sensitivity of the relationships between sound absorption characteristics and microstructure related parameters for polyurethane foams. Morvan Ouisse (Appl. Mech., FEMTO-ST Univ. of Franche-Comté, 24 rue de l'épitaphe, Besançon 25000, France, morvan.ouisse@univ-fcomte.fr), Olivier Doutres, Noureddine Atalla (GAUS, Dept. of Mech. Eng., Université de Sherbrooke (Qc), Sherbrooke, QC, Canada), and Mohamed Ichchou (LTDS, Ecole Centrale de Lyon, Ecully, France)

Straightforward semi-phenomenological models have been developed for highly porous polyurethane foams to predict the macroscopic non-acoustic parameters involved in the classical Johnson-Champoux-Allard model (i.e., porosity, airflow resistivity…) from microstructure properties (i.e., strut length, strut thickness, and reticulation rate). These microstructure properties are measured using sophisticated optical methods (i.e., optical microscope, SEM) and a large variability can be observed due to great complexity of the 3D microstructure; variability also depends on the precision of the measurement device. This work investigates how the variability associated with the model inputs affects the model outputs (i.e., non-acoustic parameters, surface impedance, and sound absorption coefficient). The sensitivity analysis is based on the Fourier Amplitude Sensitivity Test (FAST). It helps quantify the correlation between the input parameters and identify the parameters contributing the most to output variability, thus requiring precise measurement. This study illustrates the preponderant impact of the reticulation rate (i.e., open pore content) on acoustic performances and guides the user on the required optical measurement device.

2:00

1pSA4. Multi-scale acoustics of partially open cell poroelastic foams. Minh Tan Hoang (Faurecia Interior System, Marne-la-Vallée, France), Guy Bonnet, and Camille Perrot (Laboratoire Modélisation et Simulation Multi Echelle, MSME UMR 8208 CNRS, Université Paris-Est, 5, Boulevard Descartes, Bâtiment Lavoisier, Bureau D13, Champs-sur-Marne 77454, France, camille.perrot@univ-paris-est.fr)

The present paper reports on the modeling of linear elastic properties of acoustically insulating foams with unit cells containing solid films or membranes at the junction between interconnected pores from a numerical homogenization technique. It combines fluid-flow induced microstructure identification with simulations of the effective Young's modulus and Poisson ratio from a mixture of routinely available laboratory measurements (porosity, permeability, cell size) and finite element calculations when the boundary conditions of the periodic unit cell take particular symmetric forms. This combination results in microstructural determination of the macroscopic coefficients entering into the Biot-Allard theory of wave propagation and dissipation through porous media. Precise control over pore morphology and mechanical properties of the base material renders this multi-scale approach particularly suitable for various advanced applications.

2:20

1pSA5. Modeling of the acoustic absorption of bi-modal polylactide foams. Shahrzad Ghaffari Mosanenzadeh (Mech. and Industrial Eng., Univ. of Toronto, No. 5, King's College Rd., Toronto, ON M5S 3G, Canada, shahrzad.ghaffari@yahoo.com), Olivier Doutres (Groupe d'Acoustique de Vibrations, Univ. of Sherbrooke, Sherbrooke, QC, Canada), Hani E. Naguib, Chul B. Park (Mech. and Industrial Eng., Univ. of Toronto, Toronto, ON, Canada), and Noureddine Atalla (Groupe d'Acoustique de Vibrations, Univ. of Sherbrooke, Sherbrooke, QC, Canada)

In this study, highly porous bi-modal structures were designed and fabricated from polylactide (PLA) as the main structure and utilizing polyethylene glycol (PEG) to form micro pores by compression molding combined with particulate leaching technique. The pore size of the foam structure was controlled by salt particulates and higher interconnectivity was achieved by the co-continuous blending morphology of PLA matrix with water-soluble PEG. This fabrication method makes it possible to control pore geometry and interconnectivity closely and therefore is an ideal approach to study the relation between microstructure and acoustic properties of the foams. PLA is a bio-based thermoplastic polymer derived from renewable resources. Therefore, the resulting acoustic foams are benign and environmentally friendly. Fabricated foams were characterized based on cellular, acoustic, and mechanical properties. The acoustic performance of the foams was studied by measuring the normal incident absorption coefficient in accordance with the ASTM E1050 standard. An analytical model based on Johnson–Champoux–Allard model was used to numerically simulate the acoustic performance of foams under study. Numerical results predict the absorption behavior of PLA foams with high accuracy. Through this research, open porosities close to 90% were achieved, and the effect of water soluble polymer on cellular properties, acoustic and mechanical performance of polylactide foams was studied.

2:40

1pSA6. Improving the sound absorbing efficiency of closed-cell foams using shock waves. Olivier Doutres, Noureddine Atalla (GAUS, Université de Sherbrooke, 2500 Boul. de l'Université, Sherbrooke, QC J1K 2R1, Canada, olivier.doutres@usherbrooke.ca), Martin Brouillette, Christian Hébert (Dept. of Mech. Eng., Shock wave Lab., Sherbrooke, QC, Canada), and David Begg (Woodbridge Foam Corp., Woodbridge, ON, Canada)

Producing closed-cell foams is generally cheaper and simpler than open-cell foams. However, the acoustic efficiency of closed-cell foam materials is poor because it is very difficult for the acoustic waves to penetrate the material. A method to remove the membranes closing the cell pores (known as reticulations) and thus to improve the acoustic behavior of closed-cell foam material is presented. The method is based on the propagation of shock waves inside the foam aggregate where both the shock wave generator and the foam are in air at room conditions. Various shock treatments have been carried out on a polyurethane foams, and the following conclusions were drawn: (1) the reticulation rate increases and thus the airflow resistivity decreases while increasing the amplitude of the shock treatment; (2) the softness of the foam increases; (3) the process is reliable and repeatable; (4) obtained acoustic performance is comparable to classical thermal reticulation; and (5) the process can be used to control the reticulation rate along the thickness.

3:00

1pSA7. Full-band exact homogenization of one-dimensional elastic metamaterials. Min Yang, Zhiyu Yang, and Ping Sheng (Physics, Hong Kong Univ. of Sci. and Technol., Dept. of Phys., HKUST, Clear Water Bay, Kowloon, Hong Kong 852, Hong Kong, erwinstu@ust.hk)

Metamaterials extend the realm of materials' properties by carefully designed structural inclusions. By targeting the extraction of effective properties from composite materials, homogenization theory plays an important role for metamaterials in their design and characterization. However, conventional homogenization methods are limited to the long wavelength limit. Here, we introduce an exact homogenization scheme valid for one-dimensional metamaterials over the full frequency band. In this scheme, with the aid of eigenstates' characterization, a set of explicit formulas for effective mass density and effective elastic modulus are obtained by matching the surface responses properties of a metamaterial's single structural unit with a piece of effectively homogenized material. In the frequency regimes beyond the conventional homogenization theory, new features, such as the imaginary parts of the effective parameters, have been found. Applying this scheme on a layered structure, the predicted transport properties and displacement fields from the effective parameters show excellent agreement with numerical simulations.

3:20

1pSA8. Noise control using lightweight acoustic metamaterials. Christina J. Naify (National Res. Council, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil)

Metamaterials have emerged as promising solutions for manipulation of sound waves in a variety of applications. Negative dynamic mass has been explored in metamaterial applications to improve sound insulation in both three-dimensional (ball-in-rubber), and two-dimensional (membrane-type) approaches. Noise control utilizing locally resonant acoustic materials (LRAM) resulted in improved

sound insulation by 500% over acoustic mass law predictions at peak transmission loss (TL) frequencies. The LRAM contribute minimal added mass, making them appealing for weight-critical applications such as aerospace structures. In this study, an overview of LRAM for noise control applications will be presented, including potential issues associated with scale-up of the structure. TL of single-celled and multi-celled LRAM was measured using an impedance tube setup with systematic variation in geometric parameters to understand the effects of each parameter on acoustic response. Finite element analysis (FEA) was also performed to predict TL as a function of frequency for structures with varying complexity, including stacked structures and multi-celled arrays. [Work supported by the Office of Naval Research.]

3:40

1pSA9. Omnidirectional acoustic absorber with a porous core—Theory and measurements. Olga Umnova, Andy Elliott, and Rodolfo Venegas (Univ. of Salford, The Crescent, Salford m5 4wt, United Kingdom, o.umnova@salford.ac.uk)

An omni-directional acoustic absorber consisting of a porous core and the impedance matching metamaterial layer has been designed and tested in the laboratory. Semi-analytical and numerical models have been developed and validated. The numerical model takes into account the viscous losses in the matching layer. A 1.5 m demonstrator has been built and tested under acoustic and weak shock excitation. Testing with acoustic excitation showed good agreement between measurement and model, with near perfect absorption between 400 and 1000 Hz. Testing against weak single-pulse shock in an anechoic chamber also confirmed a significant reduction in peak pressure levels when compared to a conventional porous absorber without matching layer. The findings suggest that structure is equally effective when wrapped around an object like a column, pipeline, or the underside of a vehicle, as it would be when entirely filled with an absorbing porous material.

4:00

1pSA10. Reflexion of flexural waves at the end of a tapered beam of quadratic profile covered with a thin viscoelastic layer. Vivien Denis, Julien Poittevin, Adrien Pelat, Benjamin Elie, and Francois Gautier (Laboratoire d'Acoustique de l'Université du Maine, Université du Maine, Avenue O. Messiaen, Le Mans 72000, France, francois.gautier@univ-lemans.fr)

Flexural waves propagating in a beam can be efficiently absorbed if one extremity is tapered with a power law profile and covered by a very thin viscoelastic layer [Krylov, JSV **274**, 605–619 (2004)]. Such a terminaison induces an effect known as "the acoustic black hole effect" (ABH), which is resulting from properties of propagation of flexural wave in beams having non homogeneous thicknesses: if the thickness decreases locally, flexural waves slow down and the amplitude of the displacement field increases, leading to efficient energy dissipation if an absorbing layer is placed where the thickness is minimum [Georgiev *et al*., JSV 330, 2497–2508 (2011)]. Absorption of the ABH terminaison is estimated, thanks to the direct measurement of the reflexion coefficient, using a wave decomposition technique. Experimental modal analysis of a ABH beam can be performed using a "high resolution" technique, which permits to estimate the modal density. Analysis of these experimental results is performed, thanks to a model based on the finite difference method. It is shown that local transverse modes are playing an important role in the absorption properties of ABH.

4:20

1pSA11. Importance of container geometry of an elastomer particulate damper called Enidamp^{TM} into the damping of vibrations. Marcelo Bustamante, Samir N. Gerges, Júlio A. Cordioli, Ovdalio P. Martin (Vib. and Acoust. Lab., UFSC, João de Deus Machado 224, Ap. 3 - Trindade, Florianópolis, SC 88040-900, Brazil, marcelo.bustamante@lva.ufsc.br), Jeffrey N. Weisbeck, and Mark Ott (Engineering, ITT/Enidine Inc., Orchard Park, NY)

Research testing has been conducted to development of an innovative damping treatment, called Enidamp^{TM}. This treatment can add considerable damping to a structure by leading a vibration through a rigid connection, to a set of elastomer particles, which behaves as a damper. It is of interest to study the parameters involved in this mechanism of energy dissipation and to achieve optimal performance of the Enidamp^{TM} system. Particularly, this paper experimentally analyzes the importance of the container geometry which houses the elastomer particles. For this purpose, the fluidization point at which the elastomeric particles become optimally excited to maximize damping is found for different depths and widths of the Enidamp^{TM} container keeping the volume constant. Important conclusions from this experiment guide future studies for prototype improvements.

4:40

1pSA12. Structural-borne sound mitigation in small wind turbines using constrained viscoelastic layer. Baruch Pletner (Intelligent Dynam. Canada, LTD, 11 Acadia St., Dartmouth, NS B2Y 2N1, Canada, baruch.pletner@iptrade.com), Nic Strum, David Sampson, and Ali Kheirabadi (Seaforth Energy, Inc., Halifax, NS, Canada)

As the growing acceptance of small wind turbines operating in suburban and rural communities coincides with increasingly stringent regulations on the sound emitted by these turbines, the need for sound mitigation solutions becomes urgent. Small turbines need to be affordable for small business use, and thus, proposed solutions must be cost-effective and low maintenance. Easy retrofit to existing turbines is also desirable. Wind turbines generate sound via two main mechanisms: structural borne sound generated by the gearbox and generator and transmitted through the nacelle structure and aeroacoustic sound generated by the interaction of the airstream with the rotating blades and other turbine components. Current study focused on the mitigation of structural-borne sound in a 50 kW wind turbine using a constrained viscoelastic layer. The viscoelastic layer comprised of multiple tiles with normal force to the nacelle structure provided by ratcheting bands. Optimal value for the normal force was empirically determined, and the resulting reductions in generated sound were documented both in the laboratory and on a working turbine under a number of operating conditions. The result is a cost-effective solution with zero cost of ownership and easy installation on a wide range of small to medium-size wind turbines.

Session 1pSCa

Speech Communication: Mixed Effects Modeling: Applications and Practice in Speech Research

Christian DiCanio, Chair
Haskins Lab., 300 George St., Ste. 900, New Haven, CT 06511

Chair's Introduction—12:55

Invited Papers

1:00

1pSCa1. Modeling multi-level factors using linear mixed effects. Cynthia G. Clopper (Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, clopper.1@osu.edu)

Linear mixed-effects models of 2x2 designs are readily interpretable using treatment contrast coding, although their interpretation is not directly comparable to the interpretation of more traditional ANOVAs. In particular, the interpretation of the "main effect" term for one factor holds only for the baseline level of the other factor. Interpreting models of designs involving factors with more than two levels and/or interactions involving more than two factors is more complex and even less comparable to familiar interpretations of ANOVAs. Alternative methods for analyzing these more complex designs include using different contrast coding (e.g., sum, Helmert, or custom), selecting specific baseline levels for the factors, and running multiple models of the same data set with different baseline levels of comparison. These methods may return quite different results, however, such that a significant factor with treatment contrast coding may not be significant with sum contrast coding or a significant interaction term in one model may not be significant when the baseline levels of the relevant factors are changed. Thus, although linear mixed effects provide an opportunity to model complex designs with multiple sources of variability, this modeling requires careful consideration of model parameters to achieve the most appropriate interpretation of the data.

1:20

1pSCa2. Multilevel models, covariates, and controlled factors in experimental speech research: Unified analyses of highly structured data. Noah H. Silbert, Jared A. Linck (Ctr. for Adv. Study of Lang., Univ. of Maryland, 7005 52nd Ave., College Park, MD 20742, nsilbert@umd.edu), and Mark VanDam (Speech & Hearing Sci., Washington State Univ., Spokane, WA)

Experimental speech research often makes use of complex experimental designs, but even when multiple experimental factors are manipulated, measured outcomes may be influenced by non-controlled and incompletely controlled factors. Multilevel models (of which mixed-effect models are a special case) enable unified analysis of the relationships between, on the one hand, trial-level data and, on the other, experimental factors and potentially important non-controlled variables. Fitted multilevel models allow us to draw inferences simultaneously about group-level experimental effects and covariates (the typical focus of experimental work) as well as individual subject and item properties (both of which can be important in applied research). The utility of multilevel models will be illustrated with analyses of data from a number of studies. We present models of phonological structure, gender differences, and within-gender subject variability in the acoustics of spoken English consonants; simultaneous modeling of experimental factors, subject and item variability, and second language proficiency in bilingual lexical processing; and modeling of the effects of age, hearing loss, phonological/lexical properties, subject and item variability, and multiple vocabulary-related covariates in early language development.

1:40

1pSCa3. Experimentally elicited productions: Differences and similarities between mixed effects and ANOVA analyses. Matthew Goldrick (Linguistics, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, matt-goldrick@northwestern.edu)

Currently, many experimental studies of speech production use fully counterbalanced designs to examine variation in categorical (e.g., correct/incorrect) or relatively continuous measures (e.g., reaction times, voice onset times). These data present several challenges to ANOVA analyses. Some of these issues are well known to be the speech community; for example, the non-normality of dependent variables such as proportion correct. Others have been less extensively addressed; for example, many speech studies account for participant- but not item-specific contributions to variance. I'll discuss the opportunities and challenges in using linear mixed effects models to address these issues. I'll review some of the common issues that arise in using such models and discuss how to interpret and report their results.

2:00

1pSCa4. The use of mixed effects models in quantifying the dynamics of speech. Khalil Iskarous (Linguistics, Univ.of Southern California, 301P Grace Ford Salvatory, USC, Los Angeles, CA 90089-1693, kiskarou@usc.edu)

Mixed-effects models have been used often in quantifying the variability in data from experiments in speech and language. In most of these experiments, the dependent variable is measured at some landmark of a kinematic or decision process. However mixed-effects models are increasingly being used to quantify the variability of dependent variables that vary in time, such as articulator movements, formant transitions, and eye tracking data. This presentation will first provide a tutorial introduction to the use of the mixed-effects model, especially the growth-curve variant, for quantifying variability where time is an essential independent variable. It will then be argued that the model coefficients can be interpreted as dynamic coefficients of differential equations that describe the dynamics of the underlying processes.

2:20–2:40 Panel Discussion

Session 1pSCb

Speech Communication: Autocorrelation-Based Features for Speech Perception

Yoichi Ando, Cochair
Ando Yoichi, Kobe Univ., 1-4-132-105 Hiyodoridai, 1-4-132-105 Hiyodoridai, Kobe Kita 657-1123, Japan

Peter Cariani, Cochair
Biomedical Eng., Boston Univ., 629 Watertown St., Newton, MA 02460

Chair's Introduction—12:55

Invited Papers

1:00

1pSCb1. Autocorrelation function mechanism for pitch salience and cross-correlation function mechanism for sound localization revealed by magnetoencephalography. Yoshiharu Soeta (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-8-31 Midorigaoka, Ikeda, Hyogo 563-8577, Japan, y.soeta@aist.go.jp), Ryota Shimokura (Dept. of Otorhinolaryngol. - Head and Neck Surgery, Nara Medical Univ., Kashihara, Japan), and Seiji Nakagawa (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), Ikeda, Osaka, Japan)

A model of primary sensations and spatial sensations is proposed by Ando (2001). The model of the auditory-brain system includes the autocorrelation function (ACF) and the interaural cross-correlation function (IACF) mechanisms. At present, environmental noises are evaluated by sound level such as equivalent continuous A-weighted sound pressure level (LAeq). However, we sometimes feel annoyed with sound with low sound level because of the quality. Sound quality can be characterized by factors obtained from ACF and IACF of sound. For example, pitch and pitch strength can be characterized by delay time and amplitude of the maximum peak of the ACF. Directional sensation can be characterized by delay time and amplitude of the maximum peak of the IACF. To verify the model, we investigated how ACF and IACF factors are coded in our human brain. The results indicated that delay time and amplitude of the maximum peak of the ACF and IACF are coded by the latency and strength of brain activity. In addition, we applied the model to analyze a Buddhist sutra chanted in temples. The results indicated that some characteristics of the sutra could be characterized by the ACF and IACF factors.

1:20

1pSCb2. Autocorrelation-based features for speech representation. Yoichi ando (Kobe Univ., 1-4-132-105 Hiyodoridai, 1-4-132-105 Hiyodoridai, Kobe Kita 657-1123, Japan, andoy@cameo.plala.or.jp)

This study investigates autocorrelation-based features as a potential basis for phonetic and syllabic distinctions. The work comes out of a theory of auditory signal processing based on central monaural autocorrelation and binaural crosscorrelation representations. Correlation-based features are used to predict monaural and binaural perceptual attributes that are important for the architectural acoustic design of concert halls: pitch, timbre, loudness, duration, reverberation-related coloration, sound direction, apparent source width, and envelopment (Ando, 1985, 1998; Ando and Cariani, 2009). The current study investigates the use of features of monaural autocorrelation functions (ACFs) for representing phonetic elements (vowels), syllables (CV pairs), and phrases using a small set of temporal factors extracted from the short-term running ACF. These factors include listening level (loudness), zero-lag ACF peak width (spectral tilt), $\tau 1$ (voice pitch period), $\varphi 1$ (voice pitch strength), τe (effective duration of the ACF envelope, temporal repetitive continuity/contrast), segment duration, and $\Delta\varphi 1/\Delta t$ (the rate of pitch strength change, related to voice pitch attack-decay dynamics). Times at which ACF effective duration τe is minimal reflect rapid signal pattern changes that usefully demarcate segmental boundaries. Results suggest that vowels, CV syllables, and phrases can be distinguished on the basis of this ACF-derived feature set.

1:40

1pSCb3. Synthesis of the speech signals by using autocorrelation function. Shin-ichi Sato and Alejandro Bidondo (Ingeniería de Sonido, Universidad Nacional de Tres de Febrero, Varentín Gómez 4752, Caseros, Provincia de Buenos Aires 1678, Argentina, ssato@untref.edu.ar)

The running autocorrelation function (r-ACF) is obtained by the FFT method based on the Wiener–Khinchine theorem after obtaining the power density spectrum for a signal. This study attempted to reconstruct the original speech signal by using a part of its r-ACF. First, the stationary part of the vowel signals were investigated to determine until which delay time of the ACF (maximum time lag) is necessary to recognize the reconstructed signals as the original ones. Then, the continuous speech signals were investigated to determine the appropriate integration interval as well as the maximum time lag.

2:00

1pSCb4. Relationship between Intelligibility and autocorrelation factors of Japanese monosyllables. Ryota Shimokura, Sakie Akasaka, Hiroshi Hosoi, and Toshie Matsui (Dept. of Otorhinolaryngol. - Head and Neck Surgery, Nara Medical Univ., Shijo-cho 840, Kashihara, Nara 634-8522, Japan, rshimo@naramed-u.ac.jp)

Amplified speech sound by a hearing aid can improve the speech intelligibility for conductive hearing loss, while improve partly for sensorineural hearing loss. Although the indiscernible consonants in Japanese monosyllables have been identified, it is still unclear which acoustical feature in the consonants causes the difficulty on hearing. Therefore, the aim of this study is to clarify an influential factor on the speech intelligibility using autocorrelation function (ACF), which can capture temporal features of signals. The ACF factors were compared with percent articulations obtained by a Japanese speech intelligibility test (n = 144). The time-series factors were calculated by running ACF along a monosyllable, and the median values of the ACF factors were represented. As results, an effective duration (τ_e [ms]) of the ACF was correlated with the averaged percent articulations among the consonants ($r = 0.87$, $p < 0.01$). The τ_e indicates temporal fluctuation of speech signals including its fundamental frequency. The deteriorated perceptual function for temporal fluctuation may reduce the recognition ability of the consonants, so the application of the ACF analysis for a hearing aid may help the hearing of patients with sensorineural hearing loss.

2:20–2:40 Break

2:40

1pSCb5. Speaker recognition analysis using running autocorrelation function parameters. Alejandro Bidondo, Shin-ichi Sato, Ezequiel Kinigsberg, Mariano Arouxet, Sabater Andrés, Agustín Arias, Adrián Saavedra, and Ariel Groisman (Ciencia y Tecnología, Universidad Nacional de Tres de Febrero, Av. De Los Constituyentes 1426, Villa Maipú. San Martín. Buenos Aires 1650, Argentina, abidondo@untref.edu.ar)

The human brain's process associated with the recognition and identification of acoustic signals is regarded as the calculation of the distances between "sound vectors;" the ones listened in the present with the memorized vectors in previous listening. Matching vectors (minimal distance between them) would indicate sounds come from similar sound sources or same sound source. In this study, the statistical values of r-ACF (running autocorrelation function) microscopic parameters of 10 spoken words recordings (in Spanish) by the same and different speakers were calculated, and the aforementioned vector's distances were constructed by using the distances between the r-ACF parameters, being able to predict the degree of similarity between the speakers.

3:00

1pSCb6. Investigation of calculation methods of effective duration on autocorrelation-based Chinese speech recognition. Kun Peng Huang (GuoGuang Electric Co. Ltd., No.8 JingHu Rd., HuaDu, GuangZhou, GuangDong 510800, China, ohmycar@163.com) and Yoichi Ando (Grad. School of Sci. and Technol., Kobe Univ., Kobe, Japan)

The factors extracted from running autocorrelation function have phonetic and syllabic meanings, and they can be used for Chinese speech recognition. One of the factors is the effective duration of ACF envelope τe, which reveals the temporal repetitive continuity of the speech signals. The values of τe vary when different calculation methods are adopted. By adopting different calculation methods to several sets of Chinese vowels and characters, which are recorded from a group of speakers, a preferred calculation method is decided by the uniqueness of τe, that is the τe calculated by such method can mostly distinguish different vowels, characters, or even speakers.

Contributed Paper

3:20

1pSCb7. The role of normalization in phoneme recognition and speaker definition. Roy D. Patterson (Physiology, Development and Neurosci., Univ. of Cambridge, Downing Site, Cambridge, Cambridgeshire CB22 5LW, United Kingdom, rdp1@cam.ac.uk) and Toshio Irino (Faculty of Systems Eng., Wakayama Univ., Wakayama, Japan)

There is size information in speech sounds because the vocal tract and the vocal cords both grow as a child develops into an adult. Specifically, average pitch and mean formant frequency decrease as speaker size increases. Nevertheless, human speech recognition is effectively size invariant across the full range of sizes in the normal population of speakers and well beyond. It is also the case that listeners can discriminate speaker size with great accuracy; indeed, with greater accurately than they can discriminate the loudness of sound or the brightness of light. The first part of this talk describes how the peripheral auditory system normalizes speech sounds automatically to produce a size invariant representation for speech recognition. The second part presents a model of how the central auditory system transforms information in the cochlea into our perception of who is speaking and what they are saying. The model suggests that the system combines information about vocal resonator size with a small amount of contextual information to determine what the person is saying (at the phonological level), and then it adds voice pitch information to determine who is speaking (in the sense of the sex and size of the speaker).

3:40–4:40 Panel Discussion

Session 1pSCc

Speech Communication: Distinguishing Between Science and Pseudoscience in Forensic Acoustics II

Geoffrey Stewart Morrison, Cochair
Forensic Voice Comparison Lab., School of Elec. Eng. & Telecommunications, Univ. of New South Wales, Sydney, NSW 2052, Australia

James Harnsberger, Cochair
Univ. of Florida, 402 NW 24th St., Gainesville, FL 32607

Contributed Papers

1:00

1pSCc1. Mismatched distances from speakers to telephone in a forensic-voice-comparison case. Ewald Enzinger (Forensic Voice Comparison Lab., School of Elec. Eng. & Telecom., Univ. of New South Wales, Sydney, NSW 2052, Australia, e.enzinger@student.unsw.edu.au)

In a forensic-voice-comparison (FVC) case, one speaker (A) was talking on a mobile telephone, and another (B) was standing a short distance away. Later, B moved closer to the telephone. Shortly thereafter, there was a section of speech where the identity of the speaker was disputed. All material for training an FVC-system could be extracted from this single recording, but there was a near-far mismatch: Training data for A were near, training data for B were far, and the disputed speech was near. We describe a procedure for addressing the degree of validity and reliability of an FVC system under such conditions, prior to it being applied to the casework recording: Sections of recordings of pairs of speakers of known identity are used to train an A and a B model; multiple other sections from each of the A and B recordings are used as test data; a likelihood ratio is calculated for each test section; and system validity and reliability are assessed. Prior to training and testing, the A and B recordings were played through loudspeakers and rerecorded via a mobile-telephone network, B was rerecorded twice, once with the loudspeaker near and once with it far from the telephone.

1:20

1pSCc2. Exploring duration and spectral parameters of English /m/ for forensic speaker comparison. Colleen Kavanagh (Audio & Video Analysis Unit, RCMP, 1426 St. Joseph Blvd., Ottawa, ON K1A 0R2, Canada, colleen.kavanagh@rcmp-grc.gc.ca)

The speaker-specificity of five acoustic features of British English /m/ was explored from a forensic speaker comparison perspective. Normalized duration, center of gravity (COG), standard deviation (SD), and frequencies at peak and minimum amplitudes were measured for 30 adult male Standard Southern British English and Leeds English speakers. Spectral measurements were made in each of five frequency bands (0–0.5 kHz, 0.5–1 kHz, 1–2 kHz, 2–3 kHz, and 3–4 kHz) and calculated from a 40-ms window at the midpoint of each token. ANOVAs showed Speaker to be a highly significant factor for all variables. Discriminant analysis (DA) and likelihood ratio (LR) estimation assessed speaker discrimination with individual predictors and combinations thereof. Sample sizes limited the number of predictors in DA to eight; F-ratios were used to select the best predictors for analysis. The COG+SD (bands 1, 3, 4, 5) and Best 8 F-ratios (COG bands 1, 4, 5 + SD 1, 3, 4 + Peak 1, 4) tests achieved 53% and 49% correct classification, respectively. The Best 8 F-ratios and COG+SD tests also produced the best LR results, while COG+Peak performed similarly. DA and LR results for all predictor combinations will be presented and the most promising speaker comparison parameters highlighted.

1:40

1pSCc3. Examining long-term formant distributions as a discriminant in forensic speaker comparisons under a likelihood ratio framework. Erica Gold (Lang. and Linguist. Sci., The Univ. of York, Heslington, York YO10 5DD, United Kingdom, erica.gold@york.ac.uk), Peter French, and Philip Harrison (J P French Assoc., York, United Kingdom)

This study investigates the use of long-term formant distributions (LTFD) as a discriminant in forensic speaker comparisons. LTFD are the distributions calculated for all values of each formant for a speaker in a single recording. Spontaneous speech recordings from 100 male speakers of Southern Standard British English, aged 18–25 were analyzed from the DyViS Database (Nolan 2009). The recordings were auto-segmented to obtain a minimum of 50 s of vowels per speaker. The iCAbS (iterative cepstral analysis by synthesis) formant tracker was used to automatically extract and measure F1-F4 every 5 ms. To assess the evidential value of the LTFDs, likelihood ratios (LRs) were computed using a MatLab implementation of Aitken and Lucy's (2004) Multivariate Kernel-Density formula (Morrison 2007). It was found that LTFD performs well overall, but much better with different speaker comparisons than same speaker comparisons (97.76 % compared to 78% of comparisons providing correct support; Cllr = 0.9072 and EER = 5.47%). LTFD appears to be a good discriminant to include in forensic speaker comparison analyses and offers the added attraction of avoiding potential correlation problems between vowel phonemes.

2:00

1pSCc4. Establishing typicality: A closer look at individual formants. Vincent Hughes (Dept. of Lang. and Linguist. Sci., Univ. of York, Heslington, York YO10 5DD, United Kingdom, vh503@york.ac.uk)

Research into the forensic performance of individual formants has offered considerable evidence to support the traditional acoustic-phonetic view that whilst F1 and F2 encode broad phonetic contrast, higher formants may offer greater speaker-discriminatory potential (Peterson 1959, Ladefoged 2006, Clermont and Mokhtari 1998, Rose 2002). However, the comparative performance of formants has largely been assessed using posterior assessments of speaker-specificity (McDougall 2004, 2006; Clermont *et al.* 2008). Using quadratic polynomial fittings of F1 to F3 from spontaneous tokens of /ai/ extracted from all 100 speakers in the DyVis database (Nolan *et al.* 2009), this paper discusses issues relating to p(H|E)-based voice comparison analysis (particularly the use of discriminant analysis, DA). Further, DA performance is compared with an analysis based on likelihood ratios (LRs). LRs based on F3 are found to only marginally outperform F1 and F2 with regard to the magnitude of same-speaker and different-speaker strength of evidence, as well system performance metrics (EER and Cllr). The poorer than expected F3 LRs are assessed with regard to the distributions of values within- and between-speakers for the best performing F3 coefficient: the constant. The data go some way to establishing F3 population statistics, which may potentially be applied to voice comparison casework.

2:20–2:40 Break

2:40

1pSCc5. A likelihood ratio-based forensic voice comparison using formant trajectories of Thai diphthongs. Supawan Pingjai (CHL, Australian National Univ., Grad. House, Garran Rd., Canberra, NSW 0200, Australia, supawan.pingjai@anu.edu.au)

This study investigates the phonetic–acoustic properties of the three Thai diphthongs /i:a, ɨ:a, u:a/ within the context of forensic voice comparison. The likelihood-ratio approach is applied to the parameterized formant trajectories of each diphthong in order to evaluate their respective discriminatory power. The aim of this study is to assess to what extent such properties can be used to distinguish, in a probabilistic sense, two or more speech samples. Formant trajectories were fitted using both polynomial interpolation and the discrete cosine transform. Likelihood ratio values were derived by the multivariate kernel density (MVKD) estimation approach proposed by Aitken and Lucy (2004) and then calibrated by using the log-likelihood ratio cost function—Cllr (Brummer 2005) and the 95%-credible interval (Morrison *et al*., 2010). We have finished gathering all speech data for this study, and are currently processing the data using various computational tools. References: Aitken, C. G. G. and Lucy, D. "Evaluation of trace evidence in the form of multivariate data," App. Stat. **54**,109–122 (2004). Brümmer, N., FoCal Toolkit [software], 2005. See: http://www.dsp.sun.ac.za/~nbrummer/focal/. Morrison G. S., Thiruvaran T., and Epps J. (2010) "Estimating the likelihood-ratio output of a forensic-voice-comparison system," The Speaker and Language Recognition Workshop, Brno.

3:00

1pSCc6. Fusion of multiple formant-trajectory- and fundamental-frequency-based forensic-voice-comparison systems: Chinese /ei1/, /ai2/, and /iau1/. Cuiling Zhang (Dept. of Forensic Sci. & Technol., China Criminal Police Univ., Tawan St. NO.83, Huanggu District, Shenyang, Liaoning 110854, China, cuiling-zhang@forensic-voice-comparison.net) and Ewald Enzinger (Forensic Voice Comparison Lab.,School of Elec. Eng. & Telecommunications, Univ. of New South Wales, Sydney, NSW, Australia)

This study investigates the fusion of multiple formant-trajectory- and fundamental-frequency-trajectory-based (f0-trajectory-based) forensic-voice-comparison systems. Each system was based on tokens of a single phoneme: tokens of Chinese /ei1/, /ai2/, and /iau1/ (numbers indicate tones). Human-supervised formant-trajectory and f0-trajectory measurements were made on tokens from a database of recordings of 60 female speakers of Chinese. Discrete cosine transforms (DCT) were fitted to the trajectories and the DCT coefficients used to calculate likelihood ratios via the multivariate kernel density (MVKD) formula. The individual-phoneme systems were fused with each other and with a baseline mel-frequency cepstral-coefficient (MFCC) Gaussian-mixture-model universal-background-model (GMM-UBM). The latter made use of the entire speech-active portion of the recordings. Tests were conducted using high-quality recordings as nominal suspect samples and mobile-to-landline transmitted recordings as nominal offender samples. Fusion of the phoneme-systems with the baseline system via logistic regression did not lead to any substantial improvement in validity and reliability deteriorated.

3:20–4:00 Panel Discussion

MONDAY AFTERNOON, 3 JUNE 2013 — 510A, 1:00 P.M. TO 3:20 P.M.

Session 1pSPa

Signal Processing in Acoustics, Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Nearfield Acoustical Holography (NAH) Measurements and Applications

Alba Granados, Cochair
Dept. Elec. Eng., Tech. Univ. of Denmark, DTU Bldg. 352, Kongens Lyngby DK-2800, Denmark

Alan T. Wall, Cochair
Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602

Contributed Papers

1:00

1pSPa1. Regularized reconstruction of sound fields with a spherical microphone array. Alba Granados and Finn Jacobsen (Dept. of Elec. Eng., Tech. Univ. of Denmark, DTU Bldg. 352, Kongens Lyngby DK-2800, Denmark, fja@elektro.dtu.dk)

Spherical near field acoustic holography with microphones mounted on a rigid spherical surface is used to reconstruct the incident sound field. However, reconstruction outside the sphere is an ill-posed inverse problem, and since this is very sensitive to the measurement noise, straightforward implementation might lead to disastrous reconstructions. A large number of regularization tools based on singular value decomposition are available, and it has been found that the acoustic holography problem for certain geometries can be formulated in such a way that similarities to singular value decomposition become apparent. Hence, a number of regularization methods, including truncated singular value decomposition, standard Tikhonov, constrained Tikhonov, iterative Tikhonov, Landweber and Rutishauser, have been adapted for spherical near field acoustic holography. The accuracy of the methods is examined by means of simulations and measurements, which leads to practical recommendations on the use of regularization techniques regarding space and frequency.

1:20

1pSPa2. Quasi-holographic processing as an alternative to synthetic aperture sonar imaging. David J. Zartman, Daniel S. Plotnick (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, zartman.david@gmail.com), Timothy M. Marston (Naval Surface Warfare Ctr., Panama City Div., Panama City, FL), and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA)

By limiting attention to supersonic-like wavevectors, time-resolved holographic imaging was demonstrated to be helpful for identifying transient elastic responses of targets contributing to far field scattering [Hefner and Marston, Acoust. Res. Lett. Online **2**, 55–60 (2001); **3**, 101–106

(2002)]. That approach was applied to time-evolving three-dimensional wavefields scanned with a two-dimensional raster scan at a significant offset. In recent work, however, it was also demonstrated that line scan measurements of bistatic scattering could be processed quasi-holographically as an alternative to synthetic aperture sonar (SAS) imaging [Baik *et al.*, J. Acoust. Soc. Am. **130**, 3838–3851 (2011)]. The present investigation broadens the line scan approach to reversible signal extraction associated with image features, and to monostatic quasi-holographic imaging, in which the source and receiver are co-located and scanned along a line. Some applications of this approach will be illustrated such as separation of signals from edge diffraction features and specular features from those of elastic features. [Work supported by ONR and the NSWC-PCD ILIR/IAR research fund.]

1:40

1pSPa3. Optimized two-dimensional imaging of transient sound fields using a hybrid transient acoustic holography. Siwei Pan and Weikang Jiang (State Key Lab. of Mech. System and Vib., Shanghai Jiao Tong Univ., 800 Dong Chuan Rd., Shanghai 200240, China, swpan@sjtu.edu.cn)

A hybrid transient acoustic holography (HTAH) is presented to visualize the two-dimensional transient sound fields radiated from planar sources with unknown locations and dimensions, by combining the time reversal source localization with the near-field acoustic holography (NAH) based on the interpolated time domain equivalent source method (TDESM). Based on the near-field measurements with a microphone array, the time reversal source focusing algorithm is used to find out the hotspots of the sound sources on the equivalent source plane, which suggests the collocation of equivalent sources. The interpolated TDESM based NAH is then employed to reconstruct and image the transient sound field on the reconstruction plane. The proposed HTAH technique can reduce the elements number of microphone array by only collocating the equivalent sources in the vicinity of the “real” sound sources. The visualization of the transient sound fields radiated from single-planar-piston and dual-planar-piston model is studied by numerical simulations, respectively. The experiments are performed in a semi-anechoic chamber by using two baffled loudspeakers. Both the simulation and experimental results revealed that this hybrid scheme can realize a better two-dimensional imaging of transient sound fields than the original interpolated TDESM based NAH in the measurement using same amount of microphones.

2:00

1pSPa4. Modified statistically optimized near-field acoustical holography for jet noise characterization. Alan T. Wall, Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, alantwall@gmail.com)

Near-field acoustical holography has been shown to be a useful tool for visualizing jet noise fields. It has been applied to a full-scale jet on an installed military aircraft with promising results, but the source characteristics in the extreme near field have been difficult to characterize because of the interference of acoustic reflections off the rigid reflecting plane beneath the jet. To provide accurate sound field reconstructions, a modified approach to statistically optimized near-field acoustical holography (SONAH) is implemented. In conventional SONAH, the sound field is represented by a matrix of elementary wave functions at all desired spatial locations. In this modified approach, advantage is taken of the property that arbitrary, user-defined functions can be selected for this matrix. Here, two sets of cylindrical wave functions, one centered on the jet centerline and one on the image source centerline, are used to obtain an accurate near-field reconstruction.

2:20

1pSPa5. A study on the effect of sound velocity estimation for underground imaging. Ryo Toh, Takuya Sakuma, and Seiichi Motooka (Dept. of Elec. Electron. and Comput. Eng., Chiba Inst. of Technol., 2-17-1, Tsudanuma, Narashino, Chiba 275-0016, Japan, liang.tao@it-chiba.ac.jp)

An efficient technique for detecting objects buried underground is expected for archaeological exploration and civil engineering. We have proposed a three dimensional underground imaging method by using an amplitude correlation synthesis processing and an electromagnetic induction type sound source. Because the sound velocity of sand and soil depends on a lot of physical parameters that varies with the area, it is needed to be measured before imaging processing. Up to now, the sound velocity employed for signal processing is measured by receivers buried underground near the area to be detected. In order to improve the efficiency of field detection, a method of estimating the underground sound velocity by the sound velocity measured from the ground surface is studied. In this study, the comparison experimental results of the velocity of underground transverse wave, the velocity of underground longitudinal wave, and the sound velocity measured from the ground surface are introduced. Furthermore, the effect on the image of underground objects brought forth by the sound velocity estimation is discussed.

2:40

1pSPa6. Improved hydrophone calibration by combining acoustic holography with the radiation force balance measurements. Sergey Tsysar (Phys. Faculty, Lomonosov Moscow State Univ., GSP-1, 1-2 leninskie Gory, Moscow 119991, Russian Federation, sergey@acs366.phys.msu.ru), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Oleg Sapozhnikov (Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation)

Ultrasound sources are frequently characterized by the radiation force (RF) balance method that is based on the relation between the total acoustic power and RF on absorbing or reflecting targets. This relation is usually taken from the plane-wave approximation or with a geometrical correction for focused sources. However, real sources emit inhomogeneous acoustic beams. Acoustic holography is a method of recording the true field by measuring both pressure magnitude and phase over a 2D surface (a hologram). The hologram makes it possible to accurately calculate the radiation stress tensor on the surface of the absorbing target. Such measurements allow the relation of hydrophone sensitivity with measured RF based on the known exact expression for RF as a function of the 2D pattern of acoustic pressure and particle velocity. This suggests an improved approach for single-frequency hydrophone calibration that benefits from the inherent accuracy of mass balances as well as the fact that pressure approximately scales with one half of the accuracy of the RF measurement. In the current study this approach was used to calibrate a hydrophone by characterizing a 1-MHz flat piezoceramic source in water using acoustic holography and a RF balance. [Work supported by RFBR and NIH EB007643.]

3:00

1pSPa7. Multi-spectral acoustic imaging on object surface in air. Xinhua Guo, Yosuke Mizuno, and Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., 4259-R2-26 Nagatsuta, Midori-ku, Yokohama, JAPAN, Yokohama 226-8503, Japan, guoxinhua@sonic.pi.titech.ac.jp)

Acoustical imaging has been performed using mono-frequency or a limited number of frequencies in the previous studies. The frequency dependence, however, may provide rich information on surface profiles, structures hidden under surfaces, and material properties of objects. In this study, acoustic imaging on object surfaces was conducted over a wide frequency range with a fine frequency step. A rigid surface with different profiles and a boundary between two objects composed of different materials were illuminated by sound wave swept over the frequency range from 1 kHz to 20 kHz with a 30-Hz step. The scattered sound field was recorded two-dimensionally using a scanning microphone, and the holographic method was used to reconstruct the sound pressure distribution on the surface from the recorded data. From the experimental results, the characteristics of the surfaces with respect to the shapes and material properties were demonstrated experimentally. The depth of the holes was identified by its own resonance frequency, and the two different materials were successfully distinguished by multiple images obtained at different frequencies.

Session 1pSPb

Signal Processing in Acoustics: Acoustic Feature Extraction and Characterization

Edmund J. Sullivan, Chair
Prometheus Inc., 46 Lawton Brook Lane, Portsmouth, RI 02871

Contributed Papers

3:20

1pSPb1. Classifying sonar signals with varying signal-to-noise ratio and bandwidth. Stefan Murphy (Underwater Sensing, Defence Res. and Development Canada, 9 Grove St., Dartmouth, NS B2Y 3Z7, Canada, stefan.murphy@drdc-rddc.gc.ca)

An automatic aural classifier developed at Defence Research and Development Canada has demonstrated the ability to distinguish target echoes from clutter using perceptual-based features inspired by sonar operators. Initially, the classifier was tested with echoes from explosive sources, but more recent research involved transmitting broadband waveforms from transducer sources. In sonar transducer operation, there is a trade off between source level and bandwidth, and the goal of this paper is to study how these factors affect echo classification. Source level relates to signal-to-noise ratio (SNR), which inherently affects classification since signals with low enough SNR cannot be distinguished from noise, let alone other signals. The dependence of classification performance on bandwidth is less obvious; however, the aural classification technique is based on a sub-band type of processing that mimics the basilar membrane in the human auditory system, and this model is not well adapted for narrow bands. Performance of the aural classifier is therefore expected to degrade as bandwidth is decreased. In this paper, the effect of SNR and signal bandwidth on echo classification is examined using echoes of varying SNR, and in various bands selected using band-pass filters.

3:40

1pSPb2. An automated framework for the extraction of ultrasonic echoes embedded in noise. Adam Pedrycz, Henri-Pierre Valero, Hiroshi Hori, Kojiro Nishimiya, Hitoshi Sugiyama, and Yoshino Sakata (Acoust.-Sonic Eng., Schlumberger Ltd., 2-2-1 Fuchinobe, Chuo-ku, Sagamihara-shi, Kanagawa-ken 229-0006, Japan, APedrycz@slb.com)

Proposed is an automated framework for the extraction and characterization of the arriving echo in ultrasonic signals embedded in high noise. Commonly, in order to correctly characterize the first echo hidden within a noise-ridden signal, multiple traces are stacked in a gather to improve the SNR, hence facilitating easier extraction and characterization of the recorded echo. Such first order statistical methods require multiple traces and usually fall short in the accuracy of the echo estimate when the variance of the noise does not belong to a known distribution. To mitigate this problem, a framework has been developed comprised of a multi-step procedure, i.e., pre-processing, localization, gating, and finally parameterization of the given echo. This automatic framework operates on single traces and does not require the setting of processing parameters. By means of this method, the true echo can be extracted in one-shot from other overlapping noise components. Furthermore, because the method operates on a trace-by-trace basis, it is insensitive to large non-stationarities in the baseline. Experiments conducted using synthetic as well as aluminum reflector pulse-echo lab data demonstrate the effective extraction of the true echo under the presence of noise and ringing at varying levels of severity.

4:00

1pSPb3. Extract voice information using high-speed camera. Mariko Akutsu, Yasuhiro Oikawa, and Yoshio Yamasaki (Dept. of Intermedia Art and Sci., Waseda Univ., 3-4-1 Ohkubo, Shinjuku, Tokyo, Japan, yoikawa@waseda.jp)

Conversation is one of the most important channels for human beings. To help communications, speech recognition technologies have been developed. Above all, in a conversation, not only contents of utterances but also intonations and tones include important information regarding a speaker's intention. To study the sphere of human speech, microphones are typically used to record voices. However, since microphones have to be set around a space, their existences affect a physical behavior of the sound field. To challenge this problem, we have suggested a recording method using a high-speed camera. By using a high-speed camera for recording sound vibrations, it can record two or more points within the range of the camera at the same time and can record from a distance, without interfering with the sound fields. In this study, we extract voice information using high-speed videos, which capture both a face and a cervical part of the subject. This method allows recording skin vibrations, which contain voices with individuality and extrapolating sound waves by using an image processing method. The result of the experiment shows that a high-speed camera is capable of recording voice information.

4:20

1pSPb4. Multi-stage identification for abnormal/warning sounds detection based on maximum likelihood classification. Kohei Hayashida, Junpei Ogawa, Masato Nakayama, Takanobu Nishiura, and Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, cm012063@ed.ritsumei.ac.jp)

In recent years, the methods utilizing environmental sounds have been increasingly employed for monitoring the safety of the elder who lives in distant place. Environmental sounds should consist of various sounds in daily life, and identified ones enables to detect abnormity. To detect abnormity, it is therefore required that abnormal/warning sounds are accurately identified among environmental sounds. In the past, environmental sound identification methods have generally utilized acoustic models constructed by each sound source for all environmental sounds. In our former research, we proposed multi-stage identification for detecting abnormal/warning sounds accurately. However, these methods design individual acoustic models from similar sounds. Therefore, the sound identification performance is degraded. To overcome this problem, in this study, we proposed environmental sound classification based on acoustic features for model construction. The proposed method classifies environmental sounds based on the difference of acoustic likelihood and designs acoustic models are constructed by classification results. Moreover, the proposed method detects the abnormal/warning sounds more accurately by combining it with the multi-stage identification. We carried out the evaluation experiment with environmental sound database. Experimental results of this experiment demonstrate that the identification performance for the proposed method is higher than that for the conventional methods.

4:40

1pSPb5. Abnormal events recognition and classification for pipeline monitoring systems based on vibration analysis and artificial neural networks. Bin Chen (School of Automation, Beijing Univ. of Posts and Telecommunications, Beijing, China) and Xiaobin Cheng (Key Lab. of Noise and Vib. Res., Inst. of Acoust., Chinese Acad. of Sci., 21, Beisihuanxilu Rd., Haidian District, Beijing 100190, China, xb_cheng@mail.ioa.ac.cn)

Pipelines have become the principal means of oil and gas transportation. However, pipeline leakage takes place due to some natural or artificial damages, which may cause loss of life and properties along with the environmental pollutions. A new pipeline detection and pre-warning system based on distributed optical fiber sensor is proposed, and the hardware has been accomplished. Now, its following key problem is how to recognize and classify the abnormal events, such as oil stealing, construction, artificial excavation, motor work, and train passing. This paper involves a study on this and proposes a solution method. First, original vibration signal is pre-processed and segmented according to threshold of energy within a narrower bandwidth. Then, event features in time and frequency domain are analyzed through statistical analysis and short-time Fourier transform (STFT). The energy coefficients at some bandwidth can distinguish different type of abnormal events, which are chosen as feature vectors. At classification, abnormal events are first divided into discrete and continuous events with single classifier, which can decrease classified event sets and improve recognition accuracy. Then, BP artificial neural network is applied to identify the type of abnormal events. Finally, proposed method will be verified with actual collection data sets.

5:00

1pSPb6. Backward waves and quasi-resonance of shells and invariants of the time reversal operator. Philippe D. Franck, Clorennec Dominique, Maximin Cès, Romain Anankine, and Claire Prada (Institut Langevin, ESPCI ParisTech, CNRS UMR 7587, 1 rue Jussieu, Paris 75006, France, claire.prada-julia@espci.fr)

Backward waves propagating on shell are guided modes with opposite phase and group velocities. For a shell in vacuum, backward modes are linked to zero group velocity modes and resonances, which have been the object of recent studies. For a shell embedded in water, the group velocity does not vanish because of the leakage into the fluid. However, the group velocity of the backward mode has a minimum associated to a quasi-resonance. These phenomena are studied on air filled steel and zircaloy hollow cylinders, using a 3 MHz linear array in pulse echo mode. The circumferential guided modes are generated and their radiation into water detected by the array. The modes are separated using the decomposition of the time reversal operator (TRO), each pair of counter-propagating modes being associated to two invariants of the TRO [Prada *et al*. J. Acoust. Soc. Am. (1998)]. Two resonances are revealed by the eigenvalues of the TRO, one is associated with the first longitudinal thickness resonance and the other, very high, occurring at a slightly lower frequency, corresponds to the minimum of the group velocity of the backward mode. The back-propagations of the eigenvectors of the TRO provide the phase velocities of these modes.

MONDAY AFTERNOON, 3 JUNE 2013 — 516, 1:00 P.M. TO 5:00 P.M.

Session 1pSPc

Signal Processing in Acoustics: Miscellaneous Topics in Signal Processing in Acoustics (Poster Session)

K. Thomas Wong, Cochair
Elec. Eng., Hong Kong Polytechnic Univ., Hong Kong, Hong Kong

Jens Meyer, Cochair
mh Acoust., 38 Meade Rd., Fairfax, VT 05454

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

1pSPc1. Objective analysis of higher-order ambisonics sound-field reproduction for hearing aid applications. Chris Orinos and Jorg M. Buchholz (National Acoust. Lab., Australian Hearing, 126 Greville St., Sydney, NSW 2067, Australia, chris.orinos@nal.gov.au)

The evaluation of hearing aids (HAs) inside realistic sound environments is of increasing interest. Higher-order Ambisonics (HOA) has been used for loudspeaker-based sound field resynthesis and HOA recording microphone arrays are available. Although HOA has been evaluated perceptually, it is unclear how far the results can be transferred to evaluating HA technologies (particularly multi-microphone enhancement algorithms). In order to determine the minimum HOA order required for HA testing, an HOA framework was developed, simulating the entire path from sound presented in a room, picked up by a microphone array, decoded, and received at the ears of a HA-fitted dummy head. HA directivity patterns were compared between an ideal free-field and its HOA representation to evaluate the introduced error. In-room analysis was conducted to investigate the bandwidth and performance of a directional microphone in realistic situations. For a bandwidth B, the required order M_{min} was found to be $M_{min} \geq B/600$ Hz for the anechoic (worst) case scenario. The presence of reverberation introduced natural room response variations across different source-receiver locations, suggesting that the acceptable HOA error can be increased. Hence, in reverberant environments, the required HOA order is reduced, and at least 2D HOA reproduction can be used for evaluation of HA technologies.

1pSPc2. Linearized versus non-linear inverse methods for seismic localization of underground sources. Geok Lian Oh and Finn Jacobsen (Acoust. Technol., Tech. Univ. of Denmark, 25 Jalan Sempadan #03-06, Singapore 457400, Singapore, gloh@elektro.dtu.dk)

The problem of localization of underground sources from seismic measurements detected by several geophones located on the ground surface is addressed. Two main approaches to the solution of the problem are considered—a beamforming approach that is derived from the linearized inversion problem, and the Bayes nonlinear inversion method. The travel times used in the beamformer are derived from solving the Eikonal equation. In the linearized inversion method, we assume that the elastic waves are predominantly acoustic waves, and the acoustic approximation is applied. For the nonlinear inverse method, we apply the Bayesian framework where the misfit function is the posterior probability distribution of the model space. The model parameters are the location of the seismic source that we are interested in estimating. The forward problem solver applied for the nonlinear inverse method is a finite difference elastic wave-field numerical method. In this paper, the accuracy and performance of the linear beamformer and nonlinear inverse methods to localize a underground seismic source are checked and compared using computer generated synthetic experimental data.

1pSPc3. B-format for binaural listening of higher order Ambisonics. Ryouichi Nishimura (National Inst. of Information and Commun. Technol., 2-2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-0288, Japan, ryou@nict.go.jp) and Kotaro Sonoda (Nagasaki Univ., Nagasaki, Japan)

B-format is a four-channel signal capable of rendering a sound scene with spatial information. It can be regarded equivalent to first order ambisonics. Ambisonics requires a high order to contain precise spatial information, and higher order ambisonics requires an exponentially large amount of data. This limitation comes from the fact that the original aim of ambisonics is to reproduce the whole sound field. However, as mobile devices are prevalent, users often listen to sound media through earphones. Because nowadays users can hold sound contents individually, one can assume that sound contents could be produced adaptively to each user. Here we propose a way to make B-format signals more suitable for individual binaural listening. We assume that the production side can capture a sound scene with higher order ambisonics, because it may be processed for enterprise applications. Under this assumption, the binaural signal is once generated from the higher order ambisonics, and then its B-format signal is obtained by inversely processing the signal, assuming the first order ambisonics. Computer simulations show that interaural phase differences (IPDs) are improved at a frequency region where IPD dominantly affects sound localization. Results of hearing tests are also discussed.

1pSPc4. Steering for listening area of reflective audio spot with parametric loudspeaker array. Shohei Masunaga, Daisuke Ikefuji, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is037089@ed.ritsumei.ac.jp)

A parametric loudspeaker has a high directivity by utilizing an ultrasound wave as a carrier wave. Therefore, a parametric loudspeaker can form a specific listening spot called "audio spot." Furthermore, a parametric loudspeaker can form the "reflective audio spot" by utilizing the reflected sound. The listeners in the listening area may perceive the acoustic sound image on the reflector. However, it has the problem that the reflective audio spot with a single parametric loudspeaker is narrow area. Therefore, it is difficult for several listeners to perceive the reflective audio spot at the same time. If we can expand the area of the reflective audio spot, several listeners can perceive the reflective audio spot at the same time. Thus. in this paper, we attempt to steer the area of the reflective audio spot with parametric loudspeaker array. We carried out objective and subjective evaluation experiments to confirm the effectiveness of the proposed system in a conference room. As a result, we confirmed the proposed system can expand the are of the reflective audio spot.

1pSPc5. Steerable parametric loudspeaker with preprocessing methods. Chuang Shi and Woon-Seng Gan (School of Elec. & Electron. Eng., Nanyang Technolog. Univ., 50 Nanyang Ave., S2-B4a-03, DSP Lab, Singapore 639798, Singapore, shichuang@ntu.edu.sg)

The emerging applications of the parametric loudspeaker, such as 3D audio, require both directivity control and high fidelity at the audible frequency (i.e., the difference frequency of the primary frequencies generated by the parametric loudspeaker). Although the phased array techniques have been applied and proved adequate to adjust the steering angles of the parametric loudspeaker, and preprocessing methods have been studied to reduce the harmonic distortions, there is no published work on the effectiveness of the combination of the beamsteering method and the preprocessing methods for the broadband steerable sound beam system. This paper aims to investigate on this unexplored problem. First, the relation between the phases of the primary waves and the difference frequency wave is explored to prove the feasibility of achieving a broadband steerable sound beam from the parametric loudspeaker with preprocessing methods. Second, based on the derived relation, the beamsteering structure is proposed. Lastly, preprocessing methods are proposed for the steerable parametric loudspeaker using double sideband modulation (DSBAM) and square root amplitude modulation (SRAM) methods. Spatial performances of the steerable parametric loudspeaker with preprocessing methods are presented in this paper.

1pSPc6. Evaluation of different spatial windows for a multi-channel audio interpolation system. Jorge A. Trevino Lopez (Grad. School of Information Sci./Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 9808577, Japan, jorge@ais.riec.tohoku.ac.jp), Takuma Okamoto (Universal Media Res. Ctr., National Inst. of Information and Commun. Technol., Seika, Kyoto, Japan), Yukio Iwaya (Faculty of Eng., Tohoku Gakuin Univ., Tagajo, Miyagi, Japan), and Yôiti Suzuki (Grad. School of Information Sci./Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Miyagi, Japan)

The adoption rate of multi-channel audio systems has dramatically increased in recent years. It is common to find 5.1- or 7.1-channel systems in typical home theaters. However, most users do not setup the satellite loudspeakers at the prescribed positions for aesthetic reasons or due to space constraints. Recently, we introduced a technique to optimize multi-channel contents for reproduction over non-ideal loudspeaker setups. Our proposal improves localization accuracy for sound sources reproduced by horizontal loudspeaker arrays. It can also be extended to handle full 3D contents, like those of the upcoming 22.2-channel standard. The proposed method works by applying a set of spatial windows centered at the loudspeaker positions and interpolating along the angles using the spherical harmonics. We now extend our previous results by evaluating the performance of six different spatial window functions. We consider the 5-channel distribution of ITU recommendation BS.775-2, as well as four variations that end-users are likely to deploy. Apparent sound source locations are estimated from the energy and velocity vectors at the sweet spot. Our study found that using the Slepian window with our proposal and a non-ideal loudspeaker layout leads to a reproduced sound field that is closer to that of the ideal configuration.

1pSPc7. Including frequency-dependent attenuation for the deconvolution of ultrasonic signals. Ewen Carcreff, Sébastien Bourguignon, Jérôme Idier (IRCCyN, 1 rue de la Noë, Nantes 44321, France, ewen.carcreff@irccyn.ec-nantes.fr), and Laurent Simon (LAUM, Le Mans, France)

Ultrasonic non-destructive testing (NDT) is a standard process for detecting flaws or discontinuities in industrial parts. A pulse is emitted by an ultrasonic transducer through a material, and a reflected wave is produced at each impedance change. In many cases, echoes can overlap in the received signal and deconvolution can be applied to perform echo separation and to enhance the resolution. Common deconvolution techniques assume that the shape of the echoes is invariant to the propagation distance. This can cause poor performances with materials such as plastics or composites, in particular because acoustic propagation suffers from frequency-dependent attenuation. In geophysics, biomedical imaging or NDT, various frequency-dependent attenuation models have been proposed under different formulations. This communication compares the related possible constructions in order to account for attenuation in deconvolution methods. Especially, we introduce a discrete model for the data, that includes an attenuation matrix in the standard convolution model. Experimental data acquired from Plexiglas plates show that, for this material, attenuation varies roughly linearly with frequency, leading to a unique parameter identification. Finally, we show that such an advanced model manages a better fitting of the data, and promises improvement for the deconvolution of complex ultrasonic data.

1pSPc8. Comparative signal processing analyses of a speed-dependent problem as motivated by brake judder problem. Osman T. Sen (Mech. Eng., Istanbul Techn. Univ., Inonu Cad. No: 65 Gumussuyu, Istanbul 34437, Turkey, senos@itu.edu.tr), Jason T. Dreyer, and Rajendra Singh (Mech. and Aerosp. Eng., The Ohio State Univ., Columbus, OH)

The goal of this paper is to investigate a transient problem using several digital signal processing techniques. First, a simple linear mathematical model, where a point mass is connected to a roller through a contact interface, is developed and the dynamic interfacial force is analytically calculated as a function of the speed. In this model, the contact interface is described with a linear spring and viscous damper, and the system is excited with a base excitation, as defined by the undulations on the roller surface. Due to the time-varying speed characteristics of the roller, the resulting response is transient. Second, the dual-domain analyses of the calculated system response is carried out by using short-time Fourier and wavelet transforms, since single-domain representation leads to a loss of information due to signal's transient characteristics. Third, the Hilbert transform is applied and the envelope curves of the interfacial force response are successfully obtained. Finally, this problem is briefly linked to brake judder phenomenon and its source regimes are briefly explained.

1pSPc9. Fast Bayesian hierarchical inference via sparsity enforcing *a priori* for aeroacoustic source imaging. Ning Chu, Ali Djafari (Groupe de problème inverse, Laboratoire des signaux et systèmes (l2s), SUPELEC, SUPELEC, plateau de Moulon, 3 rue Joliot-Curie, 91192 GIF-SUR-YVETTE Cedex (France), Gif sur yvette, Paris 91192, France, chuning1983@gmail.com), José Picheral (Dept. Signal et Systèmes Electroniques, SUPELEC, Paris, France), and Nicolas Gac (Groupe de problème inverse, Laboratoire des signaux et systèmes (l2s), SUPELEC, Paris, France)

Aeroacoustic imaging is a technique for mapping the positions and powers of aeroacoustic sources. We propose a novel inverse solution by applying Bayesian hierarchical inference via sparsity enforcing *a priori*. We model the sparse prior of source powers by using the double exponential distribution, which can greatly improve the spatial resolution and robustness to background noise. Hyperparameters and source powers can be alternatively estimated based on the joint maximum *a priori* optimization. To accelerate the optimization, we improve the forward model of aeroacoustic power propagation by exploring the convolution operator. Finally, our approach is compared with some classical methods on simulated and real data. And our approach is feasible to apply for aeroacoustic imaging with the 2D non-uniform microphone array in wind tunnel tests, especially for near-field monopole and extended source imaging.

1pSPc10. Low latency audio coder design for high quality audio service on server-client environment. Han-gil Moon, Nam-suk Lee, and Hyun-wook Kim (DMC R&D Ctr., Samsung Electron., 416, Maetan 3-dong, Yeongtong-gu, Suwon 443-742, South Korea, hangil.moon@samsung.com)

Low latency audio coding attracts increasing attention among high quality communication applications such as video conferencing system and server-client media applications such as cloud computing based interactive A/V service. This paper presents a low latency audio coding scheme which can achieve both low delay and high subjective audio quality at the same time. In order to guarantee low delay, the proposed coding system incorporates low overlap window while preserving the window size as same as that of conventional (AAC) long window. The supplementary signal processing tool is incorporated to enhance the audio quality. The proposed coding scheme achieves the delay of 24 ms in 48 kHz and the MUSHRA score, which is comparable to that of commercialized AAC.

1pSPc11. Security screening using ultrasound. David Hutchins, Lee Davis, and Sheldon Tsen (School of Eng., Univ. of Warwick, Gibbet Hill Rd., Coventry CV4 7AL, United Kingdom, D.A.Hutchins@warwick.ac.uk)

This work will demonstrate that it is possible to produce images of hidden objects, using ultrasound transmitted through air. For example, it can be shown that a knife can be imaged, when hidden behind a layer of clothing fabric. To achieve this, it is necessary to use coded waveforms and signal recovery techniques, in order to retrieve small signals in the presence of a much larger reflection from the outer fabric surface. In addition, ultrasound can be used in through-transmission to detect hidden objects within thin packages. This and other examples of the use of air-coupled ultrasound for security work will be demonstrated.

1pSPc12. Articulatory-based speaker recognition. Luis Rodrigues (Concordia Univ., 1515 St. Catherine W, EV12.111, Montreal, QC H3G2W1, Canada, luisrod@encs.concordia.ca) and John Kroeker (Eliza Corp., Beverly, MA)

This paper presents a new methodology for computational speaker recognition based on a mathematical model of articulatory speech production. The method, based on articulatory phonology is tested on the MOCHA database for recognizing a male speaker and a female speaker. From an engineering perspective, in articulatory phonology one is interested in the trajectories over time of a set of articulators. These time trajectories are associated with the production of speech. The basic phonological unit in articulatory phonology is the articulatory gesture, which is defined as a dynamic system specified by a characteristic set of parameters. This dynamic system receives as inputs a target state and a set of parameters that tune the system to the desired action. The output is the solution of the state equation, i.e., the state trajectory, where the sate is formed by the x-y positions of the important articulators that describe human speech. The state trajectory is then mapped to the output speech waveform by emulating the human vocal tract, through the observation equation and the MFCCs frequency description. A simplification of this model will be used in this paper for speaker recognition with 100% success in recognizing a male and a female speaker.

1pSPc13. A dynamic automatic noisy speech recognition system for a single-channel hybrid noisy industrial environment. Sheuli Paul (Univ. of Kaiserslautern, Kaiserslautern, Kaiserslautern 67663, Germany, paul@eit.uni-kl.de)

A dynamic noisy speech recognition system is developed to recognize single-channel small spoken commands in a hybrid noisy industrial environment. This hybrid system has three parts: (a) hybrid pre-processing to enhance noisy speech, (b) feature extraction for perceptual speech features, (c) classification and recognition for the DANSR's result. Here, the single-channel is only one microphone, and the hybrid noise is environmental mixed noise distinguished as: (i) strong, (ii) time varying steady-unsteady, and (iii) mild. A new adaptive feature extraction technique based on local trigonometric transformation (LTT) is introduced and examined. This is adapted with psychoacoustic quantities such as Bark scaled critical band spectrum, loudness scale, and perceptual entropy. Here the spectral analysis is done by rising cut-off function, folding operation, and discrete cosine transformation (DCT-IV) instead of Fourier transform. Then, inverse DCT-IV and unfolding operation result in perceptual LTT (PLTT) features. These are recognized by hidden Markov model (HMM). The new PLTT features are more efficient and perceptually meaningful than the standard feature extraction techniques. The DANSR system is a novel solution for small commands to a long existing hybrid noise problem.

1pSPc14. A novel noise-reduction algorithm for real-time speech processing. Frederic E. Theunissen and Tyler Lee (Psychology and Neurosci., UC Berkeley, 3210 Tolman Hall, Berkeley, CA 94720, theunissen@berkeley.edu)

We developed a new noise-reduction algorithm based on a joint spectro-temporal representation of signals. The algorithm was inspired by the discovery in our laboratory of higher-level avian auditory cortical neurons that showed invariant responses to communication signals. The algorithm consists of an analysis step and a synthesis step. In the analysis step, the sound is first decomposed into narrow band signals by a frequency filter bank. These time-frequency waveforms are then further analyzed using a spectro-temporal modulation filter bank to obtain a representation that is akin to the one generated by cortical neurons. In our algorithm, this modulation filter bank was obtained from the principal component analysis of the speech signal in the time-frequency representation. We then learned which subset of the modulation filters provided the best information to extract the signal from the noise. In the synthesis step, we then used this subset of spectral-temporal modulation feature detectors to generate a set of time-varying frequency gains that could be applied directly to the original time frequency decomposition. In this manner, we were able to perform noise reduction in real time and with minimal delay. Our algorithm yielded similar noise reduction but better quality speech quality than current state-of-the-art algorithms.

1pSPc15. Causal binary mask estimation for speech enhancement using sparsity constraints. Abigail A. Kressner, David V. Anderson, and Christopher J. Rozell (School of Elec. and Comput. Eng., Georgia Inst. of Technol., 58 6th St NE Unit 2608, Atlanta, GA 30308, abbiekre@gatech.edu)

While most single-channel noise reduction algorithms fail to improve speech intelligibility, the ideal binary mask (IBM) has demonstrated substantial intelligibility improvements for both normal- and impaired-hearing listeners. However, this approach exploits oracle knowledge of the target and interferer signals to preserve only the time-frequency regions that are target-dominated. Single-channel noise suppression algorithms trying to approximate the IBM using locally estimated signal-to-noise ratios without oracle knowledge have had limited success. Thought of in another way, the

IBM exploits the disjoint placement of the target and interferer in time and frequency to create a time-frequency signal representation that is more sparse (i.e., has fewer non-zeros). In recent work (in preparation for ICASSP 2013), we have introduced a novel time-frequency masking algorithm based on a sparse approximation algorithm from the signal processing literature. However, the algorithm employs a non-causal estimator. The present work introduces an improved de-noising algorithm that uses more realistic frame-based (causal) computations to estimate a binary mask.

1pSPc16. Objective and subjective evaluation of complementary Wiener filter for speech dereverberation. Kento Ohtani, Tatsuya Komatsu (Grad. School of Information Sci., Nagoya Univ., Furo-cho,Chikusa-ku, Nagoya, Aichi 464-8601, Japan, ohtani.kento@g.sp.m.is.nagoya-u.ac.jp), Kazunobu Kondo (Corporate Res. & Development Ctr., Yamaha Corp., Nagoya, Japan), Takanori Nishino (Information Eng., Grad. School of Eng., Mie Univ., Nagoya, Japan), and Kazuya Takeda (Grad. School of Information Sci., Nagoya Univ., Nagoya, Japan)

Acoustic distortion caused by reverberation can degrade speech quality and performance of speech-based systems. Several dereverberation techniques have been proposed in the literature. For example, a dereverberation method using a complementary Wiener filter can suppress late reverberation with few computational resources. As a method for dereverberation, the method using a complementary Wiener filter has been proposed, and for the exponential decay impulse response model, it is shown theoretically that we can suppress reverberation with few computational resources. In this report, we approximate expectation of the power spectrum. which is necessary to calculate a complementary Wiener filter as exponential moving average. We conducted dereverberation experiments using actual environment room impulse response. The results of the objective evaluation show that the suppression performances of the actual environment room impulse response can approximate from the results of the exponential decay impulse response model. Additionally, we investigated the relationship between the results of objective evaluation and the results of subjective evaluation. In a small reverberation environment, we can see strong correlation between the results of objective and subjective evaluation.

1pSPc17. Evaluation of human-phonatory radiation characteristics with a polyhedron loudspeaker. Naoki Yoshimoto, Kota Nakano, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Noji-higashi, Kusatsu 525-8577, Japan, is046081@ed.ritsumei.ac.jp)

Spoken dialog systems have been studied for car navigation systems and voice search systems. For evaluating these systems, a loudspeaker is used instead of a human because these systems require various kinds of speech samples. However, the sounds radiated by loudspeaker cannot reproduce human-phonatory radiation characteristics. Therefore, the mouth simulator is utilized to reproduce human-phonatory radiation characteristics. Although it is based on the average mouth shapes, shapes of mouth are different among phonemes. Therefore, due to the hardware structure, it cannot accurately reproduce various human-phonatory radiation characteristics affected by shapes of mouth. In this study, we developed a polyhedron loudspeaker to solve this problem. It consists of 11 loudspeakers, which are independently controlled. Controlling eleven loudspeakers makes it possible to reproduce desired radiation characteristics. By utilizing this method, we try to reproduce human-phonatory radiation characteristics of Japanese five vowels and typical consonants with digital filters which were adaptively designed. We carried out an evaluation experiment in various measuring points to verify the effectiveness of the proposed method. As a result, it was confirmed that human-phonatory radiation characteristics with the proposed method could be accurately approximated compared with the conventional mouth simulator.

1pSPc18. Optimized hermetic transform beam-forming of acoustic arrays via cascaded spatial filter arrangements derived using a chimerical evolutionary genetic algorithm. Harvey C. Woodsum and Christopher M. Woodsum (Sci. and Technol., Nergetic System Dynam., LLC, 3700 N. Charles St., Unit 903, Baltimore, Maryland 21218, cwoodsu1@jhu.edu)

Hermetic transforms are complex matrices, having particular mathematical properties, that have recently been introduced to the field of acoustic array signal processing. Cascade sequences of Hermetic transform matrices have been shown to have direct utility in accomplishing spatial filtering and beam-forming of data from oversampled arrays. The present work details the adaptation of techniques previously shown to be successful in the processing of radio-wave phased-array antenna systems [Woodsum *et al.*, *16th International Conference on Cognitive and Neural Systems* (2012)] to the processing of sampled digital data from acoustic arrays. As in our earlier work, the use of a Chimerical, Evolutionary, Genetic Algorithm, having a “feature seeking” fitness function, is retained, for deriving optimal multiplicative arrangements of non-commuting elemental transform matrices. Each elemental matrix represents a spatial “pole” or “zero,” and cascaded arrangements of these are utilized to create a desired spatial pattern response for the array. In terms of acoustic reception, the technique is especially successful in dealing with null placement in order to mitigate large numbers of interfering signals, and in achieving super-resolution beams for arrays that are “acoustically small.” Experimental results are compared to theoretical predictions of performance.

1pSPc19. A study on acoustic imaging based on beamformer to range spectra in the phase interference method. Ryota Miyake, Kohei Hayashida, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0041ki@ed.ritsumei.ac.jp)

Information on the distance to the target is very important to achieve the practical use of hands-free speech interfaces and nursing-care robots. Many distance measurement methods, which use the time-of-flight (TOF) of a reflected wave measured with reference to the transmitted wave, have been proposed. However, these methods cannot measure short distances because the transmitted wave, which has not attenuated sufficiently at the time of a reflected wave reception, suppresses reflected waves for short distances. We previously proposed an acoustic distance measurement method based on interference between the transmitted and reflected waves, which can be used for distance measurement over a short range using single microphone. This method is referred to the phase interference method. It can estimate the distance to target, but cannot estimate the direction of target. In the present paper, therefore, we propose to achieve acoustic imaging with the phase interference method by using microphone-array instead of single microphone. More specifically, we apply the beamformer to the range spectra calculated from observed signals at each microphone of microphone-array to obtain the spatial information. Finally, we confirm the effectiveness of the proposed method through evaluation experiments in real environments.

1pSPc20. Investigations into the human pinna shapes on head-related transfer functions in the median plane. Hajime Komatsu, Kota Nakano, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0016rv@ed.ritsumei.ac.jp)

The binaural reproduction system requires many accurate measurements of head-related transfer functions (HRTFs) to achieve the high-precision sound localization. However, the actual measurement of HRTFs has a heavy burden for subjects. To solve this problem, personalize HRTFs have been proposed. In the personalize HRTFs, the interaural level difference (ILD) and the interaural time difference (ITD) are utilized on the sound localization in the horizontal plane, and the spectral envelope of HRTFs is utilized on the sound localization in the median plane. In the present paper, we focus on the human pinna shapes as listener’s anthropometric parameters on the sound localization in the median plane. In order to reveal the effect of human pinna shapes on HRTFs in the median plane, we investigate the relationship between human pinna shapes and the spectrum envelope of HRTFs. More specifically, we crafted the dummy pinna for the dummy head. Also, we investigated the spectrum envelope of HRTFs in various shape conditions of the dummy pinna in the median plane. As a result of investigations, we confirmed correspondence relationship between human pinna shapes and the spectrum envelope of HRTFs.

1pSPc21. Acoustic echo cancelation in discrete Fourier transform domain based on adaptive combination of adaptive filters. Luis A. Azpicueta-Ruiz, Anibal Figueiras-Vidal, and Jeronimo Arenas-Garcia (Dept. of Signal Theory and Commun., Universidad Carlos III de Madrid, Av Universidad 30, Leganes, Madrid 28911, Spain, azpicueta@tsc.uc3m.es)

Acoustic echo cancelers (AECs) are vital to many of communication systems, including hands-free telephone and videoconference, among others. Recently, adaptive combination of adaptive filters has been presented

as an easy but effective method to improve its performance, alleviating different compromises inherent to adaptive filters (responsible to identify the room impulse response). The most important tradeoff is related with the selection of the step size, which involves speed of convergence and residual echo. However, AECs require long adaptive filters, forcing to employ frequency-domain techniques to reduce the computational load and to accelerate the algorithms convergence when colored inputs—such speech—are presents. In this paper, we present an AEC based on combination of filters in discrete Fourier transform domain. Considering that both the input signal and the cancelation scenario make the behavior of adaptive filters is frequency dependent, our proposal exploits the combination capabilities employing different mixing parameters to separately combine independent spectral regions of two frequency-domain adaptive filters with different step sizes. In this way, our scheme outperforms recent algorithms where only a single combining parameter mixes the overall outputs of two frequency-domain adaptive filters. These advantages are illustrated by means of realistic experiments.

1pSPc22. Adaptive active control of free space acoustic noise. Iman Tabatabaei Ardekani and Waleed H. Abdulla (Elec. and Comput. Eng., The Univ. of Auckland, Private Bag 92019, Auckland CBD, Auckland, Auckland 1142, New Zealand, i.ardekani@auckland.ac.nz)

This paper concerns adaptive active control of acoustic noise in free space. Traditional adaptive active noise control algorithms are efficient in acoustic ducts; however, they are very unstable and sensitive when being used in free space. An efficient adaptive active noise control algorithm for free space noise is derived based on a root locus analysis on the adaptation process performed in adaptive active noise control. The traditional algorithm and the proposed algorithm are fully implemented by using a high performance embedded controller. The controller is then used for active control of acoustic noise in a duct and, also, in free space. Different experiments show that the traditional active noise control algorithm is not stable when the setup is used in free space. However, the proposed algorithm is stable and converges at a high convergence rate until reaching steady state conditions.

1pSPc23. A detection of danger sounds based on variable-state hidden Markov models. Asako Okamoto, Kohei Hayashida, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Science and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0009sv@ed.ritsumei.ac.jp)

To detect hazardous situations with danger sounds, the acoustic surveillance system is an ideal candidate. The conventional systems recognize environmental sounds with hidden Markov model (HMM) in order to detect danger sounds. It is however difficult to accurately recognize the environmental sounds, because the optimum HMM parameters for environmental sounds have not been identified. It is important factor for accurately recognizing them to ideally determine the number of states, one of the HMM parameter. On the other hand, environmental sounds which include danger sounds have a wider characteristic as the structure, the complexity, the length, etc. The variable states should be therefore an optimum HMM structure to detect the danger sounds. We thus propose the danger sound detection based on variable-state HMMs corresponding to a number of inflection points with the delta power of environmental sounds. We first investigate the recognition performance of environmental sounds including danger sounds with various states of HMM. We then investigate the relationship between the recognition performance and a number of inflection points with the delta power of various environmental sounds. As a result of evaluation experiments, we designed an optimum variable-state HMM for environmental sounds and confirmed the effectiveness of the proposed method.

1pSPc24. Sound source measurement of magnetic resonance imaging driving sound for feedforward active noise control system. Shohei Nakayama, Kenji Muto (Elec. Eng. and Comput. Sci., Shibaura Inst. of Technol., 3-7-5, toyosu, kouto-ku, Tokyo 135-8548, Japan, ma12077@shibaura-it.ac.jp), Kazuo Yagi (Dept. of Radiol. Sci., Tokyo Metropolitan Univ., Tokyo, Japan), and Guoyue Chen (Dept. of Electron. and Information Systems, Akita Prefectural Univ., Akita, Japan)

We proposed the active noise control (ANC) system reduce the loud MRI sound. It was important for performance improvement of the system. Therefore, we estimated the sound source of MRI driving sound. The position of the sound source of MRI driving sound was between the center and the edge in the gantry of MRI equipment. MRI equipment is important for the medical inspection, which gets the tomography of the body without x-ray. The patient of the MRI inspection needs to use the ear protector because the MRI equipment generated the loud sound, which the sound pressure level was around 100 dB. Here, our study was to make good acoustical environment using the ANC system for the MRI patient. The ANC system used the feedforward type because the MRI driving sound have unsteady pulsed sound. We made the ANC system using non-magnetic devices, ear protectors, and optical microphones. Because the MRI room had high magnetic environment. We measured the sound source of MRI driving sound to set the reference microphone. In this case, we showed the reduction effect of the ANC system of the sound by the computer simulation. As a result, the system reduced the MRI driving sound by around 50 dB.

1pSPc25. Parametric loudspeaker for speech signal based on the combination of amplitude and frequency modulations. Toru Iwasaki, Daisuke Ikefuji, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0005ri@ed.ritsumei.ac.jp)

A parametric loudspeaker has been used for audio guidance to a specific area because it has a sharper directivity compared with the conventional electrodynamic loudspeakers. The parametric loudspeaker emits an ultrasound as a carrier wave, which is modulated with an audio signal and has large-amplitude. An audible sound is reproduced by the modulated ultrasound with large-amplitude distorted by the nonlinearity on the air. The conventional modulations have been proposed as the amplitude modulation and the frequency modulation. In the sound quality, the amplitude modulation is superior to the frequency modulation. However, in the sound pressure level, the frequency modulation is superior to the amplitude modulation. In the present paper, we especially focus on that the parametric loudspeaker will emit the speech signals for the audio guidance. Therefore, we propose new modulation method based on the combination of amplitude and frequency modulations, which are specialized for speech signals. More specifically, we apply the amplitude and frequency modulations to the divided frequency bands of speech signals, respectively. In order to confirm the effectiveness of the proposed method, we carried out evaluation experiments. As a result of evaluation experiments, we confirmed the effectiveness of the proposed method compared with conventional methods.

1pSPc26. Chain architecture: An efficient hardware solution for a large microphone array system. Dmitry N. Zotkin (Inst. for Adv. Comput. Studies (UMIACS), Univ. of Maryland, College Park, MD), Ross Adelman (Dept. of Comput. Sci., Univ. of Maryland, College Park, MD), Adam E. O'Donovan (VisiSonics Corp., College Park, MD), and Ramani Duraiswami (Inst. for Adv. Comput. Studies (UMIACS), Univ. of Maryland, Dept. of Comput. Sci., Univ. of Maryland, College Park, MD 20770, ramani@cs.umd.edu)

A typical microphone array system consists of a number of microphones connected to the digitization hardware and central processing unit in a parallel fashion. Such radial, hub-and-spoke architecture has multiple points of failure, suffers from electromagnetic interference, and does not scale well. In this paper, an alternative, chain-like architecture is described. In such setup, the microphones in a system are organized in a single chain. Each individual microphone board has an ADC chip and is connected to the previous and to the next microphones in the chain with short multi-wire cables carrying digital signals. A buffer board at the end of the chain converts the digital data stream into the industry-standard USB 2.0 format. In this way, the individual microphone board becomes the building block for quick and easy arbitrary-configuration microphone array assembly with minimal amount of wiring involved. A hardware implementation of the chain architecture was developed and is described. Accompanying drivers and software allow the user to perform on-the-fly data acquisition and processing in C and in MATLAB. As an example, a 64-microphone array was built, and several source localization and beamforming algorithms were implemented in MATLAB. Experimental results using the data gathered from the array are presented.

1pSPc27. A design of audio spot based on separating emission of the carrier and sideband waves. Tadashi Matsui, Daisuke Ikefuji, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is0039fx@ed.ritsumei.ac.jp)

Parametric loudspeaker, which utilizes an ultrasonic of non-linear interaction, is developed to achieve audio spot. The parametric loudspeaker has sharper directivity, but reflections and intercepts by emitted sounds become severe problems. This is because reflections and intercepts lead to an invasion of privacy, and become noise to other listeners except a target listener. Principle of the parametric loudspeaker can formulate as non-linear interaction of carrier and sideband waves in emitted ultrasonic sounds on air. This suggests that we can design audio spot by individually emitting the carrier and sideband waves. In the present paper, therefore, we propose the design method of audio spot with the separating emission of the carrier and sideband waves. More specifically, the audible sound is demodulated at an area where the carrier and sideband waves individually emitted from each parametric loudspeaker are overlapped. We carried out evaluation experiments to measure sound pressure level (SPL) of demodulated audible sound. In addition, we evaluated the speech articulation of the demodulated audible sound with the proposed method. As a result of evaluation experiments, we confirmed the effectiveness of the proposed method.

MONDAY AFTERNOON, 3 JUNE 2013 — 511AD, 1:00 P.M. TO 4:40 P.M.

Session 1pUW

Underwater Acoustics: Seabed Scattering: Measurements and Mechanisms II

Charles W. Holland, Cochair
Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16801

Gavin Steininger, Cochair
School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. (Ring Rd.), Victoria, BC V8P 5C2, Canada

Dale D. Ellis, Cochair
DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada

Contributed Papers

1:00

1pUW1. A study of the reflection coefficients and backscattering effects of one-dimensional rough poroelastic surfaces using the finite element method. Anthony L. Bonomo, Marcia J. Isakson, and Nicholas Chotiros (Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

Acoustic reflection and scattering effects of one-dimensional rough poroelastic surfaces are studied using the finite element method. The poroelastic sediment layer is modeled following the classical work of Biot as extended by Stoll, which assumes that two attenuating compressional waves and one attenuating shear wave propagate in the sediment. The rough surfaces are generated using power-law type spectra and the incident wave used is a Gaussian tapered plane wave. This work seeks to assess how the reflection coefficients and backscattering effects of a poroelastic bottom vary as a function of frequency, roughness, and sediment type. Special consideration is given to the mesh required to accurately resolve the effects of the slow compressional and shear waves, which often have wave speeds slower than the fast compressional wave by an order of magnitude or more. [Work sponsored by the Office of Naval Research, Ocean Acoustics.]

1:20

1pUW2. High frequency backscattering from sandy sediments: single or multiple scattering. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Lab., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, chotiros@arlut.utexas.edu)

As the sand grain size approaches the acoustic wavelength, the underwater backscattering strength increases rapidly. Laboratory measurements indicate that the shallow-grazing angle backscattering strength increases as the third power of the normalized grain diameter. In this regime, it has been shown that the attenuation of the sound in the sand increases as the fourth power of frequency and the speed of sound decreases with increasing frequency. The most likely explanation for the attenuation and speed dispersion is multiple scattering [Schwartz and Plona, J. Appl. Phys. **55** (1984) and Kimura, J. Acoust. Soc. Am. Express Lett. **129** (2011)]. Single and multiple scattering theories will be applied to the backscattering problem with the purpose of determining if it is a single or multiple scattering process. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

1:40

1pUW3. The impact of finite ensonified area on the scattering cross section. Derek R. Olson (Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, dro131@psu.edu) and Anthony P. Lyons (Appl. Res. Lab., The Penn State Univ., University Park, PA)

The definition of the scattering cross section for the pressure field scattered by a rough interface uses the underlying assumption that the ensonified area does not affect its shape. This assumption holds so long as the incident field is a good approximation of a plane wave, which situation occurs when the ensonified area is large compared to a wavelength. In the opposite situation, when the ensonified dimensions approaches a wavelength, the incident plane wave assumption does not hold and the cross section can depart from modeled behavior. This research uses the perturbation approximation to derive a model for the low grazing-angle behavior of the scattering cross section. The primary results are (1) the appearance of an additive Lambertian term and (2) a separation of scales imposed by the acoustic resolution. This separation can then be used as a criterion for the application of the composite roughness model. Model results are checked against direct numerical solution of the governing integral equations. Implications for inversion of seafloor parameters based on acoustic scattering experiments is discussed.

2:00

1pUW4. In situ calibration of seafloor acoustic backscatter data from swath mapping sonars. Christian de Moustier (HLS Res., Inc., 3366 N. Torrey Pines Court, Ste. 310, La Jolla, CA 92037, cpm@hlsresearch.com) and Barbara J. Kraft (HLS Research, Inc., Barrington, New Hampshire)

Geometric and radiometric corrections are necessary to convert raw seafloor acoustic backscatter data to imagery in which the intensity variations are due to the geoacoustical properties of the bottom and the angular dependence of the backscattering process. Radiometric corrections include removal of the combined effects of the sonar's transmit and receive beam patterns, and removal of the area ensonified. To this end, a method is presented to estimate the transmit/receive beam pattern of a multibeam swath mapping sonar using seafloor acoustic backscatter data collected over an entire survey area. This involves estimating the scattering area at each sounding based on local seafloor slopes. For a given sonar installation, results show that such beam pattern estimates remain stable to within $\pm$ 0.5 dB over different survey areas.

2:20

1pUW5. Non-invasive characterization of fluid mud from scalar and vector noise fields due to a small boat. Jean-Pierre Hermand and Qunyan Ren (Université libre de Bruxelles (U.L.B.), av. F.D. Roosevelt 50 - CP 165/57, B-1050 Brussels, Université libre de Bruxelles (U.L.B.), Brussels 1050, Belgium, jhermand@ulb.ac.be)

The passive geoacoustic characterization of sediment using the ratio between pressure and vector field that are measured by an easily deployable system is discussed. The ratio is very sensitive to environmental properties but independent of unknown source spectrogram, e.g., boat noise that exhibits its complex spectral shape. Noise data sets due to different runs with a small boat were recorded on two closely adjacent hydrophones offshore the Amazon Rio mouth and processed by a nonlinear inversion scheme. Global optimization based on genetic algorithms provides the geoacoustic parameters of the fluid mud and underlying mud sediment, and the marginal posterior probabilities. Good consistency among the respective inversion results demonstrates the feasibility of the method under far from ideal conditions for environmental characterization, in term of unknown range, source ship navigation data, unknown source spectra, uncertain receiver depth, tilt, etc.

2:40

1pUW6. Resolution analysis of the inverse problem for geoacoustic experiment design. Andrew A. Ganse (Appl. Phys. Lab., Univ. of WA, 1013 NE 40th St., Seattle, WA 98105, aganse@apl.washington.edu)

This work explores effects of experiment geometry and array configuration on the resolving power of a continuous ocean bottom geoacoustic inverse problem in a shallow water environment. The uncertainty and resolution of this problem, in which ocean bottom P-wave velocities as a function of depth are estimated from noisy acoustic pressure waveforms received on vertical and horizontal line arrays in the water, can be investigated before the experiment is conducted, allowing one to improve or optimize the problem parameters to best configure an experiment during its planning phase. In this work, the resolution results of complete synthetic geoacoustic inversions at varying geometries and array configurations are compared with resolution results at various candidate seabottom profiles, initially using standard techniques of linearized inverse theory. Singular value decomposition is used to interpret the tie between geometry and regularization in the inverse problem, which directly affects the resolution. Then, additional comparisons and analysis address the nonlinearity of the problem, which causes a dependence of the resolution results on the bottom profile being solved for—which is unknown, and Monte Carlo analysis is used to show where the linearity approximation breaks down in the resolution results. [Work partially funded by ONR.]

3:00

1pUW7. Inversion of seabed acoustic parameters in shallow water using the warping transform. Juan Zeng (Inst. of Acoust., CAS, No.21 BeiSi-Huan XiLu, Bei Jing, China, Bei Jing 100190, China, zengjuan01@yahoo.com.cn), N.Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Li Ma, and Yan Chen (Inst. of Acoust., CAS, Bei Jing, China)

In this paper, a method is described for inverting geoacoustic parameters of the seabed from short range field data recorded by single hydrophone. The original data in time domain are processed by a warping operator at first, and then, the dispersion curve and the mode amplitude ratios are extracted separately from the warped data. The velocity and the density in the bottom are inverted from the dispersion curve, and the attenuation from the mode amplitude ratios, respectively. The performance of the method is examined using simulated data and then experimental data from the North Sea of China. The source used in the experiment was a small explosive charge that provided good signal to noise ratio over the frequency band from 200 Hz to 1 kHz. The depth of the water was about 30 m, and the water sound speed was nearly constant with depth. The seabed geoacoustic parameters are inverted from the data received at different ranges from 2 to 14 km. The results from the different ranges are consistent with a simple half space model of the bottom. The seabed velocity is about 1600 m/s.

3:20

1pUW8. Determination of grain size distribution in water-saturated granular medium using p-wave attenuation dispersion. Haesang Yang, Keunhwa Lee, and Woojae Seong (Dept. of Ocean Eng., Seoul National Univ., Bd. 34, Rm. 306, Underwater Acoust. Lab.,1, Gwanak-ro, Gwanak-gu, Seoul 151-744, South Korea, coupon3@snu.ac.kr)

P-wave attenuation in the water-saturated granular medium depends on both the frequency and the grain size. In this study, the use of the attenuation dispersion for the determination of grain size distribution in the water-saturated granular medium is discussed. For the dense granular medium, mathematical model considering multiple scattering is used for regression algorithm by fitting model predictions to the measured attenuation data. Inversion of grain size distribution is carried out numerically, and the results are discussed and compared to measured data for the water-saturated glass beads with unimodal and bimodal distributions.

Invited Paper

3:40

1pUW9. Rough interface acoustic scattering from layered sediments using finite elements. Marcia J. Isakson and Nicholas Chotiros (Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Quantifying acoustic scattering from rough interfaces is important for reverberation modeling, acoustic sediment characterization, and propagation modeling. Most models of interface scattering on layered surfaces rely on approximations to the Helmholtz/Kirchhoff integral. These models generally make such assumptions as neglecting the local angle for reflection and disregarding multiple scattering between rough layers. In this study, a mixed boundary element/finite element model is used to calculate rough interface scattering from sediment layers including elastic solids. The finite element method, based on the Helmholtz equation, is exact within the limits of the discretization density; reflections are calculated locally and all orders of scattering among layers are included. Using this model, bottom loss and backscattering predictions will be calculated for several cases. These predictions will be compared with more traditional scattering models and finally to a full finite element model of a point source reflection from the same sediment bottom. [Work supported by ONR, Ocean Acoustics.]

4:00

1pUW10. Geo-acoustic parameter estimation using a multistep inversion technique based on normal mode method. Lin Wan, Mohsen Badiey (College of Earth, Ocean,and Environment, Univ. of Delaware, 003 Robinson Hall, Newark, DE 19716, wan@udel.edu), and David Knobles (Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX)

The geo-acoustic parameters are of great importance in determining how the sea bottom affects sound propagation in the ocean. Many inversion techniques have been developed to estimate geo-acoustic parameters. One-step inversion algorithms using a cost function defined only by energy loss may not result in a unique solution of geo-acoustic inversion problem because of the correlation between seabed sound speed and attenuation. The present paper utilizes different characteristics of normal modes, including modal dispersion, modal attenuation, and modal based spatial coherence, to define appropriate cost functions in a multistep inversion algorithm for geo-acoustic parameter estimation. This inversion scheme is applied to the long-range broadband acoustic data obtained from L-shaped arrays in the Shallow Water 2006 experiment. The seabed sound speed and attenuation are estimated by minimizing the cost function at each step. The results show a non-linear frequency dependence of attenuation, which is similar to the seabed attenuation derived from measured time series and transmission loss data at the same experimental site [Knobles *et al*., J. Acoust. Soc. Am. **124**, 2008]. The uncertainties caused by the range dependent water column variability and bathymetry are discussed. [Work supported by ONR322OA.]

4:20

1pUW11. Electrochemical basis of the card-house model of mud and its acoustical implications. Joseph O. Fayton (Mathematics, Rensselaer Polytechnic Inst., Amos Eaton, 110 Eighth St., Troy, NY 12180, faytoj@rpi.edu), Allan D. Pierce (Mech. Eng., Boston Univ., Boston, MA), and William L. Siegmann (Mathematics, Rensselaer Polytechnic Inst., Troy, NY)

A basic mud model contains thin mineral (kaolinite and smectite) particles, roughly hexagonally shaped platelets, with diameters typically 1 micron. Isomorphous substitution causes each platelet to carry a net negative charge per unit area. Because the ions in the surrounding water respond so that there is a net positive charge on both sides of the platelet, each platelet is modeled as a sheet of longitudinal electric quadrupoles aligned perpendicular to the surface. The electrical interaction between platelets is responsible for the card-house structure, whereby the edge of one platelet touches a central line along the surface of another platelet, with the platelets being at right angles to each other. When the perpendicular arrangement is perturbed, a restoring torque attempts to return the platelets to their original state. Electrostatic analysis is used to explain why the restoring torque is formally singular at the joining line when one platelet is slightly tilted from perpendicular. The singular behavior appears to arise when the corners of one platelet touch the edge of another. This singularity requires imposition of the cantilever boundary condition in order to consider the shear resistance of mud, with each platelet bending as an elastic plate. [Research supported by SMART Fellowship and ONR.]

TUESDAY MORNING, 4 JUNE 2013 | 517D, 7:55 A.M. TO 8:40 A.M.

Session 2aID

Interdisciplinary: Plenary Lecture: Basics and Applications of Psychoacoustics

Sonoko Kuwano, Chair
Osaka Univ., 2-24-1-1107 Shinsenri-Nishimachi, Toyonaka, Osaka 560-0083, Japan

Chair's Introduction—7:55

Invited Paper

8:00

2aID1. Basics and applications of psychoacoustics. Hugo Fastl (AG Technische Akustik, MMK, TU München, Arcisstr.21, München 80333, Germany, fastl@mmk.ei.tum.de)

The field of psychoacoustics studies relations between acoustic stimuli, defined in the physical domain, and the hearing sensations elicited by these stimuli. The plenary lecture will address questions of stimulus generation and presentation. Both traditional methods and more recent methods like wave field synthesis will be touched. Concerning hearing sensations, basic magnitudes as for example absolute thresholds or loudness, but also advanced topics like pitch strength will be covered. Applications of psychoacoustics will include rather different fields such as music, noise evaluation or audiology, and address also cognitive effects as well as audio-visual interactions.

TUESDAY MORNING, 4 JUNE 2013 | 513ABC, 8:55 A.M. TO 12:00 NOON

Session 2aAAa

Architectural Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Adapting, Enhancing, and Fictionalizing Room Acoustics I

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362

Alex Case, Cochair
Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854

Chair's Introduction—8:55

Invited Papers

9:00

2aAAa1. Electronically variable room acoustics—Motivations and challenges. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Steve Ellison (Meyer Sound, Sierra Madre, CA)

Electronically variable room acoustics, or "active acoustics," has become an effective solution to a variety of room acoustics challenges. The motivations for considering such systems often go beyond acoustics. Energy conservation, financial considerations, historic preservation, and balancing the needs of a venue's constituencies all can play a role in the determination to employ active acoustics. This paper will discuss some practical examples, including planned renovations to the Santa Monica Civic Auditorium, home of the Academy Awards during the 1960s, a resident orchestra, legendary rock concerts, and a unique hydraulic floor to convert the Civic from a performance space to an exhibit space. The active acoustic system objectives, design strategies, and challenges will be discussed.

9:20

2aAAa2. Sound system in a small college auditorium. Sergio Beristain (IMA, ESIME, IPN, P.O. Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

A new 220 seats college auditorium needed an electro acoustics system in order to adequately perform all its normal activities, which included live voices for lectures and conferences, a sound reproduction system working alone or together with the projection equipment, sometimes background music, and eventually some small groups with live music presentations as requested in the specs of the auditorium usage; it also needed recording capabilities for the most important lectures and presentations. A 12/8 channel with stereo output system with peripherals was chosen for the installation, where three microphones were reserved for the front table, six distributed in fixed stands located in the aisles of the audience area for questions and dialog, and the other three were movable. Background noise was not an issue because the auditorium is located in a tree full area within the university campus, away from busy streets. Budget for the acoustical conditioning and the electronic equipment was very limited.

9:40

2aAAa3. Active acoustics and sound reinforcement at TUI Operettenhaus, Hamburg: A case study. Roger W. Schwenke (Res. and Development, Meyer Sound Lab., 2832 San Pablo Ave., Berkeley, CA 94702, rogers@meyersound.com)

TUI Operettenhaus is a proscenium theater with one balcony, which is host to drama, musical theater, and concerts. The venue hosts different sound reinforcement systems for different shows, and now has a permanent active acoustic system. The physical acoustics are very dry as is appropriate for modern theater with spatial sound reinforcement, and the active acoustic system allows the reverberation time to be extended as appropriate for different performances. The active acoustic system can also pass through signals to its speakers for spatial surround reproduction. The installation of the active acoustic system in an older building posed many challenges. This case study presents the challenges that were overcome during installation, the integration of the active acoustic system with sound reproduction, and the measured performance of the system.

10:00

2aAAa4. The raw and the cooked in architectural acoustics. William L. Martens and Dagmar Reinhardt (Faculty of Architecture, Design and Planning, Univ. of Sydney, 148 City Rd., Wilkinson Bldg. G04, NSW 2006, Australia, william.martens@sydney.edu.au)

Whereas the “raw” experience of live sound events is often quite far removed from the “cooked” auditory imagery that is presented when live acoustical events are amplified by a sound reinforcement system, there are many audio signal processing tools that can be applied in the attempt to simulate the more natural auditory characteristics of live (unplugged?) musical performances. This paper builds a discussion of perceptual results of modern sound reinforcement technology based upon the Lévi-Strauss notion regarding what modern culture does to the “raw” to make it “cooked”). A key concept in evaluating the quality of a sound reinforcement system is that of the standard of reference against which the perceptual results can be compared. As there is no shared opinion nor well-established optimal acoustical character for a space upon which some consensus could be built, the question presents itself again and again. This paper will address related issues of reference, preference, and adequacy in sound reinforcement.

10:20

2aAAa5. Adapting spaciousness of artificial, enveloping reverberation in multichannel rendering based on coded sequences. Ning Xiang, Jiseong Oh, Jonas Braasch (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Greene Building, 110 8th Street, Troy, NY 12180, xiangn@rpi.edu), and Bosun Xie (School of Sci., South China Univ. of Technol., Guangzhou, China)

For virtual room environments, adapting realistic reverberation and enhancing reverberation are critical for producing a convincing immersive experience. Also, in perceptual studies of room-acoustics using virtual room environments, using the appropriate enveloping reverberance to correlate perceived room size to the virtual space is a challenging task. This research applies to both binaural rendering and a multi-channel loudspeaker reproduction that can be employed in simulating such an environment. Approaches to adapting and enhancing spaciousness within the context of artificially generated reverberation are investigated via psychoacoustics tests. The pseudo-random properties of coded signals based on reciprocal maximum-length sequences allow for a deterministic, controllable decorrelation between all reverberation channels. For this challenging task, shapes of both sound energy decay and spatial profiles have been found to be decisive for creating successful immersive environments. This paper will discuss potential values for fundamental research in room-acoustics and for educational purposes in seeking a broadened understanding of perceived spaciousness and reverberance in varying contexts.

10:40

2aAAa6. E-Venue—Affordable electronic acoustic enhancement for small venues. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

The advent of modern digital signal processing made altering the acoustical conditions in a venue using electro-acoustic tools practical. Such systems have been in constant use for many years in concert halls, opera houses performance spaces, houses of worship, and a variety of other spaces and applications throughout the world. However, the cost associated with specialized nature of these systems has put them out of the reach of many small venues that stand to benefit most from use of this technology. This paper describes a new low cost, integrated, electro-acoustic system designed specifically for use in small venues—including but not limited to; performance venues, recital halls, rehearsal spaces, houses of worship, etc.

11:00

2aAAa7. Interaction between critical listening environment acoustics and listener reverberation preference. Brett Leonard, Richard King (The Grad. Program in Sound Recording, The Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, brett.leonard@mcgill.ca), and Grzegorz Sikora (Bang & Olufsen Deutchland GmbH, Pullach, Germany)

Reverberation is a central effect in many modern music productions. In the case of classical music, it may even be the only effect used. There is, however, minimal literature concerning the interaction between reverberation preference and the listening environment used during critical mixing tasks. In order to explore this critical interaction, a group of highly trained subjects are tasked with adding reverberation to dry, premixed stereo program material in two different acoustic environments: a recording studio control room and a highly reflective room. The control room is representative of most studios, with an RT of approximately 200 ms. The reflective environment more closely approximates an untreated residential room, with an RT of over 350 ms, with a marked increase in lateral energy. Somewhat predictably, the mean preferred reverberation level is higher in a less reverberant environment, but the distributions of reverberation level preference are shown to be narrower for the more reflective mixing environment. The time it takes for subjects to reach a decision is similar in both environments, but the reflective environment shows a longer period of adaptation at the beginning of each trial set.

11:20

2aAAa8. Adapting, enhancing, and fictionalizing room acoustics. Ted Ohl (acouStaCorp, 701 E. 132 St., Bronx, NY 10454, tedohl@pdoinc.com) and Niels Adelman-Larsen (FlexAcoustics, Kgs Lyngby, Denmark)

The need for adjustable acoustics extends far beyond the Performing Arts Center or world class concert hall. The small scale of the existing market for adjustable acoustics impedes the development of products economical enough to expand the market. Expand the market by: Raising awareness of the options available for adjusting the acoustics in their room will expand the market. User and designer feedback regarding desirable features also will contribute. Broadening the concept of variable acoustics to include noise control creates opportunities in the industrial sector. Focusing on cost saving efficiencies in design and production of products making the cost of adjustable acoustic solutions accessible to more clients; developing products suitable for retrofit as well as new construction Product Development: identify critical and desirable features in collaboration with users and designers. Develop product performance data to enable acousticians to incorporate variable acoustics into their designs and predict performance for clients. Incorporate other design team members' input to anticipate conflicts such as space use, look, cost, and other systems. Available Products Available product types will be quickly reviewed. *In-situ* test data from existing projects will be presented. More innovation to reach a broader market will be encouraged

11:40

2aAAa9. Electric guitar—A blank canvas for timbre and tone. Alex Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz), Agnieszka Roginska, Justin Matthew, and Jim Anderson (New York Univ., New York, NY)

The electric guitar is a complex mechanical, electrical, and acoustic system, invented less than a century ago. While more traditional instruments such as voices and violins, trumpets and tympani, piano and piccolo might possess innate traits that most listeners easily identify, the electric guitar is a sound synthesizer capable of a vast range of sounds. The guitar, the amp, and the recording techniques used enable the performer and the engineer to define and refine elements of tone, almost without limit. Electric guitar has no single reference tone quality, but instead invites, and even inspires performers and recordists to create new sounds and explore alternative timbres as desired.

TUESDAY MORNING, 4 JUNE 2013 — 513DEF, 8:55 A.M. TO 12:00 NOON

Session 2aAAb

Architectural Acoustics, Noise, and Physical Acoustics: New Materials for Architectural Acoustics

Matthew V. Golden, Chair
Scantek, 6430c Dobbin Rd., Columbia, MD 21045

Chair's Introduction—8:55

Invited Papers

9:00

2aAAb1. National Gypsum Company's acoustically enhanced gypsum board—SoundBreak XP. Stephen A. Cusa (Sales/Marketing, National Gypsum Co., 2001 Rexford Rd., Charlotte, NC 28211, stevec@nationalgypsum.com)

Gold Bond® BRAND SoundBreak XP Gypsum Board is an acoustically enhanced gypsum board used in the construction of high STC wall assemblies. This innovative gypsum board allows for construction of high STC wall assemblies that are thinner, cost effective, and more reliable than traditional methods. In addition to achieving a higher performance wall relative to acoustics, SoundBreak XP helps to achieve a higher performance wall in general. SoundBreak XP comes standard with mold and mildew resistance and is manufactured with an abrasion resistant paper. Both the mold resistance and the abrasion resistance achieve the best possible scores relative to the standard ASTM test methods. SoundBreak XP installs and finishes like standard gypsum board.

9:20

2aAAb2. Increased scattering efficiency through cell optimization in acoustic diffusers. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

This presentation will provide test data, experiments, and analysis in showing the interaction of absorption and diffusion characteristics. It will be shown that absorption, either additive or design inherent, has an adverse effect on the efficiency of the acoustic diffusers. The effects of low frequency interference on various cell structures of quadratic residue diffusers will be highlighted as a prime example of this theory. The inclusion of additive absorption, i.e., fabric, fiberglass, and other materials, further complicates the matter of diffuser efficiency. While design characteristics of some diffusers, such as prime root and quadratic residue diffusers, can be used effectively for their absorption coefficients, overall these characteristics detract from the efficiency of the diffusion coefficients. Product designs utilizing optimized cell designs to inhibit absorption in diffusers will be shown. It will be shown that this cell optimization dramatically increases the efficiency of the diffuser. This optimization includes both higher scattering coefficients and smoother frequency responses. Comparative analysis will be provided with traditional designs and newer optimized designs.

9:40

2aAAb3. From felt to fungus: New materials and applications—Focus on innovation and exploration. Dawn Schuette and Scott Pfeiffer (Threshold Acoust, LLC, 53 W Jackson Blvd., Ste. 815, Chicago, IL IL, dschuette@thresholdacoustics.com)

A two-part presentation of new materials for use in architectural acoustics. This presentation emphasizes new materials, both commercially available and those pressed into use for acoustic benefit. The companion session is presented in "Cultivating the Sustainable in Architectural Acoustics." Contemporary architectural design often seeks to push the standard of construction, resulting in the need to explore new acoustic solutions to new architectural challenges. Innovative use of commercially available materials or exploration into the development of new materials or modifications of known materials is required to find the best solutions both acoustically and architecturally. Use of acoustical products and non-acoustical products for acoustical benefit are reviewed through case studies.

Contributed Papers

10:00

2aAAb4. Simulation of normal incidence sound absorption coefficients of perforated panels with/without glass wool by transmission line parameters in a two- port network. Takayoshi Nakai and Kota Yoshida (Dept. of Elec. & Electron. Eng., Faculty of Eng., Shizuoka Univ., 3-5-1 Johoku, Naka-ku, Hamamatsu 432-8561, Japan, tdtnaka@ipc.shizuoka.ac.jp)

This paper describes simulation of normal incidence sound absorption coefficients of perforated panels by ABCD-parameters in a two-port network. Maa and Sakagami have investigated micro perforated panels, MPP. But their theories can treat only near 1% perforation rates of perforated panels with back cavities. If sound propagates as a plane wave, sound propagation can be represented as ABCD-parameters in a two-port network. Perforated panels, back cavities, and glass wool absorption materials are represented as matrix of ABCD-parameters, respectively. ABCD-parameters of a perforated panel with a back cavity are calculated as multiplication of their matrices. An input impedance can be calculated from the calculated ABCD-parameters. A normal incident absorption coefficient is calculated from the input impedance. Holes of the perforated panels have losses of viscous friction and thermal conduction at their walls. Simulations are done in the condition of 0.25 to 5 mm diameters of holes, 0.25% to 25% perforation rates, 0.5 to 5 mm thickness of the perforated panels with back cavities in which there are or are not glass wool absorption materials. The results of these simulations are good agreements with the results of our measurements by transfer function method.

10:20–10:40 Break

10:40

2aAAb5. Sound absorption and transmission through flexible microperforated structures. Cédric Maury (Centre National de la Recherche Scientifique (CNRS), Equipe Sons - Laboratoire de Mécanique et d'Acoustique (UPR CNRS 7051), Laboratoire de Mécanique et d'Acoustique, 31, chemin Joseph Aiguier, Marseille cedex 20 13402, France, cedric.maury@centrale-marseille.fr), Teresa Bravo (Consejo Superior de Investigaciones Cientificas (CSIC), Centro de Acustica Aplicada y Evaluacion No Destructiva (CAEND), Madrid, Spain), and Cédric Pinhède (Centre National de la Recherche Scientifique (CNRS), Equipe Sons - Laboratoire de Mécanique et d'Acoustique (UPR CNRS 7051), Marseille, France)

This work presents a theoretical and experimental study on sound absorption and transmission through structures made up of single and multiple-layer micro-perforated panels (MPPs). As they contribute to improve acoustical comfort, speech intelligibility and comply with lightweight, transparency and fiberless requirements, increasing applications are found in architectural acoustics or in the aeronautic and surface transport industries. A fully coupled modal approach is proposed to calculate the absorption coefficient and the transmission loss of finite-sized layouts made up of multiple flexible MPPs separated by air gaps. Validation results are obtained for single and double-layer thin MPPs against the transfer matrix approach and against measurements performed in a standing wave tube and in an anechoic chamber. Analytical approximations are derived from coupled-mode analysis for the Helmholtz-type and structural resonance frequencies of a single layer MPP structure together with relationships on the air-frame relative velocity over the MPP surface at these resonances. Principled guidelines are provided for enhancing both the sound absorption and transmission properties of multiple-layer MPP structures through suitable setting of the design parameters.

11:00

2aAAb6. Analysis of sound absorption behavior of polyester fiber material faced with microperforated panels. Davide Borelli, Corrado Schenone, and Ilaria Pittaluga (DIME - Sez. TEC, Università degli Studi di Genova, Via all'Opera Pia 15/A, Genova, GE 16145, Italy, corrado.schenone@unige.it)

Perforated facings used in lined ducts or sound absorbing panels can have various purposes: protecting the porous sound absorbing material from dust or grazing flow, acting as a rigid support for the porous material, or also affecting the behavior of the "backing" material, modifying this way the acoustical performance of the porous layer. This paper describes the effect of perforated facings on sound absorption characteristics of samples made by polyester fiber, experimentally investigated in accordance with ASTM C384 04 standard by means of two Kundt's tubes with different diameters. The polyester (PET) fiber material had bulk density of 30 kg/m^3 and melting point at 260°C. The analysis was performed for a sample thickness equal to 100 mm. The samples were faced by means of different metal plates perforated with circular holes. The holes diameter was equal to 2 mm for all facings, while the percent open area was varied from 4.9% to 30%. The microperforated panels (MPPs) were positioned in adherence of the PET fiber material or at a distance of 2, 4, and 6 mm. The different behaviors due to the multiple combinations of percent open area and distance from the sample have been then analyzed and discussed.

11:20

2aAAb7. Structures of resonators in a cavity for improving a sound insulation of a thin double-leaf panel. Shinsuke Nakanishi (Faculty of Eng., Hiroshima Int.Univ., 5-1-1, Hiro-koshingai, Kure 737-0112, Japan, s-nakani@it.hirokoku-u.ac.jp)

The specific acoustic problem of a double-leaf panel is a less sound insulation caused by a mass-air-mass resonance. For improving the sound insulation, many studies have suggested Helmholtz resonators in the cavity, which are tuned at the resonant frequency. They have measured and analyzed this problem of double-walls spaced with 100 mm thickness of air gap. They have suggested that the resonators improve the sound insulation to the resonant transmission, and discussed its optimization for a gain by the resonators and structures set in the cavity. But it is unclear that those results can apply to sound insulation by a double grassing with 5 mm thickness of air gap, which is often seen even as a thermal insulated window, and whose air gap is quite thinner than that of the walls. Then, this study measured effects of various resonators in the cavity for improving the sound insulation of thin double-leaf panels, and discusses effects of structures and perforation ratio to the sound insulation. Moreover, for analyzing the effects of resonators, this study discusses measured results with theoretical studies of sound absorption models for resonators.

11:40

2aAAb8. Rectangular slot perforations to improve the sound absorption of walls. Umberto Berardi (DICAR, Politecnico di Bari, via Orabona 4, Bari 70125, Italy, u.berardi@poliba.it)

This paper focuses on theories to predict the sound absorption properties of a surface with rectangular perforations. First, the paper gives a synthetic review of available knowledge about theoretical and experimental works about rectangular slot perforations. A comparison of these models is also reported. Then, the adaptability of these models to 3-d structures of finite size, such as walls, is discussed. Later the paper presents a model based on the transfer matrix method, which takes into account surface, hole and cavity impedances. In particular, the surface impedance is obtained by the sum of a resistance term, a mass term, and characteristic impedance. The numerical model is hence adapted to the case of stone walls. Experimental tests have been performed in a reverberant chamber to measure the sound absorption of several configurations of slotted walls. The bricks of these walls have been mounted creating vertical slots of different dimensions, which, combined with different air gaps, allowed the wall to work as an assembly of Helmholtz resonant absorbers. Comparisons with theoretical models are carried out to explain the different mechanisms of sound absorption. Finally, this paper aims to establish practical roles to develop optimized noise control solutions through rectangular slot perforated walls.

TUESDAY MORNING, 4 JUNE 2013 — 510B, 9:00 A.M. TO 12:00 NOON

Session 2aAB

Animal Bioacoustics and Signal Processing in Acoustics: Conditioning, Segmentation, and Feature Extraction in Bioacoustic Signals

David K. Mellinger, Chair
Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Science Dr., Newport, OR 97365

Invited Papers

9:00

2aAB1. A supervised approach for segmentation of bioacoustics audio recordings. Forrest Briggs, Raviv Raich, and Xiaoli Fern (School of EECS, Oregon State University, 1148 Kelley Engineering Center, Corvallis, OR 97331-5501, raich@eecs.oregonstate.edu)

Segmentation is one of the most important tasks in preprocessing audio recordings for species recognition. For examples, bird songs or calls often consist of distinct short utterances. Correctly segmenting such utterances is an essential step in the analysis of bird songs or calls. Energy based time-domain segmentation is commonly used and can be fairly effective when dealing with high signal-to-noise ratio recordings of a single individual. We consider the scenario in which omnidirectional microphone are deployed for round-the-clock in-situ species monitoring. In such scenario, recordings may suffer from two problems: (i) low signal-to-noise ratio and (ii) simultaneous vocalizations of multiple individuals. We are interested in segmentation of such recordings. We propose a framework for supervised time-frequency segmentation of audio recordings. Using manually labeled spectrograms, a classifier is trained to separate vocalization from the background noise in the two dimensional time-frequency domain.

9:20

2aAB2. Joint classification of whistles and echolocation clicks from odontocetes. Yang Lu, David K. Mellinger, Holger Klinck (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Science Dr., Newport, OR 97365, lu.yang@noaa.gov), and Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., Newport, Oregon)

We propose to use acoustic features of both clicks and whistles to classify odontocete sounds to species. The species studied are Cuvier's beaked whales (*Ziphius cavirostris*), bottlenose dolphin (*Tursiops truncatus*), melon-headed whale (*Peponocephala electra*), and short- and long-beaked common dolphin (*Delphinus delphis* and *D. capensis*). An energy-based detector is used for echolocation click detection, and Roch's Silbido algorithm is used for whistle detection. Detected whistles are characterized by maximum and minimum frequencies, duration, slope, spectral maxima, spectral gaps, number and frequency of inflection points, number of "loop" repetitions, and other acoustic characteristics. Detected clicks are characterized by cepstral characteristics, as well as by a set of noise-resistant statistics. Clicks that occur within a certain time neighborhood of a whistle have the corresponding feature vectors merged to produce the input to the classification system. Random forest and Gaussian mixture model classifiers are tested on the resulting features and performance is characterized. [Funding from ONR.]

9:40

2aAB3. Sparse coding for scaled bioacoustics: From Humpback whale songs evolution to forest soundscape analyses. Herve Glotin (CNRS LSIS, Univ Sud Toulon, Inst. Univ. de France, USTV, avenue Université, BP20132, La Garde 83957, France, glotin@univ-tln.fr), Jérôme Sueur (MNHN, CNRS UMR, Paris, France), Thierry Artières (CNRS LIP6, UPMC, Paris, France), Olivier Adam (CNRS CNPS, Univ. Paris Sud, Paris, France), and Joseph Razik (CNRS LSIS, Univ Sud Toulon, Inst. Univ. de France, Paris, France)

The bioacoustic event indexing has to be scaled in space (oceans and large forests, multiple sensors), and in species number (thousand). We discuss why time-frequency featuring is inefficient compared to the sparse coding (SC) for soundscape analysis. SC is based on the principle that an optimal code should contain enough information to reconstruct the input near regions of high data density, and should not contain enough information to reconstruct inputs in regions of low data density. It has been shown that SC methods can be real-time. We illustrate with an application to humpack whale songs to determine stable components versus evolving ones across season and years. By sparsing at different time scale, the results show that the shortest humpack acoustic codes are the most stable (occurring with similar structure across two consecutive years). Another illustration is given on forest soundscape analysis, where we show that time-frequency atomes allow an easier analysis of forest sound organization, without initial classification of the events. These researches are developed within the interdisciplinary CNRS project "Scale Acoustic Biodiversity," with Univ. of Toulon, Paris Natural History Museum, and Paris 6, consisting into efficient processes for conditioning and representing relevant bioacoustic. Information, with examples at sabiod.univ-tln.fr.

10:00

2aAB4. Bat species identification from zero crossing and full spectrum echolocation calls using Hidden Markov Models, Fisher scores, unsupervised clustering and balanced winnow pairwise classifiers. Ian Agranat (Wildlife Acoust., Inc., 970 Sudbury Rd., Concord, MA 01742-4939, ian@wildlifeacoustics.com)

A new classification technique for the identification of bats to species from their echolocation calls is presented. Three different datasets are compiled and split in half for training and testing classifiers. Combined, the data include 9014 files (bat passes) with 226,432 candidate calls (pulses or extraneous noise) representing 22 different species of bats found in North America and the United Kingdom. Some files are of high quality consisting of hand-selected search phase calls of tagged free flying bats while others are from a variety of field conditions including both active (attended) and passive (unattended) recordings made with a variety of zero crossing and full spectrum recording equipment from multiple vendors. Average correct classification rates for the three datasets on test data are 100.0, 97.9, and 88.8%, respectively, with an average of 92.5, 72.2, and 39.9% of all files identified to species. Most importantly, classifiers in the third dataset for two species of U.S. endangered bats, *Myotis sodalis* (MYSO) and *Myotis grisescens* (MYGR) have a correct classification rate of 100 and 98.6%, respectively, and identify 67.4% and 93.8% of all files to species suggesting that the classifiers are well suited to the accurate detection of these endangered bats.

10:20–10:40 Break

10:40

2aAB5. Conditioning for marine bioacoustic signal detection and classification. David K. Mellinger (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Science Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu)

Marine acoustic signals are characterized by certain types of noise and interference. Conditioning methods applied to spectrograms can be used to reduce or even remove these sounds, making bioacoustic signals more evident and simplifying the tasks of detection and classification. One family of methods is for making a long-term estimate of noise at each frequency and subtracting this estimate from the spectrogram; this has the beneficial effects of whitening the noise spectrum and removing relatively stationary noise sources such as vessel sound, but has the detrimental effect that relative spectrum levels—important in echolocation click classification—are altered. Another method estimates the spectrum in narrow bands at each time step and subtracts this estimate from the corresponding spectrogram frame; this method is useful for tonal sound detection and classification in that it removes short-duration clicks from snapping shrimp and echolocating animals. Other methods for removing other, more rare types of noise are presented as well. Examples and performance characterization of these methods are presented. [Funding from ONR and N45.]

Contributed Papers

11:00

2aAB6. Robustness of perceptual features used for automatic aural classification to propagation effects. Carolyn M. Binder, Paul C. Hines, Sean P. Pecknold, and Jeff Scrutton (Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, carolyn.binder@drdc-rddc.gc.ca)

Previous effort has shown that a prototype aural classifier developed at Defence R&D Canada can be used to reduce false alarm rates and successfully discriminate cetacean vocalizations from several species. The aural classifier achieves accurate results by using perceptual signal features that model the features employed by the human auditory system. Current work focuses on determining the robustness of the perceptual features to propagation effects for two of the cetacean species studied previously—bowhead and humpback whales. To this end, classification results are compared for the original vocalizations to classification results obtained after the vocalizations were re-transmitted underwater over ranges of 2 to 10 km. Additional insight into the propagation effects is gained from transmission of synthetic bowhead and humpback vocalizations, designed to have features similar to the most important aural features for classification of bowhead and humpback vocalizations. Each perceptual feature is examined individually to determine its robustness to propagation effects compared to the other aural features. To gain further understanding of propagation effects on the features, preliminary propagation modeling results are presented in addition to experimental data.

11:20

2aAB7. Odontocete click train deinterleaving using a single hydrophone and rhythm analysis. Olivier Le Bot (STIC/AP, ENSTA Bretagne, 2, rue Francois Verny, Brest 29200, France, olivier.le_bot@ensta-bretagne.fr), Julien Bonnel (LabSTICC/TOM, ENSTA Bretagne, Brest, France), Jérôme I. Mars (Image and Signal, GIPSA-Lab, Saint Martin d'Hère, France), and Cédric Gervaise (Image and Signal, GIPSA-Lab, Saint Martin d'Hère, France)

Most odontocetes live in pods of several individuals, resulting in an overlapping of click trains recorded by passive acoustic monitoring systems. Localization algorithms and click classifiers are usually used for train separation. However, their performances fall down if individuals are too close to each other or if acoustical parameters vary greatly from click to click, respectively. Assuming odontocete clicks follow rhythmic patterns, we propose to use a rhythm analysis to separate mixed click trains from a single hydrophone. The proposed algorithm is based only on inter-click-intervals (ICI) to cluster clicks into trains. It uses information given by complex-valued autocorrelation to compute a histogram, which will exhibit peaks at ICIs corresponding to interleaved trains. By this technique, subharmonics corresponding to multiples of ICIs are automatically suppressed. The algorithm is then extended by a time-period analysis leading to a time-varying ICI spectrum. A threshold can be applied on this spectrum to detect the different interleaved trains. The final result is a binary time-ICI map on which trains can be fully and easily distinguished and extracted. We validate it on simulated and experimental data, and we show that the algorithm is particularly suitable as a preprocessing tool prior to localization and classification schemes.

11:40

2aAB8. Complexity index and proportional variability to study dolphin whistles. Carmen Bazúa Durán, E. Julieta Sarmiento Ponce (Facultad de Ciencias, UNAM, Circuito Exterior s/n, Ciudad Universitaria, México, D.F. 04510, Mexico, bazua@unam.mx), Brenda P. González Leal (Universidad del Mar, Oaxaca, Mexico), and Camila Rodríguez Bohorquez (Facultad de Ciencias, Universidad de los Andes, Bogotá, Colombia)

Dolphin whistles are emitted especially during social interactions and feeding activities involving group cohesion, individual recognition, and recruitment. This paper presents a new methodology to describe and compare the whistle repertoire of dolphins. It consists on first extracting the whistle contour using MATLAB BELUGA, then classifying whistles into whistle types using MATLAB ArtWARP, next classifying whistle types into four general categories (high complexity, low complexity, linear long, and linear short), and finally computing a complexity index and a proportional variability of the whistle repertoire. The method was tested with whistles from captive and wild bottlenose dolphins, *Tursiops truncatus*, and from wild Guyana dolphins, *Sotalia guianensis*. Results obtained showed that this very simple method is useful to describe the whistle repertoire and to compare it according to the general behavioral state of dolphins, and between species. It is necessary to implement new methodologies like this one to better understand how dolphins are using whistles, since acoustic communication is the most important sense in dolphin species. This is specially important in areas where dolphins are exposed to humans, and where underwater visibility is limited, like Laguna de Términos, a Marine Protected Area in Mexico. [Work supported by PAPIIT-UNAM.]

TUESDAY MORNING, 4 JUNE 2013 — 510D, 8:55 A.M. TO 12:00 NOON

Session 2aAO

Acoustical Oceanography: Seismic Oceanography

Warren Wood, Cochair
Geology and Geophysics, Naval Res. Lab., 1005 Balch Blvd., Stennis Space Ctr., MS 39529-5004

Berta Biescas Gorriz, Cochair
Oceanography, Dalhousie Univ., 1355 Oxford St., Halifax, NS B3H3k4, Canada

Chair's Introduction—8:55

Contributed Papers

9:00

2aAO1. Uncertainty of transmission loss due to small scale fluctuations of sound speed in two environments. Josette P. Fabre (Acoustics, Naval Res. Lab., 1005 Balch Blvd., Stennis Space Ctr., MS 39529, josie.fabre@nrlssc.navy.mil) and Warren Wood (Geology and Geophysics, Naval Res. Lab., Stennis Space Ctr., MS)

Seismic oceanography techniques reveal detection of small scale variations in sound speed not detectable via conventional oceanographic means, i.e., frequent XBT or CTD casts). Due to computational and practical limitations, such small scale spatial and temporal detail that exists in a real ocean environment is not typically included in acoustic ocean models. However, such measurements can provide insight to the small scale variability (uncertainty) that exists in the ocean but is not predicted by mesoscale ocean models. We show acoustic predictions made with the Range Dependent Acoustic Model (RAM) using measured seismic oceanography and CTD data at two locations in significantly different environments. Additionally, the CTD measurements are smoothed to a resolution comparable to that provided by a dynamic ocean model and acoustic predictions are computed. The Uncertainty Band (UBAND) algorithm (UBAND) [Zingarelli,"A mode-based technique for estimating uncertainty in range-averaged transmission loss results from underwater acoustic calculations," J. Acoust. Soc. Am. **124**(4) (2008)] is applied to the smoothed oceanographic data using estimates of sound speed uncertainty calculated from the high resolution measurements. We find reasonable estimates of uncertainty due to the small scale oceanography that is not characterized by mesoscale ocean models.

9:20

2aAO2. Inversion of density in the ocean from seismic reflection data. Berta Biescas Gorriz, Barry Ruddick (Oceanography, Dalhousie Univ., 1355 Oxford St., Halifax, NS B3H3k4, Canada, berta.biescas@dal.ca), and Valenti Sallares (Marine Geology, Marine Science Inst. - CSIC, Barcelona, Barcelona, Spain)

Vertical stability of the fluid particles, mixing, and mesoscale motions in the ocean interior occur mostly along-isopycnals surfaces. Therefore, potential density profiles with high lateral resolution would provide important information about the fluid dynamic and the general circulation in the ocean.

Could we observe density changes from seismic data? Is seismic oceanography able to measure density with enough accuracy? How is the relation between seismic reflectors and isopycnals surfaces? We have inverted oceanic impedance from seismic data and then derived density and potential density surfaces from the oceanic impedance. Results of the inverted potential density have been compared with digitized seismic reflectors to show the relation between isopycnals and reflectors. We have also compare the seismic profiles of the GO Survey with the space-coincident CTDs and space and time-coincident XBTs to understand the nature of the reflectivity and its relation with the physical parameters of the ocean.

9:40

2aAO3. Seismic oceanography imaging of thermal intrusions in strong frontal regions. Jeffrey W. Book, Warren T. Wood (Naval Res. Lab., Oceanogr. Div., Stennis Space Ctr., MS 39529, jeff.book@nrlssc.navy.mil), Ana E. Rice (National Res. Council, Stennis Space Ctr., MS), Sandro Carniel (C.N.R - Inst. of Marine Sci., Venezia, Italy), Richard Hobbs (Univ. of Durham, Durham, United Kingdom), Isabelle Ansorge (Univ. of Cape Town, Cape Town, South Africa), Tim Fischer (GEOMAR Helmholtz Cte. for Ocean Res., Kiel, Germany), and Hartmut Prandke (ISW Wassermesstechnik, Fünfseen, Germany)

The Naval Research Laboratory and collaborating partners carried out two dedicated seismic oceanography field experiments in two very different strong frontal regions. Adriaseismic took seismic oceanography measurements at the confluence of North Adriatic Dense Water advected along the Western Adriatic Current and Modified Levantine Intermediate Water advected around the topographic rim of the Southern Adriatic basin. ARC12 took seismic oceanography measurements in and around the Agulhas Return Current as it curved northward past the Agulhas Plateau and interacted with a large anticyclone that collided with the current. Despite one study focused on coastal boundary currents and the other focused on a major Western Boundary Current extension, the complex horizontal structures seen through seismic imaging are tied to the processes of thermal intrusions and interleaving in both systems. Seismic Oceanography provides a unique capability of tracking the fine-scale horizontal extent of these intrusions. In both systems they occur primarily along isopycnals and are largely density compensating. The formation of these structures is associated with advective processes rather than diffusive processes, despite gradients favorable for double diffusion mixing. Results from these studies also show that submesoscale eddies are playing an important role in the formation of thermal intrusions near these strong fronts.

10:00

2aAO4. Exploring the shelf-slope dynamics in the Adriatic Sea using numerical models and seismic oceanography. Andrea Bergamasco, Francesco Falcieri (Oceanography, CNR-ISMAR, Castello 2737, Venice 30122, Italy, andrea.bergamasco@ismar.cnr.it), Jeff W. Book (Oceanography, Naval Res. Lab., Stennis, MS), Sandro Carniel (Oceanography, CNR-ISMAR, Venice, Italy), Warren W. Wood (Oceanography, Naval Res. Lab., Stennis, MS), Mauro Sclavo (Oceanography, CNR-ISMAR, Venice, Italy), and Richard W. Hobbs (Univ. of Durham, Durham, United Kingdom)

Dense shelf waters are formed and spread in the Adriatic Sea during winter periods, which dynamics are usually investigated by means of sea truth campaigns and modeling efforts. The former are either based on observational approaches (moored instruments, CTD, current meters, etc.) or on more innovative techniques, e.g., employing Seismic Oceanography (SO). Recent studies have shown that SO techniques can produce maps of vertical transects along the survey lines with horizontal and vertical resolution of, respectively, 10 and 100 m, suitable to explore the horizontal structures of BBL dynamics. Elaborating on these considerations, a novel approach combining the SO dataset collected during the ADRIASEISMIC cruise and high-resolution numerical model (ROMS) results was performed in two restricted areas of the Adriatic Sea: off the Gargano promontory and off the Bari shelf break. We present the first steps along the definition of a novel methodology. On one hand, SO can help to image the existing dynamical structures and their spatial/temporal resolution; on the other, the numerical model can quantify these acoustic snapshots in terms of temperature, salinity, and density, integrating the XBTs that are acquired during SO lines, and help identifying the nature of other processes (e.g., turbulence, internal waves, etc.).

10:20–10:40 Break

10:40

2aAO5. Mapping turbidity layers using a combination of high resolution seismic oceanographic and physical oceanographic data. Ekaterina A. Vsemirnova (Geospatial Res, Ltd., Durham Univ., Durham, County Durham, United Kingdom) and Richard W. Hobbs (Dept. of Earth Sci., Durham Univ., Durham, DH1 3LE, United Kingdom, r.w.hobbs@durham.ac.uk)

Synchronized seismic and oceanographic data were acquired during the Geophysical Oceanography (GO) project cruise in the Gulf of Cadiz in April–May 2007. The small volume (117 cu-in.) mini GI-gun seismic source used during the GO calibration experiment provided high resolution seismic data, which unveiled new features of the internal structure of the ocean. The seismic acquisition design gave a usable bandwidth of 50–250 Hz with a vertical resolution of 1.25 m, which is similar to that achieved by co-located CTD casts. We focus on the reflections observed on seismic data covering the moorings area. To test the hypothesis that measurable reflections can be generated by suspended sediment, we perform forward modeling of seismic response based on the temperature, salinity, and light attenuation measurements, available from CTD casts. Forward modeling based solely on temperature and salinity profiles show that thermohaline structure does not always explain reflections in water column, but they are consistent with light attenuation measurements.

11:00

2aAO6. Characterization of thermohaline staircases in the Tyrrhenian sea using stochastic heterogeneity mapping. Grant G. Buffett (Marine Geodynamics, GEOMAR Helmholz Ctr. for Ocean Res. Kiel, Gebdude Ostufer, Wischhofstr. 1-3, Geb. 8D/217, Kiel D-24148, Germany, gbuffett@geomar.de), Richard W. Hobbs (Dept. of Earth Sci., Durham Univ., Durham, United Kingdom), Ekaterina Vsemirnova (Dept. of Earth Sci., Durham Univ. Geospatial Res. Ltd., Durham, United Kingdom), Dirk Klaeschen (Marine Geodynamics, GEOMAR Helmholz Ctr. for Ocean Res. Kiel, Kiel, Schleswig-Holstein, Germany), Charles Hurich (Dept. of Earth Sci., Memorial Univ. of Newfoundland, St. John's, NF, Canada), C. Ranero (Barcelona Ctr. for Subsurface Imaging, Instituto de Ciencias del Mar, Barcelona, Spain), and V. Sallares (Inst. of Marine Sci., CMIMA, CSIC, Barcelona, Spain)

We apply Stochastic Heterogeneity Mapping based on the band-limited von Kármán power law function to stacked migrated seismic data of thermohaline staircases in the Tyrrhenian Sea. This process allows the estimation of stochastic parameters such as the Hurst number (a measure of surface roughness) and scale length. Thermohaline staircases are regular, well-defined step-like variations in vertical profiles of temperature and salinity. In the ocean, they are thought to arise from double diffusion processes driven by the large difference in the molecular diffusivities of heat and salt. They are thought to have an anomalously weak internal wave-induced turbulence, making them suitable for the estimation of a lower detection limit of turbulent dissipation. The Tyrrhenian Sea is a natural laboratory for the study of such staircases because, due to the internal basin's dynamic stability, steps as small as 10's of meters can be seen. Lower Hurst numbers represent a richer range of higher wavenumbers corresponding to a broader range of heterogeneity in reflection events. We interpret a broader range of heterogeneity as indicative of a greater degree of turbulence.

11:20

2aAO7. Mapping non-local turbulent breakdown of oceanic lee waves offshore Costa Rica through seismic oceanography. Will F. Fortin, W. Steven Holbrook (Geology and Geophysics, Univ. of Wyoming, 1000 E. University Ave., 3006, Laramie, WY 82071, wfortin@uwyo.edu), Ray Schmitt (Physical Oceanogr., Woods Hole Oceanogr. Inst., Woods Hole, MA), Jeff Book, and Scott Smith (Naval Res. Lab., Washington, DC)

Data analysis techniques in Seismic Oceanography are rapidly becoming more complex. Beyond first-order observation of oceanic structures, it is possible to extract quantifiable information about internal wave energies and turbulent dissipation rates. We use two co-located seismic surveys taken one day apart to estimate turbulent diffusivities of lee wave breakdown with emphasis on the mid-water turbulence generated. Through a horizontal wavenumber spectral analysis of our seismic images, we estimate turbulent dissipation rates throughout the cross-section. By integrating horizontal seismic slope spectra in the wavenumber domain over the turbulent subrange, we obtain relative turbulent energies across the survey. To resolve absolute turbulent diffusivities, we scale the relative measures to known absolute energies from tracked seismic

reflectors (isopycnals). The analysis section spans 22 km laterally and full ocean depth with a resolution on turbulent diffusivity of 10 m vertically by 400 m laterally. We focus on the region of elevated turbulent diffusivity caused by the breakdown of the lee wave above its generating site. We find the turbulent diffusivities related to the lee wave breakdown to be about five times greater than surrounding waters and 15 times greater than average open ocean diffusivities. We also see increased turbulent diffusivity around the rough bathymetry.

11:40

2aAO8. Current-eddy interaction in the Agulhas Return Current region from the seismic oceanography perspective. Ana E. Rice (National Res. Council, Bldg. 1009, Rm. B137, Stennis Space Ctr., MS 39529, anaerice@gmail.com), Jeffrey W. Book, Warren T. Wood (Naval Res. Lab., Stennis Space Ctr., MS), and Tim Fischer (GEOMAR Helmholtz Ctr. for Ocean Res., Kiel, Germany)

Interleaving in the Agulhas Return Current (ARC) frontal region is commonly manifested in the form of thermohaline intrusions, as sub-tropical and sub-polar water masses of similar density meet. In Jan./Feb. 2012, the Naval Research Laboratory and collaborators carried out a field experiment in which seismic and traditional hydrographic observations were acquired to examine frontal zone mixing processes. The high lateral resolution (10 m) of the seismic observations allowed fine-scale lateral tracking of thermal intrusions, which were corroborated with simultaneous XBT casts. Between seismic deployments both salinity and temperature data were acquired via CTD, underway-CTD and microstructure profiles. This study focuses on analyzing seismic reflection data in a particular E-W transect where the northward flowing ARC interacted with the southward flowing portion of a large anticyclonic eddy. Strong reflectors were most prominent at the edge of a hyperbolic zone formed between the eddy and ARC, where sub-polar waters interacted with waters of sub-tropical origin on either side. Reflectors were shallow within the hyperbolic zone and extended to 1200 m below the ARC. The nature of the observed reflectors will be determined from comparison of seismic reflection and derived $\partial T/\partial z$ fields, and XBT and TS profiles from the available hydrographic data.

TUESDAY MORNING, 4 JUNE 2013 — 519A, 9:00 A.M. TO 12:00 NOON

Session 2aBA

Biomedical Acoustics, Physical Acoustics, and Acoustical Oceanography: Bubbles Bubbles Everywhere I

Ronald A. Roy, Cochair
Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215

Thomas J. Matula, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., WA 98105-6698

Contributed Papers

9:00

2aBA1. A method for desalination and water remediation by hydrodynamic cavitation. Larry Crum (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, lac@apl.washington.edu), Michael Skinner, and Scott Zeilinger (Globe Protect, Inc., San Francisco, CA)

Water is becoming an increasingly valuable commodity, with population growth demanding more and more amounts of this limited resource. Increased efforts are directed toward recycling and remediation, as well as desalination of the large quantities of seawater available. Dr. Bertwin Langenecker was a pioneer in utilizing hydrodynamic cavitation in a variety of applications that would remove dissolved solids from water and other liquids. His combination of intense cavitation using a rotor-stator combination, as well as simultaneously adding an adsorbent, demonstrated impressive results in desalination and waste water remediation. In this presentation, a description will be given of Dr. Langenecker's technology as well as a sampling of some of his most impressive results. Speculations as to why this approach works as well as it does will be presented.

9:20

2aBA2. A new approach to ultrasonic cleaning. Tim Leighton (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk), Peter R. Birkin, and Doug Offin (School of Chem., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

Traditional ultrasonic cleaning baths are limited in that they cannot clean objects that are too large to fit in the bath, and cannot be taken to objects with complex geometries in order to "clean in place." Furthermore, the object to be cleaned sits in a "soup" of contaminated liquid, and while cavitation fields can be set up under test conditions, immersion of the object to be cleaned can significantly degrade the bath's performance by disrupting the sound field. An alternative technique, which does not use ultrasound is the commercial pressure- or -power washer, where high speed jets of water and cleaning agent are pumped onto a surface. Although these can "clean in place," they pump large volumes of water, and produce significant volumes of contaminated run-off and contaminated aerosol, both of which are hazards for secondary contamination of users and water supplies. The momentum of the water and pump requirements mean they are difficult to scale up. This paper presents a low volume flow technique for ultrasonic cleaning in place, benefits being that it operates with low flow rates (1–2 L/min), and there is no need to expend energy on heating the water.

9:40

2aBA3. The effect of surfactant shedding and gas diffusion on pressure wave propagation through an ultrasound contrast agent suspension. Jean-Pierre O'Brien, Nick Ovenden (Dept. of Mathematics, Univ. College London, UCL, London, United Kingdom, jean-pierre.o'brien@ucl.ac.uk), and Eleanor Stride (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Interest in microbubbles as agents for therapeutic and quantitative imaging applications in biomedical ultrasound has increased the need for their accurate modeling. However, effects such as gas diffusion, the properties of the shell, and changes in bubble behavior under repeated exposure to ultrasound pulses are still not well understood. A revised equation for microbubble motion is proposed that includes the effects of gas diffusion as well as a nonlinear surface tension, which depends on a non-constant surfactant surface concentration. This is incorporated into a nonlinear wave propagation

model to account for these additional time-dependent effects in the response of microbubble contrast agent populations. The results from the model indicate significant changes in both bubble behavior and the propagated pulse compared with those predicted by existing propagation models; and show better agreement with experimental data. Our analysis indicates that changes in bubble dynamics are dominated both by surfactant shedding on ultrasonic timescales and gas diffusion over longer timescales between pulses. Therefore, incorporating such time-dependent phenomena in ultrasound imaging algorithms should lead to better quantitative agreement with experiments.

10:00

2aBA4. Inertial cavitation at the nanoscale. James J. Kwan, Susan Graham, and Constantin Coussios (Inst. of Biomed. Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford, Oxfordshire OX3 7DQ, United Kingdom, james.kwan@eng.ox.ac.uk)

Our group has recently developed novel nano-sized drug carriers that spatially target a tumor and release their payload in the presence of ultrasound-induced inertial cavitation. To maximize drug release and distribution within the tumor, co-localization of the drug carrier and cavitation nuclei is necessary. We have recently demonstrated that rough-patterned silica nanoparticles can reduce inertial cavitation thresholds to clinically relevant levels, and will extravasate in tumors alongside the liposomes by virtue of their size. We now report on the underlying mechanisms that these nanoparticles, which are orders of magnitude smaller than the acoustic wavelength, can instigate inertial cavitation. The rough surface of the nanoparticle is modelled as a plane with a crevasse that traps a nanobubble. Using this model, we predict the motion of a gas bubble as it emerges from the cavity in response to the compressional and rarefactional ultrasonic pressures. We show that cavitation occurs when the nanobubble breaks free from the surface, growing unstably before collapsing during the compressional half cycle of the acoustic wave. Calculations show that a nanoscaled cavity greatly reduces the cavitation threshold across all frequencies and geometries studied. In addition, cavitation thresholds nonlinearly decrease with increasing cavity size.

10:20

2aBA5. Cavitation-induced streaming in shock wave lithotripsy. Yuri A. Pishchalnikov (Impulse Devices, Inc., 13366H Grass Valley Ave., Grass Valley, CA 95945, yurapish@gmail.com) and James A. McAteer (Dept. of Anatomy and Cell Biol., Indiana Univ. School of Med., Indianapolis, IN)

Cavitation generated by lithotripter shock waves (SWs) in non-degassed water was studied using a 60 frames-per-second camcorder—recording the migration of microbubbles over successive SWs. Lithotripter SWs were produced using a Dornier DoLi-50 electromagnetic lithotripter at 0.5 and 2 Hz pulse repetition frequency (PRF). Cavitation was affected by PRF and by the power level (PL) of the lithotripter. At slow PRF, such as shots fired many seconds apart, cavitation was relatively sparse and bubble clouds flowed in the direction of SW propagation. When PRF was increased, the bubble clouds generated by one SW were amplified by subsequent SWs. Cloud amplification was accompanied by an apparent change in the pattern of bubble migration. Whereas bubbles continued to enter the field of view from the prefocal side, the main bubble cloud remained near the focal point. This was due to a streaming of bubbles opposite to the direction of SW propagation. Increasing the PL grew the cavitation field and enhanced the flow of bubbles opposite to the direction of SW propagation. Stepping up the PL acted to push the broad cloud progressively pre-focally (toward the SW source), shifting the position of the plane at which the opposing directional bubble flows collided. [NIH DK43881.]

10:40

2aBA6. Bubbles trapped on the surface of kidney stones as a cause of the twinkling artifact in ultrasound imaging. Oleg Sapozhnikov (Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, oleg@acs366.phys.msu.ru), Wei Lu, Michael Bailey, Peter Kaczkowski, and Lawrence Crum (Appl. Phys. Lab., Ctr. for Industrial and Med. Ultrasound, Univ. of Washington, Seattle, WA)

A twinkling artifact (TA) associated with urinary calculi has been described as rapidly changing colors on Doppler ultrasound. The purpose of this study was to investigate the mechanism of TA. Doppler processing was performed on raw per channel radio-frequency data collected when imaging human kidney stones in degassed water. Suppression of twinkling by an ensemble of computer generated replicas of a single received signal demonstrated that the TA arises from variability among the acoustic signals and not from electronic signal processing. This variability was found to be random in nature, and its suppression by elevated static pressure, and its return when the pressure was released, suggests that the presence of surface bubbles on the stone is the mechanism that gives rise to the TA. Submicron size bubbles are often trapped in crevices on solid objects, but the presence of these bubbles *in vivo* is unexpected. To further check this mechanism under conditions identical to *in vivo*, stone-producing porcine kidneys were harvested en bloc with a ligated ureter and then placed into a pressure chamber and imaged at elevated atmospheric pressure. The result was similar to *in vitro*. [Work supported by NIH DK43881, DK092197, RFBR, and NSBRI through NASA NCC 9-58.]

11:00

2aBA7. The use of twinkling artifact of Doppler imaging to monitor cavitation in tissue during high intensity focused ultrasound therapy. Tatiana D. Khokhlova (School of Med., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, tanyak@apl.washington.edu), Tong Li (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Dept. of Acoust. Phys. Faculty, Moscow State Univ., Russian Federation and Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Joo Ha Hwang (School of Medicine, Univ. of Washington, Seattle, WA)

In high intensity focused ultrasound (HIFU) therapy, it is important to monitor the presence and activity of microbubbles in tissue during treatment. The current methods—passive cavitation detection (PCD) and B-mode imaging—have limited sensitivity, especially to small-size, non-violently collapsing microbubbles. Here, a new method for microbubble detection is proposed, based on "twinkling" artifact (TA) of Doppler imaging. TA occurs when color Doppler ultrasound is used to image hard objects in tissue (e.g., kidney stones) and is displayed as brightly colored spots. As demonstrated recently, TA can be explained by irregular scattering of the Doppler ensemble pulses from the fluctuating microbubbles trapped in crevices of the kidney stone. In this work, TA was used to detect cavitation in tissue and in polyacrylamide gel phantoms during pulsed 1 MHz HIFU exposures with different peak negative pressures (1.5–11 MPa). At each pressure level, the probability of cavitation occurrence was characterized using TA and the broadband signals recorded by PCD, aligned confocally with the HIFU transducer. The results indicate that TA is more sensitive to the onset of cavitation than conventional PCD detection, and allows for accurate spatial localization of the bubbles. [Work supported by RFBR and NIH (EB007643, 1K01EB015745, and R01CA154451).]

11:20

2aBA8. Rectified growth of histotripsy bubbles. Wayne Kreider (Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Adam D. Maxwell (Univ. of Washington School of Medicine - Dept. of Urology, Seattle, WA and Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Seattle, WA), Tatiana D. Khokhlova, Julianna C. Simon (Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Seattle, WA), Vera A. Khokhlova, Oleg A. Sapozhnikov (Lomonosov Moscow State Univ.- Phys. Faculty, Moscow, Russian Federation and Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Moscow, Russian Federation), and Michael R. Bailey (Univ. of Washington - Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Seattle, WA)

Histotripsy treatments use high-amplitude shock waves to fractionate tissue. Such treatments have been demonstrated using both cavitation bubbles excited with microsecond-long pulses and boiling bubbles excited for milliseconds. A common feature of both approaches is the need for bubble growth, where at 1 MHz cavitation bubbles reach maximum radii on the order of 100 microns and boiling bubbles grow to about 1 mm. To explore how histotripsy bubbles grow, a model of a single, spherical bubble that accounts for heat and mass transport was used to simulate the bubble dynamics. Results suggest that the asymmetry inherent in nonlinearly distorted waveforms can lead to rectified bubble growth, which is enhanced at elevated temperatures. Moreover, the rate of this growth is sensitive to the waveform shape, in particular the transition from the peak negative pressure to the shock front. Current efforts are focused on elucidating this behavior by obtaining an improved calibration of measured histotripsy waveforms with a fiber-optic hydrophone, using a nonlinear propagation model to assess the

impact on the focal waveform of higher harmonics present at the source's surface, and photographically observing bubble growth rates. [Work supported by NIH EB007643 and DK43881; NSBRI through NASA NCC 9-58.]

11:40

2aBA9. Ultrasonic atomization: A mechanism of tissue fractionation. Julianna C. Simon (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu), Oleg A. Sapozhnikov, Vera A. Khokhlova (Dept. of Acoust., Phys. Faculty, Moscow State Univ. Moscow, Russian Federation and Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Yak-Nam Wang, Lawrence A. Crum, and Michael R. Bailey (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

High intensity focused ultrasound (HIFU) can be used to atomize liquid by creating a fountain on the surface exposed to air. The mechanism of atomization can be most accurately described by the cavitation-wave hypothesis wherein a combination of capillary waves excited on the liquid surface with cavitation beneath the surface produces a fine spray. Here, we show experimentally that a free tissue surface can also be atomized resulting in erosion of tissue from the surface. A 2-MHz spherically focused transducer operating at linearly predicted *in situ* intensities up to 14,000 W/cm^2 was focused at *ex vivo* bovine liver and *in vivo* porcine liver tissue surfaces without the capsule. The end result for both *in vivo* and *ex vivo* tissues was erosion from the surface. In bovine liver at the maximum intensity, the erosion volume reached 25.7 $\pm$ 10.9 mm^3 using 300 10-ms pulses repeated at 1 Hz. Jet velocities for all tissues tested here were on the order of 10 m/s. Besides providing a mechanism for how HIFU can mechanically disrupt tissue, atomization may also explain how tissue is fractionated in boiling histotripsy. [Work supported by NIH EB007643, NIH DK43881, and NSBRI through NASA NCC 9-58.]

TUESDAY MORNING, 4 JUNE 2013 — 512AE, 8:55 A.M. TO 12:00 NOON

Session 2aEA

Engineering Acoustics: Directional and Non-Directional Microelectromechanical Microphones

Gary W. Elko, Chair
mh Acoust. LLC, 25A Summit Ave., Summit, NJ 07901

Chair's Introduction—8:55

Invited Papers

9:00

2aEA1. A biologically inspired silicon differential microphone with active Q control and optical sensing. Ronald Miles (Dept. of Mech. Eng., SUNY Binghamton, Vestal, NY 13850, miles@binghamton.edu), Levent Degertekin (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Weili Cui, Quang Su, Dorel Homentcovschi (Mech. Eng., SUNY Binghamton, Binghamton, NY), and Banser Fredrick (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

A MEMS differential microphone is described in which the diaphragm design is inspired by the mechanics of directional hearing in the fly *Ormia ochracea*. The 1 mm by 3 mm diaphragm is designed to rotate about a central pivot in response to sound pressure gradients. The diaphragm is designed to have its dominant resonance mode within the audible frequency range and to have as little viscous damping as possible (to minimize the effects of thermal noise). The motion of the diaphragm is detected using an optical sensing scheme that includes a semiconductor laser (VCSEL), photodetectors, a mirror, and a diffraction grating. To minimize the adverse effects of the light damping on the response, an active feedback system is implemented to achieve active Q control. This uses the output of the optical detection scheme to drive the diaphragm through a capacitive actuator. The microphone and optoelectronics are packaged into an assembly that can be incorporated into a mock behind-the-ear hearing aid. The microphone is shown to achieve a noise floor that is approximately 17 dBA lower than what can be achieved using a pair of existing low noise hearing aid microphones to create a directional hearing aid.

9:20

2aEA2. Biomimetic flow sensors for environmental awareness. Gijs Krijnen, Harmen Droogendijk (MESA+ Res. Inst., Univ. of Twente, P.O. Box 217, Enschede 7500AE, Netherlands, gijs.krijnen@utwente.nl), Jerome Casas (IRBI, Université de Tours, Tours, France), and Ahmad Dagamseh (MESA+ Research Inst., Univ. of Twente, Enschede, Netherlands)

Crickets possess hairy organs attached to their abdomen, the so-called cerci. These cerci contain highly flow-sensitive mechanosensors that enable the crickets to monitor the flow-field around them and react to specific stimuli form the environment, e.g., air-movements generated by hunting spiders. Salient is the sensitivity of these sensors, which work at thermal noise threshold levels, and the large number of hairs which, together with the necessary neural processing, allows the cricket to use the cerci as a kind of "flow-camera." Biologists and engineers have been working together in the recent past to regenerate part of the outstanding sensing capabilities of crickets in manmade, bio-inspired flow-sensor arrays. Using micro-electromechanical systems (MEMS) technology, sensors are created that are sensitive and show a high degree of directivity. By analyzing the governing physics, the sensors have been optimized to the point that currently the electronic interfacing is the limiting factor. Nonlinear parametric effects are used to increase the range of applicability of the sensors. Stochastic resonance is investigated to further enhance sensing capabilities. Arrays of sensors, interfaced using frequency division multiplexing (FDM), have been demonstrated to enable the tracking of the movement of small spheres.

9:40

2aEA3. Small directional microelectromechanical systems microphone arrays. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com)

Directional microphone arrays that are physically small compared to the acoustic wavelength are of great interest for hand-held communication devices. Spatially directive microphones can reduce the impact of background acoustic noise without adding distortion to the signal. This talk will present some design topologies and requirements as well as a new physical design that could enable directional microphone responses while being small in size.

Contributed Papers

10:00

2aEA4. Leveraging microelectromechanical microphones inherent matching to reduce noise using multiple microphone elements. Wade Conklin (Knowles Electron., 1151 Maplewood Dr., Itasca, IL 60143, wade.conklin@knowles.com)

Signal-to-noise ratio (SNR) is a critical parameter in the adoption of small scale (~1 mm) microphones for use in hearing aids. As a result, electret microphones have dominated the market since their invention in the 1960's. Significant effort is being invested to increase the SNR of microelectromechanical (MEMs) microphones near that of electrets. This work covers the approach of using multiple microphone elements to increase SNR. It explores the theory, examines the dependence of the SNR improvement on the matching of the microphone elements, and compares measurements on a single element microphone versus a multiple element microphone. Finally, it examines why the MEMs fabrication process lends itself to this usage and compares the trade-offs in scaling elements versus scaling size.

10:20–10:40 Break

10:40

2aEA5. A novel two dimensional particle velocity sensor. Olti Pjetri, Remco J. Wiegerink, Theo S. Lammerink, and Gijs J. Krijnen (Transducers Sci. and Technol., Univ. of Twente, MESA+ Inst. for Nanotechnology, Drienerlolaan 5, Enschede 7522NB, Netherlands, o.pjetri@utwente.nl)

In this paper, we present a two wire, two-dimensional particle velocity sensor. The miniature sensor of size $1.0 \times 2.5 \times 0.525$ mm, consisting of only two crossed wires, shows excellent directional sensitivity in both directions, thus requiring no directivity calibration, and is relatively easy to fabricate. The sensor consists of two crossed beams of SiRN with a platinum layer on top. These beams are used both as heaters and sensors. Two currents with equal amplitude are injected in both terminals of one of the beams and are extracted from the terminals of the other beam. A particle velocity component in the direction of a beam will cause its temperature, and thus resistance, profile to change asymmetrically. This asymmetry in resistance will give rise to a voltage difference across that beam which is proportional to the particle velocity level. The sensor shows a frequency bandwidth between 20 Hz and 10 kHz.The two figures of eight are exactly perpendicular to each other as desired, which was difficult to obtain in earlier implementations using parallel beams. Furthermore, the structure consisting of two crossed wires increases the mechanical robustness of the beams resulting in fabrication yields of 94% as opposed to 70% in earlier implementations.

11:00

2aEA6. Characterization of directional microphones in an arbitrary sound field. Quang T. Su, Joshua H. Merlis, Daniel Antonelli, and Ronald N. Miles (Mech. Eng., Binghamton Univ., P.O. Box 6000, Binghamton, NY 13902-6000, qsu@binghamton.edu)

An acoustic characterization method for directional microphones is presented that does not require an anechoic chamber to provide a controlled plane wave sound field. Measurements of a directional microphone under test are performed in a nearly arbitrary sound field for several angles of sound incidence, and the corresponding sound pressure and pressure gradients in the vicinity of the test microphone are measured using an automated probe microphone scanning system. From these measurements, the total acoustic frequency response of the directional microphone can be decomposed into its sensitivities to sound pressure and pressure gradient using a least squares estimation technique. These component responses can then be combined to predict the directional response of the microphone to a plane wave sound field. This technique is demonstrated on a commercially available pressure gradient microphone, and also on a combination sound pressure-pressure gradient microphone. Comparisons with the plane wave responses measured in an anechoic environment show that the method gives accurate results down to 100 Hz.

11:20

2aEA7. Calibration of smartphone-based devices for noise exposure monitoring: Method, implementation, and uncertainties of measurement. Romain Dumoulin and Jeremie Voix (École de technologie supérieure, Université du Québec, 1100, rue Notre-Dame Ouest, Montreal, QC H3C 1K3, Canada, romain.dumoulin@ens.etsmtl.ca)

Standardized noise exposure campaigns have as their principle disadvantages the cost of instrumentation and the difficulties associated with practical deployment in the field. Our ongoing research evaluates the suitability of an alternate solution based on smartphone sensing: the occupational noise exposure and its associated measurement uncertainties are estimated from a spatio-temporal analysis of smartphones noise measurements and GPS data. This paper presents a diffuse field calibration method for such smartphone-based devices. The measurements methods and the calculation of expanded uncertainties for a large range of sound levels are detailed. The calibration corrections include a frequency response linearization and an A-weighted sound level correction, which is function of the C-A spectral balance of the sound pressure levels measured. To later ensure a realistic correction, these spectral balance values come from distribution of referenced industrial noise databases. An Android^{TM} "app" has also been developed to measure the noise levels and to compute the calibration and corrections factors. Finally, an laboratory validation is conducted to evaluate, on a population of calibrated smartphone-based devices, the measurement errors associated with such devices as a function of microphone directivity, linearity, and frequency response.

11:40

2aEA8. Examination of acoustic mechanism for compact acoustic reproduction systems. Kosuke Sugihara (Faculty of Eng. Sci., Kansai Univ., Suita-Shi yamate-cho 3-3-35, Osaka-hu 556-8680, Japan, sukekiyo15@gmail.com), Masashi Nakamura (Fujitsu TEN, Hyogo, Japan), Yoshinobu Kajikawa, Yasuo Nomura (Faculty of Eng. Sci., Kansai Univ., Osaka, Japan), and Takashi Miyakura (HOSHIDEN, Osaka, Japan)

In this paper, we propose a method for analyzing compact acoustic reproduction systems (e.g., mobile phones) through acoustic equivalent circuits. Measured responses of compact acoustic reproduction systems cannot be represented accurately by the analysis based on the conventional acoustic theory. Acoustic engineers consequently are obliged to design compact acoustic reproduction systems by trial and error. Moreover, the sound quality of those systems is likely to deteriorate due to the difficulty of such an acoustic design. We therefore clarify the cause of the difference between the measured response and the analysis one calculated by the finite element method (FEM) analysis and consider the possibility of obtaining new acoustic theoretical formula based on the analysis results in order to make it easier for acoustic engineers to design compact acoustic reproduction systems.

Session 2aED

Education in Acoustics: Tools for Teaching Advanced Acoustics

David T. Bradley, Chair
Phys. + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604

Invited Papers

9:00

2aED1. Summer school for acoustics graduate students. Steven L. Garrett, Anthony A. Atchley (Grad. Prog. in Acoust., Appl. Res. Lab., Penn State, State College, PA 16804-0030, sxg185@psu.edu), Logan E. Hargrove (Retired, Reston, VA), Thomas J. Matula (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Joseph R. Gladden, and Henry E. Bass (Deceased) (National Ctr. for Physical Acoust., Univ. of Mississippi, University, MS)

In addition to subject mastery and the focused effort required to complete a thesis project, graduate students also need to develop a broad understanding of their field and cultivate a familiarity with the larger community of researchers and practitioners. The “summer school” format has been shown to enhance both subject-matter breadth and build community awareness in physical acoustics. Physical Acoustics Summer School (PASS) has been held in late-May, in even-numbered years, since 1992. The format for each day is usually two three-hour lectures followed by evening discussion groups to answer questions and explore extensions of the day’s lecture topics. One lecture session is typically dedicated to acoustics demonstrations. Attendance for the full week is required of all participants who also dine together three times each day. Venues are chosen to provide isolation that minimizes distraction and maximizes interactions among all participants. Typical enrollment has been 10 distinguished lecturers (including many Silver Medal winners in Physical Acoustics), 10 discussion leaders, and 30 graduate students. This format has been successfully extended to one other ASA Technical Committee: the marine bioacoustics community has held their summer school twice (SeaBASS). PASS has now been functioning long enough that former students have become lecturers.

9:20

2aED2. Experience teaching acoustics at the senior-undergraduate and first-year graduate levels. David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133, drd@umich.edu)

Perhaps without appreciating it, college students are more fully equipped to understand and study acoustics than any other field of science. This assertion stems from the fact that most college students have two exquisite broadband receivers with impressive dynamic range (ears), and a matched multi-functional sound projector (voice). Given that nearly all college students have used their ears and voice for many years before arriving in an acoustics classroom, the advanced-acoustics instructor’s task is primarily to link theoretical results with the acoustic intuition that students already possess. Thus, a worthy pedagogical goal is to activate this submerged knowledge and connect it to mathematical results through practical examples, classroom demonstrations, and relevant homework. At the senior-level, useful demonstrations include the following: acoustic resonances of a cardboard tube, the dipole characteristics of small raw loudspeaker, directional reflection with a metal salad bowl, and sound volume changes as a loud speaker is lifted out of a cabinet. At the graduate level, useful homework assignments include boundary-element and finite-element calculations with commercial software that can be checked with established theory. In addition, worthwhile homework problems that attempt to provide sufficient reward for students who master the mathematical content have been developed for both classes.

9:40

2aED3. Combining theory and experiment to teach acoustic concepts. Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

A rigorous theoretical development is desirable to help students at both the undergraduate and graduate levels develop a deep understanding of acoustic phenomena. However, numerous students labor through the mathematics associated with the concepts without ever developing an understanding of how that translates over into the physical world. Many acoustic phenomena lend themselves to experimental demonstrations that can greatly aid students’ understanding of the physical concepts and help them connect the theoretical developments with what physically happens. These demonstrations also provide a means for introducing common issues associated with making acoustic measurements that can also be educational for students. As an example, this paper will focus on how we have developed concepts associated with vibrating strings in a class through both theoretical development and use of a relatively simple experimental apparatus. Students gain a much better understanding not only of modes associated with the string, but the relative accuracy of the underlying theory. In addition, basic signal analysis topics and measurement accuracy also surface in the process of making the measurements.

10:00

2aED4. Acoustic interference and diffraction experiments in the advanced laboratory class. Andrew Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

Many acoustic analogs to classical optics experiments can be performed with low-cost ultrasonic transducers. Mounts for the ultrasonic transducers have been designed to be produced with desktop 3D printing technology. The designs are open and available for use and modification. Examples of experiments that can be done with this system include single-slit diffraction, double-slit interference, and Lloyd’s

mirror experiments. Although simple in appearance, these experiments provide a rich opportunity for students to explore acoustic phenomena in the advanced laboratory such as the radiation pattern of the transducer and rudimentary acoustic beamforming techniques. The lab activities for use in intermediate or advanced acoustics lab classes are included in a revised laboratory manual currently in development. The lab manual has experiments appropriate for both introductory and advanced acoustics labs covering a range of acoustics subfields.

10:20

2aED5. Using your ears: A novel way to teach acoustics. Lauren Ronsse, Dominique J. Chéenne, and Sarah Kaddatz (Dept. of Audio Arts and Acoust., Columbia College Chicago, 33 E. Congress Pkwy, Ste. 601, Chicago, IL 60605, lronsse@colum.edu)

Auditory simulations of physical phenomena pertaining to acoustics have been developed to enhance student learning and understanding of these conditions. The demonstrations range from simulations of fundamental concepts, such as reverberation, flutter echoes, reflections, and room modal effects, to more applied topics, such as sound transmission through barriers, mechanical system noise spectra, and varying absorption distribution in rooms. The simulations were generated by using auralization tools and processed recordings. The demonstrations may be utilized in the classroom to introduce new acoustical concepts by having students first listen to a simulation, then write and/or discuss what they hear, providing conjectures about the parameters that could create such acoustical conditions. The goal of the demonstrations is to encourage students to use their ears as part of a quantitative and qualitative assessments of acoustical phenomena.

10:40

2aED6. Creating interactive acoustics animations using Mathematica's Computable Document Format. Daniel A. Russell (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

The computational and graphical prowess of Mathematica has long made it a powerful educational tool for creating effective animations of acoustic and vibration phenomena [Russell, J. Acoust. Soc. Am. **106**, 2197 (1999) and Sparrow and Russell, J. Acoust. Soc. Am. **103**, 2987 (1998)]. Once an animation has been created within Mathematica it is relatively easy to convert the animation to an animated GIF file for display on a website [Russell, J. Acoust. Soc. Am. **114**, 2308 (2003)]. However, such animations, while effective at conveying or illustrating complicated acoustic phenomena, are "static" in the sense that they are not interactive and a person viewing the animation cannot change parameters. Recently, Wolfram Research implemented a new Computable Document Format that allows interactive plots and animations to be inserted into webpages and electronic documents. A free CDF player from Wolfram allows viewers to interact with plots and animations by moving sliders to change values of parameters. This talk will demonstrate the process of creating a CDF animation for embedding in a webpage. Other, more complex, demonstrations will also be showcased do illustrate the potential capabilities of CDF as an educational tool.

11:00

2aED7. Teaching advanced undergraduate students principles of outdoor sound propagation using football game measurements. Kent L. Gee, Tracianne B. Neilsen, Alan T. Wall, and Nicholas J. Eyring (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

As part of a sound system evaluation at Brigham Young University's football stadium and to assist in planning for future system design, measurements were made before and during games by an upper-level undergraduate acoustics class. The measurement experience provided significant training opportunities for the students. Teams of students used sound level meters to make recordings at numerous locations both inside and outside the stadium. These measurements were then correlated with data from stationary microphones placed near the field. From the data, the predicted slow, A-weighted equivalent levels in and around the stadium were calculated relative to an assumed 90 dBA on the sideline. Straightforward outdoor sound propagation prediction methods involving geometric spreading, atmospheric absorption, barriers, etc. were successfully used to validate the measured data within 1-2 decibels at many locations, including a location in the foothills to the southeast of the stadium at a distance of approximately 3 km. The students appreciated the hands-on experiences gained by participation in the measurements and analysis.

11:20

2aED8. Spectrogram puzzles: A tool for teaching acoustic phonetics. Tessa Bent and Emily Garl (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

One of the most useful tools for the acoustic analysis of speech is the spectrogram. A spectrogram is a visual representation of speech which includes time, frequency, and amplitude information. To conduct appropriate and accurate acoustic-phonetic analyses, students must learn to identify important features of vowels and consonants on spectrograms. To help students learn to identify these features, the spectrogram puzzle exercise was developed. In this exercise, spectrograms of sentences are printed using a large-format printer and cut into phoneme sections. Students then arrange the segments into the appropriate order based on a provided sentence. Depending on students' level of knowledge and experience, task difficulty can be increased or decreased by: (1) providing phonetic transcription versus orthography, (2) including more or less easily identifiable consonants, (3) including citation-style speech versus conversational or disordered speech, and (4) having teams versus individual students complete the exercise. Through these modifications, this activity can be used with a wide range of students from beginning undergraduate to advanced graduate students. For all students, spectrogram puzzles provide a hands-on, interactive learning experience that can facilitate critical thinking, collaborative learning, and acquisition of knowledge about the representation of speech sounds on spectrograms.

11:40

2aED9. Mechanical model of the human ear. E. C. Everbach (Engineering, Swarthmore College, 500 College Ave., Swarthmore, PA 19081, ceverba1@swarthmore.edu)

Diagrams showing cutaway views of the human ear are all very well for teaching the mechanics of hearing, but a tabletop model the students can manipulate is even better. The author presents a mechanical model based upon the water-tube cochlea previously developed by Robert Keolian, but including the outer- and middle-ear components. The model allows phenomena such as the acoustic reflex, critical bands, and masking of higher-frequency by lower-frequency tones.

Session 2aMU

Musical Acoustics and Signal Processing in Acoustics: Aeroacoustics of Wind Instruments and Human Voice I

Shigeru Yoshikawa, Cochair
Grad. School of Design, Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan

Xavier Pelorson, Cochair
CNRS, 11 rue des mathematiques, Saint Martin d'Heres 38330, France

Invited Papers

9:00

2aMU1. Aeroacoustics of the panpipes. Roman Auvray, Benoît Fabre (LAM, d'Alembert, UPMC Univ Paris 06, CNRS UMR 7190, 11 rue de Lourmel, Paris 75015, France, auvray@lam.jussieu.fr), Felipe Meneses, Patricio de la Cuadra (CITA, Pontificia Universidad Catolica de Chile, Santiago, Chile), and Pierre-Yves Lagrée (FCIH, d'Alembert, UPMC Univ Paris 06, CNRS UMR 7190, Paris, France)

The generic term "flute-like instruments" includes a wide variety of instruments whose sound production is ensured by the coupling of an air jet with an acoustic resonator. Within the family, different kinds of resonator (for instance Helmholtz resonator, open-open, or open-closed tube), may be used with different kind of air supply systems such as the ones found in the recorder, the flue organ pipe, or the shakuhachi. It is common to extent the results obtained on one of the member of the family to the whole family. However, when an accurate description of the sound production mechanisms is required, small discrepancies may arise due to the wide variability in the geometries or in the air supply systems. Among other, a closed-end flute may have a different behavior than an open-open flute since the recirculation of air flow within the pipe may alter the hydrodynamics of the jet, and thus the auto-oscillation process. While most of the studies on flute-like instruments have focused on open pipes (organ pipes and recorder), the panpipes (a representative closed-end flute) has only received little attention. We present experimental data, including flow visualization and pressure signal measurement gathered on a closed pipe. A model of the flow in the pipe allows to interpret the data and compare the behavior of a closed pipe blown with a turbulent jet with that of an open pipe blown with a laminar jet.

9:20

2aMU2. Aerodynamical sounding mechanism in flue instruments: Acceleration unbalance between the jet vortex layers. Shigeru Yoshikawa (Grad. School of Design, Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, shig@design.kyushu-u.ac.jp)

According to particle image velocimetry (PIV) measurement applied to the sound production in organ flue pipes and flutes, the vortex shedding at the pipe edge proposed by Howe (1975) are not observed but the formation of the vortex layer is clearly observed along both sides of the jet flow. This has been confirmed in various sounding conditions with different blowing pressures and resulting pitches. The acceleration unbalance is generated from an incomplete cancelation of the aeroacoustical source term ($\omega\times\boldsymbol{U}$) between both sides of the jet, where $\boldsymbol{U}$ is the jet velocity and ω ($=\mathrm{rot}\boldsymbol{U}$) the vorticity. In addition, the vortex layer is essentially unstable because it is formed along the inflection point of the lateral jet-velocity profile. Therefore, the acceleration unbalance and inflection instability of the vortex layer activates the jet wavy motion to reinforce the inward or outward acoustic velocity $\boldsymbol{u}$ at the pipe mouth. Phase relations between acoustic quantities approve conventional acoustical models based on the volume-flow drive and momentum drive. Since $\omega\times\boldsymbol{U}$ can also activate the jet movement in edge-tone generation, the vortex-layer formation may be regarded as the fluid-dynamical mechanism common to the edge-tone generation and the pipe-tone generation.

9:40

2aMU3. Effective techniques and crucial problems of numerical study on flue instruments. Kin'ya Takahashi, Takuya Iwasaki, Takahiro Akamura (The Phys. Lab., Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, takahasi@mse.kyutech.ac.jp), Yuki Nagao (Res. Information Sci. and Elec. Eng., Kyushu Univ., Fukuoka, Japan), Ken'ichiro Nakano (The Phys. Lab., Kyushu Inst. of Technol., Iizuka, Japan), Taizo Kobayashi, Toshiya Takami, Akira Nishida, and Mutsumi Aoyagi (Res. Inst. for Information Technol., Kyushu Univ., Fukuoka, Japan)

In the last several decades, there have been many important proposals for study on flue instruments from theoretical and experimental points of view. Analyses based on aerodynamic sound theory are crucial for understanding the sounding mechanism of flue instruments. According to the growth of computer power and the improvement of numerical schemes, numerical simulations based on fluid dynamics now become important tools for the study of aerodynamics sound. In this talk, we will discuss accuracy, efficiency, and reliability of numerical calculations with large-eddy-simulation (LES) of compressible flow and we will show to what extent LES can reproduce the fluid and acoustic behavior of flue instruments observed experimentally. Furthermore, we will consider how to calculate the important theoretical formulae of aerodynamics sound theory, e.g., Lighthill's quadrupole, Powell-Howe vortex sound source, Howe's formula that allows us to estimate the energy transfer between the acoustic field and the hydro-dynamic field. Actually, those quantities given by the theoretical formulae play an important role in the analyses of sounding mechanisms of flue instruments. M. Miyamoto *et al*., "Numerical study on acoustic oscillations of 2D and 3D flue organ pipe like instruments with compressible LES," Acta Acustica (accepted for publication).

10:00

2aMU4. Experimental and numerical characterization of aerodynamic noise applied to moderate Reynolds number airflow. Yo Fujiso, Annemie Van Hirtum (GIPSA-Lab, Grenoble Univ., 11 rue des Mathematiques, Grenoble Campus, Saint Martin d'Heres 38402, France, yo.fujiso@gipsa-lab.grenoble-inp.fr), Kazunori Nozaki, and Shigeo Wada (Grad. School of Eng. Sci., Osaka Univ., Toyonaka-city, Japan)

In the study of aerodynamic noise, a tackling challenge is to understand the underlying aeroacoustic mechanisms leading to its generation. The current paper aims at contributing to the noise characterization by focusing on moderate Reynolds number ($100 \leq Re \leq 10000$) airflow circulating through a rectangular channel containing a trapezoidal obstacle near the outlet. The outcome of large Eddy simulation and acoustic experiments are compared for different experimental boundary conditions at inlet and outlet, and for different apertures below the obstacle.

10:20

2aMU5. Numerical analysis of the interaction between fluid flow and acoustic field at the mouth-opening of a flue instrument. Takahiro Akamura (The Phys. Lab., Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, red0ta@yahoo.co.jp), Yuki Nagao (Res. Information Sci. and Elec. Eng., Kyushu Univ., Fukuoka, Japan), Takuya Iwasaki, Ken'ichiro Nakano, Kin'ya Takahashi (The Phys. Lab., Kyushu Inst. of Technol., Iizuka, Japan), Taizo Kobayashi, Toshimi Takami, Akira Nishida, and Mutsumi Aoyagi (Res. Inst. for Information Technol., Kyushu Univ., Fukuoka, Japan)

The fluid-sound interaction is the key to understanding the sounding mechanism of flue instruments. The formula introduced by Howe allows us to estimate the energy transfer between acoustic field and hydro-dynamic field. For calculation of Howe's formula, it is necessary to divide acoustic fields from fluid, but we do not have any established method to do it, yet. Recently, several authors developed approximate methods to evaluate Howe's formula and applied to experiments of cavity noise, flue instruments and so on. In this talk, we introduce a numerical method to calculate Howe's formula, which is similar to those above. Our model is a small flue-organ like instrument with an end-stop. We use compressible large-eddy simulation (LES), which is able to reproduce the fluid flow and acoustic field, simultaneously. First, fluid flow and acoustic oscillation excited in the pipe by a jet-injection from the flue are reproduced by LES. Next, an acoustic field is reproduced by LES without the jet-injection but with driving at the far end, pressure driving, particle velocity driving or oscillating wall driving (like a loudspeaker). Combining those results allows us to calculate Howe's formula and to estimate the fluid-sound interactions.

10:40

2aMU6. Numerical study on the function of tone holes of a recorder like instrument from the viewpoint of the aerodynamic sound theory. Takuya Iwasaki (Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, gyjcjc56@gmail.com), Taizo Kobayashi (Res. Inst. for Information Technol., Kusyu Univ., Fukuoka, Fukuoka, Japan), Kin'ya Takahashi (Kyushu Inst. of Technol., Iizuka, Japan), Toshiya Takami, Akira Nishida, and Mutsumi Aoyagi (Res. Inst. for Information Technol., Kusyu Univ., Fukuoka, Japan)

We have investigated properties of tone holes of a small recorder like instrument by using compressible large-eddy simulation (LES), which reproduces fluid flow and acoustic field, simultaneously. When an acoustic flow of strong sound pressure passes through the junction between a tone hole and the main body of the bore, vortex shedding occurs, which induces unpleasant noises due to re-radiation of aerodynamics sound from the vortices. We have succeeded in reproducing this process and attempted to explain its mechanism from aerodynamic sound theory. We have also investigated how the position of the pad of a key above a tone hole, i.e., the distance between the pad and the top of the tone hole, affects the pitch of the excited wave in the pipe. Furthermore, we attempt to numerically reproduce the function of tone holes, namely change of notes. Opening and closing tone holes change the topology of bore geometry, which yields a moving boundary problem accompanied by topological change of numerical grids. We have developed a LES solver resolving the moving boundary problem and are planning to apply it to the problem of opening and closing tone holes.

11:00

2aMU7. Sound radiation of trained vocalizers. Braxton B. Boren and Agnieszka Roginska (Music and Audio Res. Lab., New York Univ., 30-91 Crescent St., 5B, Astoria, NY 11102, bbb259@nyu.edu)

Current research at NYU has focused on computational estimation of vocal loudness of George Whitefield, an Anglican preacher in the 18th century who reportedly spoke to crowds of 30,000 or more. After having established an overall level for his voice, we have begun investigating how his voice would have radiated spatially. Existing literature on the radiation of the spoken voice has focused extensively on the context of conversation in workspaces. These studies typically examine one soft, one normal, and one raised voice condition. Trained actors and orators, however, employ more methods of projection than are used in conversational speech and can achieve higher loudness as well. The radiation patterns from these types of communication have not been quantifiably studied yet. This paper investigates the radiation patterns of different methods of projection in trained vocalizers using relative intensity levels at 1 m from the front of the speaker. The results are compared to the existing data for conversational speech, and the implications to open-air oratory are discussed.

11:20

2aMU8. Vibrato rate variability in three professional singing styles: Opera, rock, and Brazilian country. Guilherme Pecoraro (Otolaryngology, Univ. of São Paulo Med. School, Rua Machado Bittencourt 361, Rua Marcelo Muller 1297, Sao Paulo, Sao Paulo 03223060, Brazil, guifono@hotmail.com), Daniella Curcio (Dept. of Morphology, Santa Casa School of Med. Sci. of São Paulo, São Paulo, SP, Brazil), and Mara Behlau (CEV, Ctr. for Voice Studies, São Paulo, SP, Brazil)

Vibrato is one of the most expressive aesthetic characteristics of singing voice. Indicative of good voice quality is typical of lyrical singing but it is also found in others styles of popular music. Acoustically, vibrato is defined as a long-term periodic modulation of the fundamental frequency. It occurs as a result of the laryngeal muscular system and is comprised of three main parameters: rate, extent, and amplitude variation. The main controversy refers to the physiological mechanism of vibrato production, specifically concerning its neurological conscious control, as well as the intra-subject variability of its acoustic parameters. In this study, we compare the characteristics related to vibrato rate (VR), assessing 423 emissions, from recorded samples, produced by 15 professional singers, publicly and artistically acclaimed in occidental culture, to represent three music styles: opera, rock and Brazilian country (sertanejo). We analyzed the samples through GRAM 5.01 and found that the VR was kept constant intra-subject, independently of the singing style. The mean values for the VR for opera and Brazilian country singers were higher than for rock singers. Effects of vocal training, kinship and aging on the vibrato rate, as well as technical skills to control it are objects of our future studies.

11:40

2aMU9. Phase-space visualization and real-time pitch tracking of vocal sounds. Dmitry Terez (SDI, 264 Eleanor, Cherry Hill, NJ 08003, dmitry-terez@gmail.com)

Novel techniques for sound waveform visualization and real-time pitch tracking are presented. The use of techniques is demonstrated on vocal sounds such as singing and speech. The visualization of a sound waveform in two- or three-dimensional reconstructed phase space obtained via signal time-delay embedding provides a compelling alternative to traditional spectral envelope, for example, for looking at a timbral structure of a sound. This phase-space visualization is performed continuously on a sample-by-sample basis and results in a moving trajectory on computer display—a living and breathing picture of voice in real time. In addition, pitch of voiced

sounds is detected and displayed in real time using original pitch tracking algorithm. The algorithm has minimal possible latency which allows obtaining reliable fundamental frequency estimates in less than two cycles of a quasi-periodic waveform, before human auditory system can register pitch sensation. The techniques are efficiently implemented in real-time software using ASIO drivers. The software can be used as a tool for teaching singing or vocal intonation patterns as it provides immediate visual feedback on minute changes in pitch, timbre, or loudness.

TUESDAY MORNING, 4 JUNE 2013 511BE, 9:00 A.M. TO 12:00 NOON

Session 2aNSa

Noise: Transportation Noise

Kenneth Kaliski, Chair
RSG Inc., 55 Railroad Row, White River Junction, VT 05001

Contributed Papers

9:00

2aNSa1. Automatic classification of road vehicles considering their pass-by acoustic signature. Xavier Valero Gonzalez and Francesc Alías Pujol (La Salle-Universitat Ramon Llull, Quatre Camins, 30, Barcelona 08022, Spain, xvalero@salleurl.edu)

In order to assess the impact of environmental noise on a community, it is essential to accurately describe all the aspects and characteristics of the encountered noises. In this context, it would be of special interest to dispose of environmental noise monitoring stations capable of not only measuring the noise levels but also identifying the sources producing those levels. To offer such functionality, an algorithm to automatically recognize the noise sources is required. According to previous works, designing algorithms able to optimally distinguishing between road vehicle noise sources (i.e., light vehicles, heavy vehicles, and motorbikes) is a challenging issue. This paper proposes a recognition scheme that takes into account the perceived characteristics of road vehicles pass-by, which may be divided into different phases: approaching, passing and receding. By taking independent decisions for the pass-by phases, the proposed recognition scheme is able to improve the recognition of road traffic vehicles with respect to a traditional recognition scheme, specifically in 7% for light vehicles and in 4% for heavy vehicles.

9:20

2aNSa2. An acoustic based method for jointly estimating speed and wheelbase length of two-axle road vehicles as they pass by. Patrick Marmaroli, Xavier Falourd, and Lissek Hervé (Ecole Polytechnique Fédérale de Lausanne (EPFL), EPFL IEL STI LEMA, ELB 033, Station 11, Lausanne 1015, Switzerland, patrick.marmaroli@epfl.ch)

This paper focuses on acoustic road traffic monitoring and looks, more specifically, into the problem of speed and wheelbase length estimation of two-axle vehicles as they pass by. It is known that both front and rear axle trajectories may be dissociated using cross-correlation based methods in conjunction with a well designed two-element microphone array placed on the roadside. This is mainly due to the broadband nature of the tyre/road noise which makes two peaks appear, one per axle, in the correlation measurement when the vehicle is in the broadside direction. This paper aims at analyzing such a "bimodal'' observation in order to automatically extract the position, speed, and wheelbase length of passing-by vehicles. We propose to conduct this tracking problem using a particle filter that model the position-variant bimodal sound source nature of the vehicles. The theoretical developments presented in this paper are experimentally assessed through real *in-situ* measurements.

9:40

2aNSa3. The effect of the shadowing phenomenon on emergency vehicle siren noise. Peter J. D'Angela, Frankie Angione, Colin Novak, and Helen Ule (Univ. of Windsor, 2313 St. Clair, Windsor, ON N9E4S7, Canada, dangelp@uwindsor.ca)

It has been observed by some that emergency siren noise has gone unnoticed by drivers due to a shadowing phenomenon where the propagating siren noise is blocked from a receiver vehicle. The event is postulated to occur when a large vehicle is positioned between the emergency responder and a receiving vehicle. The sound of the siren is projected along the surface of the large vehicle and does not fall in time to reach the receiving vehicle. This situation is common at controlled intersections where the smaller vehicle is traveling perpendicular to the emergency vehicle but can also occur when the vehicles are in a common line on the road. The intent of this study is to investigate this phenomenon and quantify the resulting hindrance of a driver's ability to detect an approaching emergency vehicle. Included will be the use of the electrical "wail" siren and accompanying air horn commonly employed by Fire and Rescue Services. The outcome will be a determination of what frequency spectra are most affected by shadowing with an eventual goal to improve emergency siren design.

10:00

2aNSa4. Detectability study of warning signals in urban background noises: A first step for designing the sound of electric vehicles. Nicolas Misdariis, Anais Gruson, and Patrick Susini (UMR STMS Ircam-CNRS-UPMC, IRCAM, 1, place Igor Stravinsky, Paris F- 75004, France, misdarii@ircam.fr)

Electric vehicles tend to become a growing category of today's human means of transport. But, because these kind of vehicles are actually quiet, or even silent, the question of a dedicated sound design arises almost inevitably in order to make them more present—then secure—both for their proximity (pedestrians) and their users (driver). This being, current issues for a sound design research framework is then to exploit and explore sound properties that, first, will fix a goal of functionality (emergence, recognition, acceptance) and, second, will define guidelines for the development of new aesthetics to be included in a general design approach. Thus, a first study focusing on detection of warning signals in urban environments was achieved. Based on the state-of-the-art, a corpus of elementary signals was built and characterized in a time/frequency domain for representing basic temporal and spectral properties (continuous, impulsive, harmonic, etc.). A corpus of representative urban environments was also recorded and realistic sequences were mixed with a dynamic approaching-source model. A

reaction time experiment was conducted and leads to interesting observations: especially, specific properties promoting the emergence. Moreover, a seemingly significant learning effect also rises from the data and should be further investigated.

10:20

2aNSa5. Detectability and annoyance of warning sounds for electric vehicles. Etienne Parizet and Ryan Robart (Laboratoire Vibrations Acoustique, INSA-Lyon, 25 bis, av. Jean Capelle, Villeurbanne 69621, France, etienne.parizet@insa-lyon.fr)

Electric or hybrid vehicles are very quiet at low speeds, which represents a very good opportunity to reduce traffic noise annoyance in cities. On the other hand, this may be very hazardous for vulnerable pedestrians (e.g., visually impaired people). The aim of the eVADER project is to propose solutions in order to add warning sounds to such cars, while fulfilling two contradictory goals: sounds should be detectable but should not contribute to traffic noise annoyance. Different perceptual experiments have been conducted: the first one evaluated the influence of various timbre parameters on sound detectability. It was shown that an electric vehicle equipped with one particular sound was as easily detected as a diesel one, while keeping a very low level. Then, the influence of some timbre parameters (pitch and temporal modulation frequency) on the distance and speed as perceived by listeners was measured. These two experiments were conducted with sighted and visually impaired subjects. Finally, a third one evaluated the consequence on traffic noise annoyance of such warning sounds.

10:40

2aNSa6. Measurement, evaluation, and analysis of noise and vibrations produced by an Argentinean medium tank. Alan Chorubczyk, Francisco Ruffa (Sound Eng., Tres de Febrero Univ., Amenabar 1819, Ciudad Autónoma de Buenos Aires 1428, Argentina, alanchoru@gmail.com), Nicolás Urquiza (Sound Eng., Tres de Febrero Univ., Caseros, Argentina), Pablo Margaretic (Sound Eng., Tres de Febrero Univ., Bernal, Buenos Aires, Argentina), Damián Morandi (Sound Eng., Tres de Febrero Univ., Bella Vista, Argentina), Andrés Piegari, and Sebastián Ausili (Sound Eng., Tres de Febrero Univ., Ciudad Autónoma de Buenos Aires, Argentina)

In the present paper it is presented the procedure of measurement, evaluation and results' analysis of an Argentinean medium tank T.A.M. Since there is no regulation on noise emissions and vibrations caused by combat units either inside or outside the vehicle, standards that evaluate similar situations were used. Then, noise and vibrations inside the moving vehicle and noise emissions of acceleration and static situations were assessed. Consequently, the procedure and results were analyzed in order to propose a proper heavy combat units assessment procedure.

11:00

2aNSa7. Contribution analysis of vehicle exterior noise with operational transfer path analysis. Jakob Putner (AG Technische Akustik, MMK, Technische Universität München, Arcisstraße 21, Munich 80333, Germany, putner@tum.de), Martin Lohrmann (Müller-BBM VibroAkustik Systeme GmbH, Planegg, Germany), and Hugo Fastl (AG Technische Akustik, MMK, Technische Universität München, Munich, Germany)

Vehicle development regarding the emitted exterior noise is a challenging task. In addition to stringent legal requirements to reduce noise exposure, also high expectations of the sound quality have to be considered during the development process. In order to manipulate the vehicle exterior noise in a manner more efficient than trial and error, knowledge about the vehicle's sound sources, and their contributions to the overall noise is essential. In order to analyze the contributions of the several sound sources of a vehicle to the exterior noise Operational Transfer Path Analysis is used in the presented experiment. Therefore, transfer characteristics are estimated from measurements of the vehicle in typical operating conditions on an acoustic roller dynamometer. These data are used to synthesize the contributions at the response positions, i.e., the microphones of a simulated pass-by array, which also allow the simulation of the contributions during a pass-by measurement. Outcomes of the Operational Transfer Path Analysis are comprehensible contributions of the dominant sound sources to the vehicle exterior noise. The validation of the analysis results shows very good accordance between the simulated and measured overall vehicle exterior noise.

11:20

2aNSa8. Improving the acoustic performance of low noise road surfaces using resonators. Manuel Männel (Müller-BBM, Robert-Koch-Str. 11, Munich 82152, Germany, manuel.maennel@muellerbbm.de), Jens Forssén, and Bart van der Aa (Appl. Acoust., Chalmers Univ. of Technol., Gothenburg, Sweden)

Road surfaces made of porous asphalt are widely used to reduce the tire-road-noise generated during the rolling process of passenger cars and trucks. As the engine noise was reduced significantly in the last decades the tire-road-noise is the main sound source for driving speeds of 40 km/h (25 mile/h) and higher for passenger cars. This means that low noise road surfaces may not only be used on highways but also on inner-city main roads to generate a significant reduction on traffic noise. However, the acoustic performance of road surfaces made of porous asphalt is limited as a result of the trade-off between acoustic properties and road surface durability. By including resonators, e.g., of Helmholtz type in the porous road surface, it is possible to improve its absorbing performance without loss in durability. The paper describes recent research activities on such resonators in porous road surfaces made in the European project HOSANNA. The acoustic properties in terms of insertion loss have been calculated for different arrays of resonators. Measurements on realized porous road surfaces including resonators were carried out. The results show that resonators can improve the acoustic performance of porous road surfaces substantially.

11:40

2aNSa9. Guidance for new policy developments on railway vibration. Eulalia Peris, James Woodcock, Gennaro Sica, Calum Sharp, Andy Moorhouse, and David Waddington (The Univ. of Salford, Flat 2, 2 Claremont Grove, Manchester M202GL, United Kingdom, E.Peris@salford.ac.uk)

Vibration is one of the main problems associated with railways in residential areas. To ensure quality of life and well being of inhabitants living in the vicinity of route paths, it is important to evaluate, understand, control, and regulate railway noise and vibration. Much attention has been focused on the impact of noise from railway but the consideration of railway-induced vibration has often been neglected. This paper aims to provide policy guidance based on results obtained from the analyses of relationships estimated from ordinal logit models between human response and vibration exposure. This was achieved using data from case studies comprised of face-to-face interviews and internal vibration measurements ($N = 755$) collected within the study "Human Response to Vibration in Residential Environments" by the University of Salford. First, the implications of neglecting vibration in railway noise policies are presented. Second, the influence of different times of day when residents are exposed to railway vibration are presented and compared to current standards. Finally, the main factors that were found to influence railway vibration annoyance are presented and expressed as weightings. This work will be of interest to researchers and environmental health practitioners involved in the assessment of vibration complaints, as well as to policy makers, planners, and consultants involved in the design of buildings and railways.

Session 2aNSb

Noise: Distinguished Lecture

Victor Sparrow, Chair
Grad. Program in Acoust., Penn State, 201 App. Sci. Bldg., University Park, PA 16802

Chair's Introduction—8:55

Invited Paper

9:00

2aNSb1. The work of the Committee on Aviation Environmental Protection and the Development of International Noise Standards. Jane Hupe (Environ. Branch, Air Transport Bureau, ICAO, 999 Rue Univ., Montreal, QC H3C 5H7, Canada, jhupe@icao.int)

Environmental Protection is one of the Strategic Objectives of ICAO. The overall aim is to minimize the adverse environmental effects of global civil aviation activity. One of the key objectives is to establish noise Standards to limit and reduce the number of people affected by aircraft noise. This mandate is carried out by the Committee on Aviation Environmental Protection (CAEP), which, as a technical committee of the ICAO Council, is a recognized international forum of environmental experts from both member and observer States, intergovernmental organizations, including airlines, aircraft and engine manufacturers, airports, environmental non-governmental organizations, and UN bodies. ICAO has set International Standards for aircraft noise certification since the 1970s, and the purpose of this talk is to describe the process of developing these Standards while providing some details on recent developments, including the key outcomes of three years' worth of research leading up to the ninth meeting of the CAEP.

TUESDAY MORNING, 4 JUNE 2013 511CF, 10:00 A.M. TO 11:00 A.M.

Session 2aNSc

Noise and ASA Committee on Standards: International Aviation Noise Standards

Victor Sparrow, Chair
Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802

Invited Papers

10:00

2aNSc1. Setting noise stringency by international consensus. Yves R. Cousineau (Civil Aviation, Transport Canada, Place de Ville, 330 Sparks St., Ottawa, ON K1A 0N8, Canada, yves.cousineau@tc.gc.ca)

The paper will evoke the background of aviation noise and examine the international consensus process used to set aircraft noise stringency requirements and present the role of controlling the noise at the source within the context of the overall community noise issue. The paper will also examine the role of technology in this process and examine the growing interdependencies of noise reduction technology on CO_2 emissions, and on other emissions that impact air quality.

10:20

2aNSc2. Developing noise standards for future supersonic civil aircraft. Robbie Cowart (Gulfstream Aerosp. Corp., POB 2206, M/S R-07, Savannah, GA 31402, robbie.cowart@gulfstream.com)

With renewed interest in civil supersonics, NASA and industry researchers continue to make progress toward enabling quiet civil supersonic aircraft. Gulfstream Aerospace has long been interested in the development of an economically viable supersonic business jet; however, many regulatory challenges still remain for routine supersonic operation. Gulfstream's approach includes unrestricted supersonic flight over land to enable the same operational flexibility of its subsonic fleet. The largest technical barrier to achieving this end is mitigating the sonic boom created by flying at cruise speeds greater than Mach 1.2. At present, the United States and many other countries prohibit supersonic flight over land due to the loudness and public annoyance associated with sonic boom noise. In the United States, the FAA prohibits supersonic flight under FAR 91.817. Although the FAA has shown interest in reconsidering its position, the agency supports the noise and emissions standards setting process through the International Civil Aviation Organization and its

Committee on Aviation Environmental Protection. Development of future standards for sonic boom noise is a key component to enabling continued investment in civil supersonic research. This paper will outline the steps currently underway to assess the viability of defining low amplitude supersonic signature acceptability.

10:40

2aNSc3. Aircraft noise technology review and medium and long term noise reduction goals. Brian J. Tester (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom), Dennis Huff (Aeropropulsion, John H. Glenn Res. Ctr., Cleveland, OH), and Luc Mongeau (Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada, luc.mongeau@mcgill.ca)

This presentation will summarize the recommendations of the second ICAO Committee on Aviation Environmental Protection noise technology independent expert panel. The technologies being developed for reducing aircraft community noise in the mid term (10 years) and the long term (20 years) were reviewed. The review consisted of detailed presentations on aircraft noise reduction by various industry and government representatives on component noise reduction programs, highlighting novel noise reduction concepts being pursued and the progress that has been demonstrated or projected so far.

TUESDAY MORNING, 4 JUNE 2013 — 519B, 8:55 A.M. TO 10:20 A.M.

Session 2aPAa

Physical Acoustics: Nanoacoustics

Srikantha Phani, Chair
Dept. of Mech. Eng., The Univ. of British Columbia, 6250 Appl. Sci. Lane, Vancouver, BC V6T1Z4, Canada

Chair's Introduction—8:55

Invited Paper

9:00

2aPAa1. Bandgap formation mechanisms in periodic materials and structures. Lalitha Raghavan and Srikantha Phani (Dept. Mech. Eng., The Univ. of British Columbia, 6250 Appl. Sci. Lane, Vancouver, BC V6T1Z4, Canada, srikanth@mech.ubc.ca)

Bandgaps are frequency intervals of no wave propagation for mechanical waves. Spatial periodicity provides one mechanism for the emergence of bandgaps in periodic composite materials such as lattice materials, phononic crystals, and acoustic metamaterials. Coupling a propagating wave in a periodic medium with a local resonator provides an alternate mechanism for band-gap emergence. This study examines these two band-gap formation mechanisms using a receptance coupling technique. The receptance coupling technique yields closed-form expressions for the location of bandgaps and their width. Numerical simulations of Blochwaves are presented for the Bragg and sub-Bragg bandgaps and compared with the bounding frequency predictions given by the receptance analysis of the unitcell dynamics. It is observed that the introduction of periodic local resonators narrows Bragg bandgaps above the local resonant bandgap. Introduction of two fold periodicity is shown to widen the Bragg bandgap, thus expanding the design space. The generality of the receptance technique presented here allows straightforward extension to higher dimensional systems with multiple degrees of freedom coupling. Implication of this study for nano-electro-mechanical systems (NEMS) based filters will be discussed.

Contributed Papers

9:20

2aPAa2. Multi-resonance transduction near acoustic Brillouin zone in microscale ferroelectric. Igor Ostrovskii (Phys. and NCPA, Univ. of Mississippi, Oxford, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu) and Lucien Cremaldi (Phys. and Astronomy, Univ. of Mississippi, University, MS)

Excitation of the acoustic plate waves (PAW) in ferroelectric wafers with microscale domains and nanoscale interdomain walls is investigated theoretically and experimentally. The periodically poled structures were fabricated in the Z-cut 0.5-mm-thick $LiNbO_3$ wafers. Rf-current applied along the X-axis generates PAW in the fabricated multidomain acoustic superlattices having stop-bands and acoustic Brillouin zones. Two main effects are observed. First, a frequency of maximum acoustic amplitude does not coincide with the domain resonance frequency when a half-wavelength equal to domain length. Second, instead of known single-frequency domain resonance such as in bulk crystals, the series of two or more transduction resonances do exist. The theory, simulation, and experiments allow concluding that the transduction multi-resonances occur due to ultrasound diffraction by the interdomain nano-walls when the acoustic wavelength is close to multidomain period; it happens near acoustic Brillouin zone boundaries. Since different PAW-modes may be excited, the transduction multi-resonances appear at different frequencies. For example, 300-micron multidomain structure in $LiNbO_3$ demonstrates multiple transduction peaks in the frequency range of 5.6 to 8.3 MHz, just for lowest four symmetric and anti-symmetric PAW modes. The experimental results are in agreement with theory. The findings may be applied for designing new MEMS and therapeutic ultrasonic transducers.

9:40

2aPAa3. Using nonlinear ultrasound to measure microstructural changes due to radiation damage in steel. Kathryn H. Matlack (G.W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 788 Atlantic Dr., Sustainable Education Bldg., Ste. 306, Atlanta, GA 30332, katie.matlack@gatech.edu), Jin-Yeon Kim (School of Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA), James J. Wall (Elec. Power Res. Inst., Charlotte, NC), Jianmin Qu (Dept. of Civil and Environ. Eng., Northwestern Univ., Evanston, IL), and Laurence J. Jacobs (School of Civil and Environ. Eng., Georgia Inst. of Technol., Atlanta, GA)

This work demonstrates how nonlinear ultrasound (NLU) can be used to monitor radiation damage in nuclear reactor pressure vessel (RPV) steels. Radiation damage is a crucial concern in the nuclear industry since many nuclear plants throughout the United States are entering a period of life extension, meaning the RPV will be exposed to higher levels of neutron radiation than it was originally designed to withstand. Currently, there is no nondestructive evaluation (NDE) method to unambiguously characterize radiation damage in RPV steels, the development of which would enable the assessment of the integrity of the vessel, allowing operators to determine if they can continue to safely operate. NLU is an NDE technique that is sensitive to microstructural features in metallic materials. The physical effect monitored by NLU is the generation of higher harmonic frequencies in an initially monochromatic ultrasonic wave, arising from the interaction of the ultrasonic wave with microstructural features. Recent research has demonstrated that NLU is sensitive to the same microstructural changes that are produced in radiation damage, such as precipitate formation and changes in dislocation density. Current experimental and modeling results are presented that relate the nonlinear ultrasonic parameter to the amount of radiation damage in RPV steel materials.

10:00

2aPAa4. Synthesis and frequency dependent ultrasonic characterization of NiO-EG nanofluids. Meher Wan, Satyendra K. Verma (Phys. Dept., Univ. of Allahabad, Allahabad, UP 211002, India, meherwan24@hotmail.com), Dharmendra K. Pandey (Phys. Dept., PPN College, Kanpur Univ., Kanpur, UP, India), and Raja R. Yadav (Phys. Dept., Univ. of Allahabad, Allahabad, UP, India)

In the present paper, we have synthesized the NiO nanoparticles with chemical route. Further, the uniform suspensions of NiO nanoparticles in ethylene glycol of different concentrations have been prepared. Samples were characterized with Acoustical Particle Sizer (APS-100) for the frequency dependent ultrasonic attenuation in the respective samples; subsequently, the particle size determination and their distribution have been calculated with the help of ultrasonic attenuation. The structural parameters were also investigated with the microscopic techniques. There is good agreement between data produced by ultrasonic spectroscopy and the microscopic measurements.

TUESDAY MORNING, 4 JUNE 2013 — 519B, 10:20 A.M. TO 12:20 P.M.

Session 2aPAb

Physical Acoustics: Atmospheric Acoustics

D. Keith Wilson, Chair
U.S. Army Cold Regions Res. Lab., 72 Lyme Rd., Hanover, NH 03755

Contributed Papers

10:20

2aPAb1. Source localization results for airborne acoustic platforms. Vladimir E. Ostashev (Cooperative Inst. for Res. in Environmental Sci./Univ. of Colorado at Boulder and NOAA Earth System Res. Lab., Boulder, CO), Sylvain Cheinet (French-German Res. Inst. of Saint-Louis (ISL), 5 Rue General Cassagnou, Saint-Louis 68300, France, sylvain.cheinet@isl.eu), Sandra L. Collier, Chris Reiff, David A. Lygon (U. S. Army Res. Lab., Adelphi, MD), D. Keith Wilson (U.S. Army Cold Regions Res. and Eng. Lab., Hanover, NH), John M. Noble, and William C. Alberts, II (U. S. Army Res. Lab., Adelphi, MD)

Acoustic sensors are being employed on airborne platforms for source localization. Under certain atmospheric conditions, airborne sensors offer a distinct advantage over ground sensors. Among other factors, the performance of airborne sensors is affected by refraction of sound signals due to vertical gradients in temperature and wind velocity. A comprehensive experiment in source localization with an aerostat-mounted acoustic system was carried out in July 2010 at Yuma Proving Ground (YPG). Acoustic sources on the ground consisted of one-pound TNT denotations and small arms firings. The height of the aerostat was approximately 1 km above the ground. In this paper, horizontal, azimuthal, and elevation errors in source localization and their statistics are studied in detail. Initially, straight-line propagation is assumed; then refraction corrections are introduced to improve source localization and decrease the errors. The corrections are based on a recently developed theory [Ostashev *et al*., J. Acoust. Soc. Am. (2008)] that accounts for sound refraction due to vertical profiles of temperature and wind velocity. During the 2010 YPG field test, the vertical profiles were measured only up to a height of approximately 100 m. Therefore, the European Center for Medium-range Weather Forecasts (ECMWF) is used to generate the profiles for July of 2010.

10:40

2aPAb2. A numerical approach to the climatology of near-surface sound levels. Sylvain Cheinet (French-German Res. Inst. of Saint-Louis (ISL), 5 Rue General Cassagnou, Saint-Louis 68300, France, sylvain.cheinet@isl.eu)

The near-surface sound levels propagated at distance from a known source show a large variability on the long term. This variability is essentially caused by the weather-dependence of the refractive characteristics: wind and temperature stratifications, turbulence. An approach to document this variability is to simulate the sound propagation under these varying characteristics at the selected site. This study uses a numerical model which physically describes the sound propagation including in presence of turbulence. The model is based on the parabolic equation, it ingests standard atmospheric parameters as input. The predicted sound levels for an example 40 Hz-frequency sound propagating at a 1.5 km-range are shown to combine the impacts of stratification and turbulence. The results are used to form the sound level climatology at several sites over the globe, based on existing climatological data. The obtained statistics are modulated by the dominant wind regimes, the seasonal and diurnal cycles. The sensitivity of these results to turbulence assessment is discussed.

11:00

2aPAb3. Statistical moments of broadband acoustic signals propagating in a refractive, turbulent atmosphere with spatial-temporal fluctuations in temperature and wind velocity. Vladimir Ostashev, D. Keith Wilson, Sergey N. Vecherin (U.S. Army Cold Regions Res. and Engineering Lab., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@noaa.gov), and Sandra L. Collier (U. S. Army Res. Lab., Adelphi, MD)

Propagation of broadband acoustic signals in a refractive, turbulent atmosphere with spatial-temporal fluctuations in temperature and wind velocity is considered. Starting from a parabolic wave equation, and using the Markov approximation and the hypothesis of locally frozen turbulence, closed-form equations for the statistical moments of arbitrary order of the sound-pressure field are derived for both sound propagation above an impedance ground and line-of-sight propagation. These equations generalize those obtained previously [Wilson and Ostashev, J. Acoust. Soc. Am. **109**, 1909–1922 (2001)], where the case of monochromatic sound waves and spatial fluctuations in temperature and wind velocity was considered. The general theory can be used for analysis of many statistical characteristics of broadband acoustic signals propagating in the atmosphere, e.g., temporal coherence, frequency decorrelation, and the pulse spread and wonder. Using this theory, the spatial-temporal coherence function of a broadband acoustic signal is calculated and analyzed. Knowledge of the theoretical coherence function is important for performing source localization with physics-based estimators, particularly maximum-likelihood estimators.

11:20

2aPAb4. A physical model for predicting the sound speed and attenuation coefficient in Titan's atmosphere based on Cassini-Huygens data. Andi Petculescu (Physics, Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504, andi@louisiana.edu)

NASA and ESA are discussing plans for a collaborative mission to use montgolfieres to gather long-duration data in Titan's atmosphere. Acoustic sensors can listen for thunder, bolide explosions, wind noise, cryo-volcanoes, and many other phenomena. This emphasizes the need for accurate acoustic predictions for Titan. In 2005, during the descent of the Huygens probe on Titan, an active ultrasonic sensor measured the speed of sound over the last 12 km. Using the ambient pressure, density, temperature, and methane concentration measured by Huygens as inputs, as well as temperature- and pressure-dependent transport parameters extracted from NIST, a theoretical model has been developed to predict the sound speed and attenuation coefficient in Titan's atmosphere. Based upon non-ideal equations of state, the sound speed predictions agree quite well with Huygens measurements in the lower troposphere. The effect of measured zonal winds on tropospheric propagation is presented via ray-tracing, showing quiet zone predictions. The model can be extended to the upper atmospheric layers (since ambient data are available); nevertheless care must be taken to account for altitude dependent processes such as winds, clouds, aerosols, chemistry, gravity waves, etc. in order to increase the accuracy.

11:40

2aPAb5. Prediction of sound levels from high-altitude, broadband sources: Is there a Lloyd's mirror effect? D. Keith Wilson (U.S. Army Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), Vladimir E. Ostashev, and Sergey N. Vecherin (U.S. Army Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., Hanover, NH)

For a high-altitude, narrowband sound source, propagation models predict the presence of pronounced rings of constructive and destructive interference for listeners near the ground. This is an example of the Lloyd's mirror effect. But even when propagation predictions are made for realistic, broadband aircraft spectra, by partitioning the spectrum into octave or one-third octave bands, running the model at the center frequency of each band, and then summing, a residual Lloyd's mirror effect can still be apparent. By varying the approach to frequency selection in the calculation, it is shown that the rings actually have random locations and generally disappear when enough frequencies are sampled, thus implying that they are numerical artifacts. This outcome unfortunately implies that coherent calculations at many individual frequencies are required to perform a broadband calculation. Some techniques for improving convergence, such as importance sampling and performing calculations incoherently, are discussed.

12:00

2aPAb6. Pneumatic infrasound source: Theory and experiment. Justin Gorhum, Thomas Muir, Charles Slack, Martin Barlett, Timothy Hawkins (Appl. Res. Lab., The Univ. of Texas at Austin, P.O. Box 8029, Austin TX 78713, Austin, TX 78713, jgorhum@arlut.utexas.edu), Charles Tinney, and Woutjin Baars (Dept. of Aerosp. Eng., The Univ. of Texas at Austin, Austin, TX)

In prior work [J. Acoust. Soc. Am. **132**(3), 2074 (2012)], we experimentally demonstrated the feasibility of releasing compressed air through a rotating ball valve to create infrasound. The present paper seeks to analyze and model those as well as new outdoor measurements, with a view toward establishing a viable theoretical model of the process. Functions involving propagation and the response to frequency, source pressure, and signal type (tone burst and transient) are examined, as is the potential utility of the method in calibration and test applications. [Work supported by ARL:UT Austin.]

Session 2aPAc

Physical Acoustics: General Physical Acoustics I

Annie Ross, Chair
Dept. Mech. Eng., Ecole Polytechnique Montreal, CP 6079 Succ CEntre Ville, Montreal, QC H3C 3A7, Canada

Contributed Papers

9:00

2aPAc1. Guided waves scattering by discontinuities near pipe bends. Mihai V. Predoi (Mechanics, Univ. Politehnica Bucharest, Splaiul Independentei nr. 313, sect. 6, Bucharest, Bucharest 060042, Romania, predoi@cat.mec.pub.ro) and Cristian C. Petre (Strength of Mater., Univer. Politehnica Bucharest, Bucharest, Romania)

Guided waves became in recent years an useful tool in nondestructive testing of pipes used in many industrial applications. The torsional and longitudinal waves in pipes are the main choice for integrity inspection. The first step is the computation of the dispersion curves, for straight pipes. Since most pipes have bends, the problem of guided modes in the toroidal segment remains of interest. Various methods have been applied to solve this problem. The most promising numerical method to obtain the dispersion curves for a torus is based on finite elements (FE), using a standing waves model. Based on these dispersion curves, transmissions of longitudinal and torsional waves through a bend were also investigated. The present paper presents the scattering process produced by geometrical discontinuities such as circumferential welds before and after a pipe bend. Longitudinal L(0,2) mode is sent along the straight pipe in FE simulations, toward the bend. Reflected and transmitted modal amplitudes are determined for frequencies of interest. The capability of detecting a defect close to one of the two welds is thus assessed. The modes transmitted past the bend are also characterized. Comparisons with results obtained by other researchers are used to validate the method.

9:20

2aPAc2. Shear waves in inviscid compressible fluids. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

While elastic solids support compressional and shear waves, waves in ideal compressible fluids are usually thought of as compressional waves. Here, a class of acoustic-gravity waves is studied in which the dilatation is identically zero, and the pressure and density remain constant in each fluid particle. An exact analytic solution of linearized hydrodynamics equations is obtained that describes the shear waves in inhomogeneous, inviscid, compressible fluids with piece-wise continuous parameters in a uniform gravity field. The solution is valid under surprisingly general assumptions about the environment and reduces to some classical wave types in appropriate limiting cases. Free shear waves in bounded and unbounded domains as well as excitation of the shear waves by a point source are considered. Edge waves propagating along vertical and inclined rigid boundaries are found in rotating and non-rotating fluids. A possible role of the shear acoustic-gravity waves in a coupled ocean-atmosphere system is discussed.

9:40

2aPAc3. Photo-acoustic and ultrasonic investigation of the mixtures of water and several glycols. Wioletta Żwirbla, Bogumil B. Linde (Inst. of Experimental Phys., Univ. of Gdańsk, Wita Stwosza 57, Gdansk 80-952, Poland, fizbl@univ.gda.pl), and Ewa B. Skrodzka (Inst. of Acoust., Adam Mickiewicz Univ., Poznań, Poland)

In the paper, the results of the ultrasonic velocity and absorption measurements were presented, as well as the thickness the "ro" in water mixtures the polyethylene glycols (PEG-s), the ethylene glycol and diethylene glycol. The experiments were provided in the temperature range from 291.15 to 303.15 K for whole molar fraction. Adiabatic compressibilities were calculated from Laplace's equation based on the experimental results obtained. Variations of these values with concentration and temperature were studied. Structural interactions and the formation of a compact pseudostable structure at very low concentrations of ethylene glycol and polyethylene glycols were observed. The plots of the adiabatic compressibility versus the mole fraction of PEG and EG display two characteristic points at low concentrations: the intersection of the isotherms and their minimum. Such relations between adiabatic compressibility, concentration and temperature are usually attributed to the formation of pseudo-stable molecular structures. To formulate a model of local structures present in the investigated molecular systems it is indispensable to get an insight into hydration of molecules and the formation of hydrogen bonds. Therefore, the attention was focused particularly on these problems.

10:00

2aPAc4. Numerical simulations of evolution of weak disturbances in vibrationally excited gas. Igor Zavershinskii, Vlavimir Makaryan (Physics, Samara State Aerosp. Univ., Moskovskoe Sh., Samara 443086, Russian Federation, ipzav63@mail.ru), and Nonna Molevich (Theoretical Phys., P.N. Lebedev Physical Inst. of RAS, Samara Branch, Samara, Russian Federation)

We consider a model of gas with an exponential law of relaxation of vibrational states of molecules excited by the external energy source (Joule heating in discharges, exothermic chemical reactions, optical pumping, etc). In such a medium, the second (bulk) viscosity coefficient inversion can take place due to the positive feedback between the sound perturbation and the nonequilibrium heat release. Such a medium with the negative viscosity is acoustical active. The existence of stationary nonlinear acoustical structures that are different from the step- or saw -wise shock wave structures are discussed basing on the solutions of general nonlinear acoustical equation [Molevich, Klimov, Makaryan, Int. J. Aeroacoust. No. 3–4 (2005)]. Using the numerical simulation of full one -dimensional (1D) system of relaxing gas dynamics, we show that any weak localized acoustical disturbance transforms into the sequence of self-sustained solitary pulses. Collisions of such pulses lead to their full reconstruction after the initial stage of the nonlinear increase of summarized amplitude. Using the 2D-system of relaxing gas dynamics, we consider the evolution of the noise signal into the non-stationary quasi-regular system of colliding self-sustained solitary pulses.

10:20

2aPAc5. Device for ultrasound imaging of standing trees. Andres Arciniegas (Aix-Marseille Université, CNRS, LMA UPR 7051, 31 chemin Joseph-Aiguier, Marseille 13009, France, arciniegas@lma.cnrs-mrs.fr), Loïc Brancheriau, Philippe Gallet (PERSYST/DIR, CIRAD, Montpellier, France), and Philippe Lasaygues (Aix-Marseille Université, CNRS, LMA UPR 7051, Marseille, France)

The aim of ARB'UST project is to develop an ultrasonic device for parametric imaging of standing trees. The device is designed to perform both transmission and reflection measurements, used for quantitative tomographic imaging. It allows various automatic acquisitions since the angular position of sensors can be precisely adjusted. The electronics and associated configuration enable particularly the measurement of velocity and attenuation of the

ultrasonic waves during their propagation within the medium. Two tomography experiments were conducted on a plane tree sample (before and after drilling a hole) and tomograms were calculated by the "Layer Stripping" algorithm. Our first results show that the artificial defect is detected.

10:40

2aPAc6. Cell structure in waves diffracted by a wedge in three-dimensional space and its implications to the field theory. Mitsuhiro Ueda (Predio Meguro Sci. Lab., 4-20-13 Meguro, Meguro-ku, Tokyo 153-0063, Japan, ueda-mt@nifty.com)

In the previous meeting, we have reported that if the aperture angle of a wedge-like region in 2D space bounded by perfectly reflecting planes is expressed as πN/M where N and M are relatively prime integers, the region can be divided into N cells and diffracted waves in a cell can be reconstructed by the sum of those in N-1 cells remained. This property holds for any positions of observation and source points. In this paper, it is shown that the same cell structure exists in the 3D wedge-like region bounded by rigid planes. In this case, the time dependent explicit solution for diffracted waves is available in closed form involving elementary functions only. Consequently the existence of cell structure can be shown analytically whereas in the 2D case it was shown numerically since the stationary solution for diffracted waves is expressed in asymptotic form involving Bessel functions. It is not easy to notice the cell structure without the new physical principle of diffraction, that is, virtual discontinuity principle of diffraction that has been proposed by us. Some physical implications of the principle to the field theory are mentioned lastly.

TUESDAY MORNING, 4 JUNE 2013 514ABC, 9:00 A.M. TO 11:00 A.M.

Session 2aPPa

Psychological and Physiological Acoustics: Binaural Hearing and Binaural Techniques II

Janina Fels, Cochair
Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52074, Germany

Pablo Hoffmann, Cochair
Aalborg Univ., Fredrik Bajers Vej 7B5, Aalborg 9220, Denmark

Invited Papers

9:00

2aPPa1. Experiments on authenticity and naturalness of binaural reproduction via headphones. Janina Fels, Josefa Oberem, and Bruno Masiero (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52074, Germany, janina.fels@akustik.rwth-aachen.de)

Binaural stimuli presented via headphones need to be plausible in localization and sound coloration for a successful reproduction of an acoustic scene, especially for experiments on auditory selective attention. The goal is to provide artificially generated acoustic scenes in a way that the difference between a real situation and an artificially generated situation has no influence in psychoacoustic experiments. The quality and reliability of binaural reproduction via headphones comparing two different microphone setups (miniature microphone in open dome and ear plug) used for individualized head-related transfer functions and headphone transfer function measurements is analyzed. Listening tests are carried out focusing on authenticity, naturalness, and distinguishability in a direct comparison of real sources and binaural reproduction via headphones. Results for three different stimuli (speech, music, pink noise) are discussed. Furthermore, approaches to perform experiments on auditory selective attention with binaural reproduction versus dichotic reproduction are made.

9:20

2aPPa2. Perceptual equalization of artifacts of sound reproduction via multiple loudspeakers. Bernhard U. Seeber (Audio Information Process., Technische Universität München, Arcisstrasse 21, Munich 80333, Germany, seeber@tum.de) and Ervin R. Hafter (Dept. of Psychology, Univ. of California at Berkeley, Berkeley, CA)

Several techniques for reproducing spatial sounds via multiple loudspeakers have been developed in recent years. A key problem for such techniques are comb filter effects caused by the uncertainty of the receiver position when playing coherent sounds from multiple loudspeakers (spatial aliasing). Here we studied if panning between two closely spaced loudspeakers can create a virtual source that resembles that of a true source. This requires not only that panned direction and speaker position correspond, but also that source width, loudness, timbre, and temporal aspects are reproduced without perceptual error. A listening experiment in an anechoic chamber showed that panned sources differ primarily in loudness and timbre from a real source at the panned location. The artifacts are caused by effects of the head, and we investigated if they can be compensated by filtering the sounds. Compensation filters were derived from simulations of the sound field at the ears. Listening tests showed that compensation filters reduced panning errors to be nearly inaudible and level roving or reflections in the reproduction room made errors inaudible. We conclude that a simple equalization is sufficient to render panned sources from nearby speakers perceptually equivalent to real sources.

9:40

2aPPa3. Physical correlates of loudness transfer functions in binaural synthesis. Florian Völk and Hugo Fastl (AG Technische Akustik, MMK, Technische Universität München, Arcisstraße 21, München 80333, Germany, florian.voelk@mytum.de)

The frequency dependent level correction necessary for a binaural synthesis system to elicit via headphones the reference scene loudness of narrow-band signals is referred to as loudness transfer function. An ideal binaural synthesis system provides frequency independent loudness transfer functions for every listener. The frequency dependence of the average of a binaural synthesis system's individual

loudness transfer functions has been shown to depend on the degree of individualization of the binaural synthesis system. In this contribution, perceptually acquired loudness transfer functions are compared from an auditory-adapted perspective to physical parameters of signals involved in the binaural synthesis process. The results provide quantitative relations between individual physical cues of the binaural synthesis output signals and the resulting loudness transfer functions.

10:00

2aPPa4. Toward a listening in spatialized noise test using complex tones. Jorg M. Buchholz, Harvey Dillon, and Sharon Cameron (Australian Hearing, National Acoust. Lab., 126 Greville St., Chatswood, NSW 2067, Australia, jorg.buchholz@nal.gov.au)

The Listening in Spatialized Noise-Sentences (LiSN-S) test has been widely applied to diagnose spatial processing disorder in both normally hearing and hearing impaired listeners who are proficient in English. The overall goal of the present study is to develop a spatial listening test that assesses similar spatial auditory processes as the LiSN-S test but does not rely on speech input and thus is language independent. Therefore, a three-alternative forced choice (3AFC) stream segregation task was implemented using a series of continuously in- or decreasing tone-complexes as targets and random tone-complexes as distractors and foils. Similar to the LiSN-S test the signals were either spatially co-located or separated using non-individualized HRTFs and the difference in thresholds defined the spatial release from masking (SRM). In order to achieve similar large SRM effects (of up to 14 dB) as observed with the LiSN-S test in normal hearing listeners, temporal jitter had to be introduced. The effect of the amount of temporal jitter was investigated on the SRM as a function of tone-complex duration. The results revealed that a jitter of about 30ms in combination with a tone-complex duration of about 30ms is sufficient to elicit the desired SRM.

10:20

2aPPa5. Auditory discrimination on the distance dependence of near-field head-related transfer function magnitudes. Yu Liu and Bosun Xie (Acoust. Lab., Phys. Dept., School of Sci., South China Univ. of China, Wushan Rd. 381#, Tianhe District, 301 Bldg. 18, Guangzhou, GuangDong 510641, China, janworc@gmail.com)

Distance dependence of head-related transfer functions (HRTFs) for nearby sound sources within 1 m is regarded as one of auditory distance perception cues. Accordingly, the near-field HRTFs have been used to synthesize sound sources at different distances in virtual auditory display. The present work analyzes the audibility of variation in near-field HRTF magnitudes with source distance. The calculated near-field HRTFs from KEMAR artificial head with a distance resolution of 0.01 m and a binaural model are used in analysis. The changes in loudness level spectra and interaural level difference are used as audible criteria. The result indicates that, as source distance decreases, the variation in HRTF magnitude with source distance become prominent and thereby audible. A psychoacoustic experiment is also carried out to validate the analysis. This work provides insight into the distance resolution of near-field HRTFs required in binaural virtual source synthesis.

10:40

2aPPa6. Issues in binaural hearing in bilateral cochlear implant users. Alan Kan, Heath Jones, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, ahkan@waisman.wisc.edu)

Despite the success of bilateral cochlear implants (CIs) in restoring sound localization abilities in profoundly deaf individuals, their localization accuracy is still poorer than that of normal hearing listeners. One factor could be the behind-the-ear location of the microphones. However, when CI users were tested with stimuli filtered through individualized head-related transfer functions (HRTFs), they showed very little difference in sound localization performance with different microphone locations (behind-the-ear versus in-the-ear). Another factor is the different implantation depths of the electrode arrays at the two ears. Since CIs are typically fitted independently in each ear at the clinic, it is likely that binaural information at a particular frequency can be presented mismatched across the ears. By simulating different amounts of interaural frequency mismatch at single electrode pairs, CI users showed poorer fusion and lower binaural sensitivity with increasing interaural mismatch. Good lateralization and fusion was achieved on or near a pitch-matched pair of electrode. Additionally, results from a separate study showed lateralization performance was typically maintained with simultaneous stimulation of multiple, pitch-matched pairs of electrodes. These results demonstrate methods beyond just changing the microphone position are needed to improve sound localization performance in CI users. [Work supported by NIH-NIDCD (R01-DC003083) and NICHD (P30-HD03352).]

Session 2aPPb

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Laurel H. Carney, Chair
Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642

Chair's Introduction—11:15

Invited Paper

11:20

2aPPb1. Physiological and behavioral studies of sound localization. Tom C. Yin (Neuroscience, Univ. of Wisconsin, 290 Med. Sci. Bldg., Madison, WI 53706, tcyin@wisc.edu)

A critical job of the auditory system is to localize sounds, which depends upon spectral cues provided by the filtering of the pinna for vertical localization and interaural time (ITDs) and level disparities (ILDs) for horizontal localization. We found anatomical and physiological specializations in the circuits that encode these cues. Cells in the medial superior olive (MSO) function as high resolution coincidence detectors or cross-correlators and their inputs have enhanced temporal synchronization compared to auditory nerve fibers, and the speed of the calyx of Held synapse helps to convey the inhibitory input from the contralateral ear to the LSO synchronously with the excitatory input from the ipsilateral ear even though it has to travel farther with an additional synapse. We have also been studying the psychoacoustics of sound localization in the cat by training them to look at sounds. Cats demonstrate high accuracy and precision when localizing with their head unrestrained. Their mobile ears have a reflex in response to head movement that keeps the ears pointed toward the sound source despite head movements. Cats also experience the precedence effect and a physiological correlate of the effect can be seen in recordings from the inferior colliculus.

TUESDAY MORNING, 4 JUNE 2013 512CG, 8:55 A.M. TO 10:40 A.M.

Session 2aSA

Structural Acoustics and Vibration: History and Application of Constrained Layer Damping

J. Gregory McDaniel, Cochair
Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215

Kenneth G. Foote, Cochair
Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543

Chair's Introduction—8:55

Invited Papers

9:00

2aSA1. Analysis and optimization of constrained layer damping treatments using a semi-analytical finite element method. James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, jgm@bu.edu) and Elizabeth A. Magliula (Vehicle Dynam. and Signature Control Branch 8233, Naval Undersea Warfare Ctr. Div. Newport, Newport, RI)

The present work investigates the physics of constrained layer damping treatments for plates and beams by a semi-analytical finite element method and presents applications of the method to the optimization of damping treatments. The method uses finite element discretizations in the thickness coordinate and propagating wave solutions in the remaining coordinates and therefore provides more generality and accuracy than existing analytical approximations. The resulting dispersion equation is solved for complex-valued wave numbers at each frequency of interest. By choosing sufficiently fine discretizations in the thickness coordinate, the method gives accurate estimates of wave numbers. The numerical implementation of the method is an efficient yet general tool for optimizing damping treatments with respect to material properties and dimensions. It explicitly allows for the possibility of analyzing structures with several layers where the material of each layer may be isotropic or orthotropic. Examples illustrate the numerical efficiency of the implementation and use this efficiency to provide optimizations of constrained layer damping treatments.

9:20

2aSA2. An overview of constrained-layer damping theory and application. Benjamin Shafer (Building Acoust., Conestoga-Rovers & Assoc., Inc., 1117 Tacoma Ave. South, Tacoma, WA 98402, bshafer@craworld.com)

Beginning in the early 1930s a variety of theoretical and experimental research has been published regarding the development and use of damping. What began as an experiment to reduce noise and vibration in metals and plastics has become a common treatment in an amalgam of applications. Constrained-layer damping (CLD) is a specific method of treatment commonly used in the aerospace and military industries. CLD may be described as a type of shear-related energy dissipation achieved by interconnecting two or more structural materials using a relatively thin viscoelastic layer. Among the advantages of using CLD as a damping treatment are the ability to obtain high loss factors with relatively thin configurations and that the stiffness of the composite system is not markedly increased. The analytic development of constrained-layer damping will be presented along with a brief discussion of the applications of CLD throughout history.

Contributed Papers

9:40

2aSA3. Numerical prediction of the vibroacoustic of sandwich panels with add-on damping. Imen Rzig and Noureddine Atalla (Mechanical, Univ. of Sherbrooke, E3-2115, 2500 Boulvard de l'université, Sherbrooe, Qc, QC J1K2R1, Canada, imen.rzig@usherbrooke.ca)

This paper discusses the numerical modeling of the vibroacoustic response of sandwich-composite panels with added-on damping, under mechanical and acoustical excitations. The studied damping is in the form of a viscolaetsic layer located within the panel. A modal synthesis approach is used for the calculation of the structural response and the Rayleigh's integral is used for the acoustic response (the panel is assumed flat and baffled). Since the panel has a viscoelastic core, a methodology is presented to handle efficiently the modeling of the frequency depended properties of the viscoelastic layer. A direct frequency response is used to validate the proposed approach. Next, a parameters study on the effect of the viscoelastic layer location is presented, In particular, three locations are compared: within the Honeycomb core, within the skins and added to the skin with a constraining layer. The effects of the excitation type on the vibration and acoustic response are also discussed.

10:00

2aSA4. A dynamic response of a laminated windshield with viscoelastic core—Numerical vs experimental results. Kaiss Bouayed (ESI Group, 20, rue du Fonds Pernant, Compiègne 60200, France, kaiss.bouayed@esi-group.com) and Mohamed-Ali Hamdi (Laboratoire ROBERVAL, Université de Technologie de Compiègne, Compiegne, France)

The dynamic response of a laminated windshield with a viscoelastic core is computed using a simplified modal method combined with a quadratic sandwich finite element. The method is based on a modal expansion of the displacement field using a constant young modulus of the core layer. The frequency dependence of the complex modulus of the core is taken into account using the residual dynamic stiffness matrix. The method is applied to predict the frequency response of two types of laminated windshield using a standard and acoustic PVB cores. Numerical results are compared in a first step with those obtained using the direct solver of Nastran software, and in a second step with experimental results obtained by a laser vibrometer. Comparisons show a very good agreement between experimental and numerical results and demonstrate the efficiency of the simplified modal solving method and the developed parabolic sandwich element. The method will be applied to compute the coupled vibro-acoustic frequency response of a full vehicle body integrating a laminated windshield and glass surfaces.

10:20

2aSA5. Nonlinear moduli estimation for rubber-like media with local inhomogeneities elastography. Timofey Krit, Valeriy Andreev, and Victor Kostikov (Dept. of Acoust., Faculty of Phys., Moscow State Univ., Leninskie Gory, Bldg. 1/2, Moscow, Moscow 119991, Russian Federation, timofey@acs366.phys.msu.ru)

Static shear deformations of a plane-parallel layer of rubber-like material created simultaneously with the uniaxial compression are considered. The layer is fixed between the rigid plates. Displacement of one plate relative to the other resulted in shear strain of the layer. This strain could reach 0.6 of the layer thickness. At such strain, effects due to the cubic nonlinearity arise. It is shown that measuring the dependence of the shear stress on the shear strain along one axis at different compression along the perpendicular axis one could determine nonlinear Landau parameters. The measurements were performed in two layers of polymeric material plastisol of 7 mm thickness with a rectangular base 8.9×8.9 cm, mounted between three aluminum plates. The upper plate was loaded with masses ranging from 0 to 25 kg and was fixed in each series of the stress-strain measurements. The values of the Landau coefficient A were measured in layers with different value of linear shear modulus. [Work supported by the Russian Foundation for Basic Research (Grant Nos. 12-02-00114 and 12-02-31418), and grant of the Government of the Russian Federation 11.G34.31.0066.]

Session 2aSC

Speech Communication: Linking Perception and Production (Poster Session)

Meghan Clayards, Chair
McGill Univ., 1085 Ave. Dr. Penfield, Montreal, QC H3A 1A7, Canada

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

2aSC1. The effect of accommodation on perceived vocal aesthetics. Grant L. McGuire (Linguistics, UC Santa Cruz, 1156 High St., Stevenson Acad. Services, Santa Cruz, CA 95064, gmcguir1@ucsc.edu), Molly Babel, and Jamie Russell (Linguistics, Univ. of British Columbia, Vancouver, BC, Canada)

We conducted an auditory naming task (n = 20) using eight model talker voices previously rated for attractiveness and prototypicality such that the most attractive, least attractive, most typical, and least typical voice for each gender served as a model talker. Female shadowers accommodated more than males, particularly to the Most Attractive Female model. This finding led us to question if in the course of accommodation to an attractive female voice, female shadowers themselves become more vocally attractive. We then conducted an AX task where listeners judged whether shadowers' baseline or shadowed productions were more attractive. Our results suggest that shadowers do modulate their perceived attractiveness in the course of accommodating; in particular, the more females accommodated to the Most Attractive Female model, the more attractive her own voice became. We are currently running a second study exploring whether shadowers' voices change in perceived typicality when accommodating the Most Typical and Least Typical voices, both of which also garnered large amounts of accommodation in the original auditory naming task. In general, our results demonstrate that the process of accommodation involves the mirroring of multidimensional speech characteristics, which in turn signal vocal aesthetics to listeners.

2aSC2. Coordinating conversation through posture. Martin A. Oberg, Eric Vatikiotis-Bateson, and Adriano Barbosa (Linguistics, UBC, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, martin.oberg@alumni.ubc.ca)

Conversation is dynamic and interactive. The importance of coordinated movement in conversation has been studied through perceptual measures of synchrony and recently through quantitative analyses of multi-dimensional motion data. The present study describes the postural system as being integrated with the communication process through an analysis of interlocutors' coordination of rigid-body head motion, postural shifts on forceplates, and motion computed from audio-visual recordings. Coordination is measured two ways: (1) holistically, as the scaling of speakers' motion over the duration of a conversation (i.e., the presence of movement encourages more movements) and (2) through analyses of the instantaneous correlation between motion signals from each speaker (i.e., a search for similar patterns of movement across time). These two approaches are evaluated in their ability to categorize conversation types. Preliminary results show that a stability emerges in the amount of correlation across conversations. Variations in the pattern of stability are analyzed as evidence of differences between general interactional coordination and linguistic coordination.

2aSC3. The role of voice similarity in accommodation. Sophie A. Walters, Molly Babel (Linguistics, Univ. of British Columbia, Totem Field Studios 2613 W Mall, Vancouver, BC V6T 1Z4, Canada, sophia.alex.walters@gmail.com), and Grant McGuire (Linguistics, Univ. of California Santa Cruz, Santa Cruz, CA)

Studies of accommodation show that some talkers are perceived as accommodating more than others. One possibility is that the similarity of the shadower's voice to a model talker's can account, in part, for the amount of perceived accommodation. To determine this, we conducted an auditory naming task having eight model talker voices previously rated for attractiveness and prototypicality, such that the Most Attractive and Least Attractive and Most Typical and Least Typical voices for each gender were used as models. Twenty participants completed an auditory naming task with these eight voices. A separate group of 20 listeners rated the similarity of model tokens and shadower's baseline productions using a visual analog scale. The results of this task were compared to the perceived accommodation results from a separate AXB rating task. Overall, female voices that were more different from the models showed more accommodation. This effect was not found for males, who generally showed less accommodation overall. These findings suggest that talkers either accommodate more when their voice is more distinct from the model talker's voice, or perhaps more likely, that such changes are more perceptible to listeners. Further explorations of the data are underway to tease apart these possibilities.

2aSC4. Training Korean second language speakers on English vowels and prosody. Dong-Jin Shin and Paul Iverson (Univ. College London, Room 326, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, dj.shin.09@ucl.ac.uk)

This study trained 36 Korean L2 speakers on vowel identification and prosody recognition (focus and lexical stress), with the aim of investigating the extent to which training improves general speech perception abilities or specific underlying processes. Vowel training was accomplished with a high-variability identification training technique (multiple talkers and words), and prosody training was accomplished using a category discrimination task in which they needed to choose sentences based on focus or words based on syllable stress. The results demonstrated that both trainers reduced vowel epenthesis and improved syllable stress perception, vowel training improved vowel identification more, and prosody training better improved focus perception in sentences. Both types of training can thus work in a complementary fashion to improve overall speech recognition.

2aSC5. Computer-based English /r/-/l/ perceptual training for Japanese children. Yasuaki Shinohara and Paul Iverson (Speech, Hearing and Phonetic Sci., Univ. College London, Rm. 326, 2 Wakefield St., London WC1N 1PF, United Kingdom, y.shinohara@ucl.ac.uk)

Computer-based perceptual training has proven successful for improving English /r/-/l/ perception by Japanese adults, but this has not been tested with younger age groups, who presumably have greater perceptual plasticity. The present study examined phonetic training for children 6–8 years old. The training program included identification and discrimination tasks with word-initial English /r/-/l/ minimal pairs (e.g., rock–lock), with each participant completing ten sessions. The results demonstrated that children improved their English /r/-/l/ identification, although identification in untrained positions such as medial and consonant clusters did not improve as much as in the trained word-initial position. In addition, older children in this age range improved more than did younger children, suggesting that the

ability to use this kind of program may improve with age, even though perceptual plasticity for speech presumably declines with age.

2aSC6. The role of acoustic/perceptual salience in directional asymmetry in infant stop/fricative contrast perception. Young-Ja Nam and Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, young.nam@mail.mcgill.ca)

The presence of stops in a language implicates the presence of fricatives but the reverse is unattested. Similarly, infants' producing fricatives implies that they acquired stops. The privileged status of stops influences infants' stop/fricative perception. For example, Altvater-Mackensen and Fikkert (2010) reported that Dutch-learning 14-month-olds noticed a fricative to stop change but not vice versa. These findings were interpreted in terms of phonological specifications while dismissing acoustic/perceptual factors. In this study, we assessed whether pre-linguistic infants show perceptual asymmetry. We tested English and French 4–5-month-olds using the look-while-listen procedure in which they were presented native nonsense syllables -/bas/ and /vas/. A mixed ANOVA showed a significant interaction between trial type and group ($p = 0.027$). Infants in /vas/-habituated group noticed the switch when the habituated fricative changed to a stop but infants in /bas/-habituated group did not notice the switch when the habituated stop changed to a fricative. This perceptual asymmetry in infants before babbling and word recognition stage indicates the potential role of acoustic/perceptual factors. We suggest that the above-mentioned privileged status of stops may reflect the possibility that stops are acoustically/perceptually more salient than fricatives. This salience difference is predicted to induce directional asymmetry in stop/fricative contrast perception.

2aSC7. Effects of acoustic variability on infant speech perception. Stephanie L. Archer (School of Commun. Sci. and Disord., McGill Univ., 6255 Rue Sherbrooke Ouest, Apt #5, Montreal, QC H4B 1M6, Canada, stephanie.archer2@mail.mcgill.ca), Suzanne Curtin (Psychology, Univ. of Calgary, Calgary, AB, Canada), and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

From birth, infants are capable of discriminating many of the speech sounds that occur cross-linguistically [Werker and Tees (1984)]. However, there are cases where the direction of presentation of contrasts reveals asymmetries in the discrimination of some speech contrasts, suggesting that some phonetic categories are more salient than others. These asymmetries may be due to an inherently uneven perceptual space [Polka and Bohn (2011)] or shaped by distributional properties of the input [Anderson *et al*. (2003)]. We explore whether acoustic variability also cause perceptual asymmetries. Six- and 9-month-olds participated in a discrimination task comparing English legal and illegal stop-liquid onsets (e.g., /kla/-/tla/ & /pla/-/tla/). Infants discriminated coronal versus bilabial onsets ($p < 0.05$), but not coronal versus velar ($p > 0.05$). Analysis of adult productions revealed that velar stop-liquid onsets showed more variability in their production, suggesting acoustic variability affects infants' perception. The current study provides a more direct test of the hypothesis that acoustic variability drives perceptual biases. Using the same clusters, we are exploring 9-month-olds' directional asymmetries. Preliminary results show that 9-month-olds successfully discriminate /dla/ after familiarization to /bla/ ($p = 0.05$; $n = 5$). Further investigation will reveal whether presentation direction affects infants' sensitivity to acoustic variability.

2aSC8. Infant recognition of infant vocal signals. Matthew Masapollo, Linda Polka (Commun. Sci. & Disord., McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca), Lucie Menard (Dept. of Linguistics, Univ. of Quebec at Montreal, Montreal, QC, Canada), and Athena Vouloumanos (Dept. of Psychology, New York Univ., New York, NY)

Most of the speech accessible to infants who are not yet babbling will not have infant vocal properties; yet their perception of infant vocal signals is critical for speech development. We report three experiments designed to assess whether 4- to 5-month-olds recognize and preferentially attend to vowels produced by infant talkers over vowels produced by adult talkers. Infants were tested in a sequential preferential looking procedure using isolated vowels synthesized by VLAM. In experiment 1, infants listened significantly longer to vowels produced by an infant talker than an adult (female). In experiment 2, infants failed to show any listening preference for infant versus adult vowels synthesized with matching (infant-appropriate) f0 values, suggesting that infants either recognized the un-natural pairing of f0 and formant structure in these adult vowels or are attracted to high f0 values. Failing to support the latter interpretation, infants in experiment 3 showed no listening preference when presented infant vowels with different (infant-appropriate) f0 values. Together, these findings suggest that young infants recognize the converging vocal (source and filter) properties that specify an adult and an infant talker. These recognition skills appear to be available prior to babbling, and thus are available to support early vocal learning.

2aSC9. Infants' categorization of vowels with infant vocal properties. Linda Polka, Matthew Masapollo (McGill Univ., 1266 Pine Ave. West, Montreal, QC, Canada, linda.polka@mcgill.ca), and Lucie Menard (Dept. of Linguistics, Univ. of Quebec at Montreal, Montreal, QC, Canada)

Prior research shows that infants can recognize phonetic equivalence among vowels produced by adult men, women, and children. It is unknown whether this ability extends to infant vowel productions, which have unique properties due to the size and geometry of the infant vocal tract. The present study was undertaken to determine whether infants recognize infant vowel productions as phonetically equivalent to vowels produced by adults and children. Infants (4–6 months) were tested in a look-to-listen procedure using isolated vowels, /i/ and /a/, synthesized to simulate productions by men, women, children and a 6-month-old. Infants were first habituated to diverse productions of the same vowel produced by several adult male, female, and child speakers while they fixated on a checkerboard. Following habituation, infants were then presented infant productions of the same vowel (familiar) and the other vowel (novel) in four test trials. A novelty effect (novel > familiar) was observed showing that infants recognized the familiar infant vowel to be similar to the habituation vowel. The findings are discussed in terms of the emergence of perceptual constancy in the development of vowel perception raising issues about how and when such knowledge is acquired in relation to the infant's own productions.

2aSC10. Estimation of vocal tract area functions in children based on measurement of lip termination area and inverse acoustic mapping. Kate Bunton, Brad H. Story (Speech, Language, and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, bunton@u.arizona.edu), and Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT)

Although vocal tract area functions for adult talkers can be acquired with medical imaging techniques such as magnetic resonance imaging (MRI), similar information concerning children's vocal tracts during speech production is difficult to obtain. This is largely because the demanding nature of the data collection tasks is not suitable for children. The purpose of this study was to determine the feasibility of mapping formant frequencies measured from the [i, ae, a, u] vowels produced by three children (age range 4 to 6 years), to estimated vocal tract area functions. Formants were measured with a pitch-synchronous LPC approach, and the inverse mapping was based on calculations of acoustic sensitivity functions [Story, J. Acoust. Soc. Am. **119**, 715–718]. In addition, the mapping was constrained by measuring the lip termination area from digital video frames collected simultaneously with the audio sample. Experimental results were augmented with speech simulations to provide some validation of the technique. [Research supported by NIH R01-DC011275.]

2aSC11. Acoustic characteristics of Danish infant directed speech. Ocke-Schwen Bohn (Ctr. on Autobiographical Memory Res., Dept. of Psychology and English, Aarhus Univ., Sejts Alle 20a, Risskov DK-8240, Denmark, engosb@hum.au.dk)

Danish presents several challenges for language learners, such as a very densely packed upper portion of the acoustic vowel space, and a sibilant contrast that is acoustically less distinct than in, e.g., English. The present study examined whether Danish caregivers enhance Danish contrasts when speaking to their 18 month old children (infant directed speech—IDS) as opposed to an adult (adult directed speech—ADS). Caregivers were recorded talking about toy animals in conversations with their child and

with an adult interlocutor. The toy names were designed to elicit Danish contrasts differing in voice onset time and in place of articulation for sibilants, and vowels which are close neighbors in the crowded Danish vowel space. The dependent variables for the comparison of IDS to ADS were as follows: VOT differences for homorganic stop consonants, the frequency at the amplitude peak for the sibilants, the Euclidean F1/F2 differences between vowels, F0 of the stressed (first) syllable in the toy name, as well as the duration of the stressed syllable, the vowels, and the fricatives. Results of the acoustic differences between ADS and IDS were compared to the results of parents' reports on the children's productive and receptive vocabulary knowledge. [Work supported by the Danish National Research Foundation –Danmarks Grundforskningsfond.]

2aSC12. Inter-rater agreement on Mandarin tone categorization: Contributing factors and implications. Puisan Wong, Lingzhi Li, and Xin Yu (Otolaryngol.–Head and Neck Surgery, The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212, pswResearch@gmail.com)

Factors that may/may not influence inter-rater reliability in assessing the accuracy of monosyllabic Mandarin tones produced by children and adults were examined in three experiments. Experiment 1 investigated inter-judge reliability in two groups of Mandarin-speaking adults—one group from China and the other from Taiwan—on their categorization of filtered tones produced by adults and children. The results showed that the magnitude of inter-rater agreement was associated with the production accuracy of the speakers; the judges attained lower agreement in categorizing children's tones than adults' tones. All judges who indicated that Mandarin was their strongest language and that they had learned and used Mandarin since birth performed similarly in their tone categorization despite the fact that they came from and were residing in different countries. Similar results was found in experiment 2, in which one group of the judges in experiment 1 categorized tones produced by a new and larger group of adults and children, and in experiment 3, in which a different group of adults categorized another new set of tones produced by a different group of speakers. Implications of the findings in research design will be discussed. [Work supported by NIH-NIDCD (1 F31 DC008479-01A1) and NSF (OISE-0611641).]

2aSC13. Effects of phonological training on tone perception for English listeners. Chang Liu and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, changliu@utexas.edu)

The goal of this study was to examine the extent to which phonological training improves categorical perception of Mandarin Chinese tones in native speakers of American English. Two sets of F0 continuums were generated from rising to level tones and from falling to level tones. Participants underwent identification and discrimination tasks before (pre-training) and after (post-training) phonological training. Pre-training tests showed that the tone identification shifted gradually from contoured tones (rising/falling) to level tones as a function of F0 frequency, while tone discrimination was near a chance rate across and within tone boundary. Phonological trainings were provided to listeners in two consecutive days with each session lasting for one hour. In phonological training, listeners were provided immediate feedback (correct/incorrect) after making response in Mandarin tone patterns. Results showed that tone identification function became significantly steeper with phonological training. Although no prominent peaks were found across tone boundaries in the discrimination function, the accuracy rate and response time of tone discrimination improved after training. Ongoing work is now testing the extent to which a longer training regiment can enhance categorical perception of tones.

2aSC14. Are two-year-olds sensitive to anticipatory coarticulation? Caterina Minaudo and Elizabeth K. Johnson (Psychology, Univ. of Toronto, 3359 Mississauga Rd. N. CCT 4110, Mississauga, ON L5L1C6, Canada, c.minaudo@mail.utoronto.ca)

Eyetracking studies have shown that adults are highly sensitive to subphonemic detail in speech [e.g., Shatzman and McQueen (2006)]. In some circumstances, adults use subphonemic coarticulatory information to anticipate which word will occur next in the speech stream [McDonough *et al*. (2009)]. In the current study, we ask whether two-year-old children use anticipatory coarticulation in a similar manner. Sixteen children were presented with pairs of images. In half of the trials, the names of the images presented on the screen had matching phonological onsets (e.g., doggy and ducky) that also matched in syllable length (e.g., monosyllabic or disyllabic). In the remaining trials, the names of the images had mismatching phonological onsets (e.g., cake and strawberry). In addition, a portion of each trial type was identity spliced (e.g., informative anticipatory coarticulation) and a portion was cross-spliced (e.g., misleading anticipatory coarticulation). We predicted that if two-year-olds are sensitive to anticipatory coarticulation, then they should be slowest to recognize named targets when the heard label was cross-spliced and the two objects on the screen had mismatching phonological onsets. However, all children looked to the named targets equally fast regardless of trial condition. Thus, no evidence of sensitivity to anticipatory coarticulation was observed.

2aSC15. Perception of speaker age in children's voices. Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, MS GR 41, Box 830688, Richardson, TX 75075, assmann@utdallas.edu), Santiago Barreda, and Terrance M. Nearey (Dept. of Linguistics, Univ. of Alberta, Edmonton, AB, Canada)

To study the perception of speaker age in children's voices, adult listeners were presented with vowels in /hVd/ syllables either in isolation or in a carrier sentence, and used a graphical slider to register their estimate of the speaker's age. The data showed a moderate correlation of perceived age and chronological age. Age estimation accuracy was fairly constant across age up to about age 11, but there was a systematic tendency for listeners to underestimate the ages of older girls. This tendency was actually enhanced when listeners were informed of the speaker's sex. Age estimation accuracy was higher for syllables embedded in a carrier sentence. Linear regression analyses were conducted using acoustic measurements of the stimuli to predict perceived age. These analyses indicated significant contributions of fundamental frequency, duration, vowel category, formant frequencies, as well as certain measures related to the voicing source. The persistent underestimation of age for older girls, and the effect knowledge of speaker sex has on this underestimation suggest that acoustic information is combined with expectations regarding speakers of a given sex in arriving at an estimate of speaker age.

2aSC16. Serial order recall for rapid auditory presentations of vowel sequences: The effect of age. Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1621 Greene St., Columbia, SC 29208, fogerty@sc.edu) and Larry E. Humes (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Temporal processing declines with age may reduce memory of rapidly presented auditory sequences. The current experiment investigated vowel sequence recall for two- and four-item vowel sequences presented at six different stimulus onset asynchronies (SOA) that spanned identification performance at 50% correct. Young, middle-age, and older adults participated in all tasks. For two-item sequences, a functional difference was observed between the age groups. Older and younger listeners had a qualitatively different pattern of recall, while performance for the middle age group approximated performance of either the young or older group, dependent upon the presentation rate (i.e., SOA). For the four-item sequences, results demonstrated the standard serial position curve. Increasing the rate of presentation by decreasing the SOA had the most profound effect on the middle items of the sequence for which subjects had the poorest retention. Overall, when temporal order performance was equated at the presentation rate corresponding to each individual's 50% threshold, recall accuracy for each position across the age groups was highly similar. These results suggest that declining temporal order performance of rapid sequences for older listeners is not the result of poorer recall performance, but is more related to sensory processing declines of rapidly presented temporal sequences.

2aSC17. Compensatory articulation in amyotrophic lateral sclerosis: Tongue and jaw interactions. Sanjana Shellikeri, Yana Yunusova (Speech Language Pathology, Univ. of Toronto, 253 South Park Rd., PH2, Thornhill, ON L3T0B4, Canada, sanjana.shellikeri@mail.utoronto.ca), Danielle Thomas (Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), Jordan Green (Special Education and Commun. Disord., Univ. of Nebraska-Lincoln, Lincoln, NE), and Lorne Zinman (Sunnybrook Health Sci. Ctr., Toronto, ON, Canada)

Previous acoustic studies on speech deterioration in amyotrophic lateral sclerosis (ALS) demonstrated that those at more advanced stages of disease show reduced F2 (second formant) slopes presumably due to disease-related changes in the tongue. Other studies have shown that patients with ALS use their jaw to compensate for decreased tongue function in speech. However, no study to date

has examined the compensatory role that the jaw has on maintaining the acoustic signatures of vocalic segments. This study will report F2 slope differences in vowels and diphthongs produced with and without jaw stabilization via a bite block. Based on previous studies, I hypothesized that the bite block will affect F2 slope measures in individuals with significant tongue impairment on the oral-motor examination and low speech intelligibility scores. Thirty participants repeat a carrier phrase "Say _ again" with words "wax, sip, yo-yo, sight" three times with and without the bite block. Kinematic measures of the distance, time, and speed, and acoustic measures of F2 slope are reported. The data will be discussed in the context of the role of the tongue in vowel production.

2aSC18. Relationship between articulation and mispronunciation detection in children with speech delay: Perception of unfamiliar speech vs. their own speech. Mark Paullin, Kyoko Nagao, and H Timothy Bunnell (Ctr. for Pediatric Auditory and Speech Sci., Nemours Biomed. Res., CPASS Ste., 1701 Rockland Rd., Wilmington, DE 19803, paullin@asel.udel.edu)

We examined the relationship between speech production and mispronunciation detection ability in children with speech delay (SD). Thirty-three SD children aged between 6;0 and 10;0 participated in a mispronunciation detection test using three types of stimuli: words pronounced correctly by other unfamiliar children (OTHERS); words mispronounced by OTHERS; and the participant's own speech (SELF) pronounced either correctly or incorrectly. The participant's articulation was assessed by the standardized GFTA-2 scores. Results indicated that SD children made significantly more errors when judging SELF speech than when judging OTHERS speech. Multiple regression analyses revealed that accuracy of detecting OTHERS mispronounced words was a significant predictor of GFTA-2 scores in these SD children. Interestingly, in the regression model, accuracy for detecting SELF mispronunciations made a significant independent contribution in addition to accuracy at detecting OTHERS mispronunciations. Overall these two measures accounted for a significant proportion of the variance in GFTA-2 scores (R-squared = 0.45). These findings suggest that children with SD may have more coarse phonological representations of their own speech than the speech of other children.

2aSC19. An EMA-based articulatory feedback approach to facilitate L2 speech production learning. Atsuo Suemitsu (Japan Adv. Inst. of Sci. and Technol., 1-1 Asahidai, Nomi 9231292, Japan, sue@jaist.ac.jp), Takayuki Ito, and Mark Tiede (Haskins Lab., New Haven, CT)

When acquiring a second language (L2), learners have difficulty in achieving native-like production even if they receive instruction on how to position the speech articulators for correct production. A principal reason is that learners lack information on how to modify their articulation to produce correct L2 sounds. A visual feedback method using electromagnetic articulography (EMA) has been previously implemented for this application with some success [Levitt *et al*. (2010)]. However, because this approach provided tongue tip position only, it is unsuitable for vowels and many consonants. In this work, we have developed a more general EMA-based articulatory feedback system that provides real-time visual feedback of multiple head movement-corrected sensor positions, together with target articulatory positions specific to each learner. We have used this system to improve the production of the unfamiliar vowel /ae/ for Japanese learners of American English. We predicted an appropriate speaker-specific /ae/ position for each Japanese learner using a model trained on previously collected kinematic data from 49 native speakers of American English, based on vowel positions for the overlapping /iy/, /aa/, and /uw/ vowels found in both languages. Results comparing formants pre- and post-feedback training will be presented to show the efficacy of the approach.

2aSC20. Articulatory phonetics of coronal stops in monolingual and simultaneous bilingual speakers of Canadian French and English. Francois-Xavier Brajot, Fateme Mollaei (School of Commun. Sci. and Disord., McGill Univ., 1266 des Pins Ouest, Montreal, QC H3G 1A8, Canada, fx.brajot@mail.mcgill.ca), Megan Callahan (Ctr. for Res. on Brain, Lang. and Music, McGill Univ., Montreal, QC, Canada), Denise Klein (Cognit. Neurosci. Unit, Montreal Neurological Inst., Montreal, QC, Canada), Shari R. Baum, and Vincent L. Gracco (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada)

Previous studies of bilingual speech production have relied on individuals whose age of acquisition of their second language varies. In the proposed research, we take advantage of the unique multilingual environment of Montreal and examine speech production in individuals who have acquired two languages from birth and compare the results to monolingual speakers. Electromagnetic recordings of single-word productions were carried out on three groups of female Canadian speakers (French monolingual, English monolingual, French-English simultaneous bilingual). Spectral moment and formant transition analyses of coronal burst segments showed cross-linguistic differences across vowel contexts. Tongue place of articulation and shape likewise indicated cross-linguistic differences in static articulatory positions. Kinematic analyses further identified language-specific movement patterns that helped clarify certain results from the acoustic analyses, namely that spatiotemporal characteristics of coronal articulation help enhance vocalic dimensions important to the respective language. Similar patterns were observed among the bilingual subjects, with the notable exception that acoustic and kinematic spaces narrowed considerably, resulting in reduced overlap between languages. It appears that simultaneous bilingual speakers not only follow language-appropriate articulatory and acoustic patterns, but further minimize areas of cross-linguistic convergence otherwise found among monolingual speakers.

2aSC21. Vowel production in sighted adults and blind adults: A study of speech adaptation strategies in high-intensity background noise. Pamela Trudeau-Fisette, Christine Turgeon, and Dominique Côté (Département de Linguistique, Université du Québec à Montréal, 405, rue Sainte-Catherine Est, Montréal, QC H2L 2C4, Canada, trudeau-fisette.pamela@courrier.uqam.ca)

Recent studies have shown that congenitally blind speakers have greater auditory discrimination acuity than sighted speakers [Ménard, Dupont, Baum, and Aubin, J. Acoust. Soc. Am. **126**, 1404–1414 (2009)]. At the production level, however, blind speakers produce smaller displacements of the lips (visible articulator) than their sighted peers. In order to further investigate the impact of visual experience on the articulatory gestures used to produce intelligible speech, adaptation strategies in background noise was studied in blind and sighted speakers. Ten sighted and 10 congenitally blind adult French participants were recorded during the production of the vowels /i/, /y/, /u/, /a/ in a CVC context. Two conditions were elicited: with high-intensity noise heard through headphones and without noise. Synchronous acoustic and articulatory data were recorded using the Carstens AG500 Electromagnetic Articulograph system. Formant measures and movements of the lips and tongue were analyzed. Results reveal that blind speakers produced smaller ranges of lip movement than sighted speakers in the noisy condition, suggesting that the blind subjects made less use of visible articulators to improve intelligibility. Results are discussed in light of multimodal production-perception relationships in speech.

2aSC22. Token-to-token variability and anticipatory coarticulation as indicators of maturity of speech motor control in 4-year-old children. Guillaume Barbier, Pascal Perrier (Speech and Cognition Dept., GIPSA-lab, 11, rue des Mathématiques, Saint Martin d'Hères 38402, France, guillaume.barbier@gipsa-lab.grenoble-inp.fr), Lucie Ménard (Laboratoire de Phonetique, UQAM, Montréal, QC, Canada), Mark Tiede (Haskins Lab., New Haven, CT), and Joseph S. Perkell (Massachusetts Inst. of Technol., Cambridge, MA)

Children's gestures do not appear to be executed with the same dexterity as adults'. Studies of arm movements have shown that young children's gestures are less accurate, more variable and slower than those of adults. This difference in behavior can be explained by a lack of experience with the sensory consequences of motor acts and still-developing forward models for the control of those acts. The hypothesis of immature and incomplete sensori-motor representations for speech in 4-year-old native speakers of Canadian French is addressed here through the analysis of ultrasound recordings of tongue contour kinematics and the speech signal from a corpus of isolated vowels and vowel-consonant-vowel sequences. Special attention is devoted to the analysis of vowel variability from two perspectives. Variability across repetitions in a single context provides information about the accuracy of the control. Contextual variability provides insights into the planning process as reflected in anticipatory coarticulation. Analysis of the observed lingual gestures will lead to improved understanding of the development of speech motor control and refinement of sensori-motor representations of speech. [Work supported by FQRNT project N° 147877 and ANR project ANR-08-BLAN-0272.]

2aSC23. Developmental aspects of American English diphthong trajectories in the formant space. Sungbok Lee (Elec. Eng., Univ. of Southern California, 3601 Watt Way, GFS-301, Los Angeles, CA 90089, sungbokl@usc.edu), Alexandros Potamianos (Electron. and Comput. Eng., Tech. Univ. of Crete, Chania, Greece), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern Califronia, Los Angeles, CA)

Formant trajectories of five American English diphthongs embedded in the target words BAIT (/EY/), BITE (/AY/), POUT (/AW/), BOAT (/OU/), BOYS (/OY/) are investigated in the first two formant space as a function of age and gender. Age range considered is from 5 to 18 years. In this report, the focus is given on the differences in position between the start/end points of diphthongs and nine monophthongs. Averaged formant data across subjects in each age group are examined for this purpose. Two findings are worth mentioning. First, across all age groups, the start and end positions of diphthongs hardly match with the monophthongs that are typically used to transcribe the diphthongs across all age groups [cf. Holbrook and Fairbanks (1962)]. For instance, the start position of /EY/ is closer to /I/ than to /e/, and the end points of /EY, AY, OY/ are significantly different with respect to each other. Second, in addition to the larger size of vowel space, an overshoot trend toward the nominal end points of diphthongs is the most prominent developmental trend. That is, formant values of diphthongs produced by younger age children are closer to the nominal monophthongs used to transcribe the diphthongs.

2aSC24. What you see is what you hear: How visual prosody affects artificial language learning in adults and children. Jaspal K. Brar (Psychology, Univ. of Toronto, 13 Dovehaven Cres, Brampton, ON L6P 2N8, Canada, pauly.brar@mail.utoronto.ca), Michael D. Tyler (Psychology, Marcs Inst., Univ. of Western Sydney, Sydney, NSW, Australia), and Elizabeth K. Johnson (Psychology, Univ. of Toronto, Mississauga, ON, Canada)

Speech perception is a multimodal phenomenon, with what we see impacting what we hear. In this study, we examine how visual information impacts English listeners' segmentation of words from an artificial language containing no cues to word boundaries other than the transitional probabilities (TPs) between syllables. Participants ($N = 60$) were assigned to one of three conditions: Still (still image), trochaic (image loomed toward the listener at syllable onsets), or Iambic (image loomed toward the listener at syllable offsets). Participants also heard either an easy or difficult variant of the language. Importantly, both languages lacked auditory prosody. Overall performance in a 2AFC test was better in the easy (67%) than difficult language (57%). In addition, across languages, listeners performed best in the trochaic condition (67%) and worst in the Iambic condition (56%). Performance in the still condition fell in between (61%). We know English listeners perceive strong syllables as word onsets. Thus, participants likely found the Trochaic Condition easiest because the moving image led them to perceive temporally co-occurring syllables as strong. We are currently testing 6-year-olds ($N = 25$) with these materials. Thus far, children's performance collapsed across conditions is similar to adults (60%). However, visual information may impact children's performance less.

2aSC25. What palatalized consonants can tell us about theories of loanword adaptation. Allan J. Schwade (Linguistics, UCSC, 401 Pacific Ave. Apt 227, Santa Cruz, CA 95060, allanschwade@gmail.com)

Phonology- and perception-based theories of loanword adaptation clash over two different assumptions: what language background the adapter has and what cognitive component handles adaptation. Phonology-based theories argue that borrowers know both the source and borrowing language and that the phonology determines output forms; perception-based accounts argue that the borrower does not know the source language and that the phonetic decoder guides adaptation. Since there is no reason to believe that either population of borrowers cannot adapt words, a production experiment was carried out to test both populations/approaches. Four monolingual English and three bilingual English-Russian speakers were played currently unborrowed Russian words containing palatalized consonants and asked to repeat them aloud in an American English accent. Since palatalized velar and coronal stops are often articulated with some degree of affrication and monolinguals are unaware of this, it was predicted that they would sometimes adapt said consonants as affricates ($t^ju > tʃu$). However, since bilinguals are familiar with the co-articulatory affrication, they were not predicted to adapt palatalized stops as affricates ($t^ju > tu$). The results corroborated the hypothesis in that bilinguals never affricated while monolinguals affricated a tenth of palatalized stops—demonstrating that both theories make the correct predictions for their respective populations.

2aSC26. A developmental study of vowels spoken in syllables and in sentence context. Daniel J. Hubbard (School of Behavioral and Brain Sci., GR4.1, Univ. of Texas at Dallas, P.O. Box 830688, Richardson, TX 75083, dhubbard@utdallas.edu), Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., Halifax, NS, Canada), Shaikat Hossain, and Peter F. Assmann (School of Behavioral and Brain Sci., GR4.1, Univ. of Texas at Dallas, Richardson, TX)

This study examined the effects of context on the production of vowels spoken by children of different ages in isolation and in a carrier sentence. Vowels were extracted from a database of hVd syllables produced by 207 native English talkers from the North Texas region, ranging in age from 5 to 18 years with approximately equal numbers of males and females. Preliminary analysis of a subset of the database (around 25% of talkers) showed a systematic reduction in vowel duration with increasing age for syllables in isolation. Vowels in sentence context were on average 30% shorter than in isolated syllables, and durations were less closely linked to age group. Formant frequencies (F1–F3) showed similar patterns for vowels in isolated syllables and sentences, and decreased as a function of age as expected. However, measures of formant movement across the vowel (from 20 to 80% of the vowel duration) revealed increased F1 and F2 movement for syllables in isolation compared to those produced in carrier sentences. A comprehensive analysis of the database will be presented and implications for vowel recognition will be discussed.

2aSC27. An ultrasound study of the acquisition of North American English /ɹ/. Lyra V. Magloughlin (Dept. of Linguistics, Univ. of Ottawa, 70 Laurier Ave. East, Rm. 401, Ottawa, ON K1N 6N5, Canada, lyra@uottawa.ca)

I report an acoustic and articulatory study of North American English /ɹ/ production in typically developing English-speaking children during early and later-stage acquisition. North American English /ɹ/ is of interest in adult populations because it exhibits acoustic stability (e.g., low F3) despite considerable articulatory variability both within and between speakers [Delattre and Freeman (1968)]. North American English /ɹ/ is also often one of the last sounds to be acquired by children [Smit (1993), Schriberg (1993)], especially in prevocalic position (Smit *et al*. (1990), McGowan *et al*. (2004)]. Tiede *et al*. (2011) have argued that children might attempt different vocal tract configurations during acquisition, particularly in contexts where the articulatory demands are greater. While there is a growing body of literature on articulatory variability in adult production of /ɹ/ [e.g., Mielke *et al*. (2010), Campbell *et al*. (2011)], there remains virtually no articulatory data on typically developing children's production during acquisition. This study uses ultrasound imaging to investigate the articulations of four typically developing English-speaking children, aged between 3 and 6 years, during production of familiar lexical items. Children's early-stage articulations are examined and compared with their later-stage productions, and with adult variability patterns.

2aSC28. Six- and ten-month-old infants' perception of non-contrastive variation. Dena Krieger and Elizabeth K. Johnson (Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, dena.krieger@mail.utoronto.ca)

Recent evidence suggests that infants do not perceive all existing speech sounds from birth. For example, the velar and alveolar nasal place contrasts are so subtle that infants require experience to perceive it [e.g., Narayan *et al*. (2012)]. Here, we examine English-learning infants' perception of another subtle contrast: pre-voicing on stop consonants. Six- and 10-month-olds' ability to discriminate between voiced and voiceless stops (phonemically contrastive in English) as well as voiced and pre-voiced stops (allophonic in English, but contrastive in other languages such as Dutch) was tested using a variant of the stimulus alternation paradigm (SAPP). Six-month-olds ($N = 34$) distinguished between voiced and voiceless stops ($p < 0.05$), but not between voiced and pre-voiced stops. Ten-month-olds ($N = 32$) failed to discriminate either contrast. We conclude that (1) English pre-voicing may be a subtle contrast requiring experience to perceive, and (2) this version of the SAPP might not be an ideal methodology to examine discrimination abilities in 10-month-olds. Overall, our findings thus far fit well with the notion that some contrasts require experience to perceive, as well as with past studies reporting mixed results regarding English-learning infants' ability to perceive pre-voicing contrasts [e.g., Aslin *et al*. (1981), Lasky *et al*. (1975)].

2aSC29. Acoustical cues versus top-down bias in infants' parsing. Mireille Babineau and Rushen Shi (Psychology, Université du Québec à Montréal, Département de Psychologie, Université du Québec à Montréal, C.P. 8888 succursale Centre-ville, Montreal, QC H3C 3P8, Canada, babineau.mireille@courrier.uqam.ca)

French liaison involves the surfacing of an underlying consonant as the onset of the following vowel-initial word (e.g., les amis - /le/ /zami /), creating misalignment. However, acoustic cues that support vowel-initial parsing may exist. We tested French-learning 30-month-olds using a preferential looking procedure. Familiarization stimuli in experiment 1 were sentences each containing a determiner preceding a vowel-initial non-word (e.g., ces onches). Infants' parsing of the non-word was assessed. The vowel-initial condition presented the vowel-initial non-word versus another non-target (onches - èque). The syllabic condition tested the consonant-initial parse (zonches - zèque). Infants in the vowel-initial, but not the syllabic condition, showed discrimination ($p = 0.008$), i.e., they correctly parsed the vowel-initial target, possibly using acoustic cues. However, knowledge of underlying liaison consonants can also explain these results. In experiment 2, we removed acoustic cues to vowel-initial parsing by using a consonant-initial non-word following a determiner as the familiarization stimuli (e.g., un zonches). Infants were tested with the same two conditions as in experiment 1. Infants yielded the same results as in experiment 1, showing discrimination only in the vowel-initial condition ($p = 0.047$). Taken together, 30-month-olds perceived /z/ as an underlying floating element; they used this liaison knowledge, rather than possible acoustical cues, for parsing.

2aSC30. Effect of talker sex on infants' detection of spondee words in a two-talker or a speech-shaped noise masker. Lori Leibold, Crystal Taylor, Andrea Hillock-Dunn (Allied Health Sci., UNC Chapel Hill, 3122 Bondurant Hall, CB#7190, Chapel Hill, NC 27599, leibold@med.unc.edu), and Emily Buss (Otolaryngol.—Head and Neck Surgery, UNC Chapel Hill, Chapel Hill, NC)

Speech recognition performance in the presence of competing speech is typically better for adults when the target and masker talkers are different sexes than when the target and masker talkers are the same sex. One explanation for this result is that the acoustic differences between male and female speech productions promote segregation of the two streams of speech, thus leading to a reduction in informational masking. In this study, an observer-based psychophysical procedure was used to compare infants' (7–13 months) masked speech detection thresholds for spondee words produced by a male or a female talker in either a two-female-talker or a speech-shaped noise masker. Infants were assigned to a single testing condition. Maskers were presented continuously throughout testing at an overall level of 50 dB SPL, fixed throughout testing. Following training to an 80%-correct criterion, thresholds for the target word were measured adaptively using a 2-down, 1-up procedure. Infants' thresholds in the two-female-talker masker were higher for the female compared to the male target word. In contrast, infants' thresholds were similar for the female and male target words in the speech-shaped noise masker. These results suggest that introducing a different sex between the target and masker aids in the segregation of sounds for infants, as has previously been shown for adults. [Work supported by the NIH.]

2aSC31. The effects of voicing and position in infants' perception of coda consonants. Kathleen Engel (Speech Development Lab, Psychology, Univ. of Calgary, 2500 University Dr. NW, Calgary, AB T2N 1N4, Canada, knengel@gmail.com), Stephanie L. Archer (School of Commun. Sci. and Disord., McGill Univ., Montreal, Alberta, Canada), and Suzanne Curtin (Psychology, Univ. of Calgary, Calgary, AB, Canada)

Infants' ability to discriminate contrasting acoustic information has been demonstrated with many of the speech contrasts found in the world's languages. However, this ability seems to be positionally constrained. Contrasts in onsets are discriminated by young infants, but coda contrasts are not discriminated until around 16- to 20-months [Zamuner (2006)]. Here we examine whether the contrast and the position influence discrimination in younger infants. We tested 64 12-month-olds' discrimination of voiceless (/p/, /k/), or voiced stops (/b/, /g/) in either word-final (VC; Exp. 1) or word-medial (VCCV; Exp. 2) position. Experiment 1 habituated infants to ap or ak (voiceless) or to ab or ag (voiced). At test, infants heard the same token presented during habituation (same) and a novel token (switch). Planned comparisons revealed that only infants in the voiced condition looked longer to the switch than the same trial ($p < 0.05$). In experiment 2, infants heard apta or akta (voiceless) or abta or agta (voiced), but no effects of trial type were found, suggesting that the perceptual advantage in word-final position does not exist in word-medially. Thus, in word-final coda position voiced stops are more acoustically salient than voiceless stops, providing infants with added information to aid in discrimination.

2aSC32. Preliminary comparison of second-formant discrimination thresholds in cochlear implant users and young normal-hearing listeners. Catherine L. Rogers, Gail S. Donaldson, Amanda J. Cooley, and Benjamin A. Russell (Dept. of Commun. Sci. and Disord., Univ. of South Florida, USF, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

Formant discrimination thresholds (FDTs) may provide insights regarding factors limiting vowel perception by cochlear implant (CI) users, but have not been systematically studied in this population. In the present study, estimates of second-formant (F2) FDTs obtained in three CI users were compared to FDTs obtained from three young normal-hearing (YNH) listeners. Procedures and stimuli were modeled after Kewley-Port and Watson [J. Acoust. Soc. Am. **95**, 485–496 (1994)] but employed fewer trials and an expanded F2 frequency range. Stimuli were formant-synthesized versions of three target vowels. FDTs were estimated using an adaptive 3AFC task with feedback and based on six consecutive 80-trial stimulus blocks. FDTs for the three YNH listeners were comparable to previously reported FDTs (2.4% of reference frequency versus 1.5% in Kewley-Port and Watson). FDTs for two of the CI users were about 70% larger than the average for the YNH listeners. FDTs for the third CI user approached YNH average values in one frequency region but were enlarged in another region. Data for this CI user could not be explained by place-pitch thresholds (obtained in a previous study) and suggest that CI users' spectral acuity for complex stimuli may not be directly predictable from measures of spectral acuity for simple stimuli.

2aSC33. Toddlers' comprehension of noise-vocoded speech and sine-wave analogs to speech. Rochelle S. Newman (Dept. Hearing & Speech Sci., Univ. of Maryland, 0100 Lefrak Hall, College Park, MD 20742, rnewman1@umd.edu), Monita Chatterjee (Auditory Prostheses & Percept. Lab., Boys Town National Res. Hospital, Omaha, NE), Giovanna Morini, and Molly Nasuta (Dept. Hearing & Speech Sci., Univ. of Maryland, College Park, MD)

A great deal of research has investigated listeners' ability to compensate for degraded speech signals such as noise-vocoded speech (a signal with reduced spectral structure but intact amplitude envelope information) and sine-wave analogs to speech (a signal that maintains the global dynamic spectral structure of the signal at the expense of amplitude envelope information). Nittrouer and colleagues found developmental changes in the ability to comprehend such signals, reporting that while adults perform more accurately with sine-wave analogs than with noise-vocoded speech, school-aged children show the opposite pattern [e.g., Nittrouer Lowenstein and Packer (2009)]. In a series of studies, we tested toddler's comprehension of these degraded signals. Twenty-seven-month-old children saw two images on each trial (e.g., cat, dog), and heard a voice instructing them which image to look at ("Find the cat!"). Sentences were presented either in full speech or were degraded. Toddlers ($n = 24$ per condition) looked at the appropriate object equally long with vocoded speech of 24 channels (60.2%) or 8 channels (62.4%) as with full speech (62.6%), but performed barely above chance with 4 channels (53.6%) and at chance for 2 channels (49.8%). Preliminary results suggest that performance with sine-wave analogs is poorer than 8-channel vocoded speech (56.1%), but testing is ongoing.

2aSC34. Exploring auditory aging can exclusively explain Japanese adults' age-related decrease in training effects of American English /r/-/l/. Rieko Kubo and Masato Akagi (School of Information Sci., JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, rkubo@jaist.ac.jp)

Age-related decrease in training effect was shown by training of American English /r/-/l/ contrasts on Japanese speakers. This study examined whether the decrease can be explained exclusively by auditory aging, or other, compensatory cognitive processing should be taken into account. Japanese speakers aged 60's participated the experiment. Hearing threshold and spoken word perception test of participants' first language were used to estimate their auditory aging. The word perception test was composed of low-familiar words, high-familiar words, and mono syllables. The audiograms showed low threshold at high frequencies. The result of the perception test showed that low intelligibility for phonemes with high frequency or short duration, and confusion between contracted sounds and basic sounds. These were particular for low-familiar words and mono syllables. These results suggest that participants had auditory aging emerging as high frequency loss and time-frequency-resolution degradation. Nonetheless, the acoustic features to distinguish /r/ and /l/ have long duration, low frequencies and wide frequency distance which are supposed to be unaffected by these auditory aging. The effect of word familiarity suggested compensatory cognitive processing involved. These suggest that age-related decrease cannot be explained exclusively by auditory aging, compensatory cognitive processing should be taken into account.

2aSC35. Listener judgments of age in a single-talker 48-year longitudinal sample. Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), Eric J. Hunter (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT), Catherine A. Mellum, and Lydia R. Rogers (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT, UT)

Numerous studies have demonstrated that listeners can make relatively accurate judgments of a talker's age from hearing the talker's voice. Materials in these previous studies have included sustained vowels (phonated and sometimes whispered), sentences, and passages of discourse. The number of talkers has ranged from 4 to 150, but in nearly all cases talkers were recorded only once. In the present study, the materials were recorded from a single talker who gave regular public speeches over a period of 48 years. Young adult listeners performed age judgments on samples extracted from 20 speeches chosen at 2–3 year intervals spanning the 48-year period, three samples per speech. Samples lasted 5 to 10 s and were chosen to minimize content that would identify the talker or link samples from the same speech to each other. In separate experiments, listeners listened to these 60 samples and after each one judged the talker's age either by choosing from three categories (50–66, 67–83, or 94–100 years) or by making a direct age estimate. Accuracy of these estimates will be compared to previous studies and examined as a function of both the talker's chronological age and acoustic measures performed for this talker in a separate experiment.

2aSC36. Examining the relationship between the interpretation of age and gender across languages. Andrew R. Plummer (Linguistics, Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, plummer@ling.ohio-state.edu), Benjamin Munson (Speech-Lang.-Hearing Sci., Univ. of Minnesota, Minneapolis, MN), Lucie Ménard (Linguistics, Univ. of Québec at Montréal, Montréal, QC, Canada), and Mary E. Beckman (Linguistics, Ohio State Univ., Columbus, OH)

Speech signals vary substantially in a number of their key properties, with the variability deriving from, among other things, talkers' age and gender. Speech processing requires resolution of this variation, necessitating interpretation of age and gender information in the signal. In some signals, the age and gender are not clear from acoustic information alone. In these cases, there may be substantial individual variation in judgments of age and gender. This study examined the interplay between the interpretation of age and gender across language communities. Corner vowel stimuli ([i], [u], [a]) generated by an age-varying articulatory synthesizer set at seven different ages (6 months, 2, 4, 5, 10, 16, and 21 years) were presented to native speakers of Cantonese, English, and Japanese. Listeners assigned an age (in years) and a gender (along a visual analog scale ranging from "definitely male" to "definitely female," or the equivalent in Japanese or Cantonese) to each stimulus. Analysis revealed a bifurcation in the interpretation of age and gender for the age 10 stimuli, which subjects rated as either a younger male or older female, suggesting a nonuniformity in the resolution of variability during speech processing. Preliminary analysis further suggests that this nonuniformity may be culture-specific.

2aSC37. Effects of vocal training on voluntary responses to pitch-shifted voice auditory feedback. Sona Patel, Cristina Nishimura, and Charles Larson (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, sona.patel@northwestern.edu)

Control of voice fundamental frequency (F0) relies on the interaction between various forms of sensory feedback and neural motor control mechanisms. Several studies have shown that unexpected changes in pitch in the auditory feedback lead to reflexive compensatory vocal responses. We examined voluntary vocal responses to unpredictable perturbations in pitch auditory feedback. Ten subjects were trained over a five-day period to change their voice F0 in the opposite direction to the pitch-shifted feedback ($\pm$100 cents, 1000 ms) and 10 in the same direction as the feedback. Results showed that responses that followed the stimulus direction had significantly shorter latencies (200 ms) than opposing responses (324 ms). The reduced latency of the following responses suggests a switch from a feedback to a feedforward control strategy. The feedback strategy requires monitoring feedback and correcting for errors between the feedback signal and the intended vocal goal. The feedforward strategy relies less on auditory feedback and more on an internal model of the desired vocal pitch goal. Furthermore, feedback systems generally are slower than feedforward strategies, which would explain the shorter latencies of the responses that followed the stimulus direction. Results of this study will be discussed in terms of the differing strategies that may be used in various vocal contexts.

2aSC38. Segment imitation in unfamiliar language varieties. Julia Forsberg (Philosophy, Linguist. and Theory of Sci., Univ. of Gothenburg, Box 200, Gothenburg 405 30, Sweden, julia.forsberg@gu.se)

Phonetic imitation has been explored in various research: in studies on impersonation [Zetterholm (2001); spontaneous phonetic imitation [Babel (2012)]; and as part of studies investigating overall imitation [Papcun (1988), Babel (2011)]. There are fewer studies focusing on specific phonemes, and the context required for successful imitation. Babel [**186** (2012)] (Am. English) and Zetterholm [**281** (1997)] (Swedish) found spontaneous imitation was more common in the open vowels than in close. A case study by Zetterholm [275 (1997)] of one impersonator shows formant measurements as closer to the target voice than that of the impersonator. This paper presents a comparative acoustic analysis of imitation of unfamiliar phonemes by untrained imitators, based on a small study containing recordings from SWEDIA and SUF. Vowel tokens are edited to three lengths with varying degrees of context. Listeners are asked to imitate the sound, and attempt to use it in a word. Acoustic analysis of F0, 1, 2, and 3 will be compared to the original recordings as well as the speakers' own speech. The conclusion and discussion includes indications of how easy it is to produce phonemes when they are not native to a speaker's own variant, as well as its relation to the forensic context.

2aSC39. Imitability of contextual vowel nasalization and interactions with lexical neighborhood density. Georgia Zellou (Linguistics, Univ. of Pennsylvania, 800 N. 48th St., #26, Philadelphia, PA 19139, gzellou@sas.upenn.edu), Rebecca Scarborough (Linguistics, Univ. of Colorado, Boulder, CO), and Kuniko Nielsen (Linguistics, Oakland Univ., Rochester, MI)

This study investigates the imitability of contextual vowel nasalization in English. Unlike other phonetic features reported to be imitable [e.g., vowel formants (Babel, 2012), VOT (Nielsen, 2011)], vowel nasality is non-contrastive in English. Nasality is, however, systematically variable: words from dense lexical neighborhoods (high-ND words) are produced with greater nasality than words from sparse neighborhoods [Scarborough (2004), (2012)]. Two experiments were conducted to test (1) whether (experimentally manipulated) nasality can be imitated in a shadowing task, and (2) whether direction of manipulation (more or less nasality, enhancing or countering natural neighborhood-conditioned patterns) affects shadowing behavior. Subjects shadowed 16 high-ND words (which are naturally more nasal) containing a vowel-nasal sequence and modified by spectral mixing to exhibit either greater-than-natural (experiment 1) or less-than-natural (experiment 2) nasality. Both the increase and the decrease in nasality were imitated (though not overall degree of nasality, as our imitation model was more nasal in both conditions than any of our subjects). This change persisted into a post-shadowing task for just the less-nasal condition. These results indicate that speakers are sensitive to non-contrastive phonetic detail in nasality, affecting their subsequent production. Further, the naturalness of nasality (reflecting neighborhood-conditioned variation) may affect the pattern of imitation.

2aSC40. Individual variability in phonetic convergence of vowels and rhythm. Gayatri Rao (Dept. of Psych., Univ. of Texas, 108 E. Dean Keeton, 1 University Station A8000, Austin, TX 78712, raog@utexas.edu), Rajka Smiljanic (Linguistics, Univ. of Texas, Austin, TX), and Randy Diehl (Dept. of Psych., Univ. of Texas, Austin, TX)

Phonetic convergence (PC) has been demonstrated for segmental (vowels, voice onset time) and suprasegmental (stress, intonation) properties [Nielsen (2008), Delvaux and Soquet (2007), Phillips and Clopper (2010), Rao, Smiljanic, and Diehl (2011)]. Since PC is subject to large individual differences [Ni Chiosáin (2007)], the current study examined individual variability in PC in both segmental and suprasegmental domains for native speakers of American English. Six female and six male pairs read CVC syllables and a short paragraph before and after an interactive map task. For each dyad, convergence in vowels was measured using formants and the cosine similarity metric for individual vowels and for the entire vowel space. Convergence in rhythm was measured using the centroid of the envelope modulation spectrum [EMS + centroid, Rao and Smiljanic (2011)]. Overall, speaker pairs converged to different extents in both measures. Vowel type, dialect background, and gender were found to influence the degree of convergence. In general, men were more likely to converge in rhythm whereas women were more likely to converge in vowels. This supports the findings that gender-based differences in convergence are due to perceptual sensitivity to indexical features [Nami *et al.* (2002), Babel (2009)] and particular sound features in spoken utterances.

2aSC41. What is the default behavior in accommodation: Convergence or maintenance? Bethany MacLeod (Carleton Univ., 211 Bruyere St., Ottawa, ON K1N 5E4, Canada, beth_macleod@carleton.ca)

This study, a secondary analysis of data from a phonetic accommodation study, considers the default behavior of speakers in accommodating to another speaker in an interaction. Should convergence or maintenance be considered the default behavior? There is inherent acoustic variation in our speech. Every time we produce a sound, such as a voiceless stop in English, it varies along some phonetic dimension, such as voice onset time (VOT). We might expect that, in the absence of any external influence, these voiceless stops will be realized with VOT longer than their overall mean 50% of the time and with VOT shorter than their overall mean 50% of the time. During interaction with another person, however, studies in social-psychology have suggested that lack of adjustment (maintenance) may be akin to divergence [Tong *et al.* (1999)]. In addition, convergence is a fairly robust finding in studies of accommodation and imitation [e.g., Nielsen (2011), Babel (2012)], suggesting that perhaps the default behavior in interaction is convergence. The purpose of this talk is to introduce these points of view, discuss the factors that may affect our interpretation, and facilitate discussion on this issue, which has implications for the growing body of research investigating accommodation.

2aSC42. Investigating developmental changes in phonological representation using the imitation paradigm. Kuniko Nielsen (Linguistics, Oakland Univ., 320 O'Dowd Hall, Rochester, MI 48309-4401, nielsen@oakland.edu)

This study investigates the developmental changes in phonological representation, by examining the word- and phoneme-level specificity of phonetic imitation by children. Prevailing linguistic theories assume three levels of phonological representations: word, phoneme, and feature. Previous research suggests that phonological representations develop throughout childhood, and that phonological awareness develops from larger to smaller units [e.g., Edwards *et al.* (2004), Treiman and Zukowski (1996)]. It has been shown that adult speakers implicitly imitate the phonetic properties of recently heard speech [e.g., Goldinger (1998)], and recently, Nielsen (2011) showed the sub-phonemic generalizability and word- and phoneme-level specificity of imitation, indicating that three levels of phonological representations simultaneously contribute to the observed patterns of phonetic imitation. In order to test whether young children manifest similar patterns of imitation and specificity, an experiment with a modified imitation paradigm with a picture-naming task was conducted, in which participants' VOT was compared before and after they were exposed to target speech with artificially increased VOT. Our preliminary results reveal that two groups of children (5 year-olds and 8 year-olds) show greater imitation than adults, while word-and phoneme-level specificity was greater for 8 year-olds than 5 year-olds. These results provide support for the continued development of phonological representations.

2aSC43. Phonetic accommodation in Spanish-English and Korean-English bilinguals. Stephen Tobin (Univ. of Connecticut, 406 Babbidge Rd., Storrs, CT 06269-1020, stephen.tobin@uconn.edu)

Preliminary results from eight participants in a cross-linguistic investigation of phonetic accommodation in speech production and perception are presented. The finding that synchronous actions are more stable than asynchronous ones has been reported in studies of general [Kelso (1981)] and speech-specific [Browman and Goldstein (1992), Byrd *et al.* (2009)] motor control. With reference to glottal-oral timing, near-zero VOTs (voice onset times) are representative of near-synchronous timing, whereas long-lag VOTs are representative of asynchronous timing [Sawashima and Hirose (1980), Dixit (1984), Lofqvist and Yoshioka (1989), Fuchs (2005)]. These observations served as a basis for the prediction that native speakers of Korean, with its long-lag aspirated stops (~120 ms), would more readily accommodate to typical English voiceless stop VOT (~70 ms) than native speakers of Spanish, with its short-lag voiceless stops (~20 ms). Spanish-English and Korean-English bilinguals were recorded reading voiceless stop-initial English words, before and during a task in which participants shadowed recorded productions of a native speaker of American English. Preliminary analysis of the production data provides some support for these hypotheses. The results contribute to our understanding of the conditions that promote phonetic accommodation.

2aSC44. The vowel spaces of Southern Californian English and Mexican Spanish as produced by monolinguals and bilinguals. Criccely Grijalva, Page E. Piccinini (Linguistics, Univ. of California San Diego, 3425 Lebon Dr. #918, San Diego, CA 92122, crgrijal@ucsd.edu), and Amalia Arvaniti (English Lang. & Linguist., Univ. of Kent, Kent, United Kingdom)

The vowel spaces of Southern Californian English and Mexican Spanish were investigated using three groups of speakers: 11 English monolinguals (8 females), 11 Spanish monolinguals (9 females), and 10 Spanish-English bilinguals (7 females). Speakers produced six repetitions of the ten American English vowels [i, ɪ, ɛ, æ, ɑ̃, ɔ, ʊ, u, ʌ, and ɜr] and six repetitions of the five Spanish vowels [i, e, a, u, and o]. Monolinguals produced vowels in one language; bilinguals produced vowels in both languages. Preliminary analysis shows Southern Californian English back vowels were less fronted compared to the results of Hagiwara (1997) from Southern Californian English, but more fronted than those of Hillenbrand *et al.* (1995) on General American English. Mexican Spanish back vowels [u] and [o] were substantially fronted compared to Castilian Spanish vowels [Bradlow (1995)], while [i] was lower and less fronted. In general, Mexican Spanish vowels were produced higher and more backed than Southern Californian English vowels in monolingual productions. Bilinguals produced their two vowel spaces closer together but with less dispersion than monolinguals, showing how bilinguals keep both language categories distinct.

2aSC45. Acoustic analysis of perceived accentedness in Mandarin speakers' second language production of Japanese. Peipei Wei and Kaori Idemaru (East Asian Lang. and Lit., Univ. of Oregon, 2250 Patterson St., 222, Eugene, OR 97405, peipei@uoregon.edu)

Many second language (L2) learners, particularly adult learners, retain foreign accent on their L2 production. What acoustic sources give rise to the perception of foreign accent? This study examines beginning and intermediate Chinese learners' production of Japanese in terms of segmental and suprasegmental features and investigates the relationship between acoustic characteristics of the L2 production and accentedness ratings provided by native Japanese listeners. Results of acoustic examination indicated that learners' production varied considerably from that of native speakers in terms of

durational features of stops, spectral features of some vowels, pitch (F0) peak alignment, and F0 contour. Multiple regression analysis identified the second formant of /u/, F0 peak alignment and contour as the strong predictors of perceived accent, accounting for nearly 90% of variance. These findings confirmed Flege's Speech Learning Model hypothesis—L2 sounds that are similar to L1 sounds, while subphonemically distinct, seem to pose greater difficulty for acquisition than dissimilar sounds. Moreover, longer classroom experience was found to show limited effects in reducing perceived accent, with slightly greater effects on segmental than suprasegmental variables.

2aSC46. Interaction of long-term acoustic experience and local context information on the perceptual accommodation of talker variability. Caicai Zhang (Haskins Lab., Yale Univ., The Chinese Univ. of Hong Kong, Hong Kong N/A, Hong Kong, yzcelia@gmail.com), Gang Peng, and William Shi-Yuan Wang (Dept. of Linguist. and Modern Lang., The Chinese Univ. of Hong Kong, Hong Kong, Hong Kong)

How do listeners recover speech content from acoustic signals, given the immense variability between talkers? In this study, two experiments were conducted on Cantonese level tones, comparing the perception of multi-talker speech stimuli in isolation and within a speech context. Without prior knowledge of a talker's pitch range, listeners resort to the population-average pitch range as a default reference for perception. This effect is attested by the significant correlation between the distance from population-average pitch range and identification accuracy in the isolation condition ($r = -0.24$, $p < 0.01$). The closer a talker's pitch range is to the population-average, the higher the identification accuracy is. The population-average reference is gender-specific, showing separate accommodation scales for female and male talkers. Such default reference is presumably built from one's long-term acoustic experience, reflecting the dense distribution of talkers in a community whose pitch is close to the population-average. Above the effect of long-term experience, the presence of a speech context allows listeners to tune to talker-specific pitch range, boosting the identification accuracy from 43% (in isolation) to 86%. Our findings demonstrate that listeners have built-in knowledge of population-average pitch and can shift from the default reference to talker-specific reference with the facilitation of context information.

2aSC47. Acoustic and articulatory information as joint factors coexisting in the context sequence model of speech production. Daniel Duran, Jagoda Bruni, and Grzegorz Dogil (Institut für Maschinelle Sprachverarbeitung (Inst. for Natural Lang. Process.), Stuttgart Univ., Pfaffenwaldring 5B, Stuttgart 70569, Germany, danielduran@ims.uni-stuttgart.de)

This simulation study presents the integration of an articulatory factor into the Context Sequence Model (CSM) [Wade *et al*. (2010)] of speech production using Polish sonorant data measured with the electromagnetic articulograph technology (EMA) [Mücke *et al*. (2010)]. Based on exemplar-theoretic assumptions [Pierrehumbert (2001)], the CSM models the speech production-perception loop operating on a sequential, detail-rich memory of previously processed speech utterance exemplars. Selection of an item for production is based on context matching, comparing the context of the currently produced utterance with the contexts of stored candidate items in memory. As demonstrated by Wade *et al*. (2010), the underlying exemplar weighing for speech production is based on about 0.5 s of preceding acoustic context and following linguistic match of the exemplars. We extended the CSM by incorporating articulatory information in parallel to the acoustic representation of the speech exemplars. Our study demonstrates that memorized raw articulatory information—movement habits of the speaker—can also be utilized during speech production. Successful incorporation of this factor shows that not only acoustic but also articulatory information can be made directly available in a speech production model.

2aSC48. Longitudinal changes of formant values and vowel space in two Mandarin-speaking children before 9 years of age. Li-mei Chen, Fan-Yin Cheng, and Wei Chen Hsu (Foreign Lang. & Lit., National Cheng Kung Univ., 1 University Rd., Tainan City 701, Taiwan, goodgoodgoodjob@yahoo.com)

Vowel productions of two Mandarin-speaking children were audio recorded in their homes with picture naming tasks once every 3 months, from birth to 9 years old. The present study is the ninth year of a longitudinal observation. Major findings in this stage are as follows: (1) The trend of decrease in formant values was continuously observed in the boy subject. As for the girl subject, it is not until 9 years old, the obvious decrease in formant values was found, especially in F1; (2) F1 values are more stable than F2 values in both subjects. They appeared to acquire jaw movement sooner than tongue movement; (3) Throughout these 9 years, the variability of F1 is around 200–300 Hz, and the variability of F2 is 500–700 Hz in both subjects. No trend of decrease was found; (4) The trend of shrinkage in F1-F2 vowel area continues from 7 to 9 years old for the boy subject, but not for the girl subject; (5) There is a clear decline in fundamental frequencies at 8-9 years of age in the boy subject. Longitudinal data of vowel formant values from the same group of subjects provide important references for assessment and treatment of articulation disorders in children.

2aSC49. Is the mora rhythm of Japanese more strongly observed in infant-directed speech than in adult-directed speech? Keiichi Tajima (Dept. of Psych., Hosei Univ., 36-511, 77 Massachusetts Ave., Cambridge, MA02139, tajima@hosei.ac.jp), Kuniyoshi Tanaka, Andrew Martin, and Reiko Mazuka (Lab. for Lang. Development, RIKEN Brain Sci. Inst., Wako-shi, Saitama, Japan)

Japanese has traditionally been called "mora-timed," but studies have shown that this intuition is based not on durational tendencies but rather on phonological, structural factors in the language. Meanwhile, infant-directed speech (IDS) is said to "exaggerate" certain properties of adult-directed speech (ADS), including rhythm. If so, then it is possible that the mora rhythm of Japanese is more strongly observed in IDS than ADS. To investigate this possibility, the present study utilized the RIKEN Japanese Mother-Infant Conversation Corpus, which contains approximately 11 h of IDS by 22 mothers talking with their 18-to-24-month-old infants, and 3 h of ADS by the same mothers. Results from durational analyses showed that aspects of mora rhythm, such as the distinction between phonemically short and long vowels and singleton and geminate consonants, and the tendency toward isochrony of moras, were not greater in IDS than ADS. Mora duration in IDS was instead more variable, partly stemming from greater phrase-final lengthening and non-phonemic, emphatic lengthening. Results from structural analysis, however, showed that non-CV moras such as moraic nasals that characterize Japanese rhythm occurred more frequently in IDS than ADS. These results suggest that even in IDS, Japanese rhythm is manifested structurally, not durationally. [Work supported by JSPS.]

2aSC50. Voice-onset time in infant-directed speech over the first year and a half. Evamarie Cropsey (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Jessica Gamache (Dept. of Linguist., Michigan State Univ., East Lansing, MI), Tonya Bergeson (Dept. of Otolaryngol.-Head and Neck Surgery, Indiana Univ. School of Med., Indianapolis, IN), and Laura Dilley (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI 48824, ldilley@msu.edu)

Previous research in small-N studies has indicated conflicting findings regarding whether mothers modify voice-onset time (VOT) of word-initial stop consonants in speech to infants compared to speech to adults, as well as the nature of any such modification. In a large-scale study, VOT was measured for voiced and voiceless stop consonants in speech of 48 mothers of infants in one of four cross-sectional age groups (0;3, 0;9, 1;1, 1;8) when they read a phonetically controlled storybook to their infant (ID speech) or an adult (AD speech). VOT measurements showed enhanced clarity (i.e., longer VOTs) in ID speech compared with AD speech for voiceless stop consonants only. An effect of infant gender was also found, showing that enhanced clarity was only produced by mothers of female infants ($N = 19$). Infant age was not found to be a significant factor in VOT production. The results have implications for understanding the nature of linguistic development in young children, specifically by elucidating factors apparently related to phonetic modification for clarity, including speech style and gender. [Work supported by NIH-NIDCD grant R01DC008581.]

2aSC51. Pitch affects voice onset time: A cross-linguistic study. Chandan Narayan (Linguistics, Univ. of Toronto, Sidney Smith Hall, 4th Fl., 100 St. George St., Toronto, ON M5S 3G3, Canada, chandan.narayan@utoronto.ca) and Mary Bowden (Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON, Canada)

Recent research into the acoustics of infant-directed speech (IDS) in English and Korean suggests that voice-onset time (normalized for varying speech rate) in word-initial aspirated stops are shorter than in adult-directed speech. The present study reports results from experiments conducted to explore the source of this VOT variation in IDS. Female English speakers ($n = 10$) and female Korean speakers ($n = 10$) recorded sentences with words varying in laryngeal condition (English: voiced/voiceless; Korean: plain, tense, aspirated) at three different pitches (low, normal, and falsetto). Results suggest that as speakers' pitch increases, the duration of VOT in aspirated stops decreases. Voiced stops (in English) and plain and tense stops) in Korean showed no difference in VOT across varying pitch conditions. Regression models suggest that VOT becomes less predictive of laryngeal state as speaker pitch increases. Results are discussed in terms of the physiological explanation of the pitch-VOT effect as well as the implications for the development of sound systems in infants.

2aSC52. /oy/ as a marker of local identity in Berlin. Stefanie Jannedy (ZAS Berlin, Schützenstr. 18, Berlin 10117, Germany, jannedy@ling.ohio-state.edu) and Melanie Wierich (Friedrich-Schiller-Universitat, Jena, Germany)

A fairly recent observation of multi-cultural urban German speech as spoken in Berlin is that the diphthongs /oy/ and /ey/ are realized more closed and fronted compared to more standard varieties of German. For this pilot study, spontaneous speech data were collected through standardized interviews from five young female speakers from two different neighborhoods in Berlin: Wedding is more Arab-dominant while Kreuzberg is more Turkish dominant. Their speech was orthographically transcribed and added to a database that allows for searching for all occurrences of the two diphthongs under investigation in their naturally occurring context in unscripted speech. So far, 250 occurrences of these vowels have been analyzed. Formant measurements were taken at five equally distanced points throughout the diphthong. A linear mixed effects model with the midpoint of the F2-formant value as the dependent variable were run, showing that speakers from the arab neighborhood (Wedding) significantly differ in their productions compared to speakers from the Turkish neighborhood ($p < 0.05$) in Kreuzberg. Moreover, there was a significant effect of language spoken around them ($p < 0.01$) on the production even though German is their dominant language. We argue that speakers use the production of these diphthongs as markers of their local urban identity.

2aSC53. Articulatory compensation to second formant perturbations. Chris Neufeld (Oral Dynam. Lab., Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON M5G 1V7, Canada, christopher.neufeld@mail.utoronto.ca), Pascal van Lieshout (Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON, Canada), and David Purcell (School of Commun. Sci. and Disord., Univ. of Western Ontario, London, ON, Canada)

There is a fast-growing literature examining speakers' response to real-time alterations of auditory feedback. The majority of these studies examine the response of the subject in acoustic terms. Since many subjects fail to (acoustically) compensate for the perturbation, the current experiment examines whether there are systematic articulatory responses to formant perturbation in the absence of compensation at the level of acoustics. Articulatory data are collected using a 3D electro-magnetic-articulograph. F2 is gradually shifted up or down and preliminary results from three English-speaking subjects showed that two subjects show no response in their acoustics or articulation. However, the remaining speaker who did not show compensation at the level of acoustics displayed a systematic response in some articulatory variables. The acoustic effects of his response were masked because the other articulators behaved in a more variable way, making the second formant vary randomly from trial to trial. Based on these results we expect to see a spectrum of response patterns from a larger population of speakers, from total non-compensation in both acoustics and articulation, partial compensation in articulation, and global articulatory compensation, which induces the appropriate compensation at the level of acoustic output.

2aSC54. Does compensation in auditory feedback require attention? Agnes Alsius, Takashi Mitsuya (Psychology, Queen's Univ., 62 Arch st., Humphrey Hall, Kingston, ON K7L 3N6, Canada, aalsius@gmail.com), and Kevin G. Munhall (Psych. & Otolaryngol., Queen's Univ., Kingston, ON, Canada)

When speakers receive auditory feedback with a real-time perturbation of formant structure, they hear themselves produce a vowel slightly different form the one intended. In response, they spontaneously change the formant structure to make the feedback more consistent with the intended sound. This compensatory behavior was reported to be automatic [Munhall *et al*. (2009)] because speakers are not able to suppress it even when they are informed about the perturbation and are instructed not to change their articulation. However, whether and to which extent attentional resources are utilized for this behavior have not been directly investigated. In the current study, speakers performed a speech production task where they pronounced a monosyllable whose formant structure was perturbed, while concurrently performing another task (i.e., dual-task paradigm). The preliminary results showed that, when attention was diverted to an unrelated auditory detection task, the magnitude of compensation remained the same as in the single task condition. Follow-up experiments will manipulate the nature and difficulty of the concurrent task to examine whether compensation in speech production is affected, and if so, what levels of the error feedback system are more susceptible to attentional manipulations.

2aSC55. Speech sensorimotor learning through a virtual vocal tract. Jeffrey J. Berry (Speech Pathol. & Audiol., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, jeffrey.berry@marquette.edu), Cassandra North, Benjamin Meyers, and Michael T. Johnson (Elec. & Comput. Eng., Marquette Univ., Milwaukee, WI)

Studies of speech sensorimotor learning often manipulate auditory feedback by modifying isolated acoustic parameters such as formant frequency or fundamental frequency using near real-time resynthesis of a participant's speech. An alternative approach is to engage a participant in a total remapping of the sensorimotor working space using a virtual vocal tract. To support this approach for studying speech sensorimotor learning, we have developed a system to control an articulatory synthesizer using electromagnetic articulography data. Articulator movement data from the NDI Wave System are streamed to a Maeda articulatory synthesizer. The resulting synthesized speech provides auditory feedback to the participant. This approach allows the experimenter to generate novel articulatory-acoustic mappings. Moreover, the acoustic output of the synthesizer can be perturbed using acoustic resynthesis methods. Since no robust speech-acoustic signal is required from the participant, this system will allow for the study of sensorimotor learning in any individuals, even those with severe speech disorders. In the current work, we present preliminary results that demonstrate that typically functioning participants can use a virtual vocal tract to produce diphthongs within a novel articulatory-acoustic workspace. Once sufficient baseline performance is established, perturbations to auditory feedback (formant shifting) can elicit compensatory and adaptive articulatory responses.

2aSC56. Exploring production-perception relationships in normal hearing and cochlear implant adults: A lip-tube perturbation study. Christine Turgeon and Amélie Prémont (Linguistique, UQAM, 320 Ste-Catherine Est, Montréal, QC H2X 1L7, Canada, turgeon.christine.2@courrier.uqam.ca)

It has been claimed that auditory feedback mechanisms enable monitoring and calibration of feedforward commands in speech production. Therefore, lack of auditory feedback may interfere with adequate compensation strategies in perturbed situations. This study investigates the effect of hearing status and a lip tube perturbation on vowel production. Eleven normal-hearing controls, and seventeen cochlear implant (CI) users (7 prelingually, 10 postlingually) were recorded during the production of the vowel /u/. Acoustic analyses were conducted with and without a 15-mm-diam tube inserted between the lips. Recording sessions were also made before and after the perturbation, with and without auditory feedback. Deaf participants' auditory feedback was provided by the CI and interrupted by switching off their implant devices. Separate analyses were conducted on the first (F1), the second formant (F2), and the fundamental frequency (F0). Results revealed a main effect of group and an interaction between condition and hearing status. Together, results suggest that auditory feedback plays an important role in speech compensation.

2aSC57. Acoustic vowel space and speech rate in Mandarin-speaking children with cerebral palsy. Wei Chen Hsu, Li-mei Chen, and Fan-Yin Cheng (Foreign Lang. & Lit., National Cheng Kung Univ., 1 University Rd., Tainan City 701, Taiwan, leemmay@gmail.com)

This study examines the variability in speech production in four Mandarin-speaking children: two with cerebral palsy (CP) and two typically developing (TD) from 4 to 5 years of age. Recordings collected from the picture-naming task and spontaneous interaction with adults was analyzed.

Acoustic vowel space and speech rate in their production were investigated. Study findings indicated the following: (1) Due to defect in speech motor control, children with CP have a smaller overall vowel space than TD children; (2) In CP group, there are more variability of formant values of individual vowels and the vowel space of individual vowels thus overlap more; (3) There is a trend of decrease of vowel formant values in both TD and CP; (4) Children with CP tend to spend more time in speech production because of their impaired speech-motor control, in terms of syllable per minute and intelligible syllable per minute; (5) Slower speech rate seems to increase speech intelligibility in CP. However, this needs to be verified in further studies. Extended longitudinal observation can provide more complete profile of individual differences in the development of vowels and speech rate to verify these preliminary findings. The variability features in the production of children with CP provide important references in speech therapy.

TUESDAY MORNING, 4 JUNE 2013 — 510A, 9:00 A.M. TO 11:40 A.M.

Session 2aSP

Signal Processing in Acoustics, Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Array Signal Processing for Three-Dimensional Audio Applications II

Yang Hann Kim, Cochair
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Jung-Woo Choi, Cochair
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Invited Papers

9:00

2aSP1. Spherical array processing with binaural sound reproduction for improved speech intelligibility. Noam R. Shabtai and Boaz Rafaely (Dept. of Elec. and Comput. Eng., Ben Gurion Univ. of the Negev, 17 Sheizaf St., Omer 84965, Israel, shabtai.noam@gmail.com)

In telecommunication applications, interfering sounds and reverberation can have a detrimental effect on speech intelligibility. For this reason, microphone arrays have been recently employed in telecommunication systems for natural environments. Currently applied array processing methods typically aim to produce array output, which is optimal on signal-based measures, e.g., signal-to-noise ratio (SNR). These measures may be particularly appropriate when the receiver is a machine. However, in order to enhance speech intelligibility when the receiver is another human, it may be desired to trigger spatial hearing capabilities of the human auditory system, such as the cocktail party effect. In particular, spatial-release from masking has been investigated. This work presents a spherical array signal processing framework in which array output is generated binaurally using the head-related transfer function. In this framework both target direction is enhanced and spatial information of all sources are perceived by the listener. The performance of the proposed binaural beamformer is compared to the performance of a non-binaural maximum directivity beamformer based on a spatial reproduction listening tests. The average percentage correct decision is calculated over five subjects and is shown to be higher when the binaural beamformer is used for every tested SNR.

9:20

2aSP2. Improvement of accuracy of 3D sound space synthesized by real-time "SENZI," a sound space information acquisition system using spherical array with numerous microphones. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Satoshi Hongo (Dept. of Design and Comput. Applications, Sendai National College of Technol., Sendai, Japan), Takuma Okamoto (National Inst. of Information and Commun. Technol., Soraku-gun, Japan), Yukio Iwaya (Dept. of Elec. Eng. and Information Technol., Tohoku Gakuin Univ., Tagajo-shi, Japan), and Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan)

We proposed a sensing method of 3D sound-space information based on symmetrically and densely arranged microphones mounted on a solid sphere. We call this method SENZI [Sakamoto *et al.*, ISUC2008 (2008)]. In SENZI, the sensed signals from each of the microphone is simply weighted and summed to synthesize a listener's HRTF, reflecting the listener's facing direction. Weighting coefficients are calculated for individual listeners based on their HRTFs. These coefficients are changed according to the listeners' head movement, which is known to provide important dynamic perceptual cue for sound localization. Therefore, accurate sound space information can be presented to unlimited number of listeners not only beyond the distance but also beyond the time. Recently, we realized this method as a real-time system using a 252-ch spherical microphone array and FPGAs. By using this system, accurate sound space information up to around 10 kHz can be synthesized to any listeners. However, the SNR of microphones affected to the accuracy of synthesized sound-space information, especially under low frequency region. To avoid the effect, we used condition numbers as an index to synthesize accurate sound-space information in the low frequency region.

9:40

2aSP3. Spatial audio coding with spaced microphone arrays for music recording and reproduction. Archontis Politis, Mikko-Ville Laitinen, and Ville Pulkki (Dept. of Signal Process. and Acoust., Aalto Univ., Otakaari 5A, Espoo 02150, Finland, archontis.politis@aalto.fi)

Spaced microphone arrays are commonly used in multichannel recording of music, due to their inherent quality of natural incoherence between the surround channels at reproduction, at the expense of accurate localization. Recent methods in parametric spatial audio coding, such as Directional Audio Coding, exploit coincident microphone patterns to extract directional information and reproduce it in a perceptually optimal way. In this study, we present how Directional Audio Coding can be adapted for spaced arrays, offering improved localization cues at reproduction, without compromising the qualities of the spaced microphone recording techniques. Examples are presented for some well-established array configurations.

10:00

2aSP4. Investigating perceptual attributes associated with reproduction of early reflections via virtually elevated loudspeaker. Sungyoung Kim (ECTET, Rochester Inst. of Technol., ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623, sungyoungk@gmail.com)

Reproducing early reflections related to "height" information through elevated loudspeakers delivers enhanced presence of auditory images and integrates with a three-dimensional visual content homogeneously. Nonetheless, it is practically difficult for consumers to place loudspeakers required for the height-channel reproduction in a listening room. To overcome this limitation, many academic or commercial institutions propose various methods that render vertical sound images and reproduce them with smaller number of loudspeakers that are typically located in the horizontal plane. The rendered image then could deliver vertically extended impression of a sound field, which is likely related to listeners' perception of enhanced presence. To better understand this relationship, this paper investigated idiosyncratic difference between one surround sound field and another with early reflections that is virtually elevated. The elicitation result revealed that listeners used four salient attributes—ASW, LEV, Powerfulness, and Clarity—to describe the difference. The subsequent result showed that perceived magnitudes of those percepts were accounted for by a physical parameter, correlation coefficient between the elevated signal and the loudspeaker signal that is to feed to the closest loudspeaker in the horizontal plane.

Contributed Papers

10:20

2aSP5. Approximate convolution using partitioned truncated singular value decomposition filtering for binaural rendering. Joshua Atkins, Adam Strauss, and Chen Zhang (Beats Electron., LLC, 1431 Ocean Ave., Apt. 1018, Santa Monica, CA 90401, joshatkins@ieee.org)

In conventional binaural rendering, a pair of head-related impulse responses (HRIR), measured from source direction to left and right ears, is convolved with a source signal to create the impression of a virtual 3D sound source when played on headphones. It is well known that using HRIRs measured in a real room, which includes a natural reverberant decay, increases the externalization and realism of the simulation. However, the HRIR filter length in even a small room can be many thousands of taps, leading to computational complexity issues in real world implementations. We propose a new method, partitioned truncated singular value decomposition (PTSVD) filtering, for approximating the convolution by partitioning the HRIR filters in time, performing a singular value decomposition on the matrix of filter partitions, and choosing the N singular-vectors corresponding to the N largest singular values to reconstruct the HRIR filters. We will show how this can be implemented in an efficient filter-bank type structure with N tapped delay lines for real-time application. We also show how improvements to the method, such as modeling the direct path HRIR separately can lead to improved rendering at minimal computational load.

10:40

2aSP6. The influence of regularization on anechoic performance and robustness of sound zone methods. Philip Coleman, Philip Jackson, Marek Olik (Ctr. for Vision, Speech and Signal Process., Univ. of Surrey, Guildford GU2 7XH, United Kingdom, p.d.coleman@surrey.ac.uk), Martin Olsen, Martin Møller, and Jan Abildgaard Pedersen (Bang & Olufsen a/s, Struer, Denmark)

Recent attention to the problem of controlling multiple loudspeakers to create sound zones has been directed toward practical issues arising from system robustness concerns. In this study, the effects of regularization are analyzed for three representative sound zoning methods. Regularization governs the control effort required to drive the loudspeaker array, via a constraint in each optimization cost function. Simulations show that regularization has a significant effect on the sound zone performance, both under ideal anechoic conditions and when systematic errors are introduced between calculation of the source weights and their application to the system. Results are obtained for speed of sound variations and loudspeaker positioning errors with respect to the source weights calculated. Judicious selection of the regularization parameter is shown to be a primary concern for sound zone system designers—the acoustic contrast can be increased by up to 50 dB with proper regularization in the presence of errors. A frequency-dependent minimum regularization parameter is determined based on the conditioning of the matrix inverse. The regularization parameter can be further increased to improve performance depending on the control effort constraints, expected magnitude of errors, and desired sound field properties of the system.

11:00

2aSP7. Sound field planarity characterized by superdirective beamforming. Philip J. Jackson (CVSSP, Dept. of Electron. Eng., Univ. of Surrey, Guildford, Surrey GU2 7XH, United Kingdom, p.jackson@surrey.ac.uk), Finn Jacobsen (Acoust. Technol., Tech. Univ. of Denmark, Lyngby, Denmark), Philip D. Coleman (CVSSP, Univ. of Surrey, Guildford, United Kingdom), and Jan Abildgaard Pedersen (Acoustics, Bang & Olufsen a/s, Struer, Denmark)

The ability to replicate a plane wave represents an essential element of spatial sound field reproduction. In sound field synthesis, the desired field is often formulated as a plane wave and the error minimized; for other sound field control methods, the energy density or energy ratio is maximized. In all cases and further to the reproduction error, it is informative to characterize how planar the resultant sound field is. This paper presents a method for quantifying a region's acoustic planarity by superdirective beamforming with an array of microphones, which analyzes the azimuthal distribution of impinging waves and hence derives the planarity. Estimates are obtained for a variety of simulated sound field types, tested with respect to array orientation, wavenumber, and number of microphones. A range of microphone configurations is examined. Results are compared with delay-and-sum beamforming, which is equivalent to spatial Fourier decomposition. The superdirective beamformer provides better characterization of sound fields and is effective with a moderate number of omni-directional microphones over a broad frequency range. Practical investigation of planarity estimation in real sound fields is needed to demonstrate its validity as a physical sound field evaluation measure.

11:20

2aSP8. Analysis of pre-echo artifact in generating a focused source. Min-Ho Song (Grad. School of Culture Technol., KAIST, YuseongGu Gusung Dong 373-1, Daejeon 373-1, South Korea, godspd@kaist.ac.kr), Jung-Woo Choi, and Yang-Hann Kim (Ctr. for Noise and Vib. Control, Dept. of Mech. Eng., KAIST, Daejeon, South Korea)

A focused source is a virtual source that can provide an auditory illusion of sound radiating between the loudspeaker array and listener. When generating a focused source, a listener cannot avoid undesired pre-arriving direct waves from the control sources, which is known as pre-echo artifact. Investigation of the artifact can be seen in several researches; however, no mathematical definition of the pre-echo artifact is given so far and only the observation in limited case is known with computer simulation. The

objective of this paper is to observe the cause and effect of pre-echo artifact analytically. The paper defines the pre-echo artifact mathematically, and the artifact at the arbitrary listening position is formulated with integral form based on Kirchhoff-Helmholtz integral equation. From the definition of the pre-echo, it is shown that the convergent wave of a focused source can be regarded as a special case of a pre-echo artifact. Furthermore, the derivation shows that the pre-echo artifact occurs in the case of using continuous array and is evolved due to the time-reversed nature of the solution.

TUESDAY MORNING, 4 JUNE 2013 — 511AD, 9:00 A.M. TO 12:00 NOON

Session 2aUW

Underwater Acoustics and Acoustical Oceanography: Wave Propagation in a Random Medium

John A. Colosi, Chair
Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943

Contributed Papers

9:00

2aUW1. Time-varying three-dimensional mapping of internal waves during the Shallow Water 2006 experiment. Mohsen Badiey, Lin Wan, and Aijun Song (College of Earth, Ocean, and Environ., Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Formation and propagation of internal waves have recently become of interest to ocean acousticians, since the propagation of internal waves in shallow water waveguide plays an important role in intensity fluctuations of acoustic signals [J. Acoust. Soc. Am. **112**(2), 747–760 (2007)]. Modeling the acoustic field in these regions requires detailed knowledge of sound speed and its spatial and temporal variability resulting from propagating internal waves. Although satellite imagery can provide snapshots of the surface impressions of the internal waves, due to low sampling in time (limited images in each orbit) other techniques to obtain the time varying, three-dimensional (3D) internal wave field are desirable. An example to obtain a time-varying, 3D internal wave field is presented in this paper. The internal wave fine structure is reconstructed using simultaneous measurement of temperature data using thermistor arrays and the surface impressions of a moving internal wave packet using a ship's radar during the Shallow Water 2006 experiment (SW06). The parameters of the internal wave train, such as wave speed, propagation direction, and amplitude of the first wave front, are determined. The resulting temperature field induced by the internal waves is used as environmental input to a 3D acoustic model to study the effects of internal wave on acoustic propagation. [Work supported by ONR322OA.]

9:20

2aUW2. The effect of surface and linear internal waves on higher order acoustic moments in shallow water. Kaustubha Raghukumar and John A. Colosi (Oceanography, Naval Postgrad. School, 315 B Spanagel Hall, 833 Dyer Rd., Monterey, CA 93943, kraghuku@nps.edu)

Acoustic fields in shallow water have a statistical nature due to complex, time-evolving sound speed fields, and scattering from rough boundaries. Previously, coupled-mode transport theory [Raghukumar and Colosi (2012)] was applied to high frequency acoustic fluctuations in an environment typical of the Shallow Water 2006 (SW06) experiment on the New Jersey continental shelf. As a consequence of the strong adiabatic component in SW06 propagation, a hybrid approach was used to calculate mode coherences where mode energies from the Dozier-Tappert approach were combined with adiabatic phase terms. Mode energies, coherences and acoustic intensities were examined, and it was found that internal and surface waves preferentially couple low and high modes respectively. Here, we extend that study to include higher moments such as scintillation index and shift focus to modes that are coupled by both internal and surface waves. Oceanographic and sea surface measurements are used to constrain the internal wave and sea surface models. The relative importance of linear internal waves and surface scattering effects are studied using transport theory and Monte Carlo simulations.

9:40

2aUW3. The effects of internal tides on phase and amplitude statistics in the Philippine Sea. John A. Colosi, Tarun Chandrayadula, Weston Coby, Jacob Fischer (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu), Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Moored oceanographic sensors and satellite altimetry has revealed energetic diurnal and semi-diurnal internal tides in the Western Philippine Sea. Because the internal tides have a complex spatio-temporal pattern and large vertical displacements, these waves have the potential for causing strong acoustic variability. This talk will present a tidal analysis of signal fluctuations from the PhilSea09 experiment in which broadband signals with a center frequency of 275 Hz and a bandwidth of 50 Hz were transmitted at the sound channel axis to a large aperture vertical array 180-km distant. Signal phase and amplitude statistics along distinct branches of the observed wavefronts will be analyzed and compared to ray-based model predictions using internal tide information obtained from moored oceanographic instruments at the source and receiver. Key issues are the acoustic effects of the internal tide nonlinearity, temporal stability, high mode structure, and complex horizontal interference patterns.

10:00

2aUW4. Comparison of transport theory predictions with measurements of the decrease in shallow water reverberation level as the sea state increases. Eric I. Thorsos, Jie Yang, W. T. Elam, Frank S. Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu), Fenghua Li, and Jianjun Liu (State Key Lab. of Acoust., Inst. of Acoust., Beijing, China)

Transport theory has been developed for modeling shallow water propagation and reverberation at mid frequencies (1–10 kHz) where forward scattering from a rough sea surface is taken into account in a computationally efficient manner. The method is based on a decomposition of the field in terms of unperturbed modes, and forward scattering at the sea surface leads to mode coupling that is treated with perturbation theory. Reverberation measurements made during ASIAEX in 2001 provide a useful test of transport theory predictions. Modeling indicates that the measured reverberation was dominated by bottom reverberation, and the reverberation level at 1 and 2 kHz was observed to decrease as the sea surface conditions increased from a low sea state to a higher sea state. This suggests that surface forward scattering was responsible for the change in reverberation level. By modeling the difference in reverberation as the sea state changes, the sensitivity to

environmental conditions other than the sea surface roughness is much reduced. Transport theory predictions for the reverberation difference are found to be in good agreement with measurements. [Work supported by the U.S. Office of Naval Research, Ocean Acoustics.]

10:20

2aUW5. Scales of time and space variability of sound fields reflecting obliquely from underwater slopes. Timothy Duda, Ying-Tsong Lin (Woods Hole Oceanogr. Inst., AOPE Dept MS 11, Woods Hole, MA 02543, tduda@whoi.edu), and Bruce D. Cornuelle (Scripps Inst. of Oceanogr., La Jolla, CA)

Ocean flows impart a time dependence to interference patterns of sound reflecting at low horizontal angle from sloping bathymetry. This is a higher level of complexity than interference patterns within vertical slices in spherically symmetric environments. The time-space statistics may be co-dependent and non-normal, making conventional methods inapplicable. Here, patterns that occur when 100 to 1000 Hz sound reflects at slopes are simulated with three-dimensional methods. Multiple statistics of sound at virtual arrays are computed over the domain, including incoherent power, beam power, spatial correlation length, and array gain assuming idealized noise conditions. These depend on source and receiver locations with respect to bathymetric features and can be computed for instantaneously sampled ocean conditions to study their evolution within the time frame of the dominant flow, or computed via averaging over a few periods of the dominant flow features (tides, for example). Here, the spatial complexity of patterns found in 100-Hz (and upward) simulations at a slope area off San Diego, CA, in a time varying flow are linked to the imposed seafloor roughness as well as geometry, with mean intensity, scintillation index, and correlation scale used to quantify the effect. Implications for statistics employed in detection and tracking algorithms are discussed.

10:40

2aUW6. The effect of random fluctuations in bottom bathymetry on acoustic coherence in shallow water. Harry A. DeFerrari and Jennifer Wylie (Appl. Marine Phys., Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, hdeferrari@rsmas.miami.edu)

The loss of temporal coherence after long range propagation in shallow water is often studied as a consequence of sound speed variability from internal waves. Here, we add the complication of small amplitude and very long wavelength random fluctuations of bottom bathymetry. It is shown that the same range dependent sound speed fluctuations result in markedly different coherence times depending on acoustic wavelength and mode number—a first order effect. A range dependent PE code (MMPE) is used to predict temporal coherence for individual surface reflected-bottom-reflected (SRBR) mode arrivals. Here, a mode coherence calculation is developed and compared for varying RMS bathymetry. Temporal coherence is inferred from mode coherence. We find first order and /or low frequency modes are insensitive to the bottom but when the (sine of the mode angle approaches 1/10 of an acoustic wavelength) the modes structure in amplitude and phase is randomized and the signal decorrelate rapidly in time from just the slightest temporal variations in sound speed. It does not take much; just 1 m in 200 m of range will randomize all but the first mode at mid frequencies (0.5 to 1 kHz). Predictions are in close agreement with SW06 mode coherence measurements.

11:00

2aUW7. Comparison of statistics of controlled source tones and single ship noise in the deep ocean. Brianne M. Baxa, Gerald L. D'Spain, Peter F. Worcester, Matthew A. Dzieciuch (Scripps Institution of Oceanography, UC San Diego, 2498 Manzana Way, San Diego, CA 92139, bmoskovitz@gmail.com), Kevin D. Heaney (OASIS Inc., Fairfax Station, VA), Jim A. Mercer (Univ. of Washington, Seattle, WA), and Art B. Baggeroer (Massachusetts Inst. of Technol., Cambridge, MA)

The deep ocean experiment, PhilSea09, was conducted April-May, 2009, in the central part of the northern Philippine Sea (22d N, 126d E). During one period in the experiment, the R/V Melville was station-keeping 35 km from the Distributed Vertical Line Array (DVLA) while seven tones, from 79 Hz to 535 Hz, were transmitted from a controlled source suspended below the ship. Recordings on the 1000-m section of the DVLA centered on the surface conjugate depth at 5026 m were dominated by the noise of this ship except at the controlled source tone frequencies. Using non-parametric statistical tests, the statistics of the spectral envelope at the tone frequencies are compared to the statistics of those for the nearby ship-noise-dominated frequency bins. When a high tone-signal to ship-noise ratio exists, the statistics of the tones differ from those of the ship noise at the 5% level of significance by the Kolmogorov-Smirnov two-sample test. Tone statistics are seen to be Gaussian distributed at frequency bands of low tone-signal to ship-noise ratios, whereas at high signal to noise ratios, the controlled source tones are non-Gaussian. When the signal to noise ratio is high, the statistics of the tone and of the noise are from different distributions. Both the tones and shipping noise travel approximately the same path to the DVLA, so these differences in the received field statistics represent differences in the statistical properties of these two acoustic sources themselves, not of the environment.

11:20

2aUW8. Probability distribution for energy of saturated broadband ocean acoustic transmission: Results from Gulf of Maine 2006 experiment. Duong D. Tran and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, d.tran@neu.edu)

The probability distribution of ocean-acoustic broadband signal energy after saturated multipath propagation is derived using coherence theory. The frequency components obtained from Fourier decomposition of a broadband signal are each assumed to be fully saturated with energy spectral density that obey the exponential distribution with 5.6 dB standard deviation and unity scintillation index. When the signal bandwidth and measurement time are larger than the correlation bandwidth and correlation time respectively of its energy spectral density components, the broadband signal energy obtained by integrating the energy spectral density across the signal bandwidth then follows the Gamma distribution with standard deviation smaller than 5.6 dB and scintillation index less than unity. The theory is verified with broadband transmissions in the Gulf of Maine shallow water waveguide in the 300–1200 Hz frequency range. The standard deviations of received broadband signal energies range from 2.7 to 4.6 dB for effective bandwidths up to 42 Hz, while the standard deviations of individual energy spectral density components are roughly 5.6 dB. The energy spectral density correlation bandwidths of the received broadband signals are found to be larger for signals with higher center frequencies and are roughly 10% of each center frequency.

11:40

2aUW9. Analysis of horizontal wave number statistical characteristics to the problem of sound propagation in two-dimensional-fluctuating shallow sea with losses. Oleg Gulin and Igor Yaroshchuk (Ocean Acoust., V.I. Il'ichev Pacific Oceanological Inst. FEB RAS, 43, Baltiyskaya St., Vladivostok, Primorskiy Kray 690041, Russian Federation, gulinoe@rambler.ru)

On the basis of the approach previously proposed by authors, a problem of the middle-frequency sound propagation in 2D-fluctuating shallow sea with losses is considered. Statistical characteristics of horizontal wave numbers corresponding to modal ones are studied. Fluctuations of horizontal wave numbers determine statistical features of wave field in random sea medium if wave field is sought by modal expansion. Within the framework of adiabatic approximation, we present calculations of sound field statistical moments, which demonstrate an effect of transmission loss attenuation along the horizontal distance. There are no references in acoustic literature for this fact. Some estimation has been carried out to explain new effect associated with two reasons that are the medium losses and sound speed fluctuations. They together influence wave numbers of modes in such a way attenuating losses.

Session 2pAAa

Architectural Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Adapting, Enhancing, and Fictionalizing Room Acoustics II

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362

Alex Case, Cochair
Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854

Invited Papers

1:00

2pAAa1. Amplified music and measurement results regarding inflatable membrane absorber technology. Niels W. Adelman-Larsen (Flex Acoust., Diplomvej 377, Kgs. Lyngby 2800, Denmark, nwl@flexac.com)

Previous studies and experience have shown that what distinguishes the best from the less well liked venues for pop and rock music is a shorter reverberation time in the 63–250 Hz octave bands. Since a longer reverberation time in these bands is needed in order to obtain warmth at classical music concerts, variable acoustics must address these frequencies in order to provide the best results in multi-purpose halls. This paper will expand on research on recommendable acoustics for amplified music. Certified measurements from reverberation chambers and installed systems on a patented, inflatable, on/off absorption technology are presented. Since the technology can be used in the entire ceiling area, the T30 of a hall can be lowered by almost 50% in the important octave bands for pop and rock music. Absorption coefficients are almost constant across the frequency bands from 63 to 1000 Hz. The technology, which is the only passive solution to enable variability also at the important lower frequencies, is meant to be used in any hall where both classical as well as amplified music is being played, such as in music schools and performing arts centers.

1:20

2pAAa2. The role of high-frequency cues for spatial hearing in rooms. Hari Bharadwaj, Salwa Masud, and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol. (CompNet), Boston Univ., 19 Euston St., 1B, Brookline, MA 02446, hari@nmr.mgh.harvard.edu)

The ability to attend to a sound source of interest while ignoring competing sounds is vital to navigating everyday acoustic scenes. Commonly, this ability depends on the ability to focus on a sound source using acoustic spatial cues, particularly interaural time differences (ITDs) and interaural level differences (ILDs). Based on past studies of localization in anechoic settings, low-frequency ITDs have been thought to dominate perception of source location. However, reverberant environments differentially degrade ITDs and ILDs, which may affect their relative influence on localization. Moreover, a recent study suggests that ILDs play a bigger role on spatial perception in reverberant settings than in anechoic settings. Here, in a series of localization and spatial attention experiments using high-pass, low-pass and broadband sounds, we tested the hypothesis that high-frequency ILD and envelope ITD cues are important for spatial judgments in reverberant rooms. We also measured the brainstem frequency following responses (FFRs) of individual subjects in response to click trains and spoken syllables. Results suggest that compared to in anechoic space, in reverberant settings, high-frequency cues are more reliable and influential on perception and that the strength of FFR phase locking to stimulus envelope predicts how well individual listeners can direct spatial attention.

1:40

2pAAa3. Internet rooms from internet audio. Chris Chafe and John Granzow (CCRMA/Music, Stanford Univ., CCRMA/Music, Stanford, CA 94305, cc@ccrma.stanford.edu)

Music rehearsal and concert performance at a distance over long-haul optical fiber is a reality because of expanding network capacity to support low-latency, uncompressed audio streaming. Multichannel sound exchanged across the globe in real time creates “rooms” for synchronous performance. Nearby connections work well and musicians feel like they are playing together in the same room. Larger, continental-size, distances remain a challenge because of transmission delay and seemingly subtle but perceptually important cues which are in conflict with qualities expected of natural rooms. Establishing plausible, room-like reverberation between the endpoints helps mitigate these difficulties and expand the distance across which remotely located musicians perform together comfortably. The paper presents a working implementation for distributed reverberation and qualitative evaluations of reverberated versus non-reverberated conditions over the same long-haul connection.

2:00

2pAAa4. Designing a stackable diffusor/absorber tailored to a violin and cello practice room. Myoung woo Nam, Kyogu Lee (Trans-Disciplinary Studies, Seoul National Univ., D406, Iuidong 864-1, Yeongtonggu, Suwon, Gyeonggi 443-270, South Korea, mnam@snu.ac.kr), and Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts, Lowell, MA)

As distinct from a much larger concert hall, the typical practice room directs added acoustic emphasis to small room challenges such as room resonances and unwanted reflections. Although rich reverberation is not easily achieved in a small space, the proposed diffusor/absorber seeks to make practice more acoustically comfortable and rewarding. The treatment is designed to attenuate the spectral portion

of the violin and cello sound that engages the room resonances. When musicians play violin and cello, omni-directional low frequencies are primarily produced in lower elevation of the room, while more directional higher frequencies of interest to the performer are directed more to upper area. This diffusor/absorber is designed to provide more absorption for bottom area and more diffusion for upper area. In this configuration, the diffusor/absorber gives comfortable acoustical conditions for musicians to practice. Based on sound propagation characteristics [J. Meyer, "The sound of the orchestra," J. Audio Eng. Soc. **41**(4) (1993)] and formant information of violin and cello [M. Nam and K. Lee, "Analyzing string instrument formant," in *Proceeding of Acoustical Society of Korea Conference* (2011)], this design proposes a moveable acoustic panel-box suitable for a typical musician's home practice room and a small sized recording studio.

Contributed Papers

2:20

2pAAa5. Improving the indoor sound quality by using cymatic shapes. Alaa S. Algargoosh (College of Design, Univ. of Dammam, 3140 Alabbas bin Ali, Dammam 32433-4614, Saudi Arabia, alaa_algargoosh@yahoo.com), Hany Hossam Eldien (College of Architecture and Planning, Univ. of Dammam, Dammam, Saudi Arabia), and Hala El Wakeel (College of Design, Univ. of Dammam, Dammam, Saudi Arabia)

Acoustic diffusers are important components in enhancing the quality of room acoustics. This paper investigates a new type of 2D diffusers obtained by the Cymatics phenomena. Cymatics is the study of sound and vibration made visible, typically on the surface of a plate, diaphragm, or membrane. Four shapes of the diffusers were designed by the Cymatic shapes and modeled by using a quadratic residue sequence. The polar response of the diffusers was measured using DIRAC software. Polar response results were generally consistent with expectations. This type of diffusers can generate a uniform polar response over the frequency range we are interested in (400–4000 Hz). It is found that this type of acoustic diffusers can be used to maintain the acoustic energy in a room and at the same time can treat unwanted echoes and reflections by scattering sound waves in many directions.

2:40–3:00 Break

3:00

2pAAa6. Sound concentration caused by curved surfaces. Martijn Vercammen (Peutz, Lindenlaan 41, Molenhoek 6584 AC, Netherlands, m.vercammen@mook.peutz.nl)

In room acoustics, the focusing effect of reflections from concave surfaces is a well-known problem. The occurrence of concave surfaces has tended to increase in modern architecture, due to new techniques in design, materials, and manufacturing. Focusing can cause high sound pressure levels, sound coloration, or an echo. Although the problem is well known, the amount of amplification that occurs in the focusing point and the sound field around the focusing point are not. The pressure in the focusing point can only be calculated using wave-based methods. An engineering method that is based on the Kirchhoff Integral is presented to approximate the reflected sound field in and around the focusing point for a few basic geometries. It will be shown that both the amplification and the area of the focusing is strongly related to wavelength. The focusing caused by surfaces that are curved in two directions (sphere, ellipsoid) is much stronger than that caused by surfaces that are curved in only one direction (cylinders). This method enables designers to evaluate and thereby improve or redesign the geometry. The method is illustrated with a few examples.

3:20

2pAAa7. Can we use the standard deviation of the reverberation time to describe diffusion in a reverberation chamber? Margriet Lautenbach and Martijn Vercammen (Peutz, Paletsingel 2, Zoetermeer 2718 NT, Netherlands, m.lautenbach@zoetermeer.peutz.nl)

It is generally assumed that the limited diffusion properties of reverberation rooms, especially with a strongly sound absorbing sample, is the main reason for the bad reproducibility values for the sound absorption between laboratories. Reverberation rooms should be made much more diffuse to reduce the inter laboratory differences. Although there are practical ways to achieve this, it is most important that there will be a requirement in the ISO 354 standard on the diffusing quality of the sound field. One possibility is to use the standard deviation of the reverberation time for different source-microphone combinations in the reverberation room. Measurements are performed to investigate the influence of different settings of a reverberation room on the standard deviation of the reverberation time, compared to the theoretical standard deviation. This is done with the interrupted impulse method and the integrated impulse method. The results will be presented in this paper. The usefulness of this qualification method for the ISO standard will be discussed.

3:40

2pAAa8. A new third generation time variant electro-acoustic enhancement system. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

This paper describes the hardware and software implementation in a new generation of time variant electronic acoustic enhancement systems designed for use in medium and large sized venues. Examples of currently installed systems and applications will be discussed, as well as capabilities for sound file storage and playback, multi-channel film surround sound, 2D effects panning, and 3D live tracking.

4:00

2pAAa9. Optimization of electroacoustic resonators for semi-active room equalization in the low-frequency range. Etienne Rivet (Laboratoire d'Électromagnétisme et d'Acoustique, École Polytechnique Fédérale de Lausanne, EPFL - STI - IEL - LEMA, Station 11, Lausanne, Vaud 1015, Switzerland, etienne.rivet@epfl.ch), Romain Boulandet, and Hervé Lissek (Laboratoire d'Électromagnétisme et d'Acoustique, École Polytechnique Fédérale de Lausanne, Lausanne, Switzerland)

At low frequencies in listening rooms, standing waves cause large frequency-response variations within the whole space. These unwanted phenomena have a significant impact on the sound quality of an audio system rendering. Unfortunately, state-of-the-art soundproofing solutions cannot efficiently handle such low-frequency sound energy. To alleviate this problem, electroacoustic resonators can be used to damp room modes. This concept is based on the connection of direct-radiator loudspeakers to synthetic electrical loads allowing the passive dissipation of a certain part of the incoming acoustic energy of the sound field. Through judicious control of acoustic impedance, and depending of the placement of the electroacoustic resonators in the room, a significant damping of the dominant natural resonances can be achieved in order to meet the specifications of sound reproduction. This paper presents the design of prototypes of electroacoustic resonators and investigates their optimization and spatial arrangement in the perspective of semi-active room equalization.

4:20

2pAAa10. Physical and subjective factors of spatial envelopment impression of surround sound reproduction. Toru Kamekawa and Atsushi Marui (Musical Creativiry and the Environ., Tokyo Univ. of the Arts, 1-25-1, Senju, Adachi-ku, Tokyo 120-0034, Japan, kamekawa@ms.geidai.ac.jp)

How we feel the envelopment of reproduced sound? For the evaluation of spatial impression, "envelopment" is one of the key factors among several attributes in audio reproduction. In the authors' past research, three attributes were elicited from participants regarding surround sound reproduction using triadic elicitation procedure. The three attributes are "brightness," "temporal separability," and "spatial homogeneity of envelopment." In this paper, authors focused on "spatial homogeneity of envelopment" and considered related physical parameters such as ESC (ear signal correlation; defined as the correlation between left and right ears' signal). From comparison between the result of subjective test and the data measured with a dummy-head facing toward different

angles, it was found that the correlation between the ESC of different head angles and the amount of differences of ESC from focused sound to diffused sound contribute the sense of "spatial homogeneity of envelopment."

4:40

2pAAa11. Influence of low-order room reflections on sound zone system performance. Marek Olik (CVSSP, Dept. of Electron. Eng., Ctr. for Vision Speech and Signal Process. (CVSSP), Univ. of Surrey, Guildford GU2 7XH, United Kingdom, m.olik@surrey.ac.uk), Philip Jackson, Philip Coleman (CVSSP, Dept. of Electron. Eng., Univ. of Surrey, Guildford, United Kingdom), Martin Olsen, Martin Møller, and Søren Bech (Bang & Olufsen, Struer, Denmark)

Studies on sound field control methods able to create independent listening zones in a single acoustic space have recently been undertaken due to the potential of such methods for various practical applications, such as individual audio streams in home entertainment. Existing solutions to the problem have shown to be effective in creating high and low sound energy regions under anechoic conditions. Although some case studies in a reflective environment can also be found, the capabilities of sound zoning methods in rooms have not been fully explored. In this paper, the influence of low-order (early) reflections on the performance of key sound zone techniques is examined. Analytic considerations for small-scale systems reveal strong dependence of performance on parameters such as source positioning with respect to zone locations and room surfaces, as well as the parameters of the receiver configuration. These dependencies are further investigated through numerical simulation to determine system configurations which maximize the performance in terms of acoustic contrast and array control effort. The design rules for source and receiver positioning are suggested, for improved performance under a given set of constraints such as a number of available sources, zone locations, and the direction of the dominant reflection.

TUESDAY AFTERNOON, 4 JUNE 2013 — 513DEF, 12:55 P.M. TO 5:20 P.M.

Session 2pAAb

Architectural Acoustics and Noise: Dah-You Maa—His Contributions and Life in Acoustics

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180

Jing Tian, Cochair
Inst. of Acoust., Chinese Acad. of Sci., 21 Beisihuanxilu, Beijing 100190, China

Chair's Introduction—12:55

Invited Papers

1:00

2pAAb1. Dah-You Maa, friend and scholar. Leo L. Beranek (Westwood, MA) and Ning Xiang (Grad. Prog. in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY, Xiangn@rpi.edu)

Maa, Dah-You's life as a scholar and a close friend of many both in China and North America is the theme. A number of his scholarly achievements from his UCLA and Harvard time, to the post-Cultural Revolution period, up to recent years, is reviewed. His two years as a research fellow at UCLA and Harvard which ended with his receiving his Ph.D. at Harvard University in 1940 is detailed. Then began his life in Kunming during World War II where he was a professor in the E.E. Department of the National S.-W. Associated University. He returned to Beijing and became the first Dean of Engineering in the National Peking University (now Beijing University). During the Cultural Revolution, 1966–1976, he was under house arrest. His sponsorship of the first All-China Acoustics Conference after the Cultural Revolution in 1979 heralded his position as China's leading acoustician. Excerpts from his many letters through the years complete the presentation.

1:20

2pAAb2. Prof. Dah-You Maa's Contribution to Acoustics. Jing Tian (Inst. of Acoust., Chinese Acad. of Sci., 21 Beisihuanxilu, Beijing 100190, China, tian@mail.ioa.ac.cn)

In his life-long career pursuing, Professor Dah-You Maa contributed greatly in many areas of acoustics, not only in acoustical research and development, but also in the promotion of acoustical education, application, and legislation. Professor Maa presented a simplified method for calculation of normal modes for room acoustics, invented micro-perforated panel absorbers and micro-perforation jet mufflers, gave a formula of jet noise power via air pressure, and acoustically designed the first and biggest Congress Hall, built the first set of acoustical laboratories, and established the acoustical standard system in China. He also supervised tens of postgraduate students working in environmental acoustics, building acoustics, speech signal processing, nonlinear acoustics, and active noise control. In this paper, the main contribution of Professor Maa is introduced. Several of his typical research works, such as micro-perforated panel absorbers and micro-perforation jet mufflers, are explained in detail, from which we can well appreciate the physical insights and theoretical skills of Professor Maa.

1:40

2pAAb3. Dah-You Maa: Most senior academic brother. David T. Blackstock (Appl. Res. Labs. & Mech. Eng. Dept., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, dtb@austin.utexas.edu)

F. V. Hunt's ONR-supported acoustics lab at Harvard turned out 30 PhD graduates after WW II. As one of those students in the late 1950s, I gradually became aware that Hunt had had a prewar group of graduate students as well. Leo Beranek was Hunt's first Ph.D., Dah-You Maa his second (both in 1940). Maa's two 1939 JASA articles on room acoustics, one coauthored with Beranek and Hunt, were benchmark papers of the day. I became fascinated with Maa's story, partly because he seemed so completely unreachable. He had returned to China during the war; afterward the Cold War intervened. I finally met my much older academic brother in 1980, at the 10th ICA in Sydney, Australia. Thus began a warm and rewarding relationship. In 1987, he began representing China on the International Commission on Acoustics, and for 7 years, we saw each other annually at Commission meetings. I learned of his work on nonlinear standing waves, a problem in which I too shared a keen interest. The high point was at the 14th ICA in Beijing. I was finally able to see his laboratory and meet his doctoral student Ke Liu. A memorable dinner followed that evening.

2:00

2pAAb4. Dah-You Maa and the many facets of modal density. Richard H. Lyon (MIT, 60 Prentiss Lane, Belmont, MA 02478-2021, rhlyon@lyoncorp.com)

Maa's formula for the modal density of a room is known to everyone in acoustics. The wonderful part of it is that it so logical and direct that you don't have to memorize or look it up - you can simply re-derive it on the spot. When it was published by Maa in 1939, there was a competing formulation by Dick Bolt that had the same volume dependent term, but was quite different in the surface area and edge length terms. But by the time of the landmark paper by Morse and Bolt in 1944 ["Sound waves in rooms", Rev. Mod. Phys. **16**(1) (1944)], Maa's formula had won and was the relationship cited by Morse and Bolt. The uses of modal density in the response statistics of acoustical spaces and structures for the purposes of estimating impedances, energy flow, and phase statistics have grown as recognition of this fundamental property of resonant systems has grown.

2:20

2pAAb5. Maa Dah-You and the design of reverberant rooms for determination of sound power. George C. Maling (60 High Head Rd., Harpswell, ME 04079, maling@alum.mit.edu)

Reverberation rooms are useful tools for the determination of sound power emitted by machines and other sound sources. One critical factor in the design of such rooms is the mode spacing at low frequencies. A seminal contribution to our understanding of mode spacing was made by Maa in 1939, and his paper was published in the Journal of the Acoustical Society of America. Richard Bolt also made a contribution at about the same time. In this paper, we begin with Maa's work and trace the development of mode spacing statistics through the work of Richard H. Bolt and Ludwig Sepmeyer. A computer study by the author of mode spacing statistics in 200 cubic meter rooms is described. This work led to recommendations for room dimensions to be included in an international standard on determination of sound power in reverberation rooms.

2:40

2pAAb6. Remembering one of Maa Dah-You's last essential contributions to acoustics. Jiqing Wang (Inst. of Acoust., Tongji Univ., 1239 Siping Rd., Shanghai 200092, China, wongtsu@126.com)

The book *The fundamentals of Modern Acoustic Theory* written by Maa Dah-You was published (in Chinese) in March 2004 at his age of 88. This 421 pages volume with 15 chapters covers almost all main acoustic fields from basics to many extended areas, such as linear to non-linear acoustics, common acoustical measures to active control, and the aerodynamic sound and thermo-acoustics. References listed in each chapter include many Maa's pioneering work of the related acoustic fields. This book systematically reflects his deeper understanding of acoustics theory, and further account new developments of acoustic technology. Moreover, the history of acoustics, particularly of the ancient China in the beginning sections of the first chapter, and the overview of the future developments in the afterword are also the exclusive items in this book. A well prepared exercise at the end of each chapter is most helpful for better understanding and thinking of the related topics. Therefore, it is widely used as a text or main reference book in many universities and research community in China.

3:00–3:20 Break

3:20

2pAAb7. Three months in Beijing—Micro-perforated absorbers, eigenvalues, and other topics. Christian Nocke and Catja Hilge (Akustikbuero Oldenburg, Katharinenstr. 10, Oldenburg 26121, Germany, nocke@akustikbuero-oldenburg.de)

In this contribution, a personal and scientific report on a three month exchange stay at the Institute of Acoustics in Beijing will be reported. Micro-perforation and low frequencies in small rooms have been the background of an investigation under supervision of Professor Daa-You Maa in 1999. Maa's ideas from the 1930s about the distribution on eigenmodes in rooms in combination with micro-perforated absorbers as reference absorbers had been the basis of this work. Abacus and computer calculations as the basis of old/new ideas formed the background of a stay as a student. Even after more than 12 years, micro-perforation is part of the daily work as consultant often dealing with new product ideas and developments. The presentation will give a very brief overview on this work that started in Beijing. Furthermore, in remembrance of a once in a life time experience personal experiences with Professor Daa-You Maa, his wife, and his colleagues will be reported.

3:40

2pAAb8. Surface waves over rigid-porous and rough surfaces. Keith Attenborough, Imran Bashir, and Shahram Taherzadeh (DDEM, The Open Univ., Walton Hall, Milton Keynes MK7 6AA, United Kingdom, k.attenborough@open.ac.uk)

Surface waves are created by near grazing sound propagation from a point source over either a rigid-porous layer or a (slightly) rough surface. In theory, they have similar origins. Frequency- and time-domain measurements have been made on surfaces composed from parallel periodically spaced rectangular strips (width 0.0126 m, height 0.0253 m) on an acoustically hard surface. The edge-to-edge spacing between the strips has been varied between 0.003 and 0.06 m. Frequency domain predictions show that when the spacing is substantially smaller than the strip height these surfaces may be regarded as locally reacting rigid-framed hard-backed porous layers with an effective depth slightly larger than the strip height. When the spacing is comparable to the strip height or greater the surfaces behave as periodically rough surfaces. Both frequency- and time-domain results show that surface waves of comparable magnitudes are created over the range of strip spacings studied but the main frequency content of these acoustically-induced surface waves is lowered as the mean spacing is increased. These data suggest that the surface waves have a similar physical origin and that they can be created also over a micro-perforated surface.

4:00

2pAAb9. Investigation of sound scattering by a surface using the acoustical wave propagator method. Jie Pan, Hongmei Sun, and James Leader (Mech. and Chem. Eng., The Univ. of Western Australia, 35 Stirling Highway, Crawly, WA 6009, Australia, jie.pan@uwa.edu.au)

The time domain acoustical wave propagator (AWP) method is used as a tool to study the sound scattering properties of a surface. As the method allows the modeling of the excitation, absorption, and scattering of sound in a vicinity of the surface within a very short period of time, the effect of other surfaces on these process can be ignored, and the acoustical property of a specific surface can be investigated in detail without solving the sound wave equation in the whole space. Sound scattering by a finite impedance strip mounted on a rigid baffle in two different conditions is solved by the method. The result is used to illustrate the efficiency of the method.

4:20

2pAAb10. A composite sound absorber with micro-perforated panel and shunted loudspeaker. Jiancheng Tao, Qijing Jiao, Xiaojun Qiu, and Ning Han (Inst. of Acoust., Nanjing Univ., Rm. 307, Acoust. West Bldg., No.22 Hankou Rd., Nanjing, Jiangsu, China, Nanjing, Jiangsu 210093, China, jctao@nju.edu.cn)

Micro-perforated panel (MPP) backed by a rigid cavity is a widely used clean sound absorber; however, its application at low frequency is limited because a deep cavity is required to achieve good sound absorption in the low frequency range. In the present paper, a composite absorber composed by an MPP and a shunted loudspeaker is proposed. The loudspeaker is installed at the back wall of the air cavity and the acoustic impedance can be optimized by adjusting the parameters of the loudspeaker and the electronic components in the shunt to improve the sound absorption performance. The prediction model of such a finite-sized composite absorber is established based on the mode analysis solution and the equivalent circuit of the loudspeaker. Both numerical simulations and experiments show that the thickness of the proposed composite absorber can be much smaller than that of traditional MPP constructions.

4:40

2pAAb11. Active noise control in rooms. Jiri Tichy (Penn State Univ., 5552 N Citation Rd., Toledo, OH 43615, tichy@engr.psu.edu)

Maa published his ideas and achievements in the early days of the new branch of noise control based on using secondary sources to reduce the sound level of the primary sources by negative interference. He concentrated on active control of the reverberant sound in a room by a loudspeaker placed in one of the room corners and a microphone in its close vicinity. He examined the role of the complete solution of the wave equation consisting of a direct wave radiated from the sources and the reverberant field created by room modes. His work resulted in deriving a general formula for possible noise reduction that is independent of the shape, size, and the content of the room. He has shown noise reduction of 8 dB for a noise band centered at 100 Hz.

5:00

2pAAb12. A hybrid modal analysis for enclosed sound fields and its applications. Buye Xu (Signal Process. Res., Starkey Hearing Technol., 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com), Scott D. Sommerfeldt, and Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In 1939, Dr. Dah-You Maa presented a paper entitled "The distribution of eigentones in a rectangular chamber at lower frequency ranges" [J. Acoust. Soc. Am. **10**, 258 (1939)]. Since then, his interest in room acoustics has not diminished. More than six decades later, Maa proposed an idea of adding a monopole solution to the normal modal expansion to improve the accuracy in simulating the near-field sound field in rooms [D. Y. Maa, Acta Acust. (Beijing) **27**, 385–388 (2002)]. Following this idea, a hybrid model that combines the free field Green's function and a modal expansion has been proposed by the authors based on a rigorous mathematical derivation [Xu *et al.*, J. Acoust. Soc. Am. **128**, 2857–2867 (2010)]. The hybrid modal expansion can be further extended for complex sound sources by introducing the multipole expansion to the solution. In this talk, the theoretical derivation of the hybrid modal expansion will be reviewed, followed by examples demonstrating the use of the hybrid modal expansion in real-world applications.

Session 2pAB

Animal Bioacoustics, Psychological and Physiological Acoustics, Signal Processing in Acoustics, and Noise: Listening in the Natural Environment

Cynthia F. Moss, Cochair
Psychology, Univ. of Maryland, Biol.-Psych. Bldg., Rm. 2123M, College Park, MD 20742

Peter M. Narins, Cochair
Integrative Biol. & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095

Chair's Introduction—12:55

Invited Papers

1:00

2pAB1. Three directions in research on auditory scene analysis. Albert S. Bregman (Psychology, McGill Univ., 1205 Doctor Penfield Ave., Montreal, QC H3A 1B1, Canada, al.bregman@mcgill.ca)

Research on auditory scene analysis (ASA) began with some simple laboratory phenomena such as streaming and illusory continuity. Subsequently, research has gone in three directions, downwards toward underlying mechanisms (by neurophysiologists), upwards toward system organization (by computer scientists), and sideways toward other species (by neurobiologists). Each direction has its problems. The downward approach sometimes takes a phenomenon-oriented view of ASA, leading to simple explanations of a single ASA demonstration, such as streaming, with no obvious connection to any larger system. Research done by the upward approach usually takes the form of a computer program to achieve ASA in a working system, often ignoring known facts about human ASA, in favor of mathematically understood principles. The sideways approach often finds that non-human animals can respond to an important sound despite the presence of other interfering sounds. However, there is no reason to believe that a frog, a fish, and a human accomplish this by means of the same mechanisms. So finding out how some animal does this, while interesting in its own right, may shed little light on how humans do it. I will describe some properties of the human ASA system that should be borne in mind when manufacturing explanations.

1:20

2pAB2. Mechanisms of perceiving communication sounds in scenes. Sarah M. Woolley (Psychology, Columbia Univ., 406 Schermerhorn Hall, 1190 Amsterdam Ave., New York, NY 10027, sw2277@columbia.edu)

Vocal communicators must perceive the vocal signals of social partners in complex auditory scenes that include distracting background sounds. The auditory system must therefore parse auditory scenes into multiple information streams and/or accurately encode individual vocalizations despite the presence of competing sounds. Explaining mechanisms whereby neural representations of vocalizations are extracted from neural representations of scenes is an important part of understanding how auditory processing leads to perception of communication signals in complex scenes. We study how songbirds recognize individual vocalizations (songs) in scenes of conspecific choruses. We combine behavioral studies with neurophysiological studies of song and scene coding in midbrain and cortex. We find dramatic transformations in the neural coding of songs and scenes between different regions of auditory cortex. Neural representations of individual songs are dense and non-selective in the midbrain and primary cortex, but are sparse and highly selective in higher cortex. Sparse coding neurons produce background-invariant responses to individual songs in scenes, providing a potential neural mechanism for the perception of individual communication vocalizations in complex auditory scenes. Acoustic manipulations of song and pharmacological manipulations of neural coding suggest that sparse and background-invariant representations of songs in higher cortex are due to context-dependent inhibition.

1:40

2pAB3. The search for a neural basis of communication: Learning, memory, perception. and performance of vocal signals. Jonathan Prather (Dept. of Zoology and Physiol., Univ. of Wyoming, 1000 E Univ Ave. - Dept. 3166, Laramie, WY 82071, jprathe2@uwyo.edu)

Brain mechanisms for communication must establish a correspondence between sensory perception and motor performance of individual signals. A class of neurons in the swamp sparrow forebrain is well suited for that task. Recordings from awake and freely behaving birds reveal that those cells express categorical auditory responses to changes in note duration, a learned feature of their songs, and the neural response boundary accurately predicts the categorical perceptual boundary measured in field studies. Extremely precise auditory activity of those cells represents not only songs in the adult repertoire but also songs of others and tutor songs, including those imitated only very few times or perhaps not at all during development. Furthermore, recordings during singing reveal that these cells also express a temporally precise auditory-vocal correspondence, and limits on auditory responses to extremely challenging tutor songs may contribute to the emergence of a novel form of song syntax. Therefore, these forebrain neurons provide a mechanism through which sensory perception may influence motor performance to enable imitation. These cells constitute the projection from a premotor cortical-like area into the avian striatum (HVCX neurons), and data from humans implicate analogous or homologous areas in perception and performance of the sounds used in speech.

Contributed Paper

2:00

2pAB4. Call perception in mice. Erikson Neilans, David Holfoth, Kelly E. Radziwon (SUNY, Univ. at Buffalo, 207 Park Hall, Buffalo, NY 14260, eneilans@buffalo.edu), Christine V. Portfors (School of Biological Sci., Washington State Univ., Vancouver, WA), and Micheal L. Dent (SUNY, Univ. at Buffalo, Buffalo, NY)

Acoustic communication in laboratory mice is a relatively recent subject of experimental study, often yielding disparate findings. For example, researchers often manually place mouse ultrasonic vocalizations (USVs) into categories based on spectrotemporal characteristics, but the numbers and types of categories differ widely between laboratories. Here, we attempt to determine what cues CBA/CaJ mice use to discriminate between vocalizations by testing them in an operant conditioning paradigm. The mice were trained to discriminate a repeating background containing one USV from several target USVs. The targets were different call types used by Holmstrom *et al*. (2010) and manipulations of the background calls, such as removing the frequency modulation, shifting the entire call up or down in frequency, shortening or lengthening the call, or reversing the entire call. Results show that large frequency shifts were easy for the mice to discriminate, while reversing the calls and removing the frequency modulation were much more difficult. For most calls, similarity in spectrotemporal characteristics yielded poor discrimination performance. These results are the first to show that mice can discriminate between some vocalizations but not others, and that they may place different meaning to different call types, though not necessarily the call types designated by humans.

2:20–2:40 Break

Invited Papers

2:40

2pAB5. Auditory processing for contrast enhancement of salient communication vocalizations. Alex G. Dunlap, Frank G. Lin (Biomed. Eng., Georgia Inst. of Technol. and Emory Univ., Atlanta, GA), and Robert C. Liu (Biology, Emory Univ., 1510 Clifton Rd. NE, Rm. 2006, Atlanta, GA 30322, robert.liu@emory.edu)

In a natural acoustic environment, coherent representations of auditory objects and sources are streamed from the myriad sounds that enter our ears. Features of those sounds that are familiar and behaviorally salient to us are detected and discriminated into invariant precepts that inform us about our external world. Research into how this occurs is increasingly converging on the idea that there is a transformation from the auditory periphery wherein an initial acoustically faithful representation by neurons becomes progressively altered to enhance the population neural representation of perceptually relevant aspects of the sound. How this occurs may vary for sounds whose meanings are acquired in different ways, perhaps depending on what actions and decisions must be executed upon recognition. We have investigated this process in a natural social context in which mouse mothers "learn" about the meaning of pup ultrasound vocalizations through their maternal care. Here we discuss our recent studies in awake mice using electrophysiological, behavioral, immunohistochemical, and computational methods. Our results suggest that experience with natural vocalizations may alter core auditory cortical neural responses so that the contrast in activity across the neural population enhances the detection and discrimination of salient calls.

3:00

2pAB6. Influences of perceptual continuity on everyday listening. Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu), Golbarg Mehraei (Speech and Hearing Biosci. and Technol., Harvard/MIT, Cambridge, MA), Scott Bressler, and Salwa Masud (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

In the natural environment, listeners face the challenge of parsing the sound mixture reaching their ears into individual sources, and maintaining attention on a source of interest through time long enough to extract meaning. A number of studies have shown that continuity of certain acoustic features (including pitch, location, timbre, etc.) allows the brain to group sound from one acoustic sound source together through time to form an auditory object or stream. This presentation reviews results demonstrating that auditory feature continuity has important consequences on how listeners maintain attention on a stream through time. For instance, continuity of a sound feature that a listener knows is irrelevant to the task at hand nonetheless impacts the ability to maintain auditory attention based on some other sound feature. Moreover, the influence of auditory feature continuity decreases as the time between events in a given sound stream increases. Taken together, these behavioral results support the idea that auditory attention operates on auditory objects, rather than on individual sound features, and that feature continuity has an obligatory influence on the formation of auditory streams, and therefore on how selective auditory attention allows us to communicate in everyday settings.

3:20

2pAB7. Bugs and bats: Neural analysis of behaviorally relevant sounds in crickets. Gerald Pollack (Dept. of Biol., McGill Univ., 1205 Dr. Penfield Ave., Montreal, QC H3A1B1, Canada, gerald.pollack@mcgill.ca)

Hearing in crickets is specialized to serve particular behavioral functions, namely intraspecific communication and predator avoidance. Male crickets produce species-specific acoustic signals (songs) that attract distant females, promote copulation, and contribute to agonistic interactions with rivals. Crickets also hear the echolocation calls of aerially hunting bats, which evoke avoidance responses. These clear behavioral functions of hearing, combined with the relative simplicity of the cricket's nervous system, make it possible to address questions about how behaviorally relevant sensory signals are analyzed at the level of single, uniquely identifiable nerve cells. Cricket songs and bat calls differ both in rhythm and in spectrum, and neurons throughout the auditory processing chain are specialized for processing these two sorts of signal. I will focus on specializations that are evident at early stages of auditory processing, i.e., primary sensory neurons and the first-order interneurons with which they interact.

Contributed Paper

3:40

2pAB8. New observations and modeling of an unusual spatiotemporal pattern of fish chorusing off the southern California coast. Gerald L. D'Spain, Heidi H. Batchelor, and Tyler A. Helble (Marine Physical Lab., Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

The purpose of this paper is to present new results on an unusual spatio-temporal pattern of fish chorusing off the southern California coast. Characteristics of this fish chorus have been reported previously; it occurs at night in the late spring and summer months in shallow, sandy bottom regions just outside the surf zone. The background sound levels increase by up to 30 dB and cycle in level with a period of 30–35 s all night long. In this paper, recent results from measurements made by a set of high spatial resolution sensor systems spanning a 50-km stretch of coastline out to 20 km offshore over a 2-month time period are presented. These data allow the spatial dependence and long-term temporal variability of the chorus to be examined at high spatial resolution. Refinements to a numerical model that predicts this chorusing behavior are required to account for some aspects of these new observations. [Work supported by the Office of Naval Research, Code 322-MMB.]

Invited Papers

4:00

2pAB9. The influence of anthropogenic noise on the evolution of communication systems. Peter M. Narins (Integrative Biol. & Physiol., UCLA, 621 Charles E. Young Dr. S., Los Angeles, CA 90095, pnarins@ucla.edu)

Many species of animals, including man, face the formidable task of communicating in noisy environments. In this talk, I shall discuss the effects of anthropogenic (man-made) noise on the calling behavior of anuran amphibians. Moreover, the role of spectral, temporal, and spatial separation in minimizing masking by background noise will be examined. For example, presenting high-level, periodic (or aperiodic) tones at the males co-note frequency to males of the Puerto Rican treefrog, *Eleutherodactylus coqui* results in a clear shift in their calling pattern in an attempt to minimize acoustic overlap with the interfering playback stimulus. Amphibians also have a remarkable ability to shift their call timing in response to small intensity shifts in the background noise. Males of *E. coqui* are capable of reliably detecting a change in interfering tone intensity as small as 2–4 dB. Finally, I shall present behavioral evidence that anthropogenic noise may act as a strong selective force in sculpting the acoustic communication systems of several species of Old World frogs. Some techniques for visualizing sound interference will be discussed. [Supported by grants from the NIDCD (Grant No. DC-00222), and the UCLA Academic Senate (3501).]

4:20

2pAB10. Active listening in a complex environment. Melville Wohlgemuth and Cynthia F. Moss (Psych. and ISR, Univ. of Maryland, Biol.-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com)

Spatially guided behaviors in echolocating bats depend upon the dynamic interplay between auditory information processing and adaptive motor control. The bat produces ultrasonic signals and uses information contained in the returning echoes to determine the direction and distance of objects in space. With this acoustic information, the echolocating bat builds a 3-D auditory representation of the world, which it uses to guide a suite of coordinated motor behaviors, including head and pinna movements, as well as the timing, duration, frequency characteristics, and directionality of sonar signals. Adaptive echolocation behaviors shape the acoustic information available to the bat's sonar imaging system and provide a window to its perception of complex scenes. In a complex environment, an echolocating bat encounters multiple reflecting surfaces that return a cascade of echoes from each sonar transmission. The work presented here will focus on adaptive echolocation behaviors of the big brown bat as it tracks a selected prey item in the presence of multiple objects, both obstacles and other prey. Data suggest that bats can successfully segregate streams of echoes from closely spaced objects through finely tuned adaptive sonar signal control.

Contributed Papers

4:40

2pAB11. Your attention, please! Determining saliency of competing audio stimuli in natural scenarios. Francesco Tordini (Dept. of Elec. and Comput. Eng., McGill, 3480 University St., McConnell Eng. Bldg., Montreal, QC H3A 2A7, Canada, tord@cim.mcgill.ca), Albert S. Bregman (Dept. of Psych., McGill, Montreal, QC, Canada), Anupriya Ankolekar, Thomas Sandholm (Hewlett-Packard Lab., Palo Alto, CA), and Jeremy R. Cooperstock (Dept. of Elec. and Comput. Eng., McGill, Montreal, QC, Canada)

Perceptual saliency is a precursor to bottom-up attention modeling. While visual saliency models are approaching maturity, auditory models remain in their infancy. This is mainly due to the lack of robust methods to gather basic data, and oversimplifications such as an assumption of monaural signals. Here we present the rationale and initial results of a newly designed experimental paradigm, testing for auditory saliency of natural sounds in a binaural listening scenario. Our main goal is to explore the idea that the saliency of a sound depends on its relation to background sounds by using more than one sound at a time, presented against different backgrounds. An analysis of the relevant, emerging acoustical correlates together with other descriptors is performed. A review of current auditory saliency models and the deficiencies of conventional testing approaches are provided. These motivate the development of our experimental test bed and more formalized stimulus selection criteria to support more versatile and ecologically relevant saliency models. Applications for auditory scene analysis and sound synthesis are briefly discussed. Some initial conclusions are drawn about the definition of an expanded feature set to be used for auditory saliency modeling and prediction in the context of natural, everyday sounds.

5:00

2pAB12. A neuroethological analysis of the information in propagated communication calls. Frederic E. Theunissen, Solveig Mouterde (Psychology, UC Berkeley, UC Berkeley, 3210 Tolman Hall, Berkeley, CA 94720, theunissen@berkeley.edu), and Nicolas Mathevon (Univ. of Lyon/Saint-Etienne, Saint-Etienne, France)

The detection and recognition of communication signals in natural soundscapes is a difficult task that animals and birds in particular excel at. We have used a neuroethological approach to quantify the recognition

performance for propagated communication signals in the zebra finch, specifically regarding the information about individual identity. The propagated signals were analyzed using a regularized discriminant function analyses on a complete spectrographic representation of the signals. We found (1) a reduction in the informative frequency range a long distances yielding a frequency band sweet-spot, (2) that call duration and pitch are important parameters at short distances, and (3) that frequency modulation gains are important parameters at longer distances. Operant conditioning experiments showed that female songbirds were able to discriminate male calls at up to 128 m but not at 256 m. Finally, neurophysiological recordings showed a similar pattern in that high neural discrimination for calls was observed at 16 m and that this information degraded as a function of distance. We are currently analyzing the tuning properties of neurons that showed the most invariant responses to propagated sounds and hypothesized that these will be tuned to the parameters that we found were the most informative in the discriminant function analysis.

TUESDAY AFTERNOON, 4 JUNE 2013 519A, 1:00 P.M. TO 5:20 P.M.

Session 2pBAa

Biomedical Acoustics, Physical Acoustics, and Acoustical Oceanography: Bubbles Bubbles Everywhere II

Ronald A. Roy, Cochair
Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215

Thomas J. Matula, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., WA 98105-6698

Contributed Papers

1:00

2pBAa1. Spatially and temporally resolved single bubble sonoluminescence and its entrainment in Rayleigh-Taylor jets. Jonathan R. Sukovich, Phillip A. Anderson (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jonsukovich@gmail.com), Ashwinkumar Sampathkumar (Frederic L. Lizzi Center for Biomed. Engineering, Riverside Research, New York, NY), and R. Glynn Holt (Mech. Eng., Boston Univ., Boston, MA)

Previous investigations of the temporal and spatial evolution of single bubble sonoluminescence (SBSL) have shown events to last on the order of tens to hundreds of picoseconds with spatial extents of less than 1 μm. Here we present observations of the temporal and spatial evolution of laser-nucleated SBSL events in a high-pressure spherical resonator. Using high-speed imaging, we observe large, long-lived SBSL events reaching diameters of up to 50 μm and lasting on the order of 30 ns. Observations of events entrained in Rayleigh-Taylor jets resulting from instabilities in the final stages of the bubbles collapses will also be presented. We observe the light emitting region entrained in these jets to reach velocities well in excess of Mach 1 and to travel up to 100 μm before being extinguished. The size and duration of events, and the velocity of those entrained in Rayleigh-Taylor jets, will be compared to the maximum radius and collapse velocity of the bubbles responsible for generating them to develop a better understanding of the dynamics leading to, and the mechanisms responsible for light emissions during highly energetic collapse events. [Work supported by Impulse Devices, Inc.]

1:20

2pBAa2. High pressure phase transitions in the fluid region surrounding the collapse point of large single bubbles in water. Jonathan R. Sukovich, Phillip Anderson (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jonsukovich@gmail.com), Ashwinkumar Sampathkumar (Frederic L. Lizzi Ctr. for Biomed. Eng., Riverside Res., New York, NY), and R. Glynn Holt (Mech. Eng., Boston Univ., Boston, MA)

Observations from imaging experiments will be presented, which have shown persistent, long-lived spherical objects to form in the fluid region surrounding large, single bubbles in highly over-pressured water. Objects have been observed to form in a region of fluid where pressures are first predicted to exceed 0.8 GPa, and to extend radially inward to where fluid pressures are predicted to reach 6 GPa. These pressures bound those requisite for transitions in water to the crystalline phases of Ice-VI and Ice-VII, at 1.1 GPa and 2.1 GPa, respectively. The objects have been observed to behave in a fashion more consistent with a highly viscous fluid. They support and recover from large shape deformations, as well as support fluid flows within them. While water does have phases which are known to exhibit properties of highly viscous fluids, they have only been observed to form at or near cryogenic temperatures, typically via hyperquenching or quasi-static pressurization at low temperatures. Here, we present evidence for a high pressure liquid-liquid phase transition in water surrounding collapsing bubbles at room temperature. [Work supported by Impulse Devices, Inc.]

1:40

2pBAa3. Experimental characterization of light emission during shock-driven cavity collapse. Phillip Anderson, Nicholas Hawker, Matthew Betney (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, phillip.anderson@eng.ox.ac.uk), Brett Tully (Oxyntix Ltd., Oxford, United Kingdom), Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and Ronald A. Roy (Dept. of Mech. Eng., Boston Univ., Boston, MA)

The authors describe experimental work examining the collapse of a cavity by a strong shockwave. A millimeter size cavity is cast in Phytagel, which is then impacted by a metallic projectile accelerated by a compressed gas gun, reaching velocities up to 500 m/s. The impact generates a strong shockwave that propagates into the gel at greater than sonic velocity. Schlieren images are presented that illustrate both this process and the subsequent cavity collapse at a sub-microsecond timescale. As the shockwave reaches the cavity, it is shown to cause a rapid asymmetric collapse process characterized by the formation of a high-speed transverse jet. The pressure of the shockwave is found to be 100+ MPa as measured via a custom-built fiber-optic probe hydrophone. Previous work examining shock-driven cavity collapse observed luminescence, postulated to be due to the high-speed impact of the transverse jet on the far bubble wall; this experimental observation is replicated. Further, the light emission is characterized as a function of impact velocity and thus of shockwave pressure. This reveals that shock-driven cavity collapse shares many of the unique features that make the more widely studied SBSL-type collapse interesting.

2:00

2pBAa4. Simulation of warm dense matter in intense bubble collapse. Brett Tully (Oxyntix Ltd., Dept. of Eng. Sci., Parks Rd., Oxford OX1 3PJ, United Kingdom, brett.tully@oxyntix.com), Nicholas Hawker, Matthew Betney, and Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Previous work by the authors includes computational study of shock-bubble interaction. This work demonstrated strong compression and heating within the bubble, with gas reaching densities of order of magnitude 1 g/cc and temperatures of 10 eV. These conditions correspond to the warm dense matter regime. This paper addresses limitations of previous work through utilization of various equations of state (EOS) appropriate for the modeling of dense plasma. This is achieved through the design and implementation of a generic interface based on tabulated EOS data. Any EOS may be utilized through the framework, requiring only knowledge of pressure and energy as functions of density and temperature. The solutions to various issues such as table interpolation, tabulated change of variables, arbitrary calculation of entropy, and calculation of thermodynamic derivatives are presented. In addition, the trade-offs between CPU time, memory requirement and computational accuracy are discussed. Validation work is presented and a comparison of different EOS is also explored. The EOS used include but are not limited to the EOS for air utilized by Moss *et al.* (1994) to study SBSL, SESAME tabulated EOS and QEOS-type formulations. Finally, conditions attained during shock-bubble interaction are re-examined.

2:20

2pBAa5. Percussoluminescence. Nicholas Hawker and Yiannis Ventikos (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, nicholas.hawker@eng.ox.ac.uk)

The phenomenon of light emission from a bubble driven to collapse via ultrasound has a long history of study. Light emission is widely believed to occur due to the formation of plasma within the bubble during the final moment of collapse and one important focus of the literature has been on understanding the exact thermodynamic conditions created. There are, however, other phenomena that demonstrate a similar light emission, including but not limited to other types of bubble collapse. This paper examines light emission in seeming disparate phenomena and discovers that a common thread exists. The terminology and fundamental theory appropriate for sonoluminescence is found to be decreasingly relevant and a new term, percussoluminescence (PCL), is established. This is substantiated through the presentation of a 1D theory, detailed numerical simulation, and illustrative experimental results. The implications of this new perspective are explored.

2:40–3:00 Break

3:00

2pBAa6. Investigating the acoustic response of gold nanoparticle coated microbubbles. Mehrdad Azmin (UCL Mech. Eng., Univ. College London, London, United Kingdom, m.azmin@ucl.ac.uk), Paul Rademeyer, Graciela Mohamedi (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Mohan Edirisinghe (Mech. Eng., Univ. College London, London, United Kingdom), Luis Liz-Marzan (Biofunctional Nanomaterials, CIC biomaGUNE, San Sebastian, Spain), and Eleanor Stride (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Recent work has shown that incorporating solid nanoparticles into the coatings of contrast agent microbubbles can be used to control their stability and to provide other functional characteristics for example in multimodality imaging. The aim of this study was to investigate the influence of nanoparticle characteristics and concentration on the response of the microbubbles to ultrasound excitation. Theoretical models were first derived to simulate the effects of adsorbing different types and concentrations of nanoparticle on to the surface of a bubble in both a monolayer and a layer of finite thickness. The results indicate that the particles modify the symmetry of the microbubble oscillations and enhance their nonlinear character. Experimentally, microbubbles coated with a surfactant and varying concentrations of gold nanoparticles of different sizes and surface properties were produced using either sonication or microfluidics. The attenuation and backscattering coefficients from the bubble suspensions and the scattered response from individual bubbles were measured for a range of frequencies (1–7.5 MHz) and pressures (50–500 kPa). The nanoparticles were found to enhance the nonlinear character of the bubble response in agreement with the theoretical results. Both the degree of enhancement and stability of the microbubbles was dependent upon the nanoparticle surface chemistry.

3:20

2pBAa7. Investigating the sensitivity of microbubble acoustic response for biosensing applications. Caroline J. Harfield (Inst. of Biomed. Eng., Dept. of Eng. Sci., Oxford Univ., Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, caroline.harfield@stcatz.ox.ac.uk), Gianluca Memoli (Acoust. Group, National Physical Lab., London, United Kingdom), Philip Jones (Dept. of Phys. and Astronomy, Univ. College London, London, United Kingdom), Nick Ovenden (Dept. of Mathematics, Univ. College London, London, United Kingdom), and Eleanor Stride (Inst. of Biomed. Eng., Dept. of Eng. Sci., Oxford Univ., Oxford, United Kingdom)

Microbubbles are currently used as contrast agents for diagnostic imaging on account of their high scattering efficiency and non-linear response to ultrasound. The exact nature of this response depends not only upon the bubble size and imposed sound field but also the bubble environment: physical properties of the surrounding liquid, bubble surface coating, ambient temperature, pressure, and proximity to other bubbles or surfaces. This dependence can potentially be exploited for the microscale interrogation of a liquid to detect, e.g., changes in viscosity or the presence of particular chemical species. To facilitate this, the sensitivity of the microbubble acoustic response to changes in its environment must be analyzed. The aim of this study was to provide a theoretical framework for this. A modified Rayleigh-Plesset equation was derived to describe the radial bubble motion, including the effects of gas diffusion and adsorption/desorption of a surfactant coating, and coupled to an equation describing microbubble translation. The presence of a rigid boundary was also included in the simulations. A sensitivity analysis was performed for the effect of each of the physical variables upon the bubble response, which indicated high sensitivity to species altering the dynamic surface tension and proximity to a boundary.

3:40

2pBAa8. Modeling of microbubbles pushed through clots via acoustic radiation force. Ascanio Guarini and E. C. Everbach (Engineering, Swarthmore College, 500 College Ave., Swarthmore, PA 19081, ceverba1@swarthmore.edu)

Previous studies have shown that thrombi, which may completely block the blood flow in a vessel, can be dissolved by ultrasound acting on echo-contrast agent microbubbles. The presumed mechanism is acoustic cavitation, the radial oscillations of the bubbles, which can exert locally large forces on the fibrin ropes that make up the clot matrix. However, the movement of the bubbles through the clot in the absence of flow suggests that acoustic radiation force also plays an important role. Because detailed mechanistic modeling of this process is not available, we present here a heuristic study in which microbubble transit times in gels of various porosities were measured and described by a simplified percolation theory. Results suggest considerations for optimizing the penetration of active microbubbles in sonothrombolysis.

4:00

2pBAa9. Use of dolphin-like pulses to enhance target discrimination and reduce clutter. Tim Leighton, Gim-Hwa Chua, Paul R. White (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk), Kenneth Tong, Hugh Griffiths (Dept. of Electron. and Elec. Eng., Univ. College London, London, United Kingdom), and David Daniels (Cobham Tech. Services, Leatherhead, United Kingdom)

Building on the earlier success of using twin inverted pulses to supress bubble scatter and so reduce clutter when detecting targets in bubbly water (proven in ship wakes in field trials), the same nonlinear processing scheme is generalized to make use of dolphin-like pulses. Their performance in reducing clutter and enhancing target discrimination is demonstrated in the laboratory, and the opportunities for using the same scheme to improve the detection of hidden electronic devices or semiconductor devices by radar are discussed.

4:20

2pBAa10. Acoustic and optical observations of methane gas seeps in the Gulf of Mexico. Thomas C. Weber, Yuri Rzhanov, Kevin Jerram, Larry Mayer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, weber@ccom.unh.edu), and Dave Lovalvo (Eastern Oceanics/NOAA Office of Ocean Exploration, West Redding, New Hampshire)

In 2011 and 2012, measurements of acoustic backscatter from natural methane seeps were made in the northern Gulf of Mexico in water depths between 1000 and 2000 m. The measurements were made using a calibrated 18 kHz echo sounder with an 11 degree beamwidth in order to estimate the depth-dependent target strength (TS). The TS data indicate a wide variation in the rate of gas seepage from the seafloor. Several of these seeps were revisited with a remotely operated vehicle in order to optically assess the bubble size distribution and to estimate the rate at which gas bubbles were exiting the seafloor. The optical data show bubble sizes between 1 and 10 mm radius, and similar rates of gas seepage ranging from a few bubbles per second to several tens of bubbles per second. Together, these data help to suggest the requirements for acoustically estimating gas flux from the seafloor over large regions.

4:40

2pBAa11. Effect of shell thickness on sound propagation through encapsulated bubbles: A resonator approach. Craig N. Dolder and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu)

The acoustics of elastic-shelled bubbles are of interest in applications ranging from biomedical imaging, to fisheries acoustics, to underwater noise abatement. Multiple models exist to predict the velocity and attenuation of sound propagating through water containing encapsulated bubbles, but existing measurements have yet been able to confidently verify model accuracy for void fractions above about 10^{-3}. The effect of shell thickness was studied in this work using tethered, rubber-encapsulated bubbles bearing shells of varying thickness, deployed within a one-dimensional resonator apparatus. In previous work, resonator modes below the individual bubble resonance frequency (IBRF) were exploited to extract inferences of sound speed, but this is a regime where there is little difference between competing model prediction. In the present work, an increased understanding of the modal field inside the resonator has extended this technique to just below IBRF and to well above IBRF, both regimes where model behavior diversifies, thus providing a new opportunity for model verification. Measurement-model comparisons will be shown for encapsulated bubbles with radii ranging from 13 mm to 30 mm, void fractions ranging from 8.4×10^{-4} to 1.1×10^{-2}, and shell thicknesses ranging from 0.085 to 0.16 mm. [Work supported by the Office of Naval Research.]

5:00

2pBAa12. Attenuation of sound in water through collections of very large bubbles with elastic shells. Kevin M. Lee (Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu) and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX)

The ultimate goal of this work is to accurately predict the attenuation through a collection of large (on the order of 10 cm radius) tethered encapsulated bubbles used in an underwater noise abatement system. Measurements of underwater sound attenuation were performed during a set of lake experiments, where a low-frequency compact electromechanical sound source was surrounded by different arrays of encapsulated bubbles with various individual bubbles sizes and void fractions. The measurements are compared with an existing predictive model [J. Acoust. Soc. Am. **97**, 1510–1521 (1995)] of the dispersion relation for linear propagation in liquid containing encapsulated bubbles. Although the model was originally intended to describe ultrasound contrast agents, it is evaluated here for large bubbles, and hence low frequencies, as a design tool for the underwater noise abatement system, and there is fairly good quantitative agreement between the data and the model. Refinements to the model to incorporate multiple scattering effects, which may be important at high void fractions, via an effective medium approach [J. Acoust. Soc. Am. **111**, 168–173 (2002)] and comparison with the data will also be discussed. [Work supported by Shell Global Solutions.]

TUESDAY AFTERNOON, 4 JUNE 2013 518C, 1:20 P.M. TO 3:20 P.M.

Session 2pBAb

Biomedical Acoustics: Imaging and Characterization in Elastic Media

Guy Cloutier, Chair

Lab. of Biorheology and Med. Ultrasonics, Univ. of Montreal Hospital, 2099 Alexandre de Sève, Montreal, QC H2L 2W5, Canada

Contributed Papers

1:20

2pBAb1. Dynamic quantitative ultrasound imaging of mimicked breast lesions during shear wave propagation to emphasize differences in tissue statistical backscatter properties. Marzieh Alavi (Lab. of Biorheology and Med. Ultrasonics, Univ. of Montreal Hospital, Univ. of Montreal Hospital Res. Ctr., Montreal, QC, Canada, marziehalavi@gmail.com), Francois Destrempes, Emmanuel Montagnon, and Guy Cloutier (Lab. of Biorheology and Med. Ultrasonics, Univ. of Montreal Hospital, Montreal, QC, Canada)

The main motivation in this study was to increase the accuracy of the breast tissue characterization by combining quantitative ultrasound (QUS) with ultrasound (US) dynamic elastography. To demonstrate that, an agar-gelatin breast mimicking phantom with two inclusions containing the same density of agar (US scatterers) but different proportions of gelatin corresponding to different mechanical properties was made. Transient plane shear waves (SW) at 200 Hz were transmitted through the phantom while the displacement of scatterers was imaged at 5 MHz with an ultrafast imaging technique. With segmented inclusions, the reciprocal (beta parameter) of the effective density of scatterers of a general distribution model of the echo envelope and its normalized range (normalized by the mean of beta during SW propagation) were estimated for each inclusion. The results showed that the relative difference of beta between the surrounding medium and both inclusions A and B were 55.6% (A) and 0.9% (B), respectively, whereas differences (in %) of the beta normalized range were 46.2% (A) and 52.6% (B), respectively. The static value beta failed to distinguish inclusion B from the surrounding; however, the dynamic range of beta succeeded in that task for the two inclusions. Thus, dynamic QUS might add information to QUS in a static framework.

1:40

2pBAb2. Shear wave elastography for characterizing muscle tissue in myofascial pain syndrome. Diego Turo, Paul Otto (Dept. of Bioengineering, George Mason Univ., 4400 University Dr., Fairfax, VA 22030, diegoturo@gmail.com), Tadesse Gebreab (National Inst. of Health, Bethesda, MD), Katherine Armstrong, Lynn H. Gerber (Dept. of Rehabilitation Sci., George Mason Univ., Fairfax, VA), and Siddhartha Sikdar (Dept. of Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Myofascial pain syndrome (MPS) affects 85% of chronic pain sufferers in a specialty pain center. Neck and low-back are commonly affected by MPS. Myofascial trigger points (MTrPs) are characteristic findings of MPS and are palpable tender nodules in the muscles of symptomatic subjects. Mechanical characterization of MTrPs and surrounding tissue can offer important insight about the pathophysiology of the MPS, which is currently poorly understood. In this study, we propose an inexpensive technique, based on ultrasound shear wave elastography, to objectively measure mechanical properties of MTrPs and surrounding tissue in the upper trapezius. In an ongoing clinical study, we recruited 34 subjects: 12 healthy controls, 10 with not spontaneously painful MTrPs (latent) and 12 with symptomatic chronic neck pain (>3 months) and active (spontaneously painful) MTrPs. Shear wave elastography was performed on the upper trapezius of all subjects using the Ultrasonix RP system and an external vibrator. Voigt' s model was used to estimate shear modulus G and viscosity μ of the interrogated tissue. Preliminary analysis demonstrates that symptomatic muscle tissue in subjects with neck pain is stiffer ($G = 8.40 \pm 8.31$ kPa, mean $\pm$ standard deviation) compared to muscle in control subjects ($G = 2.86 \pm 2.48$ kPa) ($p < 0.05$), and that active MTrPs are more viscous ($\mu = 17.10 \pm 9.46$ Pa*s) than surrounding tissue ($\mu = 10.59 \pm 5.96$ Pa*s). Latent MTrPs ($\mu = 24.31 \pm 12.72$ Pa*s) and surrounding tissue ($\mu = 20.09 \pm 7.48$ Pa*s) are more viscous than normal tissue ($\mu = 10.96 \pm 4.17$ Pa*s).

2:00

2pBAb3. Electromagnetic-acoustic imaging of stiffness and dielectric properties in gels. Ning Zhang, Robin Cleveland, and David J. Edwards (Eng. Sci., Univ. of Oxford, Inst. Biomed. Eng., Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, robin.cleveland@eng.ox.ac.uk)

Here we present a novel multi-modal imaging method, electromagnetic-acoustics (EMA) that combines acoustic radiation force and electromagnetic scattering. Experiments were carried out in gels using a focused megahertz ultrasound source, amplitude modulated at 160 Hz, to create an oscillating radiation force resulting in vibration of the gel in the focal region. At the same time an EM signal was transmitted into the gel and the backscattered EM signal recorded. The tissue that was in motion resulted in a Doppler shift of the EM signal which manifested itself as frequency modulation of the EM wave. The modulated component was detected by means of a demodulator and lock-in amplifier and the amplitude of the modulated signal we call the EMA signal. By steering the ultrasound beam through the sample, an EMA image can be created which has spatial resolution of the ultrasound but is sensitive to shear and dielectric properties. We show that the EMA signal is sensitive to changes in elasticity and conductivity in a gel. EMA may have utility in biomedical imaging by detecting diseases, which have contrast in dielectric properties without the cost and complexity of an magnetic resonance imaging system.

2:20

2pBAb4. Speckle tracking in multiple-scattered pressure field. Ayse Kalkan-Savoy (Biomed. Eng. and Technol., UMass-Lowell, 1 University Ave., Lowell, MA 01854, ayse.k.savoy@gmail.com) and Charles Thompson (Elec. and Comput. Eng., UMass-Lowell, Lowell, MA)

Speckle tracking using ultrasound B-scan image sequences to quantify myocardial strain has the potential to become a standard method in echocardiography. A pressure field for a modulated Gaussian pulse traveling in inhomogeneous media is constructed. This model is based on computation of pressure field using Kirchoff integral formulation in frequency domain. Each pressure term of Neumann Series is converted into time domain and used in computation of Pade approximants. Pressure field in time and space is constructed utilizing Pade approximants and includes multiple-scattering effects. The scatterers of interest are large enough in size and contrast to contribute to strong and multiple-scattering. The signals at transducer point for each time frame are analyzed and speckles are tracked utilizing an optical flow algorithm. A velocity field which can be utilized to quantify strain is computed and visualized. The efficiency of this method will be discussed.

2:40

2pBAb5. Validation of three-dimensional strain tracking by volumetric ultrasound image correlation in a pubovisceral muscle model. Anna S. Nagle, Ashok R. Nageswaren, Balakrishna Haridas, and T. Douglas Mast (SEEBME, Univ. of Cincinnati, 3940 CVC, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, nagleas@mail.uc.edu)

Little is understood about the biomechanical changes leading to pelvic floor disorders such as stress urinary incontinence. In order to measure regional biomechanical properties of the pelvic floor muscles *in vivo*, a 3D strain tracking technique employing correlation of volumetric ultrasound images has been implemented. In this technique, local 3D displacements are determined as a function of applied stress and then converted to strain maps. To validate this approach, an *in vitro* model of the pubovisceral muscle, with a hemispherical indenter emulating the downward stress caused by intra-abdominal pressure, was constructed. Volumetric B-scan images were recorded as a function of indenter displacement while muscle strain was measured independently by a sonomicrometry system (Sonometrics). Local strains were computed by ultrasound image correlation and compared with sonomicrometry-measured strains to assess strain tracking accuracy. Image correlation by maximizing an exponential likelihood function was found more reliable than the Pearson correlation coefficient. Strain accuracy was dependent on sizes of the subvolumes used for image correlation, relative to characteristic speckle length scales of the images. Decorrelation of echo signals was mapped as a function of indenter displacement and local tissue orientation. Strain measurement accuracy was weakly related to local echo decorrelation.

3:00

2pBAb6. Measurement of surface acoustic wave in soft material using swept-source optical coherence tomography. Yukako Kato, Yuji Wada, Yosuke Mizuno, and Nakamura Kentaro (Tokyo Inst. of Technol., 4259-R2-26 Nagatsudacho Midoriku, Yokohama 226-8503, Japan, ykato@sonic.pi.titech.ac.jp)

In endoscopic elastography, it is needed to observe small area with high spatial resolution. Optical coherence tomography (OCT) is one of the candidate imaging methods, which has the depth resolution of several 10 μm. In this study, we try to find the propagation velocity of surface acoustic wave (SAW) using a swept-source OCT (SS-OCT). The depth scanning rate in the SS-OCT is rather fast, which is determined by the wavelength sweep of the light source as fast as 20–100 kHz. However, on the other hand, the lateral scanning is limited up to 100 times per second, since it is performed using a mechanical moving mirror. We develop a theory to estimate SAW velocity of tissues ranging from 1 to 20 m/s from data taken by the slow lateral scanning of less than 1 m/s using the OCT. The present method is tested for agar samples with different concentrations and also for several tissue samples. Vibrations are excited on the sample surface using a small stick connected to a loudspeaker. The measurements are carried out at many frequencies from 500 to 1000 Hz. The dependence of the SAW velocity on the concentration successfully agreed the previous results.

Session 2pEAa

Engineering Acoustics: Computational Methods in Transducer Design, Modeling, Simulation, and Optimization I

Daniel M. Warren, Chair
Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60134

Chair's Introduction—12:55

Invited Paper

1:00

2pEAa1. Virtual prototyping of condenser microphones using the finite element method for detailed electric, mechanic, and acoustic characterization. Mads Jakob Herring Jensen (COMSOL A/S, Diplomvej 373, Lyngby 2800, Denmark, mads@comsol.dk) and Erling Sandermann Olsen (Brüel & Kjaer Sound and Vib. Measurement A/S, Naerum, Denmark)

Recent development and advances within numerical techniques and computers now enable the modeling, design, and optimization of many transducers using virtual prototypes. Here, we present such a virtual prototype of a Brüel & Kjær Type 4134 condenser microphone. The virtual prototype is implemented as a model using the finite element method with COMSOL Multiphysics and includes description of the electric, mechanic, and acoustic properties of the transducer. The acoustic description includes thermal and viscous losses explicitly solving the linearized continuity, Navier-Stokes, and energy equations. The mechanics of the diaphragm are modeled assuming a pre-stressed membrane, electrostatic attraction forces, and acoustic loads. The model includes electric description of the active and passive capacitances of the microphone cartridge as well as an external circuit model representing the preamplifier. Different modes of the system are studied, including the important first rocking mode of the membrane. The model has no free fitting parameters and results in the prediction of the frequency dependent sensitivity, capacitance, and mechanical impedance. The model results show good agreement with measured data.

Contributed Papers

1:20

2pEAa2. New planar nano-gauge detection microphone: Analytical and numerical acoustic modeling. Cécile Guianvarc'h, Thierry Verdot (LVA-Insa de Lyon, Villeurbanne, France), Jaroslaw Czarny (CEA-Leti, Grenoble, France), Emmanuel Redon, Kerem Ege, Jean-Louis Guyader (LVA-Insa de Lyon, 25 Bis, avenue Jean Capelle, Villeurbanne 69621, France, emmanuel.redon@insa-lyon.fr), Arnaud Walther, and Philippe Robert (CEA-Leti, Grenoble, France)

The miniaturization of microphones is of great interest for several fields, such as medical applications (audio implants), or consumer electronics (cell phones). Almost all existing miniature microphones rely on electrostatic transduction and offer good performances (sensitivity, frequency bandwidth). However, their sensitivity, proportional to the membranes area, would be dramatically reduced in case of extreme miniaturization. A new concept of microphones developed by CEA-LETI, which uses membranes moving in the plane of the substrate and inducing strain on piezoresistive Si nano-gauges (M&NEMS technology), seems promising for its miniaturization potential without significant decrease of sensitivity. The design and optimization of such planar piezo-resistive microphone require a deep understanding of its acoustic and vibroacoustic behavior. Regarding the small dimensions of the slits (1–100μm) and the sharp discontinuities in the microphones structure, viscous and thermal effects in the boundary layers and turbulent perturbations are of great importance, and must then be taken into account with high accuracy in device modeling. The aim of the present work is to provide accurate analytical and numerical (FEM) models able to gather all these effects in a consistent manner, and to suggest an experimental method to check their validity.

1:40

2pEAa3. Noise minimization in micromachined piezoelectric microphones. Robert Littrell (Baker-Calling, Inc., 1810 14th St., Ste. 210, Santa Monica, CA 90404, rlittrell@bakercalling.com) and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Piezoelectric MEMS microphones have been built for more than 30 years and they offer some advantages over other technologies such as improved linearity, simple construction, no need for a bias voltage or charge, and the ability to withstand high temperatures. Despite these advantages, a relatively high noise floor has always limited their utilization. Traditionally, the noise of these sensors has been minimized by viewing the preamplifier or amplifier as a black box with fixed gain and noise. This leads the designer to minimize noise by maximizing microphone sensitivity. By viewing the microphone-amplifier system together, we develop a different method of optimization, leading to lower noise. Further, by including the back cavity compliance of the package in the optimization, we can determine absolute limits on the minimum achievable noise floor with very few assumptions. To date, we have built piezoelectric MEMS microphones utilizing aluminum nitride with a 32 dBA noise floor. We can compute a minimum achievable noise floor of 24 dBA for the same sensing structure with a 2 mm^3 back cavity volume.

2:00

2pEAa4. Directional robustness of an *in-situ*, dual dipole omni microphone array for hearing instruments. Thomas Burns (Starkey Hearing Technol., 6600 Washington Ave. S, Eden Prairie, MN 55344, tburns@starkey.com)

The three critical factors for providing stable directional performance for typical microphone arrays used in hearing aids include the relative sensitivity and phase between the microphones in addition to the placement of the hearing instrument behind the user's ear. A directional system is robust if these factors can operate over a wide range of levels without degrading the directional performance. In this study, dual dipole microphones were arranged symmetrically around an omnidirectional microphone such that all inlets were collinear. Compared to an endfire array, whether it be a delay-and-sum or Blumlein configuration, this dual-dipole-omni array is remarkably more robust, yielding very little degradation in the Directivity Index for the aforementioned critical factors varying as much as +/− 3 dB, +/− 30 ms, and the directional axis of the hearing instrument varying +/− 20 degrees on the ear.

2:20

2pEAa5. The efficiency of receivers used in hearing aid devices. Michael Salameh (Research, Starkey Hearing Technol., 6600 Washington Ave. S., Eden Prairie, MN 55344, michael_salameh@starkey.com)

Current drain is one of the most important considerations in the hearing aid design. It is determined mainly by the receiver impedance. Recently, many additional features and technologies such as wireless are used in the hearing aid devices. These capabilities increase the current drain further. Efficiency evaluation becomes an important step in the receiver selection for hearing aid devices. Balanced armature receivers are widely used in hearing aids. In comparison with the moving-coil loudspeakers, these receivers are designed for better efficiency in closed acoustical loads found in different hearing aid styles. However, the receiver efficiency for hearing aid applications is not well defined, evaluated, or reported. In this paper, the receiver efficiency including the effect of the receiver size, impedance, and acoustical load is discussed. The evaluation of the receiver efficiency is revisited and a new approach is suggested. The simulation and the measurement results are presented.

2:40

2pEAa6. Cantilever mode piston transducer array. John Butler and Alexander L. Butler (Image Acoust., Inc., 97 Elm St., Cohasset, MA 02025, jbutler@imageacoustics.com)

The underwater sound cantilever mode transducer uses a piston, tail mass, dual piezoelectric cantilever benders, and mechanical arms for translating lateral bending cantilever motion to piston motion in a direction normal to the cantilever motion. This compact design provides a wide band response and exceptionally low resonance, especially under array loaded conditions. We present the operation of the cantilever mode transducer along with its unique response under array loading conditions where the resonance frequency decreases as the array gets larger. These results are demonstrated through a number of finite element array models and measured results on a 2x2 array. The single crystal PIN-PMN-PT results show a significant source level improvement over ceramic PZT-8 active material when driven up to 5 V/mil.

3:00

2pEAa7. Applications of network synthesis and zero-pole analysis in transducer modeling. Daniel Warren (R&D, Knowles Electron., 1151 Maplewood Dr., Itasca, IL 60143, daniel.warren@knowles.com)

When a transducer is an immutable component of a larger system being simulated, it is sufficient that the transducer model correctly reproduce behavior only at the available ports of the transducer. The behavior of two-port electroacoustic transducer should be completely characterized by three transfer functions related to the electrical and acoustic termination impedance and the transfer impedance. To the extent that the transducer could be represented as a analog circuit of passive linear elements, the same circuit could be exactly represented by rational polynomials of the Laplace *s*, embodied as a set of poles and zeros of the transfer functions. This invites the reverse process of identifying poles and zeros by nonlinear curve fit of rational polynomials to measured transfer data, perhaps even synthesizing a circuit directly from the identified poles and zeros. Measured transducer transfer data have been fit demonstrating both the utility and the pitfalls of this method. Curve fit transfer functions can be a compact and faithful representation of complex data over frequency, but have no predictive value outside the given data. Judicious selection of the number of poles and zeros, initial values, proper constraints, and some physical insight are necessary for stable curve fits. Further investigation into the relationship among the transfer functions of a physical system may lead to a meaningful model derived from curve fits.

3:20

2pEAa8. Improved estimation of direction of arrival of sound sources for hearing aids using gyroscopic information. Alan W. Boyd (Dept. of Electron. and Elec. Eng., Univ. of Strathclyde, 204 George St., Glasgow G1 1XW, United Kingdom, alan.boyd@strath.ac.uk), William M. Whitmer, W. Owen Brimijoin, and Michael A. Akeroyd (Inst. of Hearing Res., Med. Res. Council, Glasgow, United Kingdom)

Determining the direction of arrival (DOA) of a sound source is important in spatial audio signal processing, as it can lead to substantial improvement in noise reduction performance. Techniques such as generalized cross correlation with phase transform (GCC-PHAT) and adaptive eigenvalue decomposition (AED) perform optimally when the measurement microphones are fixed in place. However, hearing-aid microphones move with the listener's head movements, which can result in momentarily inaccurate directional estimates and noise artifacts in the output signal. Techniques such as GCC-PHAT experience degraded short-term performance in the presence of multiple signals and noise. The system presented measures instantaneous head movement velocity using a micro-electromechanical systems (MEMS) gyroscope attached to binaurally communicating hearing aids. Estimates of DOA for physically stationary sources are shifted based on the gyroscope's head-movement information. Using GCC-PHAT with gyroscopic input can produce robust *in situ* DOA estimates for several sources in reverberant environments. In addition, the gyroscope allows an adaptive beamformer to be steered to a target direction, compensating for head movements on a very short timescale during DOA estimation. Results show improved localization performance over a standard GCC-PHAT system during head movements.

3:40

2pEAa9. Miniaturized electrostatic receiver with small-sized backing electrode. Alexey Podkovskiy, Petr Honzík, Stéphane Durand, Nicolas Joly, and Michel Bruneau (LUNAM Université, LAUM (Laboratoire d'acoustique de l'université du Maine), UMR CNRS 6613, Avenue Olivier Messiaen, Le Mans F-72085, France, petr.honzik@gmail.com)

A miniaturized electrostatic receiver design, having a central cylindrical backing electrode of small radius surrounded by a flat annular cavity behind the circular membrane, can lead to both a higher sensitivity and a larger frequency bandwidth compared to the ones achieved with other designs, while bringing a geometrical simplicity which is advantageous from the point of view of microfabrication. An appropriate computational method, relying on a specific 2-D axisymmetrical simulation using an adaptive mesh and accounting for both viscous and thermal boundary layer effects, provides results against which analytical results can be tested. An analytical approach, which leads to solutions based on the eigenmode expansion of the membrane displacement, the acoustic pressure field depending on the radial coordinate in the central fluid gap but being assumed quasi-uniform in the annular cavity, is much faster in terms of running time and appears to be sufficiently accurate to achieve final optimization of this kind of devices.

4:00

2pEAa10. How a hearing aid transducer works. Noori Kim and Jont B. Allen (UIUC, 1085 Baytowner df 11, Champaign, IL 61822, nkim13@illinois.edu)

The oldest magnetic earphone, the balanced armature receiver (BAR), is the most widely used receivers in modern hearing-aid instruments, where the efficiency of the power (battery life) and the size of the device, as well as the larger frequency bandwidth, are critical parameters. Since these miniature loudspeakers remain one of the most expensive components of the hearing-aids, a detailed studying of them is a cornerstone of understanding the hearing-aid system, and we believe that the appropriate and rigorous analysis of this transducer is critical. The motivation of this study started from the modeling of a widely used commercial hearing-aid receiver ED series, manufactured by Knowles Electronics, Inc. Our proposed model includes a semi-inductor and a gyrator along with the two-port network glue which enables us an intuitive design of the electromagnetic transducer. Based on the BAR model, we will investigate and discuss the roles of each physical component in the BAR such as a coil, magnets, an armature, a diaphragm, and the rear volume of the receiver. Ultimately, this work will deliver a fundamental and innate answer for the question, "How does a hearing-aid transducer work?"

4:20

2pEAa11. Numerical study of the cross-talk effects in acoustical transducer arrays and correction. Abdelmajid Bybi, Jamal Assaad (OAE Dept., CNRS UMR 8520, IEMN, Univ. of Valenciennes and Hainaut Cambrésis, Valenciennes, France), Anne-Christine Hladky-Hennion (ISEN Dept., UMR CNRS 8520, IEMN, Lille, France), Farouk Benmeddour, Sébastien Grondel, and Frederic Rivart (OAE Dept., CNRS UMR 8520, IEMN, Univ. of Valenciennes and Hainaut Cambrésis, Campus Mont Houy, Valenciennes 59313, France, farouk.benmeddour@univ-valenciennes.fr)

Cross-talk in acoustical transducer arrays is an undesirable phenomenon which decreases seriously the performances of these sensors. Indeed, when one element of the array is driven, it generates parasitic displacement fields at the radiating surfaces of the neighboring elements, which changes the directivity of the antenna. To well understand this phenomenon a transducer array similar to those used in medical imaging and NDT applications is modeled by finite element method. The research work, investigated systematically the effects of the cross-talk. First, it inspected the acoustical and mechanical cross-talk throughout the propagating medium and the filling material. Second, it studied the influence of the matching layer and the backing on the acoustical performances of the transducer. It is shown that the filling material and the matching layer are the major factor contributing to this phenomenon. In order to cancel the cross-talk a correction method previously developed by the author has been used. This solution consisted in applying adapted electrical voltages on each neighboring element of the active one in the purpose to reduce the displacement field on their active surface. This method was tested numerically and the obtained results clearly demonstrated its ability to reduce the cross-talk.

4:40

2pEAa12. Development and performance evaluation of virtual auditory display system to synthesize sound from multiple sound sources using graphics processing unit. Kanji Watanabe (Faculty of Systems Sci. and Technol., Akita Prefectural Univ., 84-4 Ebinokuchi, Tsuchiya, Yuri-Honjo, Akita 015-0055, Japan, kwatanabe@akita-pu.ac.jp), Yusuke Oikawa (Grad. School of Systems Sci. and Technol., Akita Prefectural Univ., Yuri-Honjo, Japan), Sojun Sato, Shouichi Takane, and Koji Abe (Faculty of Systems Sci. and Technol., Akita Prefectural Univ., Yuri-Honjo, Japan)

Head-related transfer function (HRTF) is characterized as sound transmission from sound source to listener's eardrum. When a listener hears a sound that is filtered with the HRTFs, the listener can localize a virtual target (sound image) as if the sound had come from the position corresponding to that at which the HRTFs were measured. A moving sound image can be generated to switch HRTFs of successive direction in real-time. While many virtual auditory displays (VADs) based on synthesis of HRTFs have been proposed, most of them can synthesize only a few sound images due to lack of computation power. In this article, the VAD system implemented based on graphics processing unit (GPU) was introduced. In our system, the convolution of HRTFs is parallelized on GPU to realize a high-speed processing. In addition, the multiple HRTFs each of which is corresponding to sound sources at different position are processed in parallel to control multiple sound image simultaneously. In this article, the performance of our system was evaluated not only objectively but also subjectively. The results showed that our current system can present at least 40 sound images simultaneously in real-time.

TUESDAY AFTERNOON, 4 JUNE 2013 512BF, 1:00 P.M. TO 5:00 P.M.

Session 2pEAb

Engineering Acoustics: Controlling Sound Quality

Stephen Butler, Chair
NUWC, Newport, RI 02841

Contributed Papers

1:00

2pEAb1. Subjective and objective evaluation of sound quality of radio programs transmitted via Digital Audio Broadcast (DAB+) System. Andrzej B. Dobrucki and Maurycy J. Kin (Chair of Acoust., Wroclaw Univ. of Technol., Wybrzeze Wyspianskiego 27, Wroclaw 50-370, Poland, andrzej.dobrucki@pwr.wroc.pl)

The work presents results of research on the sound quality of different radio programs transmitted via Digital Audio Broadcasting (DAB+). This assessment has been provided with a use of psychoacoustic model as well as standard listening tests, using an Absolute Category Rating (ACR) method of scaling, and Comparison Category Rating (CCR) method. Results have shown that sound quality gets worse when bit-stream is of the lowest values (48 kbit/s or 24 kbit/s). Application of the Spectral Band Replication processor significantly improves the perceived quality, which is satisfying for bit streams higher than 64 kbit/s, particularly for jazz and popular music. The assessment with ACR method (recommended for broadcast by International Telecommunication Union) showed better notes than CCR one. It means that recommended method is less critical. Also results obtained with psychoacoustic model are more similar to obtained with CCR method. The attributes of spatial impression change in different ways. The greatest distortion has been observed for the perspective and spaciousness of sound image, while the sound color as well as localization stability and accuracy of phantom sources remained almost the same.

1:20

2pEAb2. Objective evaluation of sound quality for attacks on robust audio watermarking. Akira Nishimura (Media and Cultural Studies, Tokyo Univ. of Information Sci., 4-1, Onaridai, Wakaba-ku, Chiba 2658501, Japan, akira@rsch.tuis.ac.jp), Masashi Unoki (School of Information Sci., Japan Adv. Inst. of Sci. and Technol., Nomi, Japan), Kazuhiro Kondo (Grad. School of Sci. and Eng., Yamagata Univ., Yamagata, Japan), and Akio Ogihara (Faculty of Eng., Kinki Univ., Higashihiroshima, Japan)

Various attacks on robust audio watermarking have been proposed. Reversible signal processing attacks, such as sampling frequency conversion, degrade sound quality of the distributed watermarked audio (stego audio)

and disturb extraction of hidden data so that copyright detection systems using automated crawling are invalidated. Reversible signal processing of the attack can recover sound quality of the degraded audio data. In order to prove validity and security of audio watermarking system, analysis of the presumed attacks or reversible signal processing on stego audio, is required. However, these attacks on audio signal also degrade sound quality of commercial music where such pieces of music are considered to be not suitable for appreciation. Therefore, degradation of sound quality induced by various attacks should be taken into account to decide if the intensity of the attacks are realistic. In this study, objective audio quality measurement (PEAQ) was applied to the audio signals including typical perceptual coding, MP3, tandem MP3, MPEG4AAC, and reversible signal processing of sampling frequency conversion, noise addition, frequency shift, bandpass filtering, and echo addition. The results indicate requirements for robustness and criteria of the attacks on high quality and robust audio watermarking technology.

1:40

2pEAb3. Measurement of acoustic transmission properties of a handset with a piezoelectric vibrator using a head and torso simulator. Toshiharu Horiuchi and Tsuneo Kato (User Interface Lab., KDDI R&D Lab., Inc., 2-1-15 Ohara, Fujimino, Saitama 356-8502, Japan, to-horiuchi@kddilabs.jp)

This paper presents equalizing an acoustic transmission property from a handset with a piezoelectric vibrator at a pinna to an eardrum with that of a normal headphone by hearing the measured sound through a head and torso simulator (HATS). Recently, a piezoelectric vibrator that vibrates a pinna to produce sounds was adopted as a receiver of smartphones to improve perceived quality within noisy environments. The HATS, used for handset testings in accordance with ITU-T recommendations, has a silicon-rubber pinna simulator to reproduce realistic acoustic properties with its human-like shape and stiffness. However, the handset with the built-in piezoelectric vibrator is beyond the scope and was not tested on the HATS before. This paper clarifies the difference of frequency responses between the pinna simulator and the real pinna based on a subjective assessment that adjusts the loudness of pure tones through the pinna simulator to be equalized auditorily to those through the real pinna. We used B&K HATS Type 4128-D. The results indicated a flat response for the pinna simulator, while a low-pass-like response for the real pinna with a cutoff at 1.5 kHz. Thereby, the actual sound can be simulated from the sound measured by the HATS with the responses.

2:00

2pEAb4. The effect of firefighting protective equipment on head related transfer functions. Joelle I. Suits, Theodore F. Argo (Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keaton St., Austin, TX 78712, jsuits@utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX), Ofodike A. Ezekoye (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Craig A. Champlin (Austin, Texas)

Personal Alert Safety System (PASS) devices are used in the fire service to locate trapped or injured personnel. When a firefighter becomes incapacitated, the device emits an audible alarm to help rescue teams locate the downed firefighter. These devices have been successful, but there are still cases in which PASS is not effective, and the present project seeks to provide science-based guidance for improvements to PASS. One part of this complex problem is the effect of the protective equipment (helmet, eye protection, hood, coat) that is worn by firefighters on hearing. Since this has not previously been studied, it has not been accounted for in the current design of the PASS signal. To address this deficiency, head related transfer function (HRTF) measurements have been taken with a KEMAR acoustic mannequin wearing various combinations of the aforementioned equipment. Results indicate a reduced received level at the ear when the full complement of gear is worn, as might be expected, potentially causing a reduced detection range. In addition, the helmet and eye protection devices cause significant disruption of the normal HRTF patterns, which could potentially interfere with localization. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

2:20

2pEAb5. An escape guiding system utilizing the precedence effect for evacuation signal. Takahiro Fujikawa and Shigeaki Aoki (Electron. Information and Commun. Eng., Kanazawa Inst. of Technol., 7-1 Ohgigaoka, Nonoichi, Ishikawa 921 - 8501, Japan, aoki_s@neptune.kanazawa-it.ac.jp)

This research aims at building an escape guiding system using audio signals. The system utilizes the precedence effect for the listener to perceive easily the direction of an emergency exit. In one of the ordinary escape guiding systems, two or more loudspeakers are set on the ceiling of a passageway. Since the precedence effect is generated by delaying suitably the audio guidance signal radiated from the loudspeakers, the proper direction of the emergency exit is recognized. However, the effective listening area is limited. That is, the ordinary escape guiding systems is not effective in areas under loudspeakers. In this paper, a new method that the loudspeakers are set beside in a passageway is proposed. The generation and disappearance of the precedence effect of an audio signal in the new configuration are investigated. In the listening tests, the time delay and the difference of intensity between loudspeakers are parameters. The installation angle of the loudspeaker is another parameter. Male voice and female voices are used as emergency guidance. The guidance effect of the configuration in setting beside the loudspeaker is confirmed and the test results are discussed.

2:40

2pEAb6. Reduction of sound leakage in handheld devices using open loop control. GunWoo Lee, Aran Cha, SeoungHun Kim, YoungTae Kim (Samsung Electron. Co., Ltd., 416, Martan 3-dong, Yeongtong-gu, Suwon-si 443-742, South Korea, gw325.lee@samsung.com), and JungWoo Choi (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., Daejeon, South Korea)

Recently, handheld devices with sound functionality are popular. A problem encountered by using such devices is sound leakage due to inappropriate volume setting, which should be reduced not to disturb people around the user. In previous study, we presented an active control (ANC) technique to control such sound leakage in handheld devices. The least squares (LS) optimum control under various positions of error sensors was investigated. Based on the results, desirable and feasible microphone-loudspeaker setups were suggested. In this paper, we present a novel open loop leakage reduction scheme, and it is compared with adaptive noise control method. To achieve this, control available frequency bands and spatial ranges are studied. And the influence of leakage reduction performance by the receiver and control speaker's radiation pattern are analyzed. Also the effects of the physical environment including the user's hand for the leakage control performance are studied. Finally, the proposed method is implemented on the mobile phone mock-up, and the performance of actual measured leakage reduction is investigated.

3:00

2pEAb7. Active control applied to simplified wind musical instrument. Thibaut Meurisse, Adrien Mamou-Mani, René Caussé (Instrumental Acoust., IRCAM, 1 place Stravinsky, Paris, France, thibaut.meurisse@ircam.fr), and David Sharp (Acoust. Res. Group, The Open Univ., Milton Keynes, United Kingdom)

Musicians have always been interested in the evolution of their instruments. This evolution might be done either to adapt an instrument's quality to musicians' and composers' needs, or to enable it to produce new sounds. In this study, we want to control the sound quality and playability of wind instruments, using active control. The active control makes it possible to modify the input impedance (frequency, gain, and damping) of these instruments and to modify the instrument's quality. Simulations and first experiments on a simplified reed instrument are presented. We simulate a control of the modes (frequency, damping) of a cylinder using two different approaches: classic feedback and modal active control. Then, we apply these control methods on a simplified reed instrument with embedded microphone and speaker. Finally, the effects on sound and playability of the instrument is studied.

3:20

2pEAb8. Predicting speech transmissibility using ray tracing and statistical energy analysis. Sascha Merz, Vincent Cotoni, and Phil Shorter (ESI Group, 12555 High Bluff Drv, Ste. 250, San Diego, CA 92130, sascha.merz@esi-group.com)

Statistical Energy Analysis (SEA) is widely used for predicting interior noise across many different industries. SEA typically describes steady-state reverberant responses, for example the response at interior passenger locations in transportation applications due to steady-state exterior sources. However, many applications exist where the direct field is also important. This includes prediction of Speech Transmissibility for audio systems in automotive applications or public address systems in train and aircraft applications. For predicting Speech Transmissiblity, the transient response at a given receiving location due to transient excitation applied at a particular source location must be known. Geometrical methods such as ray tracing are often used for describing the first few reflections of the direct field; however, they are not well suited for describing the reverberant field and typically include approximate statistical assumptions about late time reflections. These assumptions often do not consider the detailed distribution of sound package in interior spaces. An alternative approach is to use ray tracing for predicting the direct field and low order reflections and to use SEA to predict late time reflections and background noise. This approach is computationally efficient and can use information contained in existing SEA models. The method is discussed and validation examples are presented.

3:40

2pEAb9. Comparison of precedence effect behavior in anechoic chamber with that in ordinary room. Koji Abe, Shouichi Takane, Sojun Sato, and Kanji Watanabe (Electron. and Information Systems, Akita Prefectural Univ., 84-4 Ebinokuchi Tsutiya, Yuri-Honjyo 015-0055, Japan, koji@akita-pu.ac.jp)

The precedence effect is well known as one of auditory illusions occurred by using multiple sound sources with similar sound output. When a sound is followed by similar sound separated with relatively short time delay, a single fused sound image is localized at the source position corresponding to the first-arriving sound. This feature is applicable to public address systems, which make audience perceive the sound image different from the actual sound source positions prepared for the system, with some sound reinforcement achieved. In spite of many studies in this phenomenon, the behavior of the precedence effect has been investigated for limited sound source arrangements in laboratory environments like anechoic chamber. On the other hand, this behavior in the ordinary room is not obvious, and it is effective to clarify the difference of the behavior of the precedence effect in anechoic chamber from that in the ordinary room for the application of the precedence effect to the public address system. In this study, the similar sound sources were installed both in the lecture room and in the anechoic chamber, and the behavior of the precedence effect was compared each other with the given time and level difference among sound sources.

4:00

2pEAb10. Vertical sound image control using level differences between parametric speakers. Kumi Maeda, Takanori Nishino, and Hiroshi Naruse (Grad. School of Eng., Mie Univ., 1577 Kurimamachiya-cho, Tsu 5148507, Japan, nishino@pa.info.mie-u.ac.jp)

Horizontal sound image can be controlled by using level difference between two loudspeakers; however, vertical sound localization is difficult. In this report, we propose a method of controlling a sound image with sound level differences between two parametric speakers that have a super-directivity. Our proposed system uses sounds that are reflected on a wall. In our experiments, two parametric speakers were arranged 56.6 and 226.0 cm high, respectively. Vertical sound localization was evaluated by subjective tests. Subjects were seven males and one female. Test signals were a white noise whose duration was 0.5 s. Sound level differences between parametric speakers were $-\infty$, -9, -6, -3, 0, 3, 6, 9, and ∞ dB. Test signals were presented three times to each subject in a random order. Both parametric speakers were arranged at angles of $0°$, $\pm 5°$, and $\pm 10°$ from the horizontal plane, respectively. Answers were examined by the Wilcoxon rank-sum test. From the results, good performances were obtained when parametric speakers were arranged at angles of $\pm 10°$. [Work supported by a Grant-in-Aid for Scientific Research (24500203).]

4:20

2pEAb11. Characteristics of whistle noise from mufflers with perforated pipes. Tatsuya Yamada, Takehiko Seo, Masato Mikami (Grad. School of Sci. and Eng., Yamaguchi Univ.,2-16-1 Tokiwadai, Ube, Yamaguchi, Japan, 201, 3-44 Wakamatsucho, Ube, Yamaguchi, Japan, Ube, Yamaguchi 755-8611, Japan, s045ve@yamaguchi-u.ac.jp), and Takashi Esaki (Sango Co., Miyoshi, Aichi, Japan)

As a countermeasure to reduce exhaust gas pulsation noise in a wide frequency range, the exhaust system employs expansion, resonance and sound absorbing structures with perforated pipes. However, whistle noise is generated near holes under some conditions. In straight-through-type mufflers, sound absorbing materials covering punching holes can suppress whistle noise generation. On the other hand, in the expansion-cavity-type muffler, which has inlet and outlet insertion pipes, effective countermeasures have not been clarified yet. The purpose of this study is to improve understanding of whistle noise generation mechanism in mufflers with perforated pipes. First, we measured sound pressure of whistle noise radiating from a straight-through-type sub-muffler with a perforated pipe with steady flow. Results show that the frequency of predominant whistle noise became higher stepwisely with increasing the flow velocity and was higher with smaller hole diameter. Strouhal number based on the hole diameter, the frequency of predominant whistle noise and flow velocity existed within a certain range while the hole diameter and flow velocity were varied. Next, we measured the sound pressure of whistle noise radiating from an expansion-cavity-type main muffler with a perforated pipe. The whistle noise generation is discussed in comparison with that for the sub-muffler.

4:40

2pEAb12. Back scattering attenuators (silencers). Giora Rosenhouse (89 Hagalil St., Haifa 32684, Israel, fwamtech@bezeqint.net)

Back scattering silencers are a kind of “sonic crystals.” The frequency range of “sonic crystals” modeling includes the whole audio range and ultrasound, up to phonon waves. Here we concentrate on applications in the audio domain. However, the present investigation is applicable at the higher frequencies as well¸ because of the high scalability of the system. The paper defines and analyses 2-D and 3-D back scattering silencers, made of arrays of rigid or soft obstacles (cylinders, spheres or prolate/oblate spheroids for example), in order to attenuate plane waves by multiple scattering along a wave guide. This effect is strong especially at certain band-gaps along the frequency domain. Each obstacle reflects secondary waves that are partially reflected. Thus, along each row of the array, the sound waves lose a certain amount of the energy that adds to the total amount of attenuation. Specifically, the paper analyses back scattering silencers built of meshes of either cylinders or prolate spheroids, where each obstacle is located symmetrically within a fluid cell and each cell is identical to the others.

Session 2pED

Education in Acoustics: Teaching Methods in Acoustics

Preston S. Wilson, Chair
Appl. Res. Lab., Univ. of Texas at Austin, 1 University Station, TX 78712-0292

Contributed Papers

1:20

2pED1. A design of an impedance tube for teaching acoustic material properties and laboratory techniques. Chelsea E. Good, Aldo A. Glean, Joseph F. Vignola, John A. Judge, Teresa J. Ryan, Nicole Bull (Mech. Eng., Catholic Univ. of America, 620 Michgan Ave. NE, Washington, DC 20064, 26good@cardinalmail.cua.edu), and Diego Turo (Bioengineering, George Mason Univ., Fairfax, VA)

The design and implementation of an adaptable acoustic impedance tube for instructional use will be presented. This system, which was designed for use in a graduate-level laboratory environment, is in accordance with ASTM standard E1050-98. The system is configurable for both horizontal and vertical measurements. Vertical use enables acoustic property measurement of granular materials, allowing characterization of a wide variety of media. The academic content is delivered by engaging students in measurements using the impedance tube. Specific acoustics content objectives include modeling of sound propagation in porous materials and the role of acoustic parameters such as porosity, tortuosity, and flow resistivity in quantifying acoustic behavior of materials. Instructional goals also include mastery of computer based data acquisition and processing. This aspect of modern laboratory practice is a critical part of our graduate acoustics curriculum. Students participated actively in the development of the software used to collect data and calibrate the impedance tube. The system design required the students to solve practical problems such as cutting and positioning of the sample as well as more demanding tasks such as design of a multilayer porous medium with desired acoustic properties.

1:40

2pED2. Popular papers with short case stories on acoustics and vibration for practical engineers and students. Roman Vinokur (Engineering, ResMed Motor Technol., 9540 De Soto Ave., Chatsworth, CA 91311, romanv@resmed.com)

One of the reasons for using foam wedges or cones in hemi-anechoic rooms is a gradual change of the acoustical impedance in order to reduce the reflection of incident sound waves from the sound absorbing walls. By analogy, popular papers on science (in particular, in acoustics and vibration) facilitate a smooth introduction to new theories because of their small cognitive "impedance" to understanding the written information. Such papers are relevant mostly for extramural reading but they help engineers and students to promptly perceive important effects and applications via interesting case stories and simplified physics and mathematics. Generally speaking, this approach is not new: in particular, it was successfully applied by Perelman in his book "Physics for Entertainment." But in author's opinion, for better effectiveness such texts should be limited in size and include 3–4 related short case stories from actual engineering or consulting practice, history, news, or literature. To illustrate this method, several one-page papers published in the "Sound and Vibration" magazine will be briefly discussed: "Vibroacoustic Measurements without Transducers," "A Common Myth about Mechanical Resonance," "Only the Best Will Do," and "Haunted Buildings and Other Acoustical Challenges."

2:00

2pED3. Mechanical bent-type models of the human vocal tract consisting of blocks. Takayuki Arai (Dept. of Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

In our previous work, we developed several physical models of the human vocal tract and reported that they are intuitive and helpful for students studying acoustics and speech science. Models with a bent vocal tract can achieve relatively realistic tongue movements. These bent-type models had either a flexible tongue or a sliding tongue. In the former case, the tongue was made of a flexible gel-type material so that we could form arbitrary tongue shapes. However, this flexibility meant that training is needed to achieve target sounds. In the latter case, the tongue was made of an acrylic resin, and only a limited number of vowel sounds can be achieved because so few sliding parts are available to change the tongue shape. Therefore, in this study, we redesigned the mechanical bent-type models so that they now consist of blocks. By placing the blocks at the proper positions, the block-type model can produce five intelligible Japanese vowels. We also designed a single bent-type model with sliding blocks that can produce several vowel sounds. [This work was partially supported by a Grant-in-Aid for Scientific Research (24501063) from the Japan Society for the Promotion of Science.]

2:20

2pED4. Teaching acoustics at architect students using digital tools. Delphine Bard (Eng. Acoust., Lund Univ., John Ericssonväg 1, Lund 221oo, Sweden, delphine.bard@construction.lth.se), Tina-Henriette Kristiansen (Architecture, Lund Univ., Lund, Sweden), and Eva Frühwald Hansson (Solid Mechanics, Lund Univ., Lund, Sweden)

At the School of Architecture, Lund University Sweden, courses are taught in different ways. A large part of the education during year one and two is held as "studios," doing creative (individual) project work. Usually acoustics courses rather correspond to the traditional engineering education style, using lectures, exercises, small project works, and final written examination. The problem is that many students do not know how to use the gathered information in their creative works. The aim of the study covered by this paper was to improve upon the existing teaching/learning of the fundamental acoustics principles scheme by introducing new methodologies. In order to achieve our goal, we gave the students two different assignments: In the first assignment, the students had to produce short educational movies to explain and teach acoustics principles to their peers. In the second assignment, they should implement the new knowledge gathered in the first assignment into their individual creative projects.

2:40

2pED5. Black box measurements – Using a family of electrical circuits as a tool for self-guided learning in acoustical engineering. Bernardo H. Murta, Sergio Aguirre, Jessica Lins, Stephan Paul, Eric Brandao (Departamento de Estruturas e Construção Civil, Universidade Federal de Santa Maria, Rua Dezenove de Novembro, 289, 302, Santa Maria, Rio Grande do Sul 97060160, Brazil, be.murta@gmail.com), and Pascal Dietrich (Inst. of Tech. Acoust., RWTH Aachen Univ., Santa Maria, Brazil)

A partnership between Brazil's first undergraduate program in Acoustical Engineering and the Institute of Technical Acoustics of RWTH Aachen University yielded in a didactic project that uses the engineering software MATLAB with the ITA-Toolbox to teach acoustic measurements. Simple electrical circuits are used to mimic typical behavior of acoustical systems. This low-cost solution has proven to be didactically very effective since it helps students to identify themselves with the measurement tasks. Two hardware solutions were developed-a simple oscillator circuit integrated into connectors of audio cables and a desktop box containing seven different transfer characteristics ranging from ideal linear and time-invariant to nonlinear and time-varying behavior. Undergraduate students of Acoustical Engineering used both devices in classroom experiments for self-guided learning by comparing their results to published results. Students were able to learn the fundamental concepts of acoustical measurements and to handle measurement tasks. Besides the practical experiences and the learning effect, the students were also encouraged to step into the open source routines of the software, understand the signal processing steps, adapt routines, and even write their own ones, e.g., a GUI that provides effective control of the measurement via touch-screens.

3:00

2pED6. A for play! Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

The task was straightforward; design and build a playhouse to be raffled for a charitable organization. A team consisting of students, volunteers, and faculty banded together to not only meet the requirements but to exceed the typical preconceived ideas of a totally enclosed miniature home. Based upon the needs of juvenile clients, the design team focused more on "play" than on "house" when working out conceptual ideas. The playful design was based upon the enclosure being partially open to allow air flow, sunlight, and the ability for the inhabitants to have an aural connection to the outside. The idea of having partially open space on the lower level, a mere 5'-0" x 5'-0" footprint, flanked by stepped bands of cedar and cypress yielded a particular acoustical presence. The space is not only visually unique, but the selection of materials, how they were cut and assembled, and the scale in relation to a seated child enhance the fun factor by creating an enveloping and somewhat amplified acoustic. This project provided pedagogical opportunities within an atypical learning environment. The final inhabitable playhouse exceeded our visual and acoustical expectations of a small space and prove acoustics "plays" an intrinsic role despite occupant age.

TUESDAY AFTERNOON, 4 JUNE 2013 512DH, 1:00 P.M. TO 5:40 P.M.

Session 2pMU

Musical Acoustics: Musical Preference, Perception, and Processing

Jean-François Petiot, Cochair
IRCCyN, Ecole Centrale de Nantes, 1 rue de la noe, BP92101, NANTES 44321, France

Richard L. King, Cochair
Music Res., McGIll Univ., 555 Sherbrooke West, Montreal, QC H3A 1E3, Canada

Contributed Papers

1:00

2pMU1. Modeling of the subjective quality of saxophone reeds. Jean-François Petiot, Pierric Kersaudy (IRCCyN, Ecole Centrale de Nantes, 1 rue de la noe, BP92101, NANTES 44321, France, jean-francois.petiot@irccyn.ec-nantes.fr), Gary Scavone, Stephen McAdams (CIRMMT, Schulich School of Music, McGill Univ., Montreal, QC, Canada), and Bruno Gazengel (LAUM, Université du Maine, Le Mans, France)

The subjective quality of cane reeds used on saxophones or clarinets may be very different from one reed to another even though the reeds have the same shape and strength. The aim of this work is to understand the differences in the subjective quality of reeds and to explain them with objective measurements. A subjective study, involving a panel of 10 musicians, was first conducted on a set of 20 reeds of the same strength. Second, signal recordings during saxophone playing (*in vivo* measurements) were made of the pressures in the player's mouth, in the mouthpiece and at the bell of the instrument. These measurements enable us to deduce specific parameters, such as the threshold pressure or the spectral centroid of the notes. After an analysis of the subjective and objective data (assessment of the agreement between the assessors and the main consensual differences between the reeds), correlations between the subjective and objective data were performed. To propose a model of the subjective quality, a machine learning approach was proposed using partial least-squares (PLS) regression and PLS discriminant analysis. Results show interesting performance of the model in cross validation and open the potential for an objectification of the perceived quality.

1:20

2pMU2. Perceptual evaluation of violins: A comparison of intra-individual agreement in playing vs. listening tasks for the case of richness. Charalampos Saitis, Gary P. Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., 555 Sherbrooke Str. West, Montreal, QC H3A 1E3, Canada, charalampos.saitis@mail.mcgill.ca), Claudia Fritz (Lutheries-Acoustique-Musique, Institut Jean Le Rond d'Alembert, Université Pierre et Marie Curie, Paris, France), and Bruno L. Giordano (Inst. of Neurosci. and Psychol., Univ. of Glasgow, Glasgow, United Kingdom)

In previous studies by the authors, it was shown that there is a significant lack of agreement between violinists when evaluating different instruments in terms of perceived richness in free-playing tasks. A new experiment was designed to further investigate the perceptual evaluation of richness using both a constrained-playing task, which was recorded, and a subsequent listening task (using the previously recorded sounds). The goal was to compare the evaluation of richness from playing vs. listening tasks in order to better understand whether they are based more on auditory feedback or tactile and

proprioceptive cues in the wider context of correlating audio features extracted from the recordings with richness judgments. Skilled violinists were asked to rank five different instruments by playing only certain notes on the G-string. Subsequently, the players were asked to listen to their recordings and rank the violins. Results appeared to show a higher inter-individual agreement relative to previous studies. Furthermore, the rankings in the playing task were generally different from those in the listening task, indicating that the evaluation of richness is based on different criteria in the two cases. Results from matching the tristimulus and spectral centroid of recorded tones to richness judgments will be presented at the conference.

1:40

2pMU3. Music of the body: An investigation of skull resonance and its influence on musical preferences. Jitwipar Suwangbutra, Rachelle Tobias, and Michael S. Gordon (Dept. of Psych., William Paterson Univ., 300 Pompton Rd., William Paterson U., Wayne, NJ 07470, jitwipar@gmail.com)

Musical preferences can be attributed to environmental and biological factors. This research analyzes the specific influence of body resonance, and in particular, how the resonant properties of the skull might contribute to auditory perception of music and musical preferences. To examine this issue resonances were sampled from a set of participants and analyzed using FFTs. The fundamental frequencies of each participant's head was correlated against their preference amongst a set of novel melodies presented in each of the 12 major keys. Using this method the spectral properties of the melody could be directly related to the resonant properties of a listener's skull to evaluate their influence. While results were subtle, participants were found to be influenced in their judgments of loudness and musical preference for the melodies. Conclusions from this research support speculation on an embodied model of cognition for musical interactions.

2:00

2pMU4. Preferences for melodic contours transcend pitch. Jackson Graves, Christophe Micheyl, and Andrew J. Oxenham (Psychology, Univ. of Minnesota, 1849 Washington Ave. S, Apt 435, Minneapolis, MN 55454, grave276@umn.edu)

The question of what makes a good melody has interested composers, music theorists, and psychologists alike. Many of the observed principles of good "melodic continuation" involve the melodic contour—the pattern of rising and falling pitch within a sequence. Recent work has shown that contour perception can extend beyond pitch to other auditory dimensions, such as timbre and loudness. Here we show that the generalization of contour perception to non-traditional dimensions also extends to melodic preferences and expectations. We find that subjective continuation ratings for sequences that vary in brightness or loudness generally conform to the same contour-based expectations as pitch sequences. The results support the hypothesis that contour perception is a general auditory phenomenon, and suggest that the well-known preference for narrow ranges and small intervals in melodies is not unique to the dimension of pitch. [Work supported by NIH Grant R01DC05126 and by the Undergraduate Research Opportunities Program of the University of Minnesota.]

2:20

2pMU5. Development of a new series of tests to assess the effectiveness of hearing aids for the perception of music. Martin Kirchberger (Health Sci. & Technol., ETH Zürich, Universitätsstrasse 2, Zürich 8092, Switzerland, martin.kirchberger@phonak.com), Frank A. Russo (Psychology, Ryerson Univ., Toronto, ON, Canada), Peter Derleth, and Markus Hofbauer (Sci. & Technol., Phonak AG, Stäfa, Switzerland)

A new series of tests has been designed to assess the effectiveness of hearing aids for the perception of music. Within each subtest, discrimination thresholds for low-level acoustic dimensions are determined adaptively using a 2AFC method within the context of a musical judgment regarding melody, harmony, timbre or meter. The presented test stimuli are synthesized and either unprocessed or processed by different hearing aid signal processing algorithms before being played back via loudspeaker. The battery will be used to evaluate different hearing aid algorithms with regard to their benefit for functional hearing in music. A group of six normal hearing control participants (6.7 dB HL) and five hearing impaired participants (34 dB HL) each performed the melody subtest and the harmony subtest twice. The hearing impaired subjects had higher discrimination thresholds than the control group. A comparison of the results from both administrations suggests that these two subtests have good test-retest reliability.

2:40

2pMU6. The examination of the performance motion and emotional valence by a pianist. Yuki Mito, Hiroshi Kawakami (College of Art, Nihon Univ., 2-42-1 Asahigaoka Nerima-Ku, Tokyo 176-8525, Japan, mitotic@hotmail.com), Masanobu Miura (Faculty of Sci. and Technol., Ryukoku Univ., Shiga, Japan), and Yukitaka Shinoda (College of Sci. and Technol., Nihon Univ., Tokyo, Japan)

Until now, we examined the performance motion by the snare drum. Particularly, we analyzed an association between emotion and motion using motion capture. We analyzed total of six emotions [tenderness, happiness, sadness, fear, anger, and non-emotion from Juslin (2001)]. The analysis method averaged the performance motion of six emotions. We named the method as "Motion Averaging Method (MAM)." We calculated the difference of the each emotion by the MAM and revealed the characteristic of motion. From the result, we achieved results in snare drum motion. Therefore, in this study, we considered the association between emotion and motion in the keyboard. The subject was a professional pianist. A musical piece was an etude of the music dictation. Motion capture system is the MAC 3D System of Motion Analysis Corp., which is an optical motion capture system. The measurement marker bonded 33 points to the upper body. As a result, we understood the difference in emotional valence by the center of gravity of the head, an arm, the body, and the hand. Particularly, the non-emotion was move less than other five emotions. Then, we were able to express an association between five emotions of Juslin and motion as a figure.

3:00

2pMU7. Human ability of counting the number of instruments in polyphonic music. Fabian-Robert Stöter, Michael Schoeffler, Bernd Edler, and Jürgen Herre (Int. Audio Lab. Erlangen, Am Wolfsmantel 33, Erlangen 91058, Germany, fabian-robert.stoeter@audiolabs-erlangen.de)

There are indications that humans are only able to correctly count up to three voices in polyphonic music pieces of homogeneous timbre, where each voice is played by the same instrument. A more general case, where voices are played by instruments of inhomogeneous timbre, has not been fully addressed so far. In order to approach this question we conducted a listening experiment with 62 participants to find out whether both scenarios — instrumentation by inhomogeneous or homogeneous timbre — share the same outcome. This paper describes the design of the experiment including an analysis of the results, which show that both scenarios are related. Furthermore, a detailed analysis of the error rates in correctly counting the number of instruments reveals that there are significant differences between non-musician and musician listeners, in particular regarding the upper auditory limit of the number of correctly counted instruments. Based on these results, models for the perception of instruments in auditory streams can be developed.

3:20–3:40 Break

3:40

2pMU8. Loudspeakers and headphones: The effects of playback systems on listening test subjects. Richard L. King, Brett Leonard (Music Res., Schulich School of Music, McGill Univ., 555 Sherbrooke West, Montreal, QC H3A 1E3, Canada, richard.king@mcgill.ca), and Grzegorz Sikora (Automotive Audio, Bang & Olufsen, Pullach, Germany)

Many modern listening test designers disagree on the best playback system to be used when conducting tests. The preference often tends toward headphone-based monitoring in order to minimize the possibility of undesirable acoustical interactions with less than ideal testing environments. On the other hand, most recording and mixing engineers prefer to monitor on loudspeakers, citing a greater ability to make critical decisions on level balances and effects. While anecdotal evidence suggests that differences exist between systems, there is little quantified, perceptually based data to guide both listening test designers and engineers in what differences to expect

when alternating between monitoring systems. Controlled tests are conducted with highly trained subjects manipulating the level of solo musical elements against a backing track, using both headphones and loudspeakers. This task serves to make the results equally applicable to critical mixing tasks and rigorous listening tests. The results from both playback systems are compared, showing a defined difference in the mean levels set on the two different monitoring systems. Likewise, the variance seen across subjects is larger when monitoring on headphones than on loudspeakers, lending credence to the hypothesis that tests conducted on one playback system may not be equally applicable to the other.

4:00

2pMU9. Modeling listener distraction resulting from audio-on-audio interference. Jon Francombe, Russell Mason, Martin Dewhirst (Dept. of Music and Sound Recording, Inst. of Sound Recording, Univ. of Surrey, Guildford, Surrey GU2 7XH, United Kingdom, j.francombe@surrey.ac.uk), and Søren Bech (Bang & Olufsen, Struer, Denmark)

As devices that produce audio become more commonplace and increasingly portable, situations in which two competing audio programs are present occur more regularly. In order to support the design of systems intended to mitigate the effects of interfering audio (including sound field control, noise cancelation or source separation systems), it is desirable to model the perceived distraction in such situations. Distraction ratings were collected for a range of audio-on-audio interference situations including various target and interferer programs at three interferer levels, with and without road noise. Time-frequency target-to-interferer ratio (TIR) maps of the stimuli were created using a simple auditory model. A number of feature sets were extracted from the TIR maps, including combinations of mean, standard deviation, minimum and maximum TIR taken across the duration of the program item. In order to predict distraction ratings from the features, linear regression models were produced. The models were evaluated for goodness-of-fit (RMSE) and generalizability (using a K-fold cross-validation procedure). The best model performed well, with almost all predictions falling within the 95% confidence intervals of the perceptual data. A validation data set was used to test the model, suggesting areas for future improvement.

4:20

2pMU10. Effects of audio latency in a disc jockey interface. Laurent S. Simon, Arthur Vimond (INRIA, INRIA Rennes, Bretagne Atlantique, Campus Universitaire de Beaulieu, Rennes Cedex 35042, France, laurent.s.simon@inria.fr), and Emmanuel Vincent (INRIA Nancy, Rennes, France)

This study presents an evaluation of the disturbance caused by audio latency in a DJ-ing task. An experiment was conducted, during which subjects were asked to synchronize one piece of music to a reference piece of music using a common DJ-ing interface. Synchronization was performed by adjusting the speed of one of the music pieces to that of a reference piece of music and time-aligning both pieces. Latency was introduced between the interface and the audio output, varying between 0 and 550 ms. The average synchronization time was estimated as a function of subjects, beat-per-minute difference between the pieces of music, and latency. Results showed that for trained DJs, synchronization time increased significantly above 130 ms of audio latency, whereas for naive subjects, latency had no influence on the synchronization time.

4:40

2pMU11. Evaluating the absolute volume of digital sound source measurement and standard measuring unit. Doo-Heon Kyon and Myung-Jin Bae (Electron. Eng. Dept., Soongsil Univ., Hyeongnam Eng. Hall #1212, Sangdo-dong, Seoul 156-030, Democratic People's Republic of Korea, kdhforce@gmail.com)

Listeners do not know the actual volume of sound before playing a sound source, so they have to adjust the volume through trials and errors. Moreover, they have to change the volume repetitively because each sound source has different volume. If we can identify the absolute volume of a sound source at the perspective of listener, the volume of all sound sources can be effectively standardized. This study evaluated a method to measure the absolute volume of a digital sound source and suggested the dB(N) as a measuring unit. The pink noise was used as a reference sound source, to be used for measuring the absolute volume. The pink noise was set as 60 dB(N), which is equal to sound output of 60 dB(A). The volume was adjusted until the pink noise and the target sound matches into a recognizable volume by reducing the pink noise or target sound source under the given environment. Subsequently, the difference is reflected to 60 dB(N) to determine the absolute volume. The accuracy of measured results was confirmed through a music listening test and suggested how to develop the volume system using the absolute volume.

5:00

2pMU12. Refining the stereo technique for augmented ambience gradient: Improvements in stereo image, spatial envelopment, and mixing flexibility. David J. Tagg and Kevin Fallis (Sound Recording, McGill Univ., 74 Cameron Cres, Toronto, Ontario M4G 2A3, Canada, kevin.fallis@mail.mcgill.ca)

While working on location, recording engineers are often challenged by insufficient monitoring. Poor (temporary control room) acoustics and/or mandatory headphone monitoring can make judgments regarding microphone choice and placement difficult. This compromised monitoring often leads to timbral, phase, and stereo image problems. We are often forced to choose between the improved spatial imaging of near-coincident techniques and the attractive acoustic envelopment from spaced omni-directional mics. This research reviews a new technique: Stereo Technique for Augmented Ambience Gradient (STAAG), which aims to improve stereo imaging, ambient envelopment, and flexibility in the mix. Building on a preliminary study, this research realizes ideal microphone angle/spacing combinations to promote spatial accuracy, investigates the quality of the ambient envelopment compared to omnidirectional-based techniques, and the ability of STAAG to allow an engineer to manipulate the direct to reverberant energy ratio during post-production without corrupting the stability of the stereo image.

5:20

2pMU13. Real-time concatenative synthesis for networked musical interactions. Chrisoula Alexandraki (Dept. of Music Technol. and Acoust., Technol. Educational Inst. of Crete, E. Daskalaki Perivolia, Rethymnon, Crete 74100, Greece, chrisoula@staff.teicrete.gr) and Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Hamburg, Germany)

The recent proliferation of Networked Music Performances has led to the investigation of low-latency, low-bitrate musical encoding schemes, including audio codecs and control protocols that specifically address the requirements of live musical interactions across the Internet. This work presents an alternative perspective inspired by the "synthesis by analysis" approach, tightly constrained in terms of processing latencies and rendering quality. The entire process is fully automated and involves an offline processing phase (that takes place prior to performance) and an online real-time analysis-synthesis phase. The offline phase involves processing a solo recording of each musician's part so as to acquire (a) audio segments corresponding to each note in the performance, and (b) a trained Hidden Markov Model to be later used for online analysis. During live performance, the online analysis process encodes the position of the performance on a music score and re-synthesizes the audio waveform by concatenating the audio segments of the offline phase. Although the synthesized waveform originates from an offline solo recording, it is synchronized to the live performance at note level, so as to allow for rendering a wide range of musical tempi as well as their expressive variations. The paper presents the complete methodology and reports on implementation details and preliminary evaluation results.

Session 2pNSa

Noise: Community Response to Low-Amplitude Sonic Booms

Alexandra Loubeau, Cochair
NASA Langley Res. Ctr., MS 463, Hampton, VA 23681

Juliet A. Page, Cochair
Wyle, 200 12th St. South Ste. 900, Arlington, VA 22202

Chair's Introduction—12:55

Invited Papers

1:00

2pNSa1. Overview of the waveform and sonicboom perception and response program. Juliet A. Page (Wyle, 200 12th St. South Ste. 900, Arlington, VA 22202, juliet.page@wyle.com)

The Waveform and Sonicboom Perception and Response (WSPR) Program conducted in California in November 2011 was designed to test and demonstrate the applicability and effectiveness of techniques to gather data relating human subjective response to multiple low-amplitude sonic booms. It was a practice session for future wider scale testing of communities, eventually using a purpose built low-boom demonstrator aircraft. The WSPR program addressed the following: design and development of an experimental design to expose people to low-amplitude sonic booms; development and implementation of methods for collecting acoustical measures of the sonic booms in the neighborhoods where people live; design and administration of social surveys to measure people's reactions to sonic booms; and assessment of the effectiveness of various elements of the experimental design and execution to inform future, wider-scale testing. The low boom community response pilot experiment acquired sufficient data to assess and evaluate the effectiveness of the various physical and psychological data gathering techniques and analysis methods. Results include a comparison of survey modes, techniques for correlating subjective and objective data, assessment of single event and cumulative daily sonic boom subjective and percent highly annoyed results, and methods for analysis of empirical boom data.

1:20

2pNSa2. Low amplitude sonic boom noise exposure and social survey design. Kathleen K. Hodgdon (Appl. Res. Lab., Penn State, ARL North Atherton St., P.O. Box 30, State College, PA 16804-0030, kkh2@psu.edu) and Juliet Page (Wyle, Arlington, VA)

The Waveform and Sonicboom Perception and Response (WSPR) Program conducted a field study relating subjective response to noise from multiple low-amplitude sonic booms. The team was led by Wyle and included researchers from Penn State, Tetra Tech, and Gulfstream Aerospace Corp. The test exposed residents in the Edwards Air Force Base (EAFB) Housing area to two weeks of low-amplitude sonic booms while recording their responses via surveys. The noise exposure design balanced DNL across test days, the number of low, medium, and high booms, and the separation of booms between AM and PM flight sequences. Survey instruments consisted of a Baseline survey, a Single Event survey, and a Daily Summary survey. The WSPR low boom survey included a question on strength of annoyance, followed by questions on the strength of perception of five additional variables that contribute to the annoyance response. Three modes of administration were utilized for both the single event and daily summary surveys: paper/pen, web-based, and Mobile (Apple) device. The survey followed recommendations published by The International Commission on the Biological Effects of Noise (ICBEN). The data from the low boom field test provide a measure of the acceptance of low booms in an acclimated community.

1:40

2pNSa3. A flight research overview of the Waveforms and Sonicboom Perception and Response Project, the National Aeronautics and Space Administration's pilot program for sonic boom community response research. Larry J. Cliatt, Edward A. Haering, Michael D. Holtz, Thomas P. Jones, Erin R. Waggoner (NASA Dryden Flight Res. Ctr., P.O. Box 273, M.S.2228, Edwards, CA 93523, larry.j.cliatt@nasa.gov), Scott L. Wiley (Tybrin Corp., Edwards, CA), Ashley K. Parham (NASA Dryden Flight Res. Ctr., Edwards, CA), and Franzeska F. Houtas (Tybrin Corp., Edwards, CA)

To support the National Aeronautics and Space Administration's (NASA) ongoing effort to bring supersonic commercial travel to the aerospace industry NASA Dryden, in cooperation with other government and industry organizations, conducted a flight research experiment to identify the methods, tools, and best practices for a large-scale sonic boom community human response test. The name of the project was Waveforms and Sonicboom Perception and Response (WSPR). Such tests go toward building a dataset that governing agencies like the Federal Aviation Administration and International Civil Aviation Organization will use to establish regulations for acceptable sound levels of overland sonic booms. This paper focuses on NASA's role in the project on essential elements of community response testing including recruitment, survey methods, instrumentation systems, flight planning, and operations. Objectives of the testing included exposing a residential community with sonic boom doses designed to simulate those produced by the next generation of commercial supersonic aircraft. The sonic booms were recorded with an instrumentation array that spanned the community. Human response data was collected using multiple survey methods, and was correlated to acoustic metrics from the sonic booms. The project resulted in lessons-learned and the findings of appropriate methods necessary to implement a successful large-scale test.

2:00

2pNSa4. Objective data collection and analysis for the waveform and sonic boom perception and response program. Brian Cook, Joe Salamone (Acoust. and Vib., Gulfstream Aerosp., 3 Innovation Dr., M/S R-4P, Savannah, GA 31408, brian.cook@gulfstream.com), Chris Hobbs, and Juliet Page (Wyle, Arlington, VA)

The Waveform and Sonic boom Perception and Response (WSPR) program experiment was conducted in November 2011. Low-amplitude sonic booms were created by planned NASA F-18 supersonic flights executing a unique dive maneuver. The WSPR program was designed to simultaneously collect objective sonic boom acoustic data and subjective response data from residents in the Edwards Air Force Base residential community. Sonic Boom field kits were developed for the WSPR program consisting of a digital data acquisition system with networked nodes, deployable for extended periods of time. The Sonic Boom Unattended Data Acquisition System (SBUDAS) purposely developed for sonic boom community noise testing was deployed and details of the measurement system and all aspects of the objective data collection process are described. Data analysis during testing provided vital information to the flight planners for experimental execution. This paper also explains the post-experimental analysis of the objective data achieved by creation of a measurement data archive, predictions of sonic boom exposure at subject household locations, an automated algorithm to locate sonic booms within the recorded data and computation of a variety of indoor and outdoor metrics.

2:20

2pNSa5. A comparison of survey implementation methods. Peg Krecker, Carrie Koenig (TetraTech, Madison, WI), Clifton Wilmer, and Juliet A. Page (Wyle, 200 12th St. South Ste. 900, Arlington, VA 22202, juliet.page@wyle.com)

As part of a pilot program to measure subjective response to low level sonic booms, 52 residents at Edwards Air Force Base were recruited to answer questions about their reactions to low-amplitude sonic booms. Over a two-week period, participants completed brief surveys (12 items) each time they heard a sonic boom and a short summary form at the end of each day. The study used three modes of survey administration—paper, Web, and Apple mobile device—to support analysis of the effectiveness of different approaches. Previous research on subjective response to sonic booms or other impulsive noise with similar measurement objectives has used in-person surveys, telephone surveys, or computer-assisted self-administered methods similar to a Web survey, but study designs prevented direct comparisons of the methods on the same sample of participants. We examine data quality across the three modes and the paper will present results on completion rates by survey mode, variation in completion rates over time, and differences in the timeliness of response for web and Apple participants. Qualitative interviews with a subset of participants yield further insights into each approach.

2:40

2pNSa6. Statistical analysis of community response to low amplitude sonic boom noise. Kathleen K. Hodgdon (Appl. Res. Lab., Penn State Univ., ARL North Atherton St., P.O. Box 30, State College, PA 16804-0030, kkh2@psu.edu), Juliet Page (Wyle, Arlington, VA), Trent Gaugler, Daisy Phillips, Durland Shumway, and James Rosenberger (Dept. of Stat., Penn State Univ., University Park, PA)

The Waveform and Sonicboom Perception and Response (WSPR) Program conducted a field study of subjective response to noise from multiple low-amplitude sonic booms. The team was led by Wyle and included researchers from Penn State, Tetra Tech and Gulfstream Aerospace Corp. The test exposed residents in the Edwards Air Force Base (EAFB) Housing area to two weeks of low-amplitude sonic booms while recording their responses via surveys. There were 52 participants divided across three response modes. The response instruments included Baseline Surveys, Single Event Surveys submitted each time a participant heard a boom, and Daily Surveys submitted at the end of each day. The analysis included assessments of single events and cumulative daily ratings of annoyance and categorical variables including loudness, interference, startle, vibration, and rattle. The WSPR daily annoyance data was analyzed by computing percent highly annoyed (%HA) and relating it to the cumulative noise exposure and by relating the subjective annoyance rating directly to the daily noise exposure. The WSPR design was established to cover the full range of noise exposures and annoyance factors so that sufficient data would be gathered to facilitate analyses of %HA and noise metrics. The statistical analyses examining these relationships will be presented.

3:00

2pNSa7. Relationships among near-real time and end-of-day judgments of the annoyance of sonic booms. Sanford Fidell (Fidell Assoc., Inc., 23139 Erwin St., Woodland Hills, CA 91367, sf@fidellassociates.com), Richard Horonjeff (Consultant in Acoust. and Noise Control, Boxborough, MA), and Linda Fidell (California State Univ., Emerita, Morro Bay, CA)

A recent social survey of the annoyance of low amplitude sonic booms included both prompt and delayed-response questions about the annoyance of sonic booms heard by respondents in the home over the course of two weeks. Interviews were conducted via smartphone with 49 voluntary test participants. Most of the prompt annoyance judgments were made within about a minute of notice of a sonic boom. The delayed response judgments were solicited in the evening, at a time of the respondent's choosing. Prior analyses showed that dosage-response relationships between the prevalence of high annoyance and sonic boom amplitude were well predicted by CTL analysis. The current analyses investigated how individual, within-day, prompt annoyance judgments were related to end-of-day judgments of the annoyance of sonic booms. Preliminary analyses suggest that end-of-day annoyance judgments are not simply a linear sum or averaging of the annoyances of individual sonic booms.

3:20

2pNSa8. Community response to low-amplitude sonic booms. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Sonic boom research conducted at NASA is oriented toward understanding the potential impact of sonic boom noise on communities from new low-boom supersonic aircraft designs. This research contributes to knowledge in key areas needed to support development of a new noise-based standard for supersonic aircraft certification. Partnerships with several industry, government, and academic

institutions have enabled the execution of a pilot low boom community response test to develop and assess experimental methodologies, including sonic boom data acquisition, subjective data collection, and data analysis. Areas of additional research are identified and a prioritization of issues is performed to guide design of a potential follow-on pilot test. Lessons learned from these community response tests will facilitate future community testing with actual low-boom aircraft in communities not familiar with sonic booms.

Contributed Papers

3:40

2pNSa9. The impact of including diffraction when predicting the effect of listener environment on the perceived loudness of outdoor sonic booms. Amanda B. Lind and Victor Sparrow (Grad. Prog. in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, State College, PA 16801, Amanda.Blair.Lind@gmail.com)

The human impact of sonic booms varies with listening environment. Given the incident sonic boom waveform, the specular field around a realistic geometry has been predicted via a c++ implementation of image source method (ISM) tailored to outdoor applications. This work explores the necessity of including the diffracted field when predicting time series and PLdB, both in and out of shadow zones. The impulsive nature of the excitation, and the sensitivity of the PLdB to temporal details, constrains appropriate diffraction modeling techniques to those capable of time domain accuracy. Uniform Theory of Diffraction (UTD) and Biot Tolstoy Medwin (BTM) approaches are considered. The benefits and challenges of each approach are explored, particularly with regards to scalability and bandwidth. The importance of accurately predicting diffraction in this application is evaluated through comparison with booms recorded around a building corresponding to the simulated geometry. [Work supported by the FAA/NASA/Transport Canada PARTNER Center Excellence and the Applied Research Laboratory. Experimental data courtesy of NASA.]

4:00

2pNSa10. Effect of room characteristics on perception of low-amplitude sonic booms heard indoors. Clothilde Giacomoni and Patricia Davies (Purdue Univ., 2911 Horizon Dr., Apt. 4, West Lafayette, IN 47906, cgiacomo@purdue.edu)

Supersonic flight over inhabited territories of the United States has been banned by the Federal Aviation Association. While research has been conducted to determine the effects of sonic booms on the general population when heard outdoors, little work has been done on people's perception of sonic booms as heard indoors. A sound's waveform will change in its transmission from outdoors to indoors due to several factors, one of which is the indoor acoustic environment. This can be changed using different room sizes, shapes or which materials are covering each surface (wall, ceiling, or floor). A subjective test, designed to determine which of these room characteristics has an effect on people's ratings of annoyance, has been completed. It was found that smaller rooms and square rooms are rated as more annoying than larger rooms or rooms with a corridor-like or rectangular shape, and that rooms with lower reverberation times were rated as less annoying than rooms with higher reverberation times.

4:20–5:20 Panel Discussion

TUESDAY AFTERNOON, 4 JUNE 2013 — 511CF, 12:55 P.M. TO 5:00 P.M.

Session 2pNSb

Noise and Architectural Acoustics: Soundscape and its Application

Brigitte Schulte-Fortkamp, Cochair
Technische Universität Berlin, Einsteinufer 25, Berlin 10178, Germany

Bennett M. Brooks, Cochair
Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066

Chair's Introduction—12:55

Invited Papers

1:00

2pNSb1. Soundscape workshop report: Perception Lexicon. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com) and Brigitte Schulte-Fortkamp (Tech. Univ. Berlin, Berlin, Germany)

A workshop was held at the 164th Meeting of the ASA in Kansas City on methodology standardization for the advancement of the developing field of soundscape measurement, analysis, and design. The workshop focused on the terminology lexicon used for soundscape subject interviews. Interviews of local experts, residents, and other users and inhabitants of the sonic environment can yield insights into both personal reactions and universal observations. The specific terminology used in this process may significantly affect the outcomes. As the success of a new research or development project can depend on the lessons learned from previous projects, the standardization of interview techniques becomes increasingly important. Workshop participants were invited to develop a standardized lexicon of descriptors for field use in interview questionnaires. After a review of soundscape objectives and procedures, the participants reviewed key issues and assessed available lexicon term types. The group then proposed, developed, and prioritized terms which could

describe a soundscape as it is contextually perceived by an interview subject. Two key conditions were considered for each proposed lexicon term: would the term be commonly understood, and would the term describe a reaction to the soundscape of sufficiently notable intensity. The results of the workshop are presented.

1:20

2pNSb2. Characterizing the soundscape of tranquil urban spaces. Bert De Coensel, Michiel Boes, Damiano Oldoni, and Dick Botteldooren (Dept. of Information Technol., Ghent Univ., St.-Pietersnieuwstraat 41, Ghent B-9000, Belgium, bert.decoensel@intec.ugent.be)

Tranquil spaces provide restorative environments for urban residents and visitors and are therefore essential for health and quality of life. Tranquil spaces may be characterized through a combination of acoustical criteria, such as relatively low (percentile) sound levels and the relative absence of non-fitting sounds, and non-acoustical criteria, such as the presence of natural elements within the visual scene. Public urban parks and courtyards as well as private urban backyards are typically considered to be the most tranquil spots within a city. Current state-of-the-art in distributed measurement technology allows for long-term sound monitoring at these places. In this paper, the soundscape at a number of urban parks and backyards in the cities of Ghent and Antwerp is investigated through a detailed analysis of sound measurements performed over an extended period of time. An analysis of percentile sound levels, noise events and indicators for temporal and spectral structure is presented, and novel computational methods are applied to estimate the relative occurrence of sounds arising from various sources.

1:40

2pNSb3. What will be the influence of e-mobility on soundscape? Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath-Kohlscheid 52134, Germany, klaus.genuit@head-acoustics.de)

The increasing electrification of the powertrain after 125 years of continuous development of the internal combustion engine will not only lead to a sound pressure level reduction of vehicle exterior noises but to a complete change of sound quality. With this expected development road traffic noise affected persons hope for quiet cities and a better quality of life. The creation and successful preservation of quiet zones in cities and to avoid harmful effects of noise exposure are special focuses in European noise policy. However, different surveys have shown the increased risk of accidents for pedestrians and cyclists with respect to collisions with quiet vehicles, which caused a lively discussion about acoustical warning systems for the prevention of crashes. But it is obvious, that major conflicts between quietness and safety arise. Consequently, to address this issue, on the one hand, sustainable concepts must be developed for the successful avoidance of accidents and on the other hand the general traffic noise must be minimized. The sound of electric vehicles will influence in a significant way our soundscape at places and cities in the future. This is a special challenge for psychoacoustics to provide helpful contribution besides the A-weighted sound pressure level.

2:00

2pNSb4. Plan for research to better understand visitor response to the soundscape in national parks. James H. Boyle (Univ. HS, Univ. of Illinois, 3805 Deerfield Dr., Champaign, IL 61822, jhhboyle@gmail.com) and Paul D. Schomer (Schomer and Assoc., Inc., Champaign, IL)

Since 91% of national park visitors come to a park, at least in part, for the natural sounds, analysis of the perceptions visitors have of sound in national parks is extremely significant. In the summer of 2011, several pilot surveys were tested at Rocky Mountain National Park in Colorado for their abilities to provide an accurate reflection of visitors' perceptions of the soundscape. The 2011 surveys were modified in various ways to create the current 2013 survey, which uses "pleasantness" of park sound as the main visitor response metric. The testing plan for the 2013 survey is non-labor intensive for respondents and researchers; rather than closely monitoring activities of respondents, and having them closely monitor the sounds they hear, this survey requires respondents to complete a short questionnaire every 30 to 60 min and an end-of-hike survey. Much like how day-night sound level (DNL) has traditionally been predicted for a cluster of houses around airports or highways, global sound measurements conducted along the trail will be correlated with visitor survey results in order to develop the means to predict how an average park visitor will respond to the soundscape.

2:20

2pNSb5. When do we judge sounds? Relevant everyday situations for the estimation of ecological validity of indoor soundscape experiments. Jochen Steffens (FH Duesseldorf, Josef-Gockeln-Strasse 9, Duesseldorf 40476, Germany, jochen.steffens@fh-duesseldorf.de)

This paper introduces a model of indoor soundscape evaluation, which is based on the results of listening experiments on household appliances. The model reveals, among other things, permanently changing action and attention processes and learning effects which occur in everyday settings. However, what are the relevant situations which we want to reconstruct in our experiments? When do people perceive and evaluate sounds in their everyday life like they do under test conditions? This theoretical knowledge is essential for the estimation of ecological validity of soundscape experiments in general. Hence, this contribution deals with approaches to determine meaningful real-life situations. The interviews held in the course of the studies show that many subjects construct extremely critical context situations while they are assessing sounds. These contexts are often not representative for these peoples' reality. Many participants, in fact, state that they no longer consciously perceive and evaluate the sounds of the appliances in their personal life. Decreasing attention over time can lead to a great influence of the first impression on subsequent perceptions and cognitions (primacy effect). Furthermore, the effect of memory processes on retrospective sound evaluations will be discussed.

2:40

2pNSb6. The practical SoundLab for architects: Sound parameters as a design tool. Juergen Bauer (Dept. of Architecture, Waterford Inst. of Technol., Granary, Hanover St., Waterford, Co Waterford, Ireland, jbauer@wit.ie)

The experience gained from exploring acoustics with architectural students in Waterford Institute of Technology (Ireland) suggests that a studio situation can be exploited and used as a case study for young designers to become familiar with basic sound phenomena. From phase 1 of previous research, the student "audiovisual design workshop" concluded that class rooms that perform poorly

acoustically can help to strengthen the students' understanding of the sonic environment, in a way that the students were exposed day after day to high reverberation, low speech intelligibility, and the "Lombard" effect in their own studio. However, becoming aware of poor acoustics necessitates a creative follow-up: how to design and create good acoustics, or how to develop the environment as a "soundscape"? Consequently, phase 2 of our audiovisual design workshop introduced the practical sound lab: our students were required to design and to build acoustic panels for their own studio environment. These prototypes would be analyzed as part of an Acoustic Laboratory, with specialist advise from an Acoustic Consultant. The paper summarizes the findings from the audiovisual design laboratory. It also reflects on how to compile international case studies of buildings with good acoustics that can be used as references for designing architects.

3:00–3:20 Break

Contributed Papers

3:20

2pNSb7. Evaluation of noise climate in a campus environment using geospatial technology. Rajasekar Elangovan (Ctr. for Excellence and Futuristic Developments, Larsen & Toubro Construction, L&T ECC, Manapakkam, Chennai, TamilNadu 600089, India, erajas@gmail.com), Pari YadavaMudaliar (GeoSpatial Technol., EDRC, Larsen & Toubro Construction, Chennai, India), Venkateswaran Rajan, Khaleel Elur Rahaman Anser Basha (Ctr. for Excellence and Futuristic Developments, Larsen & Toubro Construction, Chennai, India), and Ravi Muttavarapu (GeoSpatial Technol., EDRC, Larsen & Toubro Construction, Chennai, India)

This study focuses on experimental evaluation of outdoor noise climate in an office campus, which is spread across 27 acres and investigation of its impact on the adjoining office buildings. For this purpose, a base map of the campus was prepared from the recent high spatial resolution satellite image using ESRI Arc/GIS and the same was used for planning appropriate locations for capturing noise levels and its spectral characteristics. Mapping grade Trimble GPS was used to stake out the measurement locations and noise levels were recorded using 01 dB solo and Norsonic type 118 sound level meters. Captured noise levels were plotted in GIS environment and appropriate spatial interpolation was carried out in order to give a continuous graphical representation of sound levels. A wide variation of noise levels was observed across the campus with LAeq ranging from 50 dB(A) to 80 dB(A). Low frequency noise was found to be predominant compared to mid and high frequency noise. Major noise sources and the propagation pattern were determined through the mapping. The data thus obtained is used to investigate the noise impact on the office buildings. The measurements were also accompanied by a subjective evaluation of the outdoor noise annoyance.

3:40

2pNSb8. On the noise generated by park visitors along hiking trails. Lucas Newman-Johnson (Univ. HS, Univ. of Illinois, 808 w. White St., Champaign, IL 61821, newmanj1@uni.illinois.edu) and Paul D. Schomer (Schomer and Assoc., Champaign, IL)

In assessing the sound environment of a park, visitor noise along trails may be important because of its affect upon: (1) the un-altered park environment, (2) wildlife, and (3) other park visitors. Just as noise in residential communities has been correlated with population density we set out to see if the noise level along trails would correlate with visitor density. In the summer of 2011, measurements were made using two sites, one on each of two trails. These measurements included one second Leq measurements at an array of four trailside microphones, and recording of the number of park visitors entering the measurement zone during each minute. Examination of the data revealed little correlation between a 5 min measurement of the Leq and density of trail visitors. The problem was that the smallest time increment in which we could accurately portray the number of park visitors was about 5 min, using the one minute totals, and in a 5 min period visitor noise would rarely equal or exceed the measurement ambient. Thus, no relation between visitor density and trailside noise could be developed. Additional analysis was done with data to better understand the limiting factors to the measurement.

4:00

2pNSb9. Investigating soundscape affordances through activity appropriateness. Frederik L. Nielbo (Ctr. for Semiotics, Aarhus Univ., Jens Chr. Skous Vej 2, Bldg. 1485, Rm. 525, Aarhus DK-2200, Denmark, norfln@hum.au.dk), Daniel Steele (Ctr. for Interdisciplinary Res. in Music Media and Technol., McGill Univ., Montréal, QC, Canada), and Catherine Guastavino (School of Information Studies, McGill Univ., Montréal, QC, Canada)

Central to the concept of soundscape is the understanding of the acoustic environment in context. Previous research indicates that people understand soundscapes through their potential for activities. One way to look at activities is through the concept of affordances—defined as the actionable properties of an object. In this study, the object is a location and time in the city. Fifteen participants listened to stereo recordings of eight outdoor sites in Paris and Montreal. In each trial, they evaluated on a continuous scale how appropriate the soundscapes were for a given activity. Four activities were considered and presented in random order: studying for an exam, meeting up with a friend, riding a bike and relaxing. Participants justified their ratings in free-format comments. A 8(Soundscapes) x 4(Activities) factorial ANOVAs revealed significant effects of Soundscape and Activities and Soundscape*Activities on appropriateness ratings. Certain soundscapes were found to accommodate specific activities only while others were found to potentially accommodate all activities or none (prominent mechanical/traffic noise). We also analyzed comments to further understand how participants envision utilizing the soundscape/environment and attribute meanings to the various sounds present.

4:20–5:00 Panel Discussion

Session 2pPA

Physical Acoustics: Material Characterization

Noureddine Atalla, Cochair
GAUS Mech. Eng., Univ. of Sherbrooke, Sherbrooke, QC J1K 2R1, Canada

Nico Declercq, Cochair
Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France

Contributed Papers

1:00

2pPA1. Probing acoustics of liquid foams by optical diffusive wave spectroscopy. Benjamin Dollet, Marion Erpelding, Reine-Marie Guillermic, Caitlin Sample, Juliette Pierre, Arnaud Saint-Jalmes, and Jérôme Crassous (Institut de Physique de Rennes, CNRS/Université Rennes 1, Campus Beaulieu, Bâtiment 11A, Rennes 35042, France, benjamin.dollet@univ-rennes1.fr)

Sound propagation through liquid foams, which are dispersions of gas bubbles in a continuous liquid phase, is not well known yet. To characterize foam acoustics at the local scale, we have studied the effect of an external acoustic wave on bubble displacements inside an aqueous foam. We quantify these displacements by using a technique based on optical diffusive wave spectroscopy, that we specially developed to resolve tiny deformations in materials. Bubble displacements induce a modulation on the photon correlation curve. Measurements for various sound frequencies and amplitudes are interpreted using a light diffusion model. It allows us to unravel a nontrivial acoustic displacement profile inside the foam; in particular, we find that the acoustic wave creates a localized shear in the vicinity of the solid walls holding the foam, as a consequence of inertial contributions. This study of how bubbles "dance" inside a foam as a response to sound turns out to provide new insights on foam acoustics and sound transmission into a foam, foam deformation at high frequencies, and analysis of light scattering data in samples undergoing nonhomogeneous deformations.

1:20

2pPA2. Reduction of ultrasonic multiple scattering applied to flaw detection with array probes in polycrystalline materials. Sharfine Shahjahan (Site des Renardieres, EDF R&D, Moret-sur-Loing, France), Alexandre Aubry (Institut Langevin - ESPCI ParisTech & CNRS, Universite Paris Diderot, 1 rue Jussieu, Paris 75005, France), Fabienne Rupin, Bertrand Chassignole (Site des Renardieres, EDF R&D, Moret-sur-Loing, France), and Arnaud Derode (Institut Langevin - ESPCI ParisTech & CNRS, Universite Paris Diderot, Paris, France, arnaud.derode@espci.fr)

Flaw detection using ultrasonic evaluation of coarse-grain steels is perturbed by a high structural noise due to scattering. This leads to a decrease of the detection capabilities, particularly at high frequencies and large depths for which multiple scattering dominates. Recent academic studies have shown that the contribution of multiple scattering could be dramatically reduced. These results were obtained on a model random medium made of parallel steel rods immersed in water. The ability to detect a target could be significantly increased using a specific filtering method, based on the full matrix capture (F.M.C.) combined with a smart post-treatment based on random matrix theory, in supplement with the DORT method (i.e., decomposition of the time-reversal operator). Here, the same technique to separate simple and multiple scattering contributions is now applied to a real material. Experimental results were obtained on a nickel-based alloy (Inconel600®) with a thermically induced coarse grain structure and manufactured flaws (side drilled holes) at different depths. The experimental set-up used a multi-element ultrasonic array. Results are presented and compared to other detection techniques, at various depths and frequencies. Despite a dominant multiple scattering noise, a significant improvement of the detection performances is observed.

1:40

2pPA3. Negative and density-near-zero acoustic metamaterials based on quasi-two-dimensional phononic crystals. Victor Manuel Garcia-Chocano, Rogelio Graciá-Salgado, Daniel Torrent, and José Sánchez-Dehesa (Dept. of Electron. Eng., Polytechnic Univ. of Valencia, C/Camino de Vera S/N, Departamento de Ingeniería Electrónica, Valencia 46022, Spain, vicgarch@upvnet.upv.es)

A phononic crystal consisting of an array of cylindrical boreholes in a two dimensional waveguide has been fabricated and characterized. Reflectance and transmittance have been measured in a slab with seven layers of scatterers. The acoustic bands as well as the effective parameters have been extracted from experimental data, showing that the proposed structure behaves as an acoustic metamaterial with negative bulk modulus. In addition it is shown that the inclusion of anisotropic effects through angular-dependent structures inside the boreholes leads to metamaterials with double negative parameters. This feature allows the observation of interesting phenomena such as an acoustic tunneling, which has been predicted through full wave simulations in a density-near-zero metamaterial.

2:00

2pPA4. Impulse response of a medium in a three layered media. Ambika Bhatta (Elec. Eng., Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu), Charles Thompson, Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA), and Vineet Mehta (MIT, Lincoln Lab., Lexington, MA)

This paper investigates the point source response of a layered medium separating two semi-infinite impedance medium. A detailed numerical method to evaluate Laplace transform of the reflection coefficient for each boundary is presented. A description of the branch integral dominantly contributing to the reflection coefficient in Laplace domain for single and multiple reflections is given. The analysis utilizes the image theory to evaluate the order of the reflection and complex pressure amplitude at each image source location. The impulse response is also evaluated numerically from the Green's function obtained from the reflection coefficient and complex pressure due to image sources. Fresnel's plane wave reflection coefficient validates the solution when the height of layer containing the source and the observation position is relatively large. The obtained expression for reflection coefficient for large wave number and observation position is verified numerically.

2:20

2pPA5. Experimental demonstration of a three-dimensional acoustic cloak based on a cancelation effect. Jose Sanchez-Dehesa, Victor M. García-Chocano, Alfonso Climente, Francisco Cervera (Dept. of Electron. Eng., Universitat Politecnica de Valencia, Camino de vera s.n., Edificio 7F, Valencia, Valencia ES-46022, Spain, jsdehesa@upvnet.upv.es), Vicente Gomez-Lozano (Centro de Tecnologías Físicas, Universitat Politecnica de Valencia, Valencia, Spain), Lorenzo Sanchis (UMDO (Unidad Asociada al CSIC-IMM), Universidad de Valencia, Valencia, Spain), Rafael LLopis-Pontiveros, and Martínez-Pastor Juan (UMDO (Unidad Asociada al CSIC-IMM), Universidad de Valencia, Valencia, Spain)

A three-dimensional acoustic cloak has been designed, fabricated, and experimentally characterized. The cloak is made of 60 concentric tori acoustically rigid that surround a sphere of radius 4 cm, which is considered as the cloaked object. The major radii and positions of the tori along the symmetry axis are determined by a cancelation condition; i.e., the scattering cross section by the sphere and the tori must be zero. An optimization algorithm that combines a genetic algorithm and simulated annealing is employed to satisfy such a condition. The operational frequency of the one-directional cloak is 8.67 kHz with a bandwidth of about 120 Hz.

2:40

2pPA6. Ice thickness determination with audio sound. Anurupa Shaw (Georgia Tech Lorraine, 133 Rue Du Fort de Queuleu, D107, Residence Lafayette, Metz 57070, France, shaw.anurupa@gmail.com) and Nico F. Declercq (Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng., Lab. for Ultrasonic Destructive Evaluation “LUNE”, Georgia Tech-CNRS UMI2958, Georgia Tech Lorraine, Metz-Technopole, France)

The objective of this study is to determine the thickness of Ice by analyzing the sound spectrum generated by dispersion of lamb waves propagating in ice. In winters when the lakes and rivers freeze, it is important to know the thickness of the ice layer before one intends to walk on it. When we throw a stone on the ice layer, we can hear a fluting noise. This is recorded for different thicknesses of ice and the sound spectrum is compared with the results simulated using a parameterized model. This model is created using a combination of plane waves for different incident angles and frequencies to generate dispersion curves for different thicknesses of ice. The frequencies of the reflected sound are then compared with the frequencies of musical notes in order to assign different musical notes to different thicknesses of ice.

3:00

2pPA7. Shaving foam: A complex system for acoustic wave propagation. Juliette Pierre (Laboratoire Matière et Systèmes Complexes, UMR CNRS 7057 - Univ. Paris Diderot, Université Rennes 1 - CNRS UMR 6251 263 av. Général Leclerc, Rennes 35042 Cedex, France, juliet.pierre@gmail.com), Valentin Leroy (Laboratoire Matière et Systèmes Complexes, UMR CNRS 7057 - Univ. Paris Diderot, Paris, France), Arnaud Saint-Jalmes, Benjamin Dollet, Imen Ben Salem, Jérôme Crassous, Reine-Marie Guillermic (Institut de Physique de Rennes, UMR CNRS 6251 - University Rennes 1, Rennes, France), Wiebke Drenckhan (Laboratoire de Physique du Solide, UMR CNRS 8502 - Univ. Paris-Sud, Orsay, France), and Florence Elias (Laboratoire Matière et Systèmes Complexes, UMR CNRS 7057 - Univ. Paris Diderot, Paris, France)

While liquid foams have applications in an increasing number of industrial areas (food, cosmetic, or petroleum industry), it remains difficult to non-invasively probe their structure and/or composition. Since the propagation of acoustic waves is very sensitive to parameters such that the liquid fraction, the bubble size distribution, or even the nature of the liquid phase, acoustic spectroscopy could be a very powerful tool to determine the structure and/or composition of liquid foams. In this context, we present an investigation of the acoustic properties of a useful and common foam, often considered as a model system: shaving foam. Phase velocity and attenuation of acoustic waves in a commercial shaving foam (Gillette) were measured over a broad frequency range (0.5 to 600 kHz), using four different experimental setups: an impedance tube (0.5–6 kHz), an acousto-optic setup based on diffusive wave spectroscopy (1–10 kHz), and two transmission setups with narrow-band (40 kHz) and broad-band (60–600 kHz) transducers. We present the results and discuss the advantages and shortcomings of each setup in terms of a potential spectroscopy technique.

3:20

2pPA8. Lamb modes and acoustic microscopy for the characterization of bonded structures. Alaoui Ismaili Naima, De Mello Da Silva Camilla (Institut Electronique du Sud (IES), UMR CNRS 5214, University of Montpellier 2, Place Eugène Bataillon, Montpellier 34095, France, naima.alaoui-ismaili@insa-lyon.fr), Ech-Cherif El-Kettani Mounsif (Laboratoire Ondes et Milieux Complexes (LOMC), UMR CNRS 6294, Univ. of Le Havre, Le Havre, France), Despaux Gilles (Institut Electronique du Sud (IES), UMR CNRS 5214, Univ. of Montpellier 2, Montpellier, France), Rousseau Martine (Institut Jean Le Rond d’Alembert UMR CNRS 7190, Univ. of Pierre et Marie Curie, Paris, France), and Izbicki Jean-Louis (Laboratoire Ondes et Milieux Complexes (LOMC), UMR CNRS 6294, Univ. of Le Havre, Le Havre, France)

This paper is a contribution to the evaluation of the bonding by means of guided waves, when the quality of the glue is degraded by addition of grease. Experimental dispersion curves are compared to the theoretical ones obtained either from a perfect welded tri-layer model or from a second model using delaminated conditions. It is shown that the second model is more convenient when the quality of the glue is highly degraded. On another hand, the parameters of Jones are extracted from experimental data by inverse problem. It is shown that the values of these parameters strongly diminish and the variation is of logarithmic type when the quality of the glue is progressively degraded. Acoustic microscopy is also performed, and it provides images of the inner structure giving a qualitative explanation of the values of the parameters of Jones.

3:40

2pPA9. Experimental measurements of the coherent field resulting from the interaction of an ultrasonic shock wave with a multiple scattering medium. Nicolas Viard, Bruno Gianmarinaro, Arnaud Derode, and Christophe Barrière (Institut Langevin, 1, rue Jussieu, Paris 75005, France, nicolas.viard@espci.fr)

Whereas multiple scattering and shock wave formation are known to be antagonistic phenomena, this work concentrates on the interaction of an ultrasonic shock wave with a random multiple scattering medium. The shock wave is generated by long distance propagation of a short pulse (four periods at a 3.5 MHz central frequency) in water before it encounters the scattering medium (a slab-shaped random set of parallel metallic rods). Transmitted waves are recorded over hundreds of positions along the lateral dimension of the slab to estimate the ensemble-averaged transmitted field $\langle \Phi(t) \rangle$, also known as the coherent wave. Experiments are repeated for different thicknesses L of the slab and different emission amplitudes. The elastic mean free path l_e (i.e., the typical distance for the decreasing of the coherent intensity $|\langle \Phi(t) \rangle|^2$ due to scattering) is determined as well as the harmonic rate of the averaged transmitted wave. Experimental results are discussed and compared to the linear case.

4:00

2pPA10. Frequency-resolved measurements of the diffusion constant for ultrasonic waves in resonant multiple scattering media. Nicolas Viard and Arnaud Derode (Institut Langevin, 1, rue Jussieu, Paris 75005, France, nicolas.viard@espci.fr)

Experimental measurements of the diffusion constant for ultrasonic waves (around 3 MHz) propagating in water through a random set of scatterers (parallel metallic rods arranged as a slab) are presented. The slab thickness is around 10 times the transport mean free path. Transmitted waves are recorded over hundreds of emitting/receiving positions in order to estimate the ensemble-averaged transmitted intensity $\langle I(x,t) \rangle$. Focused beamforming is performed on both faces of the sample in order to mimic a set of point-like sources and receivers. In theory, under the diffusion approximation, the ratio of the off-axis intensity $\langle I(x,t) \rangle$ to the on-axis intensity $\langle I(0,t) \rangle$ shows a simple gaussian dependence on the lateral dimension x, independently from absorption or boundary conditions. This yields a simple way to estimate the diffusion constant D and therefore characterize the scattering medium. Based on that method, broadband as well as frequency-resolved measurements of the diffusion constant are presented in controllable model media, such as these forests of steel rods. Experimental results and difficulties for measuring a reliable value for D on a real sample are discussed.

4:20

2pPA11. Wavetrain-long waves interaction in a non-homogeneous, non-stationary medium. Alexander Voronovich (Physical Sci. Div., NOAA/ESRL, 325 Broadway, Boulder, CO 80305, alexander.voronovich@noaa.gov)

A weakly nonlinear wavetrain of the acoustic waves apart from higher harmonics generates also a low-frequency, large-scale motion which in turn affects both the wavetrain itself and other wavetrains, thus leading to an effective wavetrains interaction. A closed set of equations describing interaction of a wavetrain with inhomogeneous medium including a recoil effect on the background motion is derived using Hamiltonian form of the equations of motion (viscosity is neglected). Corresponding energy and momentum conservation equations are derived which account for the energy and pseudo-momentum exchange between the wavetrain and a background, large-scale motion (the latter includes the nonlinearly generated component due to the wavetrains). A large-scale motion due to a single wavetrain propagating in an initially homogeneous medium is calculated. Far from the wavetrain nonlinearly generated large-scale current is confined by an expanding sphere with the velocity aligned with the wavetrain propagation direction; the velocity appears to be constant in the planes perpendicular to this direction. It is demonstrated that interaction between acoustic wavetrains lead to their effective repulsion which in a complex way depends on mutual location and velocities of the wavetrains. Generalization of the approach to the case of integral gravity waves will be also briefly mentioned.

4:40

2pPA12. Gaussian closure technique for Bouc's hysteretic model under white noise excitation. Holger Waubke (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, holger.waubke@oeaw.ac.at)

Bouc developed a hysteretic model for materials like rubber under dynamic excitation. The response of a hysteretic system under white noise excitation is normally estimated by means of the statistical linearization or a related method. Disadvantages of this method are the assumption of a Gaussian nature of the random distributions, the high computational efforts caused by the iterations needed, and the instability of the iteration in certain parameter regions. Using the assumption of Gaussian random distributions, the Gaussian closure technique can be applied. Analytic solutions of the integrals occurring in this approximation were found and are presented. These solution allow for an explicit time step procedure for the random moments in the transient case. For the stationary case, a fast and stable iteration about a set of non linear equations is needed. Both procedures allow to calculate the moments in a fast manner and allow to solve problems with more than one degree of freedom with limited computational efforts.

5:00

2pPA13. Numerical model of a causal and fractional all-frequency wave equation for lossy media. Margaret Wismer (Bloomsburg, 400 E Second St., Bloomsburg, PA 17815, mwismer@bloomu.edu)

A numerical algorithm, to simulate a lossy acoustic wave equation with fractional time derivative terms, is presented. The inclusion of fractional derivatives yields a causal acoustic wave equation which can model and predict power law attenuation for which the level of absorption is proportional to frequency raised to a non-integer power. The fractional Zener wave equation is derived from the fractional Zener stress-strain constitutive relation and contains two absorption terms. One term which includes the Laplacian, similar to the traditional wave equation in viscous fluids, has a coefficient, which is proportional to a relaxation time. The other term is a fractional time derivative, higher than second order, and is proportional to the creep or retardation time. Both relaxation and retardation parameters will affect the frequency behavior of the attenuation. The inclusion of two terms enables the modeling of power-law attenuation in all frequency ranges. It results in more stable numerical simulations as both the overall mass and stiffness matrices of the finite element algorithm are changed according to the level of absorption. Results show the animation of diffraction patterns of focused and planar acoustic waves by inclusions with different values for the two different parameters, retardation time and relaxation time.

TUESDAY AFTERNOON, 4 JUNE 2013 514ABC, 12:55 P.M. TO 4:20 P.M.

Session 2pPPa

Psychological and Physiological Acoustics: Celebrating a "Long" Career: Explorations of Auditory Physiology and Psychoacoustics

Jungmee Lee, Cochair
Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208

Elizabeth A. Strickland, Cochair
Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907

Chair's Introduction—12:55

Invited Papers

1:00

2pPPa1. Glenis Long's contribution to animal psychoacoustics. Richard Fay (MBL, 179 Woods Hole Rd., Falmouth, MA 02540, rfay@luc.edu)

Glenis and I were psychology graduate students together in 1970–1971. She was interested in auditory psychophysics, but soon developed an interest in comparative psychophysics which she pursued in a post-doc and a faculty position in Germany, where she obtained a behavioral audiogram for the horseshoe bat and later studied masking in the same species. This audiogram was one of the very first psychophysical investigations of hearing in any bat and the very first for the horseshoe bat. It confirmed a rather complex frequency response function with sensitivity peaks at about 20 kHz and 60 kHz, and a very sharp peak at about 80 kHz. The masked thresholds indicated rather normal critical masking ratios (CR) except in a narrow region at about 80 kHz where the CRs are up to 15 dB lower than expected, suggesting a critical band of about 300 Hz at 80 kHz that is presumably used in Doppler-shift compensation. In 1981–1983, Glenis went on to investigate frequency and rate modulation discrimination in the chinchilla, for the first time, and to study tone-on-tone masking in the chinchilla. Since these early works, Glenis has maintained her comparative interests with studies on birds, kangaroo rats, frogs, and fleas on cats.

1:20

2pPPa2. Understanding subtle changes in auditory function with otoacoustic emissions. Linda J. Hood (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, MCE South, 8310, Nashville, TN 37232, linda.j.hood@vanderbilt.edu), Shanda Brashears (A. I. du-Pont Hospital for Children, Wilmington, DE), Glenis Long (The Grad. Ctr., City Univ. of New York, New York, NY), and Carrick Talmadge (Univ. of Mississippi, Oxford, MS)

Otoacoustic emissions (OAEs), a sensitive measure of cochlear processing, may be altered by subtle changes in auditory function that are not measurable by usual clinical methods. We studied auditory function in carriers of genetic mutations related to recessive hereditary hearing loss where we hypothesized that carrying a single mutation copy may compromise auditory function and be reflected when sensitive assays are used. Parents and siblings who were confirmed carriers of recessive mutations associated with GJB2 (connexin 26) or Usher syndrome, as well as obligate carriers of mutations related to recessive hearing loss of unknown genetic origin, were compared to age and gender matched control subjects. All participants had normal pure tone and middle ear responses. Metrics included transient OAEs, distortion product OAEs, and OAE fine structure. DPOAE fine structure was specifically explored based on the ability to isolate components that could be differentially affected by genetic mutations. The results support the hypothesis that carriers of gene mutations related to hearing loss display subtle auditory abnormalities that can be observed in OAEs. These findings will be related to other studies of subtle changes in OAEs in disorders affecting auditory function. [Work supported by NIH NIDCD R01-DC03679 and VU Development Funds.]

1:40

2pPPa3. Brain activity and perception of gaze-modulated tinnitus. Pim Van Dijk, Margriet J. Van Gendt, Kris Boyen, Emile De Kleine (Otorhinolaryngology, Univ. Med. Ctr. Groningen, P.O. Box 30001, Groningen 9700 RB, Netherlands, p.van.dijk@umcg.nl), and Dave R. Langers (NIHR Nottingham Hearing Biomed. Res. Unit, Univ. of Nottingham, Nottingham, United Kingdom)

We studied the correspondence between brain activity and tinnitus in subjects with gaze-modulated tinnitus. These subjects are able to modulate their tinnitus by peripheral gaze of the eyes. This is a rare form of tinnitus that primarily occurs in subjects that underwent acoustic schwannoma surgery. The voluntary control of the tinnitus allows for a controlled experiment to study the perceptual characteristics of tinnitus and the corresponding brain activity as assessed by functional MRI. Eighteen subjects with gaze-modulated tinnitus participated in the study. The effect of gaze on tinnitus was diverse. Most commonly, the largest effect on tinnitus was observed for horizontal gaze toward the surgery side. When the loudness of tinnitus changed, it was usually an increase. In addition, changes of the pitch and apparent bandwidth of the tinnitus were reported. Peripheral gaze corresponded to increase of activity in the cochlear nucleus and inferior colliculus, a decrease of activity in the medial geniculate body, and a reduction of deactivation in the auditory cortex. The inhibition of the medial geniculate body in the thalamus contrasts with the excitation that is typically observed in response to external sound stimuli. It suggests that abnormal functioning of the thalamus plays a role in tinnitus.

2:00

2pPPa4. Improving the usability of the distortion product otoacoustic emisssions-sweep method: An alternative artifact rejection and noise-floor estimation. Manfred Mauermann (Med. Phys., Univ. of Oldenburg, Carl-von-Ossietzky-Straße 9-11, Oldenburg 26111, Germany, manfred.mauermann@uni-oldenburg.de)

The DPOAE-sweep method combined with a specific least-squares-fit (LSF) analysis provides a fast method to measure and analyze distortion product otoacoustic emissions (DPOAE) with a high frequency resolution. In studies using this technique the noise reduction, artifact rejection and noise estimation are typically realized in a "classical" way, i.e., as temporal averaging with preceding elimination of time epochs exceeding an artifact threshold level and a noise estimation based upon the analysis of the difference of two buffers of epoch averages. However, the choice of an artifact threshold is arbitrary to some extent and different choices can lead to differences in the estimation of DPOAE levels. The two-buffer technique is ambiguous as well since a different grouping of the epochs into the buffers leads to rather different noise estimates. Therefore, the present study proposes an alternative approach, which provides unique noise estimators for a given set of data, a robust artifact rejection without the need to select an arbitrary rejection threshold, as well as estimators including confidence intervals for the DPOAE levels and phases. The "classical" and suggested estimators for DPOAE levels, DPOAE phases, and noise levels are compared based on Monte Carlo simulations and real measured data sets.

2:20–2:40 Break

2:40

2pPPa5. The relevance of otoacoustic emission fine structure. Sumitrajit Dhar (Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, s-dhar@northwestern.edu)

The quasi-periodic fluctuations in otoacoustic emission (OAE) amplitude and phase, known as fine structure, provide critical insight into the mechanisms of OAE generation and propagation. Whether fine structure is relevant from the perspective of hearing health remains an open question. Starting with work done under the tutelage of Professor Long, our group has conducted a decade of work investigating the clinical relevance of fine structure in general, and distortion product (DP) OAE components in particular. These studies cover a significant portion of the human lifespan (0 to ~80 yr) and the collective results provide insights into the avenues of application that hold promise. Drawing from the results of several studies conducted in our laboratory and those of collaborators and colleagues we will make the case for focused clinical application of the DPOAE components that constitute fine structure. These constituent elements appear to be ideal for monitoring of cochlear status and expose physiological changes prior to their evidence in other measures of

auditory function. However, harnessing the information encapsulated in fine structure directly is not without challenge and the examination of the constituent elements of fine structure may prove more profitable for clinical applications.

3:00

2pPPa6. Noise-induced hearing loss and strategies for its prevention in the New Zealand population: The Kiwi connection. Peter R. Thorne, Gavin Coad, Ravi Reddy, and David Welch (Audiology, Univ. of Auckland, School of Population Health, Private Bag 92019, Auckland 1142, New Zealand, pr.thorne@auckland.ac.nz)

Celebrating Glenis Long's outstanding contribution to auditory physiology and psychoacoustics, this presentation covers the New Zealand connection and particularly her involvement in our research program into the monitoring and prevention of noise-induced hearing loss (NIHL) in the country. A large multidisciplinary project is being undertaken to investigate the nature of occupational hearing loss in New Zealand and establish a national approach to prevent NIHL. This includes estimates of NIHL prevalence and the design and evaluation of education and prevention programs to reduce the impact of noise. Using a modeling approach, we have estimated that NIHL contributes to 17–25% of cases of hearing impairment in New Zealand and is therefore a significant modifiable risk factor. A key component of our project is monitoring of noise injury and we have also studied distortion product otoacoustic emissions (DPOAE) as a measure of early injury. To assess DPOAEs as a measure of injury, we recorded them using swept pure tones and extracted DPOAE components using a least-squares fit approach in noise and non-noise exposed individuals. OAE findings were compared with measures of auditory function. We found that the generator component correlated more strongly with auditory threshold and thus may be a better physiological index of noise injury. Overall, these findings have informed a national strategy involving government and community agencies to mitigate the effects of noise.

3:20

2pPPa7. Demonstration of distributed distortion-product otoacoustic emission components using onset-latency techniques. Brenda L. Lonsbury-Martin (Otolaryngology, Loma Linda Univ. Med. Ctr., Res. Service (151), 11201 Benton St., Loma Linda, CA 92357, blonsbury-martin@llu.edu), Glen K. Martin, and Bart B. Stagner (Res. Service, VA Loma Linda Healthcare System, Loma Linda, CA)

An oversimplified notion is that DPOAEs originate from a restricted region on the basilar membrane (BM). In actuality, DPOAEs are a distributed process involving the interaction of many wavelets, most likely generated over a broad region at, and basal to the overlap place of the primary tones. In the present study, DPOAEs were measured in rabbits as time waveforms by using phase rotation to cancel all components in the final average, except the 2f1-f2 DPOAE. At times, f2 was turned off for 6 ms producing a gap so that the DPOAE was no longer generated. These procedures allowed the DPOAE onset as well as the decay during the gap to be observed in the time domain. Results showed that complexities emerged near the onset of the DPOAE time waveform as the f2/f1 ratio decreased, and at the beginning of the gap when f2 was turned off. Such complexities were unaffected by interference tones (ITs) near the DPOAE. However, these complexities were removed by ITs presented above f2, which can be explained by the interactions of distributed DPOAE components with different phase relationships.

3:40

2pPPa8. Distortion-product otoacoustic emission generator and reflection components in newborns, infants, and adults. Beth Prieve (Commun. Sci. and Disord., Syracuse Univ., 805 S. Crouse Ave., Syracuse, NY 13244, baprieve@syr.edu), Glenis Long (Speech-Lang.-Hearing Program, City Univ. of New York Grad. Ctr., New York, NY), and Carrick Talmadge (National Ctr. for Physical Acoust., University, MS)

Glenis Long and colleagues (Talmadge *et al*., J. Acoust. Soc. Am. **105**, 275) were among the first to model and describe the characteristics of distortion-product otoacoustic emissions (DPOAEs) using two sub-components from different, cochlear sources. It is now accepted that there is a generator component that results from inter-modulation distortion created by nonlinearity in the outer hair cell near the f2 place. The reflection component predominantly arises from coherent reflection near the characteristic place corresponding to the frequency of the distortion product. Because the two components are generated through different mechanisms, it has been hypothesized that they may be differentially affected by human development. The goal of this presentation is to discuss ongoing research of DPOAE components in newborns, infants and adults. Analysis of the components indicates that the relationship between growth rates for generator and reflection components are significantly different between infants and adults. Furthermore, the phase functions for both components are different among groups. Possible sources for these differences will be discussed. [Funded by the March of Dimes Birth Defects Foundation].

4:00–4:20 Panel Discussion

Session 2pPPb

Psychological and Physiological Acoustics: Speech, Attention, and Impairment (Poster Session)

Jayaganesh Swaminathan, Chair
Boston Univ., 635 Commonwealth Ave., Rm. 320, Boston, MA 02215

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

2pPPb1. Factors affecting frequency discrimination in school-aged children and adults. Crystal N. Taylor (Allied Health Sci., UNC Chapel Hill, Dept. of Allied Health Sci., CB 7190, Chapel Hill, NC 27599, Crystal_Taylor@med.unc.edu), Emily Buss (Otolaryngol.—Head and Neck Surgery, UNC Chapel Hill, Chapel Hill, NC), and Lori J. Leibold (Allied Health Sci., UNC Chapel Hill, Chapel Hill, NC)

Auditory frequency discrimination is a basic ability that may limit the maturation of speech and language skills in some listeners. Despite its importance, the factors affecting frequency discrimination in school-aged children are poorly understood. The goal of the present study was to evaluate effects related to memory for pitch, musical training, and the utilization of temporal fine-structure cues. Listeners were normal-hearing children, 5.1–13.6 years old, and adults. One subgroup of children had musical training (>150 h) and the other did not. The standard stimulus was either a 500- or a 5000-Hz pure tone, and the target stimulus was either a tone of higher frequency or a frequency-modulated tone (2- or 20-Hz rate) centered on the standard frequency. As commonly observed, mean frequency discrimination thresholds tended to be elevated in younger listeners. This developmental effect was smaller for FM detection than for pure-tone frequency discrimination, consistent with an effect of memory for pitch. The child/adult difference tended to be smaller for musically trained than untrained children. Children were not particularly poor at 2-Hz FM detection for the 500-Hz standard, a condition thought to rely on temporal fine-structure cues. [Work supported by NIDCD R03DC008389.]

2pPPb2. Cognitive and auditory influences on speech recognition by younger, middle-aged, and older listeners. Karen S. Helfer and Angela Costanzi (Commun. Disord., Univ. of Massachusetts Amherst, 358 N. Pleasant St., Amherst, MA 01002, khelfer@comdis.umass.edu)

In this study, we measured hearing thresholds and cognitive abilities (working memory, processing speed, executive function, and inhibition) in our participants to determine how these factors relate to speech understanding in the presence of competing speech. Participants were younger, middle-aged, and older adults, with the older adults having hearing thresholds that ranged from normal to a moderate hearing loss. The target stimuli for this study were recordings of TVM sentences [Helfer and Freyman (2009)] presented with various types of maskers (one or two other TVM sentences spoken by same-sex maskers, samples of running speech spoken by one or two same-sex talkers, and single-channel signal-envelope-modulated noise) at several signal-to-noise ratios. This poster will discuss speech recognition performance in the framework of multimasker penalties and will detail connections among hearing thresholds, cognitive abilities, and speech understanding. [Work supported by NIH DC012057.]

2pPPb3. Effects of reverberation and spatial diffuseness on the speech intelligibility of public address sounds in subway platform for young and aged people. Yong Hee Kim and Yoshiharu Soeta (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-8-31 Midorigaoka, Ikeda, Osaka 563-8577, Japan, yh.kim@aist.go.jp)

This paper investigated how young and aged people response differently on the public address sounds in subway platform according to various reverberation and diffuseness. Both word intelligibility (WI) and listening difficulty (LD) tests were adopted as experimental method. Twelve simulated sound fields at the same position were prepared according to the combination of four different reverberation times (RT) and three different interaural cross-correlation coefficients (IACC). Auralized sounds with anechoic test words of actual station names were presented to young and aged subjects at the fixed sound pressure level. As results, LD results showed significant correlation with RT for both young and aged subjects, whereas significant correlation of WI results and RT was found only in aged subjects. Aged subjects showed worse speech intelligibility performances on public address sounds than young subjects due to their worse hearing level. RT was found as the most important factor to determine speech intelligibility for both young and aged subject, whereas aged subject showed better speech intelligibility performances with lower IACC. From the regression analysis, LD rating was estimated from the measured RT and IACC. Additionally, the effects of word familiarity, individual noise sensitivity and hearing level on the speech intelligibility were discussed.

2pPPb4. Auditory influence on tactile perception changes with age. Simon P. Landry (École d'orthophonie et d'audiologie, Université de Montréal, 7077 avenue du Parc, Montreal, QC H3N 1X7, Canada, simon.landry.4@umontreal.ca), Jean-Paul Guillemot (Département de kinanthropologie, Université du Québec à Montréal, Montreal, QC, Canada), and François Champoux (École d'orthophonie et d'audiologie, Université de Montréal, Montreal, QC, Canada)

Characteristics of auditory interaction with vision have been extensively studied. However, auditory system integration with other sensory modalities, such as the tactile system, lacks such thorough investigations. The objective of this study was to examine the effects of age on audiotactile integration in humans. Thirty-one participants between the ages of 20 and 65 were divided into three groups according to their age. Audiotactile integration was assessed using the "auditory flash illusion" in which 1, 2, 3, or 4 tactile stimuli were accompanied with 0, 1, 2, 3, or 4 auditory stimuli. Participants were asked to ignore auditory stimulations and report the number of tactile stimulations perceived. All participants were tested with task relevant auditory and tactile stimuli as a control measures and were shown to have similar abilities. However, groups differed during the experimental conditions. The youngest group reported a greater number of tactile stimuli than actually presented during the illusory experimental conditions. Participants in the middle and older age groups did not report this illusory tactile perception. These results suggest that age reduces predisposition to audiotactile integration. These results are consistent with developmental studies for multisensory integration in other sensory modalities.

2pPPb5. The influence of auditory training on measures of temporal resolution in younger and older adults. Meital Avivi-Reich (Psychology, Univ. of Toronto Mississauga, 1909-35 Charles st.W, Toronto, ON m4y 1r6, Canada, me_avv@yahoo.com), Stephen R. Arnott (Rotman Res. Inst., Baycrest, Toronto, ON, Canada), Tamara Tavares, and Bruce A. Schneider (Psychology, Univ. of Toronto Mississauga, Toronto, ON, Canada)

Deterioration in the ability to perceived rapid changes in auditory input is thought to contribute to the difficulties older adults experience when communicating in noise. Studies have demonstrated that the performance of

young adults on auditory tasks improves with training. However, few studies have tested the degree to which practice improves auditory performance in older adults. A previous study examined the extent to which younger and older adults benefited from training when the task was to detect a gap in a narrow-band noise centered at 1 kHz. Significant improvements occurred in both age groups, indicating that auditory learning can still occur later in life. The present study examines if training improves performance when more than one auditory filter is activated. Twenty-four younger and older participants were trained for 10 days to detect a gap between a 2-kHz and a 1-kHz noise. Performance was assessed one day and one month after the last training session along with the extent to which the benefits generalized to other frequencies and the untrained ear. In addition, event-related potentials (ERPs) were obtained pre- and post-training to assess cortical changes in the response to temporal gaps. Initial results show improvement throughout training in both age groups.

2pPPb6. Intelligibility of voiced and whispered speech in noise in listeners with and without musical training. Dorea Ruggles, Ariane Riddell (Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, druggles@umn.edu), Richard Freyman (Univ. of Massachusetts-Amherst, Amherst, MA), and Andrew J. Oxenham (Univ. of Minnesota, Minneapolis, MN)

Musicians have been shown to exhibit more robust neural coding of periodicity than non-musicians; they have also been reported to exhibit an advantage in understanding speech in noise. This study tested the hypothesis that the musicians' speech intelligibility advantage arises from more efficient coding of voiced (periodic) speech. This was tested by comparing intelligibility of normal speech in noise with that of whispered (unvoiced) speech in musicians and non-musicians. Listeners with less than 2 years of formal musical training were categorized as non-musicians; listeners who began musical training before age 10 and who currently play more than 10 h/wk were included as musicians. Listeners heard grammatically correct nonsense sentences that were either (1) voiced, (2) whispered, or (3) whispered with subband amplitude distributions matched to the voiced speech. Masking noise was either continuous or gated with a 16 Hz, 50% duty cycle. In contrast to the earlier study, preliminary data suggest no advantage for musicians over non-musicians in understanding voiced or whispered speech in either continuous- or gated-noise conditions. The results suggest that more investigation is needed to fully understand the nature of auditory and speech processing advantages imparted by musical training. [Wok supported by NIH Grant No. R01DC05216.]

2pPPb7. Level-dependent effects on speech intelligibility. Patricia Pérez-González and Enrique A. Lopez-Poveda (Neurosci. Inst. of Castilla y Leon, Univ. of Salamanca, Pintor Fernando Gallego 1, Salamanca, Salamanca 37007, Spain, patripg@usal.es)

The effect of noise on speech intelligibility is typically measured using fixed-level speech (or noise) and varying the speech-to-noise ratio (SNR). An assumption of this procedure is that intelligibility mostly depends on the SNR and barely depends on speech level. The effective SNR, however, (i.e., the SNR in the internal stimulus representation), possibly depends on peripheral compression. Indeed, compression could facilitate or hinder intelligibility for negative and positive SNRs, respectively. Insofar as compression varies with level, speech intelligibility might also vary with speech level. Here, we tested these hypotheses by measuring percent correct digit triplet identification as a function of speech level for fixed SNRs. Measurements were carried out for normal-hearing subjects and for hearing-impaired subjects with linear cochlear responses, as assessed using the temporal masking curve method. Results for both groups suggest that the detrimental effect of the noise on intelligibility is larger for speech levels near threshold, particularly for negative SNRs, a result that cannot easily be explained by compression. Alternative explanations for the result are discussed.

2pPPb8. Talker effects in speech band importance functions. Eric W. Healy, Sarah E. Yoho, Carla L. Youngdahl, and Frederic Apoux (Speech and Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu)

The literature is somewhat mixed with regard to the influence of (a) the particular speech material (e.g., sentences or words) versus (b) the particular talker used to create the recordings, on band-importance function (BIF) shape. One possibility is that previous techniques for creating BIFs are not sensitive enough to reveal these influences. In the current investigation, the role of talkers was examined using the compound technique for creating BIFs. This technique was developed to account for the multitude of synergistic and redundant interactions that take place among various speech frequencies. The resulting functions display a complex microstructure, in which the importance of adjacent bands can differ substantially. It was found that the microstructure could be traced to acoustic aspects of the particular talkers employed. Further, BIFs for IEEE sentences based on ten-talker recordings displayed less microstructure and were therefore smoother than BIFs based on one such talker. These results together suggest that the compound technique is sensitive enough to reveal acoustic aspects of the particular talker employed. It is further suggested that multiple talkers, rather than smoothing of the functions, be used if the goal is to describe speech more generally.

2pPPb9. Psychometric effects of adding realism to a speech-in-noise test. Virginia Best, Gitte Keidser, Jörg M. Buchholz, and Katrina Freeston (National Acoust. Lab. and the HEARing Cooperative Res. Ctr., 126 Greville St., Chatswood, NSW 2067, Australia, virginia.best@nal.gov.au)

The speech reception threshold (SRT) is routinely measured in the laboratory to assess speech understanding in noise, but is often reported to be a poor predictor of performance in real world listening situations. The overall goal of this work is to determine whether introducing realistic aspects to speech tests can better capture individual differences and ultimately produce more relevant performance measures. We examined the psychometric effects of (a) transplanting a standard sentence-in-noise test into a simulated reverberant cafeteria environment, and (b) moving from sentence recall to a new ongoing speech comprehension task. Participants included normal hearers and hearing-impaired listeners (who were tested with and without their hearing aids). SRTs in the cafeteria environment were significantly correlated with standard SRTs, but were poorer overall and more sensitive to hearing loss. The comprehension task, despite having very different demands to sentence recall, produced similar SRTs under these conditions. The benefit of hearing aids was weakly correlated across the two listening environments and the two listening tasks. These manipulations promise to be useful for the creation of realistic laboratory tests that are engaging and challenging, yet controlled enough to be useful for psychophysical experiments.

2pPPb10. The Glasgow Monitoring of Uninterrupted Speech Task: A naturalistic measure of speech intelligibility in noise. Alexandra MacPherson and Michael A. Akeroyd (MRC Inst. of Hearing Res. (Scottish section), Queen Elizabeth Bldg., 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, alex@ihr.gla.ac.uk)

When listening to speech in noisy environments parts of the speech signal are often missed due to masking, degradation, or inattention. Sometimes the message can be reconstructed, but reallocating resources to recover the missed information can affect the efficiency and speed at which the message is understood. A slowing of processing will be particularly detrimental if there are few opportunities for understanding to catch up, e.g., when listening to the radio. It is likely then that the monitoring of uninterrupted flows of continuous speech will be especially sensitive to hearing or listening impairments. We developed a new task to test this. The Glasgow Monitoring of Uninterrupted Speech Task (GMUST) requires participants to listen to a 10-min segment of continuous speech while monitoring a scrolling transcript of the speech on the screen in front of them for any word substitutions. The proportion of word substitutions listeners are able to correctly identify is then used as a measure of speech intelligibility. When compared to speech reception thresholds (SRTs) given by a standard speech-in-noise test pilot results indicate higher SRTs for the GMUST. This suggests that the GMUST could be a more sensitive measure of deficits in speech-in-noise understanding than standard speech-in-noise tests.

2pPPb11. Magnitude of speech-reception-threshold manipulators for a spatial speech-in-speech test that takes signal-to-noise ratio confounds and ecological validity into account. Filip M. Rønne, Søren Laugesen, Niels S. Jensen, Renskje K. Hietkamp, and Julie H. Pedersen (Eriksholm Res. Ctr., Rørtangvej 20, Snekkersten 3070, Denmark, fmr@eriksholm.com)

Measuring speech-reception threshold (SRT) using adaptive procedures is popular, as testing yield results with desirable statistical properties. However, SRT measures have drawbacks related to the unbounded nature of the signal-to-noise ratio (SNR) at which the SRT is achieved. Often the SRT will be a double-digit negative number, which compromises the ecological validity of the result. If testing involves hearing aids, it means that these devices and the signal-processing algorithms in them may be operating in conditions for which they were not intended. Further, the commonly observed large spread in SRT (both between- and within-group) has the possibility to cause SNR confounds that may lead to faulty conclusions. One way to address these issues is to provide the experimenter with SRT manipulators, to control the SNR at which testing takes place for the individual listener. The present work aims at developing a spatial speech-in-speech test with a selection of SRT manipulators for the experimenter to choose from. The manipulators investigated in this study are as follows: the spatial separation between target and maskers, the number of spatially separated maskers, changing the masker gender, and scoring in words versus sentences. The magnitudes of the SRT manipulators were investigated using 20 hearing-aid users as listeners.

2pPPb12. Binaural speech recognition in continuous and intermittent noises in people with hearing loss. Chantal Laroche, Jean-Grégoire Roveda, Julie Levionnois, Christian Giguère, and Véronique Vaillancourt (Audiology/SLP Program, Univ. of Ottawa, 451 Smyth Rd., Ottawa, ON K1H 8M5, Canada, claroche@uottawa.ca)

Multiple tests have been developed to quantify speech recognition in noise, but the characteristics of the masking noise vary significantly across tests and can considerably impact performance and clinical interpretation. Using the HINT, speech perception in 24 young adults with normal hearing was assessed using both the standard masker (a continuous speech-spectrum noise) and an intermittent version of the masker at an ON-OFF rate of 16 Hz. Intermittency helps in extracting speech from the "quiet" segments of the noise. Speech recognition thresholds for frontal speech were better in the intermittent than the continuous noise by an amount of 13 and 10 dB for noise maskers located in front or side, respectively, The average difference in thresholds between the noise front and side conditions, called the binaural advantage, was 4.7 and 8.4 dB for the intermittent and continuous noises, respectively. Data collected with people presenting different hearing loss profiles also show binaural and intermittency advantages, but to a lesser degree. Considering that people encounter a wide a range of fluctuating noises in daily life, these results motivate adding an intermittent noise condition to the HINT protocol to better reflect the challenges faced by individuals with hearing loss.

2pPPb13. Spatial release from masking for noise-vocoded speech. Jayaganesh Swaminathan, Christine R. Mason, Timothy M. Streeter (Boston University, 635 Commonwealth Avenue, Room 320, Boston, MA 02215, jswamy@bu.edu), Virginia Best (National Acoust. Lab., Chatswood, NSW, Australia), and Gerald Kidd, Jr (Boston Univ., Boston, MA)

Spatially separating a speech target from interfering masker(s) generally improves target intelligibility; an effect known as spatial release from masking (SRM). This study assessed the contribution of envelope cues to SRM. Target speech was presented from the front (0° azimuth) and speech maskers were either colocated or symmetrically separated from the target in azimuth ($\pm 15^\circ$, $\pm 30^\circ$, $\pm 45^\circ$ and $\pm 90^\circ$) using KEMAR head-related transfer functions. The target and maskers were presented either as natural speech or as noise-vocoded speech. For the vocoded speech, intelligibility was conveyed only by the envelopes from M frequency bands. Experiment 1 examined the effects of varying the number of frequency bands for the vocoder, and the degree of target-masker spatial separation, on SRM. Experiment 2 examined the effects of low-pass filtering the envelopes of the vocoded speech bands on SRM. Preliminary results for experiment 1 indicated that SRM improved as the number of spectral channels providing independent envelope cues increased for all spatial separations. Preliminary results for experiment 2 showed no difference in SRM between low and high envelope-frequency cutoffs. Potential implications for studying hearing-impaired and cochlear-implant subjects will be discussed. [Work supported by NIH-NIDCD and AFOSR.]

2pPPb14. Can envelope recovery account for speech recognition based on temporal fine structure? Frederic Apoux, Carla L. Youngdahl, Sarah E. Yoho, and Eric W. Healy (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com)

Over the past decade, several studies have demonstrated that normal-hearing listeners can achieve high levels of speech recognition when presented with only the temporal fine structure (TFS) of speech stimuli. Initial suggestions to explain these findings were that they were the result of the auditory system's ability to recover envelope information from the TFS (envelope recovery; ER). A number of studies have since showed decreasing ER with increasing numbers of analysis filters (the filters used to decompose the signal) while intelligibility from speech-TFS remains almost unaffected. Accordingly, it is now assumed that speech information is present in the TFS. A recent psychophysical study, however, showed that envelope information remains in the TFS after decomposition, suggesting a possible role of ER in speech-TFS understanding. The present study investigated this potential role. In contrast to previous work, a clear influence of analysis filter bandwidth on speech-TFS understanding was established. In addition, it was shown that near perfect speech recognition from recovered envelopes can be achieved with as many as 15 analysis filters. Finally, the relationship between analysis and auditory filter bandwidths was explored in ER. Taken together, the present findings suggest that a role of ER in speech-TFS understanding cannot be excluded.

2pPPb15. Complementary correlation may offer a new approach to better understand temporal fine structure coding. Adrian KC Lee (Speech & Hearing Sci. and Inst. for Learning & Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98195, akclee@uw.edu), Les E. Atlas, and Xing Li (Elec. Eng., Univ. of Washington, Seattle, WA)

Historically, sound has been separated into a slow-varying envelope and a component that rapidly varies, called the temporal fine structure (TFS). Perceptual studies suggest that users of cochlear implants can benefit from better TFS coding strategies, as they lead to improved speech understanding in noise. Yet given the standard signal processing methodology used in the auditory neuroscience community, namely the Hilbert transform, findings in such prior studies have been limited by their use of the heavily distorted estimates of the TFS information (through the Hilbert phase of the related analytic signal). Complementary correlation is a new mathematical tool with potential to advance our understanding of temporal coding in neuroscience. This new mathematical formulation can be defined simply by dropping the usual complex conjugation operation from the canonical definitions of correlation, coherence, variance, magnitude-squared estimators of power, and other similar common second-order statistical quantities and their estimators. We will show that a complementary correlation approach, which provides a more complete characterization of TFS information, will provide measurable perceptual benefits by reducing distortion in the original speech signal.

2pPPb16. The roles of temporal envelope and temporal fine structure in speech synthesis for cochlear implants for tonal language speakers. Nantaporn Saimai, Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., 99 Moo 18 Phaholyothin Rd., Khlongluang, Pathumthani 12120, Thailand, 5310030076@student.tu.ac.th), Chutamanee Onsuwan (Linguistics, Thammasat Univ., Khlongluang, Pathumthani, Thailand), and Chai Wutiwiwatchai (National Electron. and Comput. Technol. Ctr. (NECTEC), Khlongluang, Pathumthani, Thailand)

Since most cochlear implants (CIs) have been developed for non-tonal languages, their level of hearing improvement is significantly decreased when used by speakers of tonal language, e.g., Thai. Temporal envelope (TE) and temporal fine structure (TFS) are important acoustic cues for languages with lexical tones. Specifically, TE is shown to carry manner and voicing cues for consonants, while TFS correlates with vowel formant transitions. Therefore, TFS and TE are expected to enhance intelligibility of lexical tones for CI patients. We proposed the use of six-channel bandpass filters to extract spectral information. Then, TE is extracted by half-wave rectification and smoothed by lowpass filler at 500 Hz cutoff frequency. TFS is extracted by the Hilbert transform to construct carrier signals. TE from each channel is modulated with its corresponding carrier signal and then combined to generate synthesized speech. Synthesized speech tokens from this study and two others [Fu *et al*. (1998) and Chen and Zhang (2008)] are evaluated by 16 Thais with normal hearing. The results showed that the intelligibility scores from the proposed algorithm are significantly higher than the other two for initials (by 32.2%) and final consonants (by 16.7%) and significantly higher for tones (by 48.8%) than Fu *et al*.

2pPPb17. The role of peripheral spectro-temporal coding in congenital amusia. Marion Cousineau (Département de Psychologie, Université de Montréal, Pavillon 1420 boul. Mont Royal, Entrance #1430 Ste. 0-120, Outremont, QC H2V 4P3, Canada, marioncousineau@gmail.com), Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), and Isabelle Peretz (Département de Psychologie, Université de Montréal, Montreal, QC, Canada)

Congenital amusia, a neurogenetic disorder, affects primarily pitch and melody perception. Here we test the hypothesis that amusics suffer from impaired access to spectro-temporal fine-structure cues associated with low-order resolved harmonics. The hypothesis is motivated by the fact that tones containing only unresolved harmonics result in poorer pitch sensitivity in normal-hearing listeners. F0DLs were measured in amusics and matched controls for harmonic complexes containing either resolved or unresolved harmonics. Sensitivity to temporal-fine-structure was assessed via interaural-time-difference (ITD) thresholds, intensity resolution was probed via interaural-level-difference (ILD) thresholds and intensity difference limens, and spectral resolution was estimated using the notched-noise method. As expected, F0DLs were elevated in amusics for resolved harmonics; however, no difference between amusics and controls was found for F0DLs using unresolved harmonics. The deficit appears unlikely to be due to temporal-fine-structure coding, as ITD thresholds were unimpaired in the amusic group. In addition, no differences were found between the two groups in ILD thresholds, intensity difference limens, or auditory-filter bandwidths. Overall the results suggest a pitch-specific deficit in fine spectro-temporal information processing in amusia that cannot be ascribed to defective temporal-fine-structure or spectral encoding in the auditory periphery. [Work supported by Fyssen Foundation, Erasmus Mundus, CIHR, and NIH grant R01DC05216.]

2pPPb18. Comodulation masking release and monaural envelope correlation perception in listeners with cochlear hearing loss. Heather Porter, John H. Grose, Joseph W. Hall, and Emily Buss (Otolaryngol. - Head & Neck Surgery, Univ. of North Carolina at Chapel Hill, 170 Manning Dr., Chapel Hill, NC 27599-7070, heather_porter@med.unc.edu)

This study investigated the comparative dependence of comodulation masking release (CMR) and monaural envelope correlation perception (MECP) on the degree of envelope similarity in pre-senescent adult listeners with normal hearing (NH) or mild-to-moderate cochlear hearing loss (CL). A 1600-Hz pure-tone signal was used to measure CMR as a function of degree of envelope correlation in 100-Hz-wide noise bands centered at 727, 1093, 1600, 2300, and 3268 Hz. The same noise band configuration was used to measure MECP thresholds for both comodulated and independent standards. Envelope correlation was adjusted by mixing comodulated and independent maskers at variable intensity ratios. The five-band complex was 85 dB SPL. Signal thresholds improved monotonically (i.e., CMR increased) with increasing degrees of envelope correlation for all listeners. Results for CL listeners were most similar to data from previous NH listeners at a 72 dB SPL masker level. For MECP, performance patterns for the two conditions were uniform across NH listeners, whereas those for CL listeners exhibited greater individual differences. Finally, CMR and MECP performance appeared to be related in listeners with CL. The pattern of results will be discussed in terms of the effects of CL on sensitivity to envelope similarity. [Work supported by NIDCD R01DC001507.]

2pPPb19. Attentional switching when listeners respond to semantic meaning expressed by multiple talkers. Ervin R. Hafter (Dept. of Psych., Univ. of California, Berkeley, CA, hafter@berkeley.edu), Jing Xia, Sridhar Kalluri (Starkey Hearing Res. Ctr., Berkeley, CA), Rosa Poggesi, Claes Hansen, and Kelly Whiteford (Psychology, Univ. of California, Berkeley, Berkeley, CA)

The "cocktail party problem" asks how we know what one person says when others are speaking at the same time. With interest in the difficulty faced by older and hearing-impaired listeners in multi-talker environments, the present experiment looks at the speed of shifting attention between talkers. In our simulated cocktail party, a subject sits among multiple talkers who are each telling a different story. A sequence of questions are drawn from the various stories and presented visually for subjects to answer with manual responses. Pay is based on correct answers, and attention is assessed by comparing response accuracy on questions from the talker identified visually as the primary and from the other talkers. Attention is manipulated by varying the primary talker at random, from question to question. The speed of shifting attention is measured by varying the time from when the new primary is identified to the moment when the relevant information appears in that story. In a related study, bilingual subjects must shift attention to talkers speaking either English or Spanish. This allows determination of the additional time needed to switch languages within a multi-talker environment.

2pPPb20. Dividing attention between two segregated tone streams. Laurent Demany and Catherine Semal (INCIA, CNRS and Universite de Bordeaux, BP 63, 146 rue Leo Saignat, Bordeaux F-33076, France, laurent.demany@u-bordeaux2.fr)

Listeners were presented with pure-tone sequences which had a high speed (25–50 tones/s) and consisted of two interleaved melodies, M1 and M2, spanning separate frequency ranges (624–786 Hz and 1483–1869 Hz). M1 and M2 were renewed from sequence to sequence and could be either (1) perfectly correlated, or (2) perfectly anticorrelated, or (3) independent of each other. The main task, performed in a 2I-2AFC paradigm, was to discriminate sequences of type 1 (perfect correlation) from sequences of type 2 or 3. This appeared to be relatively easy when, in the type-1 sequences, each note of M2 immediately preceded or followed the corresponding note of M1. In that case, however, listeners were unable to tell whether M2 preceded or followed M1, which shows that the two melodies were perceptually segregated. In another series of experimental sessions, listeners knew that M2 would always follow M1 after a fixed delay, corresponding to one tone in some sessions and three tones in other sessions. The main discrimination task was performed better for the one-tone delay, but performance was still well above chance for the three-tone delay. Overall, the data suggest that two segregated tone streams can be attended to simultaneously.

2pPPb21. Selective auditory or visual attention reduces physiological noise in the ear canals of human subjects. Kyle P. Walsh, Edward G. Pasanen, and Dennis McFadden (Psychology, Univ. of Minnesota, 75 East River Rd., Minneapolis, Texas 55455, kpwalsh@umn.edu)

A nonlinear version of the stimulus-frequency OAE (SFOAE), called the nSFOAE, was used to measure cochlear responses from human subjects while they simultaneously performed behavioral tasks requiring selective auditory attention (dichotic or diotic listening), selective visual attention, or relatively little attention. The auditory- and visual-attention tasks both were digit-recall tasks, where the nSFOAE-stimuli were interleaved with seven spoken (or displayed) digits. Unlike many previous studies, the required motor behavior always was the same across all tasks, including the inattention tasks. A 30-ms recording in the quiet followed every nSFOAE-eliciting stimulus to provide an estimate of the magnitude of each subject's physiological noise in each experimental condition. For every subject, physiological-noise magnitudes were higher (noisier) in the inattention tasks, and lower (quieter) in the selective auditory- and visual-attention tasks. The differences in noise levels were about 3–6 dB, on average, and the effect sizes for those differences all were greater than 2.5. Our interpretation is that the efferent innervation of the cochlea is activated maximally during selective attention (be it auditory or visual), potentially to the benefit of the observer. [Work supported by NIDCD grant DC00153.]

2pPPb22. Build-up of auditory stream segregation induced by tone sequences of constant or alternating frequency and the resetting effects of single deviants. Nicholas R. Haywood and Brian Roberts (Psychology, School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk)

Three experiments investigated the dynamics of auditory stream segregation. Experiment 1 used a 2.0-s constant-frequency inducer (10 repetitions of a low-frequency pure tone) to promote segregation in a subsequent, 1.2-s test sequence of alternating low- and high-frequency tones. Replacing the final inducer tone with silence reduced reported test-sequence segregation substantially. This reduction did not occur when either the 4th or 7th inducer was replaced with silence. This suggests that a change at the induction/test-sequence boundary actively resets build-up, rather than less segregation occurring simply because fewer inducer tones were presented. Furthermore, experiment 2 found that a constant-frequency inducer produced its maximum segregation-promoting effect after only three tone cycles—this contrasts with the more gradual build-up typically observed for alternating sequences. Experiment 3 required listeners to judge continuously the grouping of 20-s test sequences. Constant-frequency inducers were considerably more effective at promoting segregation than alternating ones; this difference persisted for ~10 s. In addition, resetting arising from a single deviant (longer tone) was associated only with constant-frequency inducers. Overall, the results suggest that constant-frequency inducers promote segregation by capturing one subset of test-sequence tones into an on-going, pre-established stream and that a deviant tone may reduce segregation by disrupting this capture.

2pPPb23. Annoyance perception for hearing impaired listeners: A revisit. Susie Valentine, Martin McKinney, and Tao Zhang (Starkey Hearing Technol., 6600 Washington Ave. S., Eden Prairie, MN 55344, susie_valentine@starkey.com)

For hearing impaired (HI) listeners, it is well known that certain sounds are much more annoying than others even though they may have similar spectral shape and level. For example, HI listeners often report paper rustling noise as highly annoying. A common approach to deal with this complaint is to reduce high frequency gain. While this approach may mitigate the complaint, it can create audibility issues for speech. A more effective approach is to determine the underlying cause of annoyance and then design an algorithm to selectively reduce it. While existing literature on annoyance perception for HI listeners is scant, a previous attempt was made to investigate this perception using real-world recordings [Vishnubhotla *et al.* (2012)]. The study showed a large variability of annoyance ratings across listeners that may have been due to subjective associations with the sound sources. In this study, we use abstract psychoacoustic stimuli designed carefully to avoid possible confounding subjective associations. A magnitude estimation method was used to measure the annoyance of each stimulus in hearing impaired listeners. All stimuli were presented over headphones in a sound treated room. Results will be presented along with implications for hearing aid applications.

2pPPb24. Acoustic correlates of tinnitus-like sounds. Jennifer Lentz and Yuan He (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu)

Although many people describe their tinnitus using complex terms (such a tea-kettle, crickets, and roaring), past studies of tinnitus have focused using pure tones and noises as stimuli. Therefore, this study was developed to begin to address the usefulness of using complex, dynamic sounds in the assessment of tinnitus. In a previous study, a free-classification task was used to ascertain the perceptual dimensions of tinnitus-like sounds in normally hearing listeners. Sounds were representative of those commonly used to describe tinnitus (e.g., ringing, tonal, noisy, pulsing, and clicking sounds). Listeners placed icons associated with each sound on a grid and placed similar sounds in clusters. Multi-dimensional scaling conducted on the classification data revealed three different perceptual dimensions. This study evaluated the acoustics of the stimuli to determine the nature of the perceptual dimensions. These analyses estimated a variety of temporal and spectral stimulus properties (e.g., autocorrelation statistics, spectral statistics, envelope characteristics, etc.). The acoustic characteristics were then correlated with the ordering along the three perceptual dimensions. Results suggest a noisy versus tonal dimension, an envelope-based dimension stimulus (choppy versus smooth), and a dimension related to dynamic stimulus characteristics.

2pPPb25. Intervention for restricted dynamic range and reduced sound tolerance: Clinical trial using a Tinnitus Retraining Therapy protocol for hyperacusis. Craig Formby (Commun. Disord., Univ. of Alabama, 700 University Boulevard East, Tuscaloosa, AL 35487, cformby@as.ua.edu), Monica Hawley, LaGuinn P. Sherlock, Susan Gold (Otorhinolaryngology, Univ. of Maryland School of Medicine, Baltimore, MD), Jason Parton, Rebecca Brooks, and JoAnne Payne (Commun. Disord., Univ. of Alabama, Tuscaloosa, AL)

Hyperacusis is the intolerance to sound levels that normally are judged acceptable to others. The presence of hyperacusis (diagnosed or undiagnosed) can be an important reason that some persons reject their hearing aids. Tinnitus Retraining Therapy (TRT), a treatment approach for debilitating tinnitus and hyperacusis, routinely gives rise to increased loudness discomfort levels (LDLs) and improved sound tolerance. TRT involves both counseling and the daily exposure to soft sound from bilateral noise generator devices (NGs). We implemented a randomized, double-blind, placebo-controlled clinical trial to assess the efficacy of TRT as an intervention for reduced sound tolerance in hearing-aid eligible persons with hyperacusis and/or restricted dynamic ranges. Subjects were assigned to one of four treatment groups (2x2): Devices: NGs or placebo NGs and Counseling: Yes or No. They were evaluated at least monthly on a variety of audiometric tests, including LDLs, the Contour Test for Loudness for tones and speech, and word recognition measured at each session's comfortable and loud levels. Eighty percentage of the participants who received full treatment benefited significantly; whereas the other treatment groups demonstrated $\leq 45\%$ treatment efficacy. Treatment dynamics and examples of improved word recognition post-treatment will be described. [Work supported by NIH R01 DC04678.]

2pPPb26. Relationship between distortion and working memory for digital noise-reduction processing in hearing aids. Kathryn Arehart (Speech, Lang., Hearing Sci., Univ. of Colorado, UCB 409, Boulder, CO 80309, kathryn.arehart@colorado.edu), Pamela Souza (Dept. of Commun. Sci. and Disord. and Knowles Hearing Ctr., Northwestern Univ., Evanston, IL), Thomas Lunner (Eriksholm Res. Ctr., Oticon A/S, Linköping, Sweden), Michael Syskind Pedersen (Oticon A/S, Smørum, Denmark), and James M. Kates (Speech, Lang., Hearing Sci., Univ. of Colorado, Boulder, CO)

Several recent studies have shown a relationship between working memory and the ability of older adults to benefit from specific advanced signal processing algorithms in hearing aids. In this study, we quantify tradeoffs between benefit due to noise reduction and the perceptual costs associated with distortion caused by the noise reduction algorithm. We also investigate the relationship between these tradeoffs and working memory abilities. Speech intelligibility, speech quality, and perceived listening effort were measured in a cohort of elderly adults with hearing loss. Test materials were low-context sentences presented in fluctuating noise conditions at several signal-to-noise ratios. Speech stimuli were processed with a binary mask noise-reduction strategy. The amount of distortion produced by the noise reduction algorithm was parametrically varied by manipulating two binary mask parameters, error rate, and attenuation rate. Working memory was assessed with a reading span test. Results will be discussed in terms of the extent to which intelligibility, quality, and effort ratings are explained by the amount of distortion and/or noise and by working memory ability. [Funded by NIH, Oticon, and GN ReSound.]

2pPPb27. Evaluation of a binaurally synchronized dynamic-range compression algorithms for hearing aids. Stephan Ernst, Giso Grimm, and Birger Kollmeier (Med. Phys., Univ. of Oldenburg, Carl von Ossietzky Universität Oldenburg, Oldenburg 26111, Germany, stephan.ernst2@uni-oldenburg.de)

Binaural cues such as interaural level differences (ILD) are used, among other cues, to organize auditory perception and to segregate sound sources in complex acoustical environments. Dynamic-range compression working independently at each ear in a bilateral hearing aid, however, can alter these ILDs, potentially affecting sound source segregation. Binaural synchronization of compression algorithms might thus be necessary to preserve potentially beneficial spatial cues. This study presents a binaurally linked model-based fast-acting dynamic compression algorithm designed to approximate the normal-hearing basilar membrane input-output function in hearing-impaired listeners. Aim of the evaluation was to assess the effect of binaural synchronization on speech intelligibility and listening effort in spatial masking conditions in comparison to bilateral fitting. Spatially symmetric and asymmetric masking conditions were used. A conventional multiband dynamic compression algorithm both implemented in a bilaterally independent and in a binaurally linked version, was tested as a reference. Hearing impaired listeners were aided individually with the algorithms for both experiments. Results indicate a small preference toward the model-based algorithm in challenging masking conditions. However, no benefit of binaural-synchronization could be found even for the fast-acting compressor, suggesting a dominant role of the better ear in all experimental conditions. [Work funded by BMBF 01EZ0741 and DFG FOR1732.]

2pPPb28. Assessing the contribution of spectral cues to recognition of frequency-lowered consonants. Kelly Fitz (Signal Process. Res., Starkey Hearing Technol., 6600 Washington Ave. South, Eden Prairie, MN 55344, kelly_fitz@starkey.com), Christophe Micheyl (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), Susie Valentine (Audiology Res., Starkey Hearing Technol., Eden Prairie, MN), and Tao Zhang (Signal Process. Res., Starkey Hearing Technol., Eden Prairie, MN)

Commercially available strategies for restoring audibility of critical high frequency cues to patients with severe high frequency hearing loss translate information from high-frequency regions with unaidable hearing to lower-frequency regions with aidable hearing. Methods for synthesizing lowered spectral features, rather than generating them from the signal, have been proposed, though no commercially available hearing aid uses such a method. We assessed consonant discrimination under three configurations of a spectral feature synthesis method intended for use in frequency lowering. Lowered consonants were rendered using one or two narrowband noise components presented in a low frequency region with aidable hearing. Different configurations conveyed different spectral cues intended to distinguish among lowered consonants. In a short pilot study, preliminary analysis found no significant difference in consonant matching accuracy between the different configurations, suggesting that listeners were not making use of additional spectral cues when they were available. However, there were some observable trends in the data, as well as open questions about the possible impact of training and acclimatization on listener performance. We will present the findings of a more extensive study to determine whether listeners with training can make use of enhanced spectral cues to distinguish among frequency-lowered consonants.

2pPPb29. Spatial release from masking in simulations of cochlear implants for single-sided deafness. Joshua G. Bernstein (Audiology and Speech Ctr., Walter Reed National Military Med. Ctr., 8901 Wisconsin Ave., Bethesda, MD 20889, joshua.g.bernstein.civ@health.mil), Nandini Iyer (Battlespace Acoust. Branch, Air Force Res. Lab., Wright Patterson Air Force Base, OH), and Douglas S. Brungart (Audiology and Speech Ctr., Walter Reed National Military Med. Ctr., Bethesda, MD)

Previous studies have shown that single-sided-deaf (SSD) patients implanted with cochlear implants (CIs) can receive spatial release-from-masking (SRM), likely due to head-shadowing effects. This study investigated the possibility that SSD-CI patients might also obtain SRM benefits from binaural-integration cues when the target speech is masked by a spatially separated fluctuating masker. Experiment 1 measured psychometric functions for word-recognition performance in the presence of stationary noise, modulated noise, and one or two same- or opposite-gender interfering talkers. The first ear received an unprocessed mixture containing the target and masker(s). The second ear received no signal (SSD), an unprocessed mixture containing just the maskers (NH-Binaural), or a mixture containing just the maskers that was processed with an eight-channel noise vocoder (SSD+CI). The results show that SRM occurs in the NH-Binaural condition for all masker types, but that it only occurs in the SSD+CI condition with same-gender interfering talkers. Experiment 2 revealed that SRM occurred in the SSD+CI condition with as few as two vocoder channels, and that maximum performance occurred with six or more channels. These results suggest that CIs for SSD have the potential to produce SRM in situations where monaural cues are insufficient for concurrent speech-stream segregation.

2pPPb30. Speech intelligibility of hearing impaired participants in long-term training of bone-conducted ultrasonic hearing aid. Toshie Matsui, Ryota Shimokura, Tadashi Nishimura, Hiroshi Hosoi (Dept. of Otorhinolaryngol. - Head and Neck Surgery, Nara Med. Univ., Shijo-cho 840, Kashihara City 634-8522, Japan, tomatsui@naramed-u.ac.jp), and Seiji Nakagawa (Living Informatics Res. Group, National Inst. of Adv. Industrial Sci. and Technol., Ikeda City, Hyogo, Japan)

Bone-conducted ultrasonic hearing aid (BCUHA) system is the unique device to provide the auditory sensation to profoundly hearing-impaired persons without any surgical operations. To clarify effects of long-term hearing training with this device, two deaf participants engaged the BCUHA training for 9 months. They were trained to use BCUHA through repetition of sentences read aloud and free conversation, and then they took part in word recognition tests and monosyllabic identification tests. Both participants showed that they could recognize words above chance using auditory sensation only provided by BCUHA device if alternatives or context were presented to them during the trials. Besides, it was observed that monosyllabic intelligibility score with both of auditory and visual cue had much increased with the day of training than the score with auditory cue only and that with visual cue only. The result suggests that the long-term training with BCUHA achieves efficient integration of auditory and visual cue of speech such as cochlear implant users showed in previous studies.

2pPPb31. Development of a novel hearing-aid for the profoundly deaf using bone-conducted ultrasonic perception: Assessments of the modulation type with regard to articulation, intelligibility, and sound quality. Seiji Nakagawa, Chika Fujiyuki, Yuko Okubo, Takayuki Kagomiya, and Takuya Hotehama (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol. (AIST), 1-8-31 Midorigaoka, Ikeda, Osaka 563-8577, Japan, s-nakagawa@aist.go.jp)

Bone-conducted ultrasound (BCU) is perceived even by the profoundly sensorineural deaf. We have developed a novel hearing-aid using BCU perception (BCU hearing aid: BCUHA) for the profoundly deaf. In the BCUHA, ultrasonic sinusoids of about 30 kHz are amplitude-modulated by speech and presented to the mastoid. Generally, two sounds are perceived: one is a high-pitched tone due to the ultrasonic carrier, with a pitch corresponding to a 8–16 kHz air-conducted (AC) sinusoid, and the other is the envelope of the modulated signal. As a method of amplitude modulation (AM), double-sideband with transmitted carrier (DSB-TC) modulation had been used; however, the DSB-TC is accompanied by a strong high-pitched tone. In this study, two new AM methods, double-sideband with suppressed carrier (DSB-SC) and transposed modulation (TM), that can be expected to reduce the high-pitched tone were newly employed in the BCUHA, and their resulting articulations, intelligibilities, and sound qualities were evaluated. The results showed that DSB-TC and TM had higher articulation and intelligibility scores than DSB-SC. Further, in terms of sound quality, the TM speech was closer than other types of BCU speech to AC speech. These results provide useful information for further development of the BCUHA.

2pPPb32. Communication aid utilizing bone-conducted sound via teeth by means of mouthpiece form actuator. Mikio Muramatsu (Dept. of Intermedia Art and Sci., Waseda Univ., Tokyo, Japan), Junko Kurosawa (Information Technol. Res. Organization, Waseda Univ., Tokyo, Japan), Yasuhiro Oikawa, and Yoshio Yamasaki (Dept. of Intermedia Art and Sci., Waseda Univ., 3-4-1, Okubo, Shinjuku-ku, Tokyo, Japan, yoikawa@waseda.jp)

Since bone-conducted sound is conveyed to cochlea directly, without passing through eardrum, it is audible even for hard-of-hearing people whose inner ears are still normal. In this study, we utilize bone-conducted sound via teeth so as to support sound communication. We implement a bone-conducted actuator on a tooth, while actuators of prevalent hearing aids are attached to mastoid, forehead, or jaw in general. Teeth convey sound excitation more easily, because they are bare bones, not covered with skin. Our hearing aid is made in the form of mouthpiece, and it can be readily put on and taken off from a tooth. Plus, we carry out experiments regarding sound localization and thresholds of bone-conducted sound via teeth, using this actuator. The results offer hearing characteristic of bone-conducted sound via teeth and show that examinees can perceive right and left using bone-conducted sound via teeth. In addition, we also attempt to record vibrations of teeth through a microphone, which is embedded on the mouthpiece form actuator. The aim of this study is to realize a hybrid actuator that enables both hearing and recording simultaneously and to suggest a new communication aid system not only for hard-of-hearing people but also for the robust.

Session 2pSA

Structural Acoustics and Vibration: Memorial Session in Honor of Miguel Junger

David Feit, Cochair
ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502

Joel M. Garrelick, Cochair
505 Tremont St., Apt 209, Boston, MA 02116

Chair's Introduction—1:35

Invited Papers

1:40

2pSA1. Recollections of my father: Miguel C. Junger. Sebastian Junger (P.O. Box 906, Truro, MA 02666, sjunger5@aol.com)

Thoughts and reflections on my father as such an inspirational and unforgettable person.

2:00

2pSA2. Miguel Junger: Legacy contributions to the field of structural acoustics. David Feit (ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502, feit.d@att.net)

The field of structural acoustics, i.e., the theory of acoustic radiation and scattering from elastic structures, developed primarily in the last half of the 20th century in response to Navy needs. Miguel Junger and colleagues at Cambridge Acoustical Associates under the sponsorship of the Office of Naval Research Sound and Structures made seminal contributions to the field. This presentation reviews some of this research, which ultimately was included in the book, "Sound, Structures and Their Interaction."

2:20

2pSA3. Three curious results of radiation loading calculations. Joel M. Garrelick (505 Tremont St., Unit 209, Boston, MA 02116, joelgarrelick@aol.com)

One of the many joys in reading Miguel C. Junger's contributions to the journal and elsewhere was the physical interpretations that accompanied theoretical results, which were at times unanticipated but always enlightening. In three instances however, he found the task of offering such an explanation particularly challenging and was beguiled by the fact. This is documented in his paper "Three apparently paradoxical results of sound radiation theory" [J. Acoust. Soc. Am. **106**(3, Pt. 1), 1589–1590 (1999)]. The specific results are as follows: (1) At frequencies above coincidence, the power radiated by a point driven infinite thin plate exposed to a low impedance fluid is equal to the power radiated by that plate when submerged in a high impedance fluid. (2) At low frequencies, the entrained mass that is associated with a translating circular piston fully submerged in an acoustic medium is equal to that acting on the piston when it is baffled and exposed to the medium on one side only. (3) The low frequency admittance of a fluid loaded infinite thin plate driven by a point force exhibits a spring-like reactance and yields a phase angle magnitude that is equal to that of the plate when driven by a line force. These three topics are revisited in this paper with an attempt to distinguish between paradox and coincidence.

2:40

2pSA4. Acoustic waves in violently collapsing bubbles. Thomas L. Geers (Mech. Eng., Univ. of Colorado, Campus Box 427, Boulder, CO 80302, geers@spot.colorado.edu)

Among Miguel Junger's many contributions to acoustical science and engineering were his papers and presentations on bubble acoustics. Among his many contributions to the well being of his colleagues at Cambridge Acoustical Associates was the mentoring of this presenter during the latter years of graduate study at MIT. Hence, this presentation in this session. In an evaluation of five reduced models for spherically symmetric bubble collapse and rebound [J. Appl. Phys. **112**, 054910 (2012)], it was found that some recent models, which incorporate wave propagation in both the external fluid and internal gas, did not perform as well as the long-established model by Keller and Kolodner, which incorporates wave propagation in the fluid but not in the gas [J. Appl. Phys. **27**, 1152–1161 (1956)]. Performance was assessed through comparisons against response histories produced by finite-difference solution of the Euler equations under adiabatic conditions. Further investigation revealed that neither acoustic-wave nor shock-wave propagation in the gas was apparent, but that a standing wave in the gas was. This prompted an enhancement of the Keller and Kolodner model that accounts for the standing wave. The formulation and evaluation of the enhanced model is the subject of this presentation.

3:00–3:20 Break

3:20

2pSA5. Poisson coupling in the in vacuo dynamics of an infinite cylindrical shell. Rudolph Martinez (Acoustics, Appl. Physical Sci., 49 Waltham St., Lexington, MA 02421, rmartinez@aphysci.com)

This paper applies a perturbation analysis to the "$n = 0$" in vacuo dynamics of an infinite cylindrical shell as presented in their exact form by Junger and Feit in *Sound, Structures, and Their Interaction*. The small parameter in the asymptotic theory is the square of the Poisson ratio ν. Two sets of results emerge: (1) Relating to the system's dispersion relation and therefore irrespective of the type of loading, the approximate roots of the governing cubic polynomial display explicitly the mutual contamination and influence of axial compression and latent skin flexure, the latter becoming actual past the ring frequency ωra/cp = 1. (2) In the parametric range of small values of ωa/cp, and for a radial ring force, our asymptotic expansions establish that the shell's normal-to-surface response is one of local reaction to zeroth order in ν2. It is not until the analysis is carried out to O(ν2) that wave propagation from axial compression and nearfield skin flexure begin to assert themselves away from the driven station.

3:40

2pSA6. Sound radiation by parallel coated plates separated by a fluid layer: Now and Then. Ann Stokes (Appl. Physical Sci., 49 Waltham St., Lexington, MA 02421, astokes@aphysci.com)

In 1987, I had the opportunity to work with Miguel Junger on a paper entitled "Sound Radiation by parallel coated plates separated by a fluid layer." A critical piece of the analysis involved analytically inverting a 5x5 matrix, which resulted in an extremely complex algebraic expression for the far-field of a point-excited 5-layer configuration. In order to understand the physical mechanisms, and validate the results, Dr. Junger developed asymptotic forms of the solution for familiar configurations and made use of spring-mass models to understand resonance enhancement effects. This talk introduces our current semi-analytical waveguide analysis of sound radiation by parallel, multi-layer plates. An advantage of the waveguide approach is that finite-length structures can be modeled, and axial discontinuities, including ribs and wavebearing bulkheads, can also be included very efficiently. Physical mechanisms are interpreted in terms of the propagating and evanescent waves of the structures. A focus of this talk is the extension of the perfectly matched layer (PML) model of the exterior fluid to waveguide models.

4:00

2pSA7. Cooperating with Miguel on improvements of the acoustical product - SOUNDBLOX. Klaus Kleinschmidt (Retired, 100 Newbury Court, Concord, MA 01742, ksquare@comcast.net)

Some 15 years after joining Miguel's consulting firm Cambridge Acoustical Associates around 1960 he asked me to help him with improving the sound absorbing quality of a patented slotted concrete block. The original concept of the block, sold under the trademark, SOUNDBLOX, was create a Helmholtz resonator by providing a slot in one face of a standard concrete block whose natural frequency would match the fundamental frequency of a common noise source, transformer hum. A broader absorption spectrum was desired to expand the rather limited applications of the original design to compete with certain acoustical materials widely used in schools, gymnasiums, auditoriums, and swimming pools. This presentation will describe the nature of our cooperation and the successes and failures of various concepts. The fact that the product, first introduced in the late 1950's, is still on the market speaks well of our collaboration.

Contributed Papers

4:20

2pSA8. Sound radiation from finite surfaces. Jonas Brunskog (Acoust. Technol., DTU Elec. Eng., Elektrovej, Bldg. 352, Kgs. Lyngby DK-2800, Denmark, jbr@elektro.dtu.dk)

A method to account for the effect of finite size in acoustic power radiation problem of planar surfaces using spatial windowing is developed. Cremer and Heckl presents a very useful formula for the power radiating from a structure using the spatially Fourier transformed velocity, which combined with spatially windowing of a plane waves can be used to take into account the finite size. In the present paper, this is developed by means of a radiation impedance for finite surfaces, which is used instead of the radiation impedance for infinite surfaces. In this way, the spatial windowing is included in the radiation formula directly, and no pre-windowing is needed. Examples are given for the radiation efficiency, and the results are compared with results found in the literature.

4:40

2pSA9. Modal contributions to sound radiated from a fluid loaded cylinder. Herwig Peters, Nicole Kessissoglou (School of Mech. and Manufacturing Eng., The Univ. of New South Wales, UNSW, Sydney NSW 2052 Australia, z3268667@student.unsw.edu.au), and Steffen Marburg (LRT4 - Inst. of Mech., Universitat der Bundeswehr Munchen, Neubiberg, Germany)

A modal decomposition technique to compute the individual modal contributions to the sound radiated from a cylindrical shell submerged in water is presented. The wet structural modes are calculated by means of a polynomial approximation and symmetric linearization of the underlying nonlinear eigenvalue problem. A Krylov subspace technique is used to reduce the model size of the structural domain, while the fluid domain remains unchanged. Results for the radiated sound power and sound pressure directivity are presented for groups of circumferential modes with common mode number. Under axial and transverse excitation, the cylinder breathing and bending modes are respectively the major modes contributing to the radiated sound at low frequencies. The contribution of the rigid body modes to the radiated sound is also observed.

5:00

2pSA10. Acoustic radiation mode shapes for control of plates and shells. William R. Johnson (Mech. Eng., Brigham Young Univ., 1085 N. 1750 W., Provo, UT 84604, will.johnson@byu.edu), Pegah Aslani (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Daniel R. Hendricks (Mech. Eng., Brigham Young Univ., Provo, UT)

During the advent of active structural acoustic control, attempts were made to target and control structural vibration mode shapes to reduce radiated sound power. In the late 1980s and early nineties work on acoustic radiation mode shapes developed an alternative way to target structural acoustic radiation. By attempting to control the radiation mode shapes, contributing structural modes could be more easily targeted. Radiation mode shapes have been examined previously for rectangular plates. The method has been extended to demonstrate radiation mode shapes of circular plates and cylindrical shells. Certain spatial derivatives of plate vibration have been found to be highly correlated with the most efficiently radiating radiation mode shapes at low frequencies. A weighted sum of these spatial derivatives is proposed as a new, generalized control metric.

Session 2pSCa

Speech Communication: Variability in Speech Intelligibility: Behavioral and Neural Perspectives

Rajka Smiljanic, Cochair
Linguistics, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198

Bharath Chandrasekaran, Cochair
Communi. Sci. and Disorders, Univ. of Texas at Austin, Austin, TX 78712

Sven Mattys, Cochair
Dept. of Psych., Univ. of York, York YO10 5DD, United Kingdom

Chair's Introduction—12:55

Invited Papers

1:00

2pSCa1. Processing speech of varying intelligibility. Rajka Smiljanic (Linguistics, Univ. of Texas-Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu) and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas-Austin, Austin, TX)

In this talk, we examine how variation in intelligibility impacts speech processing with insights from behavioral and neuroimaging studies. We discuss a set of experiments that explore the extent to which listener-oriented speaking style changes, sentence context, and visual information contribute to enhanced word recognition in challenging listening conditions. We further examine whether these same enhancements impact speech processing beyond word recognition, namely recognition memory for sentences. The results show that both signal-related and contextual enhancements lead to improved speech recognition in noise and, crucially, to a substantially better sentence recall. We then discuss studies examining neural mechanisms involved in processing speech of varying intelligibility using fMRI. Previous fMRI studies have examined speech intelligibility by using artificially degraded speech stimuli. Few studies have examined natural variation in intelligibility. Here we present neuroimaging data from two studies that examine natural variations in speech intelligibility (native vs. non-native speech; audio versus audiovisual speech). Overall, combined insights from behavioral and neuroimaging studies provide important additions to our understanding of how different sources of variability in the speech signal affect speech processing and memory representations.

1:20

2pSCa2. The impact of variation in phoneme category structure on consonant intelligibility. Valerie Hazan, Rachel R. Romeo, and Michele Pettinato (Speech, Hearing and Phonet. Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, v.hazan@ucl.ac.uk)

Newman *et al*. [J. Acoustic. Soc. Am. **109**, 1181–1196 (2001)] suggested that phoneme identification accuracy and speed for a given talker was affected by the degree of variability in their production of phoneme categories. This study investigates how intra-talker variability in the production of two phoneme contrasts varies with age and gender, and how this variability affects speed of perceptual processing. Multiple iterations of tokens differing in initial consonants (/s/-/ʃ/, /p/-/b/) were collected via picture elicitation from 40 adults and 31 children aged 11 to 14; measures of within-category dispersion, between-category distance, overlap, and discriminability were obtained. While females produced more discriminable categories than males, children produced farther yet more dispersed—and thus similarly discriminable—categories than adults. Variability was contrast-specific rather than a general talker characteristic. Tokens with initial /s/-/ʃ/ from pairs of adult and child talkers varying in between-category distance or overlap were presented for identification. The presence of overlap had a greater effect on identification accuracy and speed than between-category distance, with strongest effects for adult speakers, but reaction time correlated most highly with within-category dispersion. These data suggest that talkers who are less consistent in their speech production may be perceived less clearly than more internally consistent talkers.

Contributed Paper

1:40

2pSCa3. On the tolerance of spectral blur in the perception of spoken words. Robert E. Remez, Chloe B. Cheimets, and Emily F. Thomas (Dept. of Psych. and Prog. in Neurosci. & Behavior, Barnard College, Columbia Univ., 3009 Broadway, New York, NY 10027, remez@columbia.edu)

How does a listener resolve linguistic properties conveyed by speech? Many descriptions of perception attribute a causal role to brief spectral details in narrow frequency ranges. Perceptual standards allow far more variety, revealed by the robustness of perception of many kinds of distorted speech. The present study considered the effects of spectral blur on the recognition of spoken words. Listeners heard successive presentations of noise-vocoded easy and hard words. The number of spectral channels composing the word increased with each presentation, reducing blur within a trial. Four conditions counterbalanced the number of presentations of each word in a trial, 3 or 5, and the severity of initial blur, either 1 or 5 channels. In all conditions, the final presentation had 9 bands, yielding a net blur reduction of 4 or 8 bands. These conditions were also tested with the instruction that the

words would become clearer during each trial. A control used two repetitions of each word at 9 spectral bands. Across the tests, exposure to spectral blur impaired the recognition of easy and hard words alike regardless of the listener's belief during presentation. However, intelligibility of hard words declined sharply when subjects were instructed to attend to the continuity and successive decrease in blur within a trial. The pattern of results exposes the role of attention, uncertainty, and spectral resolution in the phonetic contribution to word identification.

Invited Papers

2:00

2pSCa4. Intelligibility of interrupted speech in listeners of different age and hearing status. Valeriy Shafiro, Stanley Sheft, Robert Risley (Commun. Disord. & Sci., Rush Univ. Med. Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu), and Brian Gygi (NIHR Nottingham Hearing Biomed. Res. Unit, Nottingham, United Kingdom)

In most real-world communicative environments, speech signals are fragmented and incomplete due to masking by other concurrent sounds. Experiments in the perception of gated and time-compressed speech provide a useful approach for systematically investigating factors involved in the perception of temporally fragmented speech. In the current work, the intelligibility of spoken sentences was measured following periodic signal interruption at various square-wave gating rates with time compression applied by either omitting or doubling silent intervals during gated-off times. Across interruption rates (0.5–16 Hz), speech perception of younger and older normal-hearing and older hearing-impaired adults was similar for slow interruption rates, but differed substantially for the faster rates. This rate-dependent variation was consistently found for different interruption methods and conditions. Importantly, speech perception at fast interruption rates correlated strongly with pure-tone thresholds and performance on speech-in-noise tests, while it did not correlate with results from tests of working memory or spectro-temporal pattern discrimination. These findings suggest that different perceptual processes are involved in the perception of interrupted speech at slow and at faster interruption rates, and have implications for developing diagnostic tests applicable to real-world listening environments. [Work supported by NIH.]

2:20–2:40 Break

2:40

2pSCa5. Impaired speech recognition under a cognitive load: Where is the locus? Sven Mattys (Dept. of Psych., Univ. of York, York YO10 5DD, United Kingdom, sven.mattys@york.ac.uk)

Improving the validity of speech-recognition models requires an understanding of the conditions in which speech is experienced in everyday life. Listening conditions leading to a degradation of the signal—noise, competing talkers, disordered speech—have received most of the attention in that literature. But what about adverse conditions that do not alter the integrity of the signal, such as listening to speech under a non-auditory cognitive load (CL)? Drawing upon a variety of behavioral methods, this presentation investigates the effects of a concurrent attentional or mnemonic task on the relative reliance on acoustic cues and lexical knowledge during speech-perception tasks. The results show that listeners under CL downplay the contribution of acoustic detail and increase their reliance on lexical-semantic knowledge. However, greater reliance on lexical-semantic knowledge under CL is a cascaded effect of impoverished phonetic processing, not a direct consequence of CL. Ways of integrating CL into the functional architecture of existing speech-recognition models are discussed.

Contributed Paper

3:00

2pSCa6. Development of the auditory evoked potential to amplitude rise time and rate of formant transition of speech sounds. Antoine Shahin and Allen Carpenter (Otolaryngology, The Ohio State Univ., 915 Olentangy River Rd., Columbus, OH 43212, tonyshahin@gmail.com)

We investigated the morphology of the N1-P2 auditory evoked potential (AEP) to changes in amplitude rise time (ART) and rate of formant transition (RFT) of consonant-vowel pairs (CVs) in 4–6-year olds and adults. In the AEP session, individuals listened passively to the CVs /ba/, /wa/, and a /ba/ with a superimposed slower-rising /wa/ envelope (/ba/wa). In the behavioral session, individuals listened to the same stimuli and judged whether they heard a /ba/ or /wa/. In 6-year olds and adults, the N1-P2 amplitude reflected a change in RFT (/ba/wa and /wa/) but not in ART (/ba/ and /ba/wa). In contrast, in the 4–5-year olds, the poorly developed N1-P2 did not show specificity to changes in RFT or ART. Behaviorally, 6-year olds and adults relied more strongly on RFT cues (classified /ba/wa as /ba/) during phonetic judgments, as opposed to 4–5-year olds, which utilized both cues equally (chance level). Our findings suggest that following age 4–5, the development of the N1-P2 AEP, representing maturation of synaptic connections in the superficial layer of the auditory cortex, reflects a shift toward weighting of spectral dynamics more than amplitude dynamics during /ba/ - /wa/ phonetic categorization.

Invited Paper

3:20

2pSCa7. Cortical responses to degraded speech are modulated by linguistic predictions. Jonathan E. Peelle (Dept. of Otolaryngol., Washington Univ. in St. Louis, 660 South Euclid, Box 8115, St. Louis, MO 63110, peellej@ent.wustl.edu)

Our perceptual experience is formed by combining incoming sensory information with prior knowledge and expectation. When speech is not fully intelligible, non-acoustic information may be particularly important. Predictions about this degraded acoustic signal can be provided intrinsically (if the speech is still partially intelligible) or extrinsically (for example, by presenting a written cue). I will discuss results from studies in which the neural response to speech was measured using magnetoencephalography (MEG), with speech clarity parametrically manipulated using noise vocoding. In one study, we found that during sentence processing the phase of ongoing

cortical oscillations is matched to that of the acoustic speech envelope in the range of the syllable rate (4–8 Hz). Critically, this phase-locking was enhanced in left temporal cortex when speech is intelligible. In a separate study of single word listening, accurate predictions provided by written text enhanced subjective clarity and changed the response in early auditory processing regions of temporal cortex. Both experiments thus highlight neural responses in brain regions associated with relatively low-level speech perception. Together these findings support the ability of linguistic information to provide predictions that shape auditory processing of spoken language, particularly when acoustic clarity is compromised.

3:40–4:20 Panel Discussion

TUESDAY AFTERNOON, 4 JUNE 2013 — 516, 2:20 P.M. TO 5:00 P.M.

Session 2pSCb

Speech Communication: Speech Intelligibility (Poster Session)

Yingjiu Nie, Chair
Speech Lang. Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr., SE, Minneapolis, MN 55455

Contributed Papers

All posters will be on display from 2:20 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 2:20 p.m. to 3:40 p.m. and contributors of even-numbered papers will be at their posters from 3:40 p.m. to 5:00 p.m.

2pSCb1. Effect of speech clarity on perception of interrupted meaningful and anomalous sentences. Rajka Smiljanic (Linguistics, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu), Stan Shaft (Commun. Disord. and Sci., Rush Univ. Med. Ctr., Chicago, IL), Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas-Austin, Austin, TX), and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Med. Ctr., Chicago, IL)

The influence of speech clarity on the perception of interrupted speech was examined for sentences distinguished by the presence of semantic-contextual cues. Semantically meaningful and anomalous sentences produced in either conversational or "clear" speech were periodically interrupted at gating rates ranging from slow (0.5 Hz) to fast (24 Hz) and presented to 32 native English listeners. At slow rates, speech perception may be based on integration of whole syllables and words, with "glimpsing" of (sub)phonemic segments playing a role at faster rates. Our results show that semantic context and speech clarity had a significant rate-dependent impact on the intelligibility of interrupted speech. At the lowest rates, intelligibility differences between conditions were minimal. Overall, interruption was most deleterious for anomalous conversational sentences. Such effects were seen even at the highest gating rate of 24 Hz for which interruption effects are generally minimal. The magnitude of the clear-speech benefit varied with gating rate for the two types of sentences, starting at 1 Hz for meaningful and 2 Hz for anomalous sentences. Acoustic-phonetic enhancements of clear speech thus "shifted" contextual benefit to lower gating rates. The implications of these results for our understanding of different time scales of speech processing will be discussed.

2pSCb2. Speech understanding in babble noise and syllabic-range temporal processing in aging. Pierre Divenyi (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, pdivenyi@ccrma.stanford.edu)

The ability to understand speech in multitalker noise was measured in a group of 46 elderly (71.6 +/− 6.03 years) having at worst moderate presbycusic hearing loss, using the SPIN test with the target and an 8-talker babble presented either monaurally (left or right) or, using generic HRTF spatialization, with the target at 45 degrees right or left flanked by two independent 4-talker babble on each side. The subjects' temporal processing was measured as (1) the reverberation time yielding a 15% decrease in word comprehension and (2) as the minimum modulation index necessary to segregate two streams generated by modulating, at an average frequency of 4.375 Hz, two carriers consisting of harmonics 4, 5, and 6 having fundamental frequencies of 107 and 189 Hz, that produced two simultaneous three-pulse envelopes, one accelerating and one decelerating. With effects of hearing loss under control, the data exhibited correlations between intelligibility in babble and temporal processing that were significant for a speech-to-babble ratio of 4 dB but not for one of 8 dB. The results thus suggest that under unfavorable acoustic conditions the elderly achieve speech understanding by focusing on prosodic rhythm on the syllabic level. [Work supported by NIA and the VA Medical Research.]

2pSCb3. The effects of temporal envelope confusion on listeners' phoneme and word recognition. Yingjiu Nie, Adam Svec, Peggy B. Nelson, and Benjamin Munson (Speech-Lang.-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, niex0008@umn.edu)

Broadened auditory filters in listeners with hearing loss may result in listeners' increased reliance on temporal envelope cues for understanding speech. Previous data have shown that background noise may affect hearing-impaired (HI) listeners by negatively affecting the temporal envelope cues in speech. The current study investigates additional HI listeners' understanding of vocoded spondees in the presence of fluctuating and stationary background noise. Stimuli were 8- and 32-channel noise vocoded double spondees, high-pass filtered at 1426 Hz. New data confirmed the previous finding that temporal envelope confusion in HI listeners resulted in speech understanding that is poorer in fluctuating noise (at a rate of 4 Hz) than in stationary noise. Preliminary analysis suggests HI listeners experience significant envelope confusion for both 8- and 32-channel vocoded stimuli. Additional analysis of phoneme errors suggests that envelope confusion affects HI listeners' perception of both consonants and vowels. Further analysis of j-factors will indicate the relationship of phoneme to whole word understanding for vocoded speech in noise. Results confirm the importance of temporal envelope cues for phoneme and syllable recognition for listeners with hearing loss. [Work supported by NIH DC008306 to PB Nelson.]

2pSCb4. Quality of voices processed by hearing aids: Intra-talker differences. Ramesh Kumar Muralimanohar, Caleb Kronen, Kathryn Arehart, James Kates (Dept. of Speech, Lang., and Hearing Sci., UCB 409, Univ. of Colorado at Boulder, Boulder, CO 80309, muralima@colorado.edu), and Kathleen Pichora-Fuller (Dept. of Psych., Univ. of Toronto, Toronto, ON, Canada)

The effects of hearing aid signal processing depend on the voice characteristics of a talker. For example, we have found that the perceptual and acoustic consequences of hearing aid signal processing vary across talkers

and that these effects can be explained, in part, by the acoustic differences between the different voices. However, we find that different utterances spoken by the same talker are also differentially affected by the hearing aid signal processing. In this study, we quantified the acoustic and perceptual consequences of hearing aid signal processing on several utterances spoken by the same talker. The voices were presented in quiet and in the presence of babble noise (10 dB SNR). The voices were processed with varying amounts of frequency compression, wide dynamic range compression and noise suppression (spectral subtraction). We considered intra-talker variability using an objective quality metric (Hearing Aid Sound Quality Index) and perceptual ratings of quality. We analyzed sentences from several different speech corpora, including the Hearing-in-Noise Test (HINT), IEEE, and TIMIT. The results showed interactions between the effects of signal processing and the acoustic characteristics of specific utterances spoken by an individual. [Work supported in part by GN ReSound and NSERC.]

2pSCb5. Perceptual confusability of French vowels. Kathleen C. Hall (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, kathleen.hall@ubc.ca) and Elizabeth Hume (Linguistics, Univ. of Canterbury, Christchurch, New Zealand)

The confusability of sounds is argued to both reflect phonological structure [e.g., Boomershine *et al*. (2008)] and be a source of phonological variability and change [e.g., Ohala (1981), Hume (1998)]. We present the results of a perception task in which 25 Parisian French-speaking participants identified the French vowels [i e ɛ y ø œ ə a u o ɔ ɔ̃ ɛ̃ ɑ̃], or ∅, in an aC_Ca context, using standard orthography in key words. The durations of the vowel in the vowel-containing stimuli were manipulated to be 0, 15, 25, 50, or 100% of their original durations. We can therefore determine which vowels are most confusable with each other (and thus likely to be the target for either mergers or dissimilatory processes) and which are most confusable with zero (and thus likely to be the target of processes such as deletion, assimilation, and metathesis). Results show high accuracy for [a i y u], even with very short durations; some degree of confusability within the nasal vowels; high confusability rates within the mid-front rounded vowels; and a tendency for zero to be confused with one of the mid-front rounded vowels. These results align with observed phonological patterns in French.

2pSCb6. Effects of phonetic experience on neural processing of English /r/ and /l/ by Korean and Japanese listeners. Lee Jung An, Brett A. Martin, and Glenis R. Long (Ph.D. Prog. in Speech-Lang.-Hearing Sci., The Grad. Ctr., CUNY, 365 Fifth Ave., 7th Fl., New York, NY 10016, lan@gc.cuny.edu)

The effects of phonetic experience on behavioral and neurophysiological processing of English /r/ and /l/ by Koreans and Japanese were compared to speakers of American English. Although English /r/ and /l/ are not phonemic in both Korean and Japanese languages, Koreans have a pseudo phonetic [ɾ]-[ll] model available for perception of English, /r/-/l/ sounds in medial position, while Japanese do not. Speech stimuli were a continuum of synthetic stimuli ranging from perceived /iri/ to perceived /ili/. To date, five subjects in each language group have been tested. As predicted, behavioral results show that English medial /r/ and /l/ were perceived in a categorical manner by Americans, in a categorical-like manner by Koreans and in a non-categorical manner by Japanese. Neural responses tapped by the ACC did not differ significantly between language groups for P1-N1-P2, suggesting little effect of phonetic experience on the encoding of these sounds. In contrast, the T-complex (Tb latency and Ta morphology) differed significantly between groups. The T-complex morphology had double-peaks in the Japanese group. These findings suggest that the T-complex may index the effects of phonetic experience on speech perception.

2pSCb7. Perception of Thai distinctive vowel length in noise. Chutamanee Onsuwan (Linguistics, Thammasat Univ., 2 Prachan Rd., Pranakhon, Bangkok, Bangkok 10200, Thailand, consuwan@tu.ac.th), Charturong Tantibundhit, Nantaporn Saimai, Tanawan Saimai (Elec. and Comput. Eng., Thammasat Univ., Khlongluang, Pathumthani, Thailand), Patcharika Chootrakool, and Sumonmas Thatphithakkul (National Electron. and Comput. Technol. Ctr. (NECTEC), Khlongluang, Pathumthani, Thailand)

A forced choice identification perception experiment using 150 monosyllabic rhyming-word stimulus pairs (with identical consonants and tone) in four conditions of white Gaussian noise was conducted to explore vowel confusions in Thai, a language with nine monophthongs and length (short-long) contrast for all vowels (e.g., /i/-/i:/ and /o/-/o:/). Each stimulus containing speech and noise portions is equal in length. Perceptual results of 18 vowels from 36 Thai listeners at a noise level (SNR) of -24 dB, where the percent intelligibility is the most interpretable, showed that stimuli with short vowels are more accurately perceived than those with long vowels (93.46 vs. 85.64%) with /o:/ and /e:/ as the most confusable. Interestingly, asymmetrical confusions are observed with very few short vowels being misperceived as long vowels, but a larger number of long vowels misperceived as short. Consistent with previous studies of perception of English vowels in white noise [e.g., Benki (2003)], the findings confirm perceptual robustness of vowel height (correlating with F1) over vowel front/backness (correlating with F2). Lastly, an analysis for listeners' misidentified responses shows that the listeners generally favor short over long vowels.

2pSCb8. Effects of semantic predictability and dialect variation on vowel production in clear and plain lab speech. Rory Turnbull and Cynthia G. Clopper (Linguistics, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43215, turnbull@ling.osu.edu)

Speech addressed to a non-native or hearing impaired listener features longer, more peripheral vowels. In addition, more extreme dialect-specific forms are produced in semantically predictable contexts, and less extreme forms (more standard forms) in unpredictable contexts. This study investigated the interactions between predictability and speaking style on Southern American English monophthongization of the vowel /aj/. The Midland dialect of American English served as the comparison. Participants read a set of sentences with monosyllabic target words in sentence-final position. Target words varied in semantic predictability based on the preceding sentential context. Each set of sentences was produced twice by each participant—first as if talking to a friend ("plain" speech) and again as if talking to a non-native or hearing impaired listener ("clear" speech). The duration, dispersion, and trajectory length of the vowel in each target word were measured. Preliminary results suggest that, as expected, Southern /aj/ has a shorter trajectory length than Midland /aj/, and in both dialects, /aj/ has a shorter trajectory length in clear speech than plain speech. However, these processes do not interact with each other or with semantic predictability, suggesting that style and predictability effects are independent of the realization of some dialect variants.

2pSCb9. Continuous recognition memory for spoken words in noise. Susanne Brouwer (Linguistics, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, smbrouwer@hotmail.com)

Previous research has shown that talker variability affects recognition memory for spoken words [Palmeri *et al*., (1993)]. This study examines whether additive noise is similarly retained in memory for spoken words. In a continuous recognition memory task, participants listened to a list of spoken words mixed with noise consisting of a pure tone or of high-pass filtered white noise. The noise and speech were in non-overlapping frequency bands. In experiment 1, listeners indicated whether each spoken word in the list was "old" (heard before in the list) or "new." Results showed that listeners were as accurate and as fast at recognizing a word as old if it was repeated with the same or different noise. In experiment 2, listeners also indicated whether words judged as "old" were repeated with the same or with a different type of noise. Results showed that listeners benefitted from hearing words presented with the same versus different noise. These data suggest that spoken words and temporally overlapping but spectrally non-overlapping noise are retained or reconstructed together for explicit, but not for implicit recognition memory. This indicates that the extent to which noise variability is retained seems to depend on the depth of processing.

2pSCb10. When spectral smearing can increase speech intelligibility. James A. Bashford, Richard M. Warren, and Peter W. Lenz (Psychology, Univ. of Wisconsin-Milwaukee, P.O. Box 413, Milwaukee, WI 53201, bashford@uwm.edu)

Sentences were reduced to an array of 16 effectively rectangular bands (RBs) having center frequencies ranging from 0.25 to 8 kHz spaced at $^1/_3$-octave intervals. Four arrays were employed, each having uniform subcritical bandwidths which ranged from 40 to 5 Hz. The 40 Hz width array had intelligibility near ceiling, and the 5 Hz array about 1%. The finding of interest was that when the subcritical speech RBs were used to modulate RBs of noise having the same center frequency as the speech, but having bandwidths increased to a critical (ERBn) bandwidth at each center frequency, these spectrally smeared arrays were considerably more intelligible in all but the 40 Hz (ceiling) condition. For example, when the 10 Hz bandwidth speech array having an intelligibility of 8% modulated the ERBn noise array, intelligibility increased to 48%. This six-fold increase occurred despite the elimination of spectral fine structure and the addition of stochastic fluctuation to speech envelope cues. (As anticipated, conventional vocoding with matching bandwidths of speech and noise reduced the 10-Hz-speech array intelligibility from 8% to 1%.) These effects of smearing confirm findings by Bashford, Warren, and Lenz (2010) that optimal temporal processing requires stimulation of a critical bandwidth. [Work supported by NIH.]

2pSCb11. Can physical metrics identify noise reduction settings that optimize intelligibility? Gaston Hilkhuysen and Mark Huckvale (Speech, Hearing and Phonet. Sci., Univ. College London, 31, Chemin Joseph Aiguier, Marseille 13402, France, ghilkhuysen@gmail.com)

Noise reduction algorithms often include adjustable parameters. Their settings can have important consequences for the intelligibility of an enhanced noisy speech signal, but it is not clear how to choose the best settings. In previous work, we have found that even experienced listeners prefer non-optimal settings, which degrade intelligibility. Measuring the effect of settings directly using an intelligibility listening task is often infeasible because of time, cost, or the large number of parameter combinations. In this paper we investigate whether physical intelligibility metrics can provide an efficient mean to optimize settings for noise reduction. A number of intelligibility metrics can be found in the recent research literature. An evaluation of these metrics has shown disappointing performance in their abilities to predict the absolute intelligibility of enhanced noisy speech. However, to find optimal settings, it may be sufficient to predict the relative change in intelligibility caused by a parameter change. Five metrics were used to predict the optimal settings of two parameters of a noise reduction system. The change in each metric with parameter settings are compared to listeners' performance on an intelligibility test with the enhanced signals. We show which metrics are best suited.

2pSCb12. Perception of English vowels and use of visual cues by learners of English and English native speakers. Yasna I. Pereira (Speech, Hearing and Phonet. Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, yasnap@hotmail.com)

English vowels may be difficult to discriminate for many learners of English (L2 learners). Research in L2 speech perception has shown that the use of visual cues improves speech perception, at least for visually salient contrasts. This study investigated the use of visual cues in the perception of English vowels by L2 Advanced learners (Spanish native speakers) and English native speakers (ENS). Thirty-seven L2 learners and 20 ENS were given a vowel test that presented real CVC words in audio (A), audiovisual (AV), and video-alone (V) mode. The A and AV conditions were presented in noise (-10 dB SNR) to ENS and in quiet to L2 learners. For ENS, identification rates were significantly higher in AV than in A condition, suggesting there were visual cues to vowel identity. For L2 learners, A scores were significantly lower than for ENS, and AV scores did not differ significantly from results in A mode. This suggests low sensitivity to visual cues to vowel identification, though L2 learners achieved better than chance scores when forced to attend to visual information in the V mode. These results support previous findings of relatively poor sensitivity to visual cues to phoneme identity in L2 learners.

2pSCb13. The development of clear speech strategies in 9–14 year olds. Michèle Pettinato and Valerie Hazan (Speech, Hearing and Phonet. Sci., UCL, Chandler House, 2, Wakefield St., London WC1N 1PF, United Kingdom, m.pettinato@ucl.ac.uk)

This study investigated the development of global clear speech strategies of child talkers. Two groups of 20 talkers aged 9–10 (children) and 13–14 (teens) were recorded in pairs while they carried out spot the difference picture tasks (diapix), either hearing each other normally (NB condition) or with one talker hearing the other via a three-channel noise vocoder (VOC condition). Acoustic-phonetic analyses focused on the talker having to overcome the communication barrier. Data were compared to those for twenty of the adults in Hazan and Baker [J. Acoust. Soc. Am. **130**, 2139–2152 (2011)]. The three age groups did not differ in task transaction time for NB, but children took significantly longer to complete the task in VOC than teens or adults who took equally long. Children spoke at a slower speech rate overall than teens, while teens and adults did not differ; all groups significantly reduced their speech rate in VOC relative to NB. Adults hyperarticulated vowels in VOC, but children and teens showed only minor adaptations. These results suggest that although 9–10 year olds use some strategies to clarify their speech in difficult conditions, other strategies continue to develop into late adolescence.

2pSCb14. Consonant confusability and its relation to phonological dissimilarity. Sameer ud Dowla Khan (Linguistics, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, sameeruddowlakhan@gmail.com)

Consonant similarity can be measured indirectly through a language's phoneme inventory, lexicon (e.g. cooccurrence restrictions), or phonology (e.g., processes that take similarity or dissimilarity into account). It can also be measured more directly as confusability in a perception task. Thus far, consonant similarity in Bengali has only been measured indirectly, through the inventory, lexicon, and phonology. Previous studies [Khan (2006)] claim that Bengali speakers judge the similarity of consonants in echo reduplication, where the initial consonant of the base is systematically replaced with a phonologically dissimilar consonant in the reduplicant, e.g., kashi "cough" > kashi-tashi "cough, etc." but thonga "bag" > *thonga-tonga > thonga-fonga "bags, etc."). This measurement of similarity assumes a set of features assigned language-specific weights; for example, [voice] is weighted more heavily that [spread glottis], to explain why speakers treat the pair [t, th] as more similar than the pair [t, d]. But does the measurement of similarity inherent in the echo reduplicative construction correspond directly to the relative perceptibility of different consonant contrasts? The current study examines data collected in a perception experiment, comparing the relative confusability of Bengali consonants produced in noise with the claims of phonological notions of similarity associated with echo reduplication.

2pSCb15. Ambiguity related to French liaisons: The role of bottom-up and top-down processes. Mireille Babineau and Rushen Shi (Psychology, Université du Québec à Montréal, Département de Psychologie, Université du Québec à Montréal, C.P. 8888 succursale Centre-ville, Montreal, QC H3C 3P8, Canada, babineau.mireille@courrier.uqam.ca)

French liaison is a phonological process involving the surfacing of an underlying floating consonant as the onset of a vowel-initial word when the word is preceded by a liaison-causing word. We used vowel- and consonant-initial ambiguous targets following four liaison-related contexts /z/, /n/, /t/, /r/ (e.g., ces onches - ces zonches). Targets included nouns and pseudonouns. Quebec-French-speaking adults performed two tasks (production, discrimination). One bottom-up (acoustic cues) and three top-down (noun token frequency, word onset probability, contextual liaison knowledge) factors were investigated in the production task. Participants had to produce the last word upon hearing each phrase. Their productions thus reflected their interpretation of the onset of the liaison-ambiguous words. Perception of acoustic cues was also tested in the discrimination task: participants judged if two phrases were different or same. No perception of any acoustical distinction was revealed in the tasks: participants often produced the intended target incorrectly, and differently intended phrasal pairs were judged same (e.g., ces onches - ces zonches). Effects of top-down information were found in the production task. Among the top-down factors, liaison knowledge related to liaison-causing words had a dominant impact on participants' interpretation of the ambiguous phrases, showing the importance of contextual knowledge in lexical recognition.

2pSCb16. The identification of high pass filtered vowels. Jeremy Donai and Dwayne Paschall (Speech-Lang.-Hearing Sci., Texas Tech. Univ. Health Sci. Ctr., 3601 4th St., Lubbock, TX 79430, jeremy.donai@ttuhsc.edu)

Vowels are typically described according to their spectral prominences (i.e., formants). Previous studies have shown that the first three formants provide important information for identification (Hillebrand *et al*. (1995), Miller (1989), Molis (2005), Peterson and Barney (1952)]. The present study measured identification accuracy for six naturally produced vowels spoken by a male and female talker with these spectral prominences removed. The six hVd tokens for each talker were high-pass filtered to remove the first three formants from the vowels and then identified by 24 normal hearing listeners. Results suggest that listeners identified a majority of the tokens above chance levels. The average identification of the male vowels was 29% (range, 17%–47%), with two vowels identified with nearly 50% accuracy. Average identification for the six female vowels was 53% (range, 37%–72%), with 5 of 6 vowels being identified with over 40% accuracy.

2pSCb17. The effects of surgical masks on speech perception in noise. Kelsi J. Wittum, Lawrence L. Feth, and Evelyn M. Hoglund (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, wittum.2@osu.edu)

Surgical masks and blood shields worn by anesthesiologists and surgeons in hospital operating rooms may negatively impact speech communication and put patients at needless risk. Young adult subjects listened to sentences from the Speech Perception in Noise Test (SPIN) recorded by a male and female talker. All eight SPIN lists were recorded under three different speaking conditions: (1) speaking normally without any obstruction, (2) wearing a typical surgical mask, and (3) wearing a surgical mask with an attached blood shield. Multi-talker babble was mixed with the SPIN sentences at several signal-to-noise ratios to simulate conversation in noisy environments. Speaker gender and recording conditions were counterbalanced across listeners to control for learning and fatigue effects. SPIN test scores for each of the three types of recordings and both talker genders were compared in order to determine the degradation that blood-shields and surgical masks may have speech communication in the operating room. [Research supported by research grants from the Division of Social and Behavioral Sciences and a scholarship from the College of Arts and Sciences at The Ohio State University.]

2pSCb18. The role of high-frequency envelope fluctuations for speech masking release. Søren Jørgensen and Torsten Dau (Elec. Eng., Ctr. for Appl. Hearing Res., DTU, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, sjor@elektro.dtu.dk)

The speech-based envelope power spectrum model [sEPSM; Jørgensen and Dau (2011), Jørgensen *et al*. (2013)] was shown to successfully predict speech intelligibility in conditions with stationary and fluctuating interferers, reverberation, and spectral subtraction. The key element in the model was the multi-resolution estimation of the signal-to-noise ratio in the envelope domain (SNR_{env}) at the output of a modulation filterbank. The simulations suggested that mainly modulation filters centered in the range from 1 to 8 Hz contribute to speech intelligibility in the case of stationary maskers whereas modulation filters tuned to frequencies above 16 Hz might be important in the case of fluctuating maskers. In the present study, the role of high-frequency envelope fluctuations for speech masking release was further investigated in conditions of speech-on-speech masking. Simulations were compared to various measured data from normal-hearing and hearing-impaired listeners [Festen and Plomp (1990), Christiansen *et al*. (2013)]. The results support the hypothesis that high-frequency envelope fluctuations ($>$30 Hz) are essential for speech intelligibility in conditions with speech interferers. While the sEPSM reflects effects of energetic and modulation masking in speech intelligibility, the remaining unexplored effect in some conditions may be attributed to, and defined as, “information masking.”

2pSCb19. Regional linguistic variations in Canadian French: Do they affect performance on speech perception in noise? Josée Lagacé, Stéphanie Breau-Godwin, and Christian Giguère (Health Sciences Faculty, University of Ottawa, 451 Smyth Rd, Ottawa, ON K1H 8M5, Canada, josee.lagace@uottawa.ca)

Many audiologists working with a Canadian French population use word recognition tests to measure speech perception abilities in noise. The TMB test (“Test de Mots dans le Bruit”) includes four lists of 35 words presented in babble noise. The test is intended to measure the pre-cognitive perceptual stage of auditory processing and does not require understanding of the phonetic differences between speech sounds at a cognitive level. Previous studies examining performance on auditory tests similar to the TMB showed differences between populations speaking the same language but with different accentuations, such as the English spoken in the United States versus the United Kingdom. Variations in performance were attributed to accentuation differences between the speaker and the listener. To the authors’ knowledge, no study appears to have investigated the effect of regional linguistic variations of Canadian French on word recognition in noise. Normative data for the TMB are being collected in three regions of Canada: Moncton, Montréal, and Ottawa. Participants are all native speakers of Canadian French, but there are important linguistic variations across the three regions. Knowledge of the effect of regional linguistic variations on the TMB performance will help refine interpretation of test results in the audiology clinics.

2pSCb20. The role of across-frequency envelope processing for speech intelligibility. Alexandre Chabot-Leclerc, Søren Jørgensen, and Torsten Dau (Ctr. for Appl. Hearing Res., Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds plads, Bldg. 352, Kongens Lyngby 2800, Denmark, alech@elektro.dtu.dk)

Speech intelligibility models consist of a preprocessing part that transforms the stimuli into some internal (auditory) representation, and a decision metric that quantifies effects of transmission channel, speech interferers, and auditory processing on the speech intelligibility. Here, two recent speech intelligibility models, the spectro-temporal modulation index [STMI; Elhilali *et al*. (2003)] and the speech-based envelope power spectrum model [sEPSM; Jørgensen and Dau (2011)] were evaluated in conditions of noisy speech subjected to reverberation, and to nonlinear distortions through either a phase jitter process or noise reduction via spectral subtraction. The contributions of the individual preprocessing stages in the models and the role of the decision metrics were analyzed in the different experimental conditions. It is demonstrated that an explicit across-frequency envelope processing stage, as assumed in the STMI, together with the metric based on the envelope power signal-to-noise ratio, as assumed in the sEPSM, are required to account for all three conditions. However, a simple weighting of the across-frequency variance of the modulation power at the output of the (purely temporal) modulation filterbank is assumed to be sufficient to describe the data, i.e., a joint two-dimensional modulation filterbank might not be required.

2pSCb21. Using landmark detection to measure effective clear speech. Suzanne E. Boyce, Sarah Hamilton (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Mail Location 379, Cincinnati, OH 45267, Suzanne.Boyce@uc.edu), Joel MacAuslan (S.T.A.R. Corp., Bedford, MA), Jean Krause (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL), Rajka Smiljanic (Dept. of Linguist., Univ. of Texas at Austin, Austin, TX), and Ann Bradlow (Dept. of Linguist., Northwestern Univ., Evanston, IL)

While the relationship of speaking style to intelligibility under challenging conditions has been established, it is a common observation that some speakers seem to be more intelligible than others for most listeners. In previous work, we have reported that automatic measures based on the technique of Landmark Detection appear to track differences between Clear and Conversational speaking style. One question that remains is whether Landmark measures can be used to predict which speakers are most likely to produce highly intelligible speech. In this study, we took advantage of a set of previously acquired databases covering a total of 31 American English speakers who produced Clear and Conversational Speech to examine correlations between our Landmark-based measures and the Clear Speech productions of highly intelligible speech. Across these databases, we had data on intelligibility for 13 speakers. Results showed that speakers with high overall intelligibility in Clear Speech showed significantly different patterns on Landmark-based automatic measures, compared to speakers with more moderate performance on intelligibility measures. Applications of these results to problems in speech technology, linguistic education, and clinical practice will be discussed.

2pSCb22. Comprehending speech at artificially enhanced rates. Lucia da Silva, Adriano V. Barbosa, and Eric Vatikiotis-Bateson (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC, Canada, helena.rozario@gmail.com)

This study is part of a larger study comparing the production, perception, and long-term comprehension of natural fast speech and artificially sped-up speech. In the 1950’s, Fairbanks and colleagues made the interesting claim that artificially compressing speech to half its duration (twice the

rate) and listening to it twice is comprehended better than listening to the original natural speed utterance once. Since this finding has been disputed [T. Sticht, J. Expr. Ed. **37**, 60–62 (1969)], we focus here on providing baseline results for perception of naturally produced speech at conversational and fast rate and of speech sped up to twice its original speed. Specifically, we compare how well the different stimuli are perceived when presented once, twice in rapid succession, and twice with delays of hours and days between presentations. We evaluate perception for both audio-only and audiovisual stimuli.

2pSCb23. Influence of cochlear implantation on sentence intelligibility and duration. Olga Peskova (Commun. Sci. and Disord., Callier Adv. Hearing Res. Ctr., The Univ. of Texas at Dallas, Office JO3.104, 800 W.Cambell Rd., Richardson, TX 75080, oxp100020@utdallas.edu), Nirmal Kumar Srinivasan (Dept. of Elec. Eng., Univ. of Texas at Dallas, Richardson, TX), Sujin Shin, Madhu Sundarrajan, and Emily Tobey (Commun. Sci. and Disord., Callier Adv. Hearing Res. Ctr., The Univ. of Texas at Dallas, Richardson, TX)

Cochlear implants (CI) allow children with hearing loss (HL) to achieve speech perception and production outcomes that make their spoken speech understandable to normal hearing adult listeners. This capability is characterize by wide variability of scores. In order to understand the factors that contribute to the overall variability, we investigated the effects of duration of cochlear implantation on speech intelligibility and sentence duration over time. Participants were 105 children implanted between the ages of 2 and 4 and tested at 2 time points there they were 8 and 16 years old. Participants repeated McGarr sentences, which vary in length from 3 to 5 to 7 syllables. Recording were analyze using acoustic software to designate the beginning and end of each sentence in listeners who only heard one sentence from one child. Speech intelligibility scores were related statistically to the duration of each sentence. Durations of sentences that approximated that of normal hearing listeners were those with high intelligibility judgment. In addition, it appears that the children with the longest experience of CI use continue to improve their intelligibility. [Sponsored by NIH.]

2pSCb24. Influence of duration on the perception of consonants /x/ and /j/ in Chinese. Li Feng (Unit of Perceptual Psych., Kyushu Univ., Kyushu University, 4-9-1Shiobaru Minami-ku, Fukuoka, Japan, Fukuoka, Fukuoka 815-8540, Japan, rihou1986@yahoo.co.jp), Nakajima Yoshitaka, and Ueda Kazuo (Dept. Human Sci., Kyushu Univ., Fukuoka, Japan)

In the current study, we investigated whether Mandarin Chinese native speakers' perception of /x/ and /j/ was affected by the duration of consonant parts. Two perceptual experiments, in which the method of constant stimuli was employed, were conducted. Six normal-hearing adults (four females) took part in the experiment. Their average age was 24 yr. All subjects were native speakers of Mandarin Chinese. In the first experiment, Chinese syllables that begin with /x/ or /j/ were extracted from a speech database, and the consonant parts were manipulated in terms of duration. As the duration of /x/ was decreased or the duration of /j/ was increased, to a certain extent, the consonant which had been originally /x/ was perceived as /j/, and vice versa. Synthesized noises instead of recorded consonants were utilized in the second experiment, similar effects of the consonant duration appeared.

2pSCb25. Quality of older voices processed by hearing aids: Acoustic factors explaining inter-talker differences. Huiwen Goy, Margaret K. Pichora-Fuller (Psychology, Univ. of Toronto, 3359 Mississauga Rd. North, Mississauga, ON L5L 1C6, Canada, huiwen.goy@utoronto.ca), Pascal van Lieshout (Dept. of Speech-Lang. Pathol., Univ. of Toronto, Toronto, ON, Canada), and Kathryn H. Arehart (Dept. of Speech, Lang., Hearing Sci., Univ. of Colorado Boulder, Boulder, CO)

Hearing aid signal processing algorithms are often evaluated with professional recordings of voices. However, hearing aid users often listen to older talkers who may have poorer voices than younger talkers. The purpose of this study was to quantify the extent to which the acoustic and perceptual consequences of hearing aid digital signal processing algorithms differ for talkers that vary in their vocal characteristics. There were six older talkers (3 males and 3 females) selected from a larger database of 79 talkers; their voices had good, moderate, or poor quality based on perceptual data from younger and older listeners. The voices were presented in quiet and in the presence of babble noise (10 dB SNR). The voices were processed with varying amounts of frequency compression, wide dynamic range compression, and noise suppression (spectral subtraction). There were interactions between signal processing and voice characteristics as measured using the Hearing Aid Speech Quality Index (HASQI) and listener ratings of perceived sound quality of the processed speech. In this paper, we examine the possible acoustic sources that may explain these interactions, including inter-talker differences in formant space and pitch variation. [Research supported by NIH, GN Resound, NSERC, and the Canada Research Chairs program.]

2pSCb26. The effect of background noise on the ability to perceive and remember unrelated words in nonnative listeners. Meital Avivi-Reich, Caterina Y. Leung, and Bruce A. Schneider (Psychology, Univ. of Toronto Mississauga, 1909-35 Charles st.W, Toronto, ON m4y 1r6, Canada, me_avv@yahoo.com)

Nonnative listeners find it more difficult to meet the challenges presented by additional background noise than do native listeners [Ezzatian *et al*., Speech Commun. **52**, 919–929 (2010)], but it is not known whether it is more difficult for them to remember what was said in noisy situations than it is for native listeners. Previous studies have acknowledged that the effect of background noise on the ability to perceive and remember unrelated words is greater in older adults than younger adults. The present study investigates auditory memory performance in nonnative younger adults, using a paired-associate paradigm in three conditions: quiet, continuous babble and babble during word presentation only. Noise levels were adjusted to equate for individual differences in the ability to identify single words in noise. The initial results suggest that nonnative listeners perform similarly to native young adults in the quiet and continuous conditions but worse in the babble during the word-presentation-only condition. These results suggest that stream segregation may be slower in nonnative listeners when the masker and the target words start at the same time.

2pSCb27. Advantage of talker differences and spatial separation for speech-on-speech listening in younger and older adults with good audiograms. Jana Besser (ENT/Audiology, VU Univ. Med. Ctr., De Boelelaan 1118, Amsterdam 1081HV, Netherlands, besser.jana@gmail.com), Kathleen Pichora-Fuller (Dept. of Psych., Univ. of Toronto, Mississauga, ON, Canada), and Joost M. Festen (ENT/Audiology, VU Univ. Med. Ctr., Amsterdam, Netherlands)

Older adults have more difficulty than younger adults understanding speech when there is competing speech, even if they have good audiograms. Age-related differences in listening may be due to declines in auditory temporal processing and/or cognition. We administered the LiSN-S [Cameron *et al*. (2011)] to measure speech reception thresholds (SRTs) in younger and older adults with good audiograms. There were four test conditions, in which the target and competing speech were presented with the same or different voices at the same or different locations. Compared to younger listeners, older listeners obtained worse SRTs in all test conditions and they realized less advantage from talker differences and spatial separation between the target and competing speech. For both groups, the results obtained in the four test conditions were strongly associated with each other. We also assessed cognitive abilities and auditory temporal processing in the older adults. LiSN-S results in this group were strongly associated with measures of cognition, measures of temporal processing (tapping the use of fine structure and gap cues), as well as pure-tone averages (PTA) for 9 and 10 kHz, but not PTAs for frequencies in the standard audiometric range.

Session 2pSP

Signal Processing in Acoustics: Acoustic Signal Processing for Various Applications

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Contributed Papers

1:00

2pSP1. Ideal signals and processing for continuous active sonar. Harry A. DeFerrari (Appl. Marine Phys., Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, hdeferrari@rsmas.miami.edu)

The ideal signal for continuous active sonar would be linear in both time and Doppler and have no time or Doppler leakage. Here two signals/signal processing methods are presented that have half the requisite condition—zero time leakage. But we show a way to use the time property to eliminate Doppler interference (clutter) from zero Doppler reverberation (bottomed) sources and from direct arrivals. Here we develop an m-sequence processed with a matched filter approach and a special class of inverted binary sequences processed with a matched-inverse filter. In both cases, there is a perfect pulse correlation property; that is, a large signal when they line up in time and hard zero when they do not. This property is used to develop a Complete Ortho-Normal (CON) data sets in both waveform and pulse response space. The CON pulse response allows elimination of the easily identified zero Doppler reverberation arrivals. Then, reprocessing for Doppler, the zero Doppler signals are cleanly removed as is all of their Doppler leakage. In effect one has the ideal clutter free signal with leakage from direct arrivals and bottom reverberation removed and a target signal displayed in a time-Doppler plane against a noise only background.

1:20

2pSP2. Toward blind reverberation time estimation for non-speech signals. João F. Santos (INRS-EMT, Institute National de la Recherche Scientifique, 800, Rue de La Gauchetière Ouest, Ste. 6900, Montreal, QC H5A-1K6, Canada, jfsantos@emt.inrs.ca), Nils Peters (Qualicomm Technol. Inc., Berkeley, CA), and Tiago H. Falk (INRS-EMT, Institute National de la Recherche Scientifique, Montreal, QC, Canada)

Reverberation time (RT) is an important parameter for room acoustics characterization, intelligibility and quality assessment of reverberant speech, and for dereverberation. Commonly, RT is estimated from the room impulse response (RIR). In practice, however, RIRs are often unavailable or continuously changing. As such, blind estimation of RT based only on the recorded reverberant signals is of great interest. To date, blind RT estimation has focused on reverberant speech signals. Here, we propose to blindly estimate RT from non-speech signals, such as solo instrument recordings and music ensembles. To estimate the RT of non-speech signals, we propose a blind estimator based on an auditory-inspired modulation spectrum signal representation, which measures the modulation frequency of temporal envelopes computed from a 23-channel gammatone filterbank. We show that the higher modulation frequency bands are more sensitive to reverberation than the modulation bands below 20 Hz. When tested on a database of non-speech sounds under 23 different reverberation conditions with early decay time (EDT) ranging from 0.26 to 7.6 s, a blind estimator based on the ratio of high-to-low modulation frequencies outperformed two state-of-the-art methods and achieved correlations with EDT as high as 0.80 for solo instruments and 0.75 for ensembles.

1:40

2pSP3. Feedback active noise control in a crew rest compartment. Delf Sachau and Oliver Pabst (Mechatronics, Helmut-Schmidt-Univ., Holstenhofweg 85, Hamburg, Hamburg D-22043, Germany, sachau@hsu-hh.de)

Active systems for noise cancelation are typically used to reduce low frequency noise (below 500 Hz). In this study, a method to reduce the delay in the control loop of feedback noise control systems is discussed. There are certain applications in which we may choose to control a specific frequency range; hence, a selection of the control-band must be performed. This selection is commonly done with analog filters introducing delay in the signal path. However, in feedback systems, the reference signal is calculated from the error signal which makes their performance more sensitive to these delays. In this contribution, the filtered-error feedback filtered-reference LMS algorithm (FE-FBFxLMS) is analyzed. Consequently, control-band selection is performed in an auxiliary loop removing the additional delay required in the signal path. This method of band selection also allows a reduction of the delay due to the anti-aliasing filters by using faster low-order filters in terms of group delay. With this delay reduction, higher noise attenuation is expected at the cost of a slower convergence rate. This method is applied to control broadband noise in a single channel feedback system in simulation and experiment.

2:00

2pSP4. A polar-coordinate-discretized wave equation finite-difference time-domain simulation for controlling the emission characteristics of sound source. Kota Nakano, Masato Nakayama, Takanobu Nishiura, Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Noji-higashi, Kusatsu, Shiga 525-8577, Japan, cm010064@ed.ritsumei.ac.jp), and Toshiyuki Kimura (Universal Commun. Res. Inst., National Inst. of Information and Commun. Technol., Soraku-gun, Japan)

In sound field simulation with conventional WE-FDTD, the simulated field and wave equation are discretized with symbol time domain and spatial domain with Cartesian coordinate, and sound pressure distribution is calculated. Spatial-discretization with Cartesian coordinate easily achieves calculation and sound synthesis. However, the discretized Cartesian coordinate cannot model sound emission characteristic exactly, because the emission characteristics is under spherical diffusion and discretize Cartesian coordinate cannot define spherical. Emission characteristic of sound source is indispensable for sound field reproduction. Then, we propose a new approach for WE-FDTD method to simulate emission characteristic of sound source object. Polar coordinate discretization is employed in the new approach instead of Cartesian coordinate. The wave equation with spatial Cartesian coordinate is projected onto polar coordinates, and discretized. The discretized polar coordinate is better way to define spherical wave surface than Cartesian one, and it is expected that the emission characteristics are simulated more exactly. The objective evaluations were conducted for the proposed approach. According to the result, the sound field simulation with the

new approach can flexibly define exact emission characteristics. However, the discretized interval of polar coordinate is coarse in distance. Verifying the interval limit is important issue in future work.

2:20

2pSP5. Wind noise reduction using empirical mode decomposition. Kohei Yatabe and Yasuhiro Oikawa (Intermedia Art and Sci., Waseda Univ., 59-407 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, k.yatabe@asagi.waseda.jp)

One common problem of recorded sound outdoors is interfusion of wind noise, which has highly non-stationary characteristics. Although there are a lot of noise reduction methods which produce good results for general kinds of noises, most methods perform worse for wind noise due to its non-stationary nature. Therefore, wind noise reduction need special technique to overcome this non-stationarity. Empirical mode decomposition (EMD) is a relatively new method to decompose a signal into several nonlinear and non-stationary bases which are modeled as amplitude and frequency modulated sinusoids that represent wind noise well. Thus, EMD has a possibility to reduce wind noise from recorded sounds in entirely different way from ordinary methods. In this paper, a preliminary discussion of applying EMD to wind noise reduction is presented. Since EMD decomposes a signal into monocomponent bases, it is easier to treat them as analytic signals via Hilbert transform. Our method utilize this characteristics in order to reduce wind noise. The experiment is performed on female voice superimposed with wind noise and shows it possibility and effectiveness.

2:40

2pSP6. Efficient speech encryption using chaotic cat map for code-excited linear prediction based coders in packet networks. Fatiha Merazka (Telecommunications, Univ. of Sci. & Technol. Houari Boumediene, P.O. Box 32 El Alia, Algiers 16111, Algeria, fmerazka@usthb.dz)

The increasing importance of multimedia applications is placing a great insist on content protection and customer privacy. Communications can be intercepted, especially over wireless links. Since encryption can effectively prevent eavesdropping, its use is widely advocated. The codec G. 729 based CS-ACELP algorithm is standardized as voice codec by ITU-T for multimedia and Voice over Internet Protocol (VoIP) applications. In this paper, we introduce a speech encryption method based chaotic cat map algorithm. Cat map extended to two-dimensional NxN matrix. It takes concepts from linear algebra and uses them to change the positions of the values of the matrix. The result after applying the Cat Map will be shuffled signals that contain the same values of the original signals. We applied our encryption scheme to the standard ITU-T G.729 standard speech coder to evaluate its performance. Simulation results show that G.729 based cat map encryption is very efficient since the encrypted speech is similar to a white noise. The perceptual evaluation of speech quality (PESQ) and enhanced modified bark spectral distortion (EMBSD) tests for speech extracted from TIMIT database confirm the efficiency of our proposed scheme.

3:00–3:20 Break

3:20

2pSP7. An investigation into the relationship between sound quality and demodulation ratio for parametric loudspeakers. Daisuke Ikefuji, Masato Nakayama, Takanobu Nishiura, and Yoichi Yamashita (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, cm000074@ed.ritsumei.ac.jp)

Recently, parametric loudspeakers with a powerful ultrasound have been utilized for reproduction of sound to particular area. Amplitude of the ultrasound is modulated with a target audible sound. The emitted amplitude modulated wave is demodulated into the target audible sound in the air. Demodulation ratio affects sound quality of the reproduced sound with parametric loudspeakers. Moreover, demodulation ratio depends on the distance between the parametric loudspeaker and the listener. Therefore, the listener should utilize parametric loudspeakers at the suitable distance, which is required for sufficient demodulation of the amplitude modulated wave. Thus, we have proposed the criterion for measuring the demodulation ratio. However, it had not been conducted to investigate a relationship between subjective sound quality and demodulation ratio. In this paper, we investigate the relationship for estimation of the suitable distance. We carried out subjective evaluation experiment for confirming sound quality at each distance. As a result, we confirmed the consistency about fluctuation tendency of the subjective sound quality and demodulation ratio at each distance.

3:40

2pSP8. An adaptive noise power spectral density estimation of noisy speech using generalized gamma probability density function. Xin Dang (Information Sci. and Technol., Grad. School of Sci. and Technol., Grad. School of Eng., Hamamatsu, Shizuoka 432-8012, Japan, f5045013@ipc.shizuoka.ac.jp) and Takayoshi Nakai (Information Sci. and Technol., Grad. School of Sci. and Technol., Hamamatsu, Shizuoka, Japan)

An estimation of the power spectral density (PSD) of noise is a crucial part to retrieve speech in a noisy environment. A novel estimation method for non-white noise of noisy speech on the basis of a generalized Gamma distribution is proposed. Because of highly non-stationary nature of speech, its probability density function (PDF) is difficult to derive using any modeling technique, while a segmental noise is more stationary and can be fitted more accurately by a generalized Gamma PDF, which is a natural extension of the Gaussian modeling of a non-white components distribution. In the experiment, different types of non-white noises are added to the clean speech signal at different SNRs to study the estimation of noise using different types of PDF. It is found that non-white noise spectrums fit more accurately on the generalized Gamma PDF with adaptive parameters instead of a Gaussian distribution function. The reported generalized Gamma PDF model shows the best performance to estimate the noise spectral amplitudes as compared with Minimum Statistics (MS), Speech absence Probability (SAP), and MMSE based PSD estimation methods. The performance of the proposed noise estimation is good when it is integrated with the speech enhancement technique as demonstrated by both the subjective and objective measures.

4:00

2pSP9. Markov random field in speech enhancement: Application for tonal languages. Tanawan Saimai, Charturong Tantibundhit (Elec. and Comput. Eng., Thammasat Univ., 99 Moo 18 Phaholyothin Rd., Khlongluang, Pathumthani 12120, Thailand, 5310030068@student.tu.ac.th), Chutamanee Onsuwan (Linguistics, Thammasat Univ., Khlongluang, Pathumthani, Thailand), and Chai Wutiwiwatchai (National Electron. and Comput. Technol. Ctr. (NECTEC), Khlongluang, Pathumthani, Thailand)

This paper proposed speech enhancement algorithm based on Markov random field (MRF) model for Thai, a tonal language. Firstly, a noisy speech signal is transformed using the short time Fourier transform (STFT). In so doing, noise is removed and speech is preserved, especially harmonics information as f0 patterns are relevant perceptual cues for lexical tones. The voice activity detector is used to classify each STFT time frame into voiced and unvoiced. Harmonics information is retrieved from each voiced time frame, where four neighborhoods of the analyzed STFT coefficients include its adjacent time frames (left, right) and nearest harmonics (top, bottom). For the unvoiced, four adjacent coefficients (left, right, top, and bottom) are used. A two-state MRF model is used to classify STFT coefficients into speech and noise. Those with speech state are retained, while the rest is set to zero. The enhanced speech is estimated by the inverse STFT. Results from quality evaluation test on four sets of Thai rhyming words corrupted by white noise at SNR levels of 0, 5, and 10 dB showed that the proposed algorithm significantly improved SNR of noisy speeches compared with spectral subtraction (1.3 dB on average) and Wiener filtering (1.9 dB on average).

4:20

2pSP10. Dereverberation of a closed test section of a wind tunnel with a mult imicrophones cesptral method. Daniel Blacodon and Jean Bulté (DSNA/ACOU, ONERA, 29, avenue de la Division Leclerc, BP N° 72, Châtillon 92322, France, daniel.blacodon@onera.fr)

Today, the manufacturers of aircraft want to perform at the same time aeroacoustic and classical aerodynamic tests in the wind tunnels with closed test sections in order to reduce test duration and costs involved in new aircraft developments. However, this kind of wind tunnels is not optimal for

acoustics testing because the measurements are often masked by background noise (i.e., noise generated by the wind tunnel flow mechanism) and reverberations on the walls without acoustic liner within the test section. This is a serious problem because these unwanted phenomena limit the accuracy of identification, and quantification of acoustic sources. The reduction of background noise can be obtained with subspace techniques or noise subtraction based on a noise reference measurement. Concerning the reduction of the contamination of the echos, it can be achieved with an inverse filtering technique or with a single microphone cepstral method. A new approach is proposed in this study to remove the spurious effects of the echos. It is based on multi microphones cesptral technique. This new technique has been successfully applied on numerical simulations, and experimental data. The theoretical developments of this method and the obtained results will be presented in the full paper.

TUESDAY AFTERNOON, 4 JUNE 2013 — 511AD, 1:00 P.M. TO 5:40 P.M.

Session 2pUWa

Underwater Acoustics and Acoustical Oceanography: Ocean Ambient Noise

Rex K. Andrew, Chair
Appl. Phys. Lab., 1013 NE40th St., Seattle, WA 98105

Contributed Papers

1:00

2pUWa1. Seabed characterization using ambient noise and compact arrays on an autonomous underwater vehicle. Peter L. Nielsen (Res. Dept., STO-CMRE, V.S. Bartolomeo 400, La Spezia 19126, Italy, nielsen@cmre.nato.int), Martin Siderius (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR), Jim Miller (Research Dept., STO-CMRE, La Spezia, Italy), Steven Crocker (Sensors & SONAR Systems Dept., NUWC, Newport, RI), and Jennifer Giard (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Estimating the seabed geoacoustic properties at various fidelity levels has been a research topic for several decades. The majority of the applied seabed characterization techniques often require significant involvement of surface vessels, complex experimental setup, and human interaction. Technical advances in underwater autonomy and the development of energy efficient electronics provide new opportunities to optimize underwater environmental surveys in particular of the seabed. In 2012, the CMRE conducted the GLASS'12 experiment in the Mediterranean Sea with the objective to investigate the feasibility of utilizing a hybrid autonomous underwater vehicle equipped with a compact nose array for long-duration seabed characterization over large areas. The vehicle has the capability of operating in traditional propulsion and glider mode, and the nose-mounted array consists of a 5-element vertical and 4-element tetrahedral array. The sound sources used as information carrier were ambient noise, e.g., sea surface generated noise and loud distant sources of opportunity. The experimental setup together with the newly developed autonomous equipment will be presented and examples of inferred reflection loss and sub-bottom profiling from the ambient noise are compared to ground truth measurements. [Work supported by the STO-CMRE, ONR-G Grant No. N62909-12-1-7040, the ONR N-STAR/ILIR program.]

1:20

2pUWa2. Monterey Bay ambient noise profiles using underwater gliders. Tarun K. Chandrayadula, Chris W. Miller, and John E. Joseph (Oceanography, Naval Postgrad. School, 650 Sloat Ave., # 3, Monterey, CA 93940, tkchandr@nps.edu)

In 2012, during two separate week-long deployments, underwater gliders outfitted with external hydrophones profiled the upper 100 m of Monterey Bay. The environment contains various noises made by marine mammals, ships, winds, and earthquakes. Unlike hydrophone receivers moored to a fixed location, moving gliders measure noise variability across a wide terrain. However, underwater mobile systems have limitations such as instrument and flow noise, that are undesired. In order to estimate the system noise level, the hydrophones on the gliders had different gain settings on each deployment. The first deployment used a 0 dB gain during which the ambient noise recordings were dominated by the glider. The second used two hydrophones, one with a 0 dB gain and the other with 20 dB. Apart from system sounds, the higher-gain hydrophone also recorded far-away sources such as whales and ships. The noise recordings are used to estimate the spectrograms across depth and record time. The spectrograms are integrated with the glider engineering data to estimate histograms of noise power as a function of depth and glider velocity. The statistics from the two different deployments are compared to discuss the value of gliders with external hydrophones in ambient noise studies.

1:40

2pUWa3. Measuring the spatial characteristics of the ambient noise field from an autonomous underwater vehicle. Stephanie Fried and Henrik Schmidt (Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 5-204, Cambridge, MA 02139, eowyn@mit.edu)

For autonomous underwater vehicles (AUVs), the primary method of sensing the local environment is through acoustics. The local noise field contains a wealth of information the AUV uses—from target tracking to communication to general understanding of the environment. A measure of the spatial composition of the ambient noise field can provide details about the physical environment as well as information for the AUV to incorporate into its control decisions. The challenge is in accurately measuring the directionality of the noise field from a single line array and continuously updating this measure to reflect changes in the environment and additional information as the AUV moves. Here we present a method for continuously assessing the spatial characteristics of an ocean ambient noise field measured by an AUV with a towed hydrophone array.

2:00

2pUWa4. Synthetic-array beamforming for bottom-loss estimation using marine ambient noise. Lanfranco Muzi and Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, muzi@pdx.edu)

Previous studies have shown that, using a vertical line array, the bottom loss can be estimated in an ambient-noise field from the output power of beams steered toward the sea surface and beams reflected off the seabed. With short arrays, the low angular resolution of the bottom-loss estimate is one of the main limitations of the approach. *Synthetic-array processing* is proposed as a technique that can improve the angular resolution of the bottom-loss estimate to a level comparable to that of an array with twice as many physical sensors (at equal inter-sensor spacing). The proposed technique follows naturally from a new derivation in frequency-wavenumber

domain of the bottom-loss estimation procedure. The conditions under which the approach can be successfully applied are analyzed, with particular regard to the need for the array cross spectral density matrix to be "close" to Toeplitz. A metric is proposed for assessing this particular circumstance in practice. The technique is illustrated using synthetic data from simulations and then applied to data from several experimental campaigns. Results show that a 16-element synthetic array can achieve an angular resolution comparable to that of a 32-element array.

2:20

2pUWa5. Flow noise measurements at strong tidal current area in Uldolmok Waterway. Myungkwon Ko and Jee Woong Choi (Dept. of Marine Sci. and Convergence Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 426-791, South Korea, buymk@hanyang.ac.kr)

Flow noise is a kind of hydrodynamic noise, which is created by turbulent flow in the boundary layer around the hydrophone. Although the turbulent pressure is not a true acoustic noise in that its influence decreases rapidly with distance, it acts as a self-noise source. In general, since the spectrum level of flow noise has been reported to increase rapidly with the increment of flow speed, it is possible to monitor the current velocity from the flow noise measurements. Uldolmok waterway in Korea is one of the locations where currents are very strong, with maximum speed of about 5 m/s. The measurements of flow noise were conducted in Uldolmok waterway using two different shapes of hydrophone. In this talk, the flow noise spectra for various flow speeds will be presented for the frequency range of 20–100 Hz, and the comparison of spectra between two different-shape hydrophones will be given.

2:40

2pUWa6. On the origins of ambient biological sounds in shallow water tropical ecosystems. Simon E. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., 3725 Miramar St. Apt. C, La Jolla, CA 92037, sfreeman@ucsd.edu), Forest L. Rohwer, Allison Gregg, Laura Coleman (Rohwer Lab., San Diego State Univ., San Diego, CA), and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Although discovered more than 60 years ago, the origins of much ambient underwater biological noise remain unclear. Snapping shrimp sounds dominate some environments but elsewhere the shallow-water biological sound field is often heterogeneous. Here we show that dominant components of underwater ambient noise recorded on coral reefs around five islands in the central Pacific may be caused by the interaction of hard-shelled benthic macro-organisms with the substrate. Recordings show a consistent, nightly 4.7 to 6.9 dB increase in estimated pressure spectral density level in the 11 to 17 kHz band with a spectral peak centered between 14 to 15 kHz. Underwater time-lapse photography reveals a marked night-time increase in benthic invertebrate activity at most locations, temporally consistent with the increase in pressure spectral density level. Intensity-filtered recordings of an example species, the hermit crab *Clibanarius diugeti*, in quiet aquarium conditions reveal that transient sounds produced by interaction between the crustacean's carapace, shell, and coral substrate are spectrally consistent with the central Pacific recordings. Passive acoustic monitoring of such ambient noise may be useful as a complementary ecological survey technique to SCUBA-based visual observations, which are typically poor in estimating the abundance and diversity of cryptobenthic organisms.

3:00

2pUWa7. Prediction of underwater noise and far field propagation due to pile driving for offshore wind farms. Stephan Lippert, Tristan Lippert, Kristof Heitmann, and Otto von Estorff (Inst. of Modelling and Computation, Hamburg Univ. of Technol., Denickestr. 17, Hamburg 21073, Germany, s.lippert@tu-harburg.de)

Wind energy plays a key role toward a greener and more sustainable energy generation. Due to limited onshore areas and possible negative effects on human living space, offshore wind parks become increasingly popular. During construction by pile driving, however, high levels of underwater sound emission are observed. To avoid negative effects on marine mammals and other sea life, different approaches, like, e.g., bubble curtains or cofferdams, are currently investigated to cut down the sound pressure levels. In order to predict the expected underwater noise, both with and without sound damping measures, numerical simulation models are needed to avoid complex and costly offshore tests. Within this contribution, possible modeling strategies for the prediction of underwater noise due to pile driving are discussed. Different approaches are shown for the direct adjacencies of the pile and for the far field sound propagation. The effectivity of potential noise mitigation measures is investigated using a detailed finite element model of the surroundings of the pile. The far field propagation in the kilohertz range at distances of several kilometer from the pile, on the other hand, is computed by a model based on wavenumber integration. Finally, the model validation with corresponding offshore tests is addressed.

3:20

2pUWa8. Wind-dependence of low-frequency ambient noise in the deep-ocean sound channel. Stephen Nichols (Grad. Prog. in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu) and David L. Bradley (Appl. Res. Lab., The Penn State Univ., State College, PA)

In the low-frequency range (1–125 Hz), the deep-ocean ambient noise field is produced by seismic, marine life, ship traffic, and wind-dependent hydrodynamic noise mechanisms. This study focuses on the contribution of wind-related source mechanisms to the overall ambient noise field, as well as previous attempts to understand the physics of these mechanisms. The Comprehensive Nuclear-Test Ban Treaty Organization (CTBTO) hydroacoustic monitoring system has produced nearly continuous recordings of the low-frequency deep-ocean ambient noise field at sites in the Pacific, Atlantic, and Indian Oceans, each spanning several years in length. Additionally, wind speed data have been recorded at the host island of each station by the National Oceanic and Atmospheric Administration (NOAA). Correlation techniques are used with these two datasets to determine the relationship between wind speed and the sound level in different frequency bands, and to determine the prominence of wind-related noise in the combined ambient noise spectrum. Results from the three sites are compared to each other to assess the uniformity of wind-generated noise over the world's ocean basins.

3:40

2pUWa9. Model for underwater noise radiated by submerged wind turbine towers. Todd Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (ARL:UT, P.O. Box 8029, Austin, TX 78713, hamilton@mail.utexas.edu)

Sustained tonal noise radiated by towers supporting offshore wind turbines contains energy in frequency bands that may disturb marine mammals, or interfere with passive sonar and seismic sensors and underwater communication equipment. Understanding the generation and propagation of underwater noise due to the operation of wind farms is important for determining strategies for mitigating the environmental impact of these noise sources. An analytic model based on a Green's function approach was previously developed for the sound radiated in the water column by a pulsating cylindrical structure embedded in horizontally stratified layers of viscoelastic sediment [Hay *et al*., J. Acoust. Soc. Am. **130**, 2558 (2011)]. This model has since been adapted to include relaxation and viscous losses in seawater and empirical loss factors for the sedimentary layers. In order to validate the model simulations were compared with reported measurements collected near an operating wind turbine that include radial acceleration of the tower, taken to be the source condition, and sound pressure levels in the water column. For long-range propagation over range-dependent environments, the analytic model has been coupled to a parabolic equation code. Simulations are presented for several bathymetries, sediment types, and tower array configurations. [Work supported by Department of Energy DE-EE0005380.]

4:00

2pUWa10. A quasi-analytic model of the underwater sound signal from impact driving of an offshore semi-infinite pile. Marshall V. Hall (Marshall Hall, 9 Moya Crescent, Kingsgrove, NSW 2208, Australia, marshallhall@optushome.com.au)

A quasi-analytic model is derived for the underwater sound signal radiated when an offshore pile is struck on its face by a hammer. The pile is modeled as a semi-infinite cylindrical shell of an elastic solid. The impact generates a pulse of vibration that travels down the pile at the longitudinal

sound-speed. At a given distance below the pile face, the axial displacement after the peak has arrived decreases exponentially with time. There are two coupled equations of motion for the axial and radial displacements. A closed form expression is derived for the radiated sound pressure (which is proportional to the radial acceleration) in terms of the Poisson ratio and Young's Modulus of the pile material, the hammer velocity, contact area between hammer and pile, pile radius, hammer mass, the pile's longitudinal sound-speed, and the sound-speed and density of the external medium. This model is applied to a published scenario for which the radiated sound pressure had been computed using a Finite Element Model, but is found to produce a different result. Some assumptions used in the model are identified that may explain the difference.

4:20

2pUWa11. Implementing physical constraints for noise only normal mode shape estimation. Ian M. Rooney, John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts at Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, irooney@umassd.edu), and Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Many underwater acoustic tasks employ a vertical line array to sample a continuous wave pressure field. In the absence of strong local sources, the noise sampled by an array consists of both a spatially correlated and a spatially uncorrelated component. The correlated component is generally due to distant sources and can be described in terms of the modes of the waveguide. If the number of propagating modes is less than the array size, the diffuse noise lies within a lower dimensional subspace than the array data. The mode shapes defining this subspace can be estimated from noise measurements. Propagation physics constrain the mode shapes defining this subspace to be real, but the basis vectors obtained from a singular value decomposition of noise snapshots are generally complex. A phase rotation for each basis vector is required to rectify this. This work proposes a weighted average of the phase angles for each element that minimizes the variance of the rotation angle estimate. Simulations compare the proposed algorithm against prior approaches such as rotating each basis vector by the phase of the largest magnitude sample, rotating by the average of the phase, or taking the magnitude and ignoring the phase. [Work supported by ONR.]

4:40

2pUWa12. Analysis of the vertical structure of deep ocean noise using measurements from the SPICEX and PhilSea experiments. Kathleen E. Wage, Mehdi Farrokhrooz (Elec. and Comput. Eng. Dept., George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA)

There are open research questions about the vertical structure of low-frequency ambient noise in deep water. For example Gaul *et al.*'s [*IEEE JOE* (2007)] analysis of the Church Opal data set showed that noise decreases substantially (on the order of 20 dB) below the critical depth, whereas other researchers have reported more modest reductions [Morris, *J. Acoust. Soc. Am.* (1978)]. Two deep water experiments provided a unique opportunity to measure ambient noise using large vertical arrays. In 2004–2005, SPICEX used two arrays to sample a North Pacific environment. One array was centered on the sound channel axis, and the other array had hydrophones above and below the critical depth. In 2010–2011, the PhilSea experiment deployed a single array with 150 hydrophones spanning the full water column. Both experiments made repeated short measurements (each 2–3 min long) of the field at the arrays. This talk compares the ambient noise observed during SPICEX and PhilSea with results reported in the literature. Since these data sets contain receptions over the period of a year, we focus on the seasonal dependence of the noise field. In addition to investigating noise level as a function of depth, we consider wind dependence and vertical directionality.

5:00

2pUWa13. Modeling of underwater piling noise mitigation using an array of soft spheres in the ocean. Keunhwa Lee (Ocean Eng., Seoul National Univ., Kwanak-ro 1, Seoul 151742, South Korea, nasalkh2@snu.ac.kr), Kyungmin Baik (CFFA Div. of Physical Metrology, Korea Res. Inst. of Standards and Sci., Daejeon, South Korea), and Woojae Seong (Ocean Eng., Seoul National Univ., Seoul, South Korea)

The ocean noise generated by marine piling affects severely fish, other marine life, and fishery activities. Accordingly, a few kind of noise mitigation system are presented. Among them, the noise mitigation system using soft scatterers such as air bubbles or rubber spheres is reported to show higher noise reduction than the classical cofferdam system composed of mass-absorbing materials. In this proceeding, a numerical scheme is developed to model and design the noise mitigation system using an array of soft spheres. This scheme is originally based on self-consistent equation of Zhen Ye for multiple scattering [Z. Ye and A. Alvarez, Phys. Rev. Lett. **80**, 3503 (1998)]. We generalize the original self-consistent equation for the oceanic waveguide using the waveguide green function. This generalized self-consistent equation is useful to model the noise propagation through an array of soft spheres in the ocean and assess the ability of the noise mitigation system. The effect of the oceanic waveguide on the noise reduction is studied and the validity of effective medium approach for a bubbly layer is also analyzed numerically.

5:20

2pUWa14. Spatial filtering in ambient noise crosscorrelation. Olivier Carriere, Peter Gerstoft, and William S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr, La Jolla, CA 92093-0238, ocarriere@ucsd.edu)

Ambient noise crosscorrelation is an attractive approach for estimating the properties of a propagation medium (presence of scatterers, wave propagation speed, etc.) without having to deploy active acoustic source. According to the theory, Green's function between pair of receivers can be extracted from the crosscorrelation of the received signals, given that the noise source is diffuse. In practice, this ideal condition is rarely met in the ocean and directional sources, often related to human activity in the vicinity of the receivers, may bias the travel time estimates. Using array processing, matrix spatial filters can be built independently from the environment properties (as opposed to model-based techniques) to filter specific directions of arrival. Such filter might be useful to remove an interferer or to restrict the effective source distribution to a reduced cone area. Here we study the use of matrix spatial filters for removing unwanted contributions in the crosscorrelations. Based on theory and simulations, we discuss the effect of array size, dimension and geometry, processed frequency band, and environmental conditions in the filtered crosscorrelations. An example of application to real data from the Shallow Water '06 experiment further illustrates the potential of the method.

Session 2pUWb

Underwater Acoustics and Acoustical Oceanography: Arctic Acoustics and Applications

Stan E. Dosso, Cochair
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Juan I. Arvelo, Cochair
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Chair's Introduction—12:55

Invited Papers

1:00

2pUWb1. Under-ice shallow-water sound propagation and communication in the Baltic Sea. Erland Sangfelt, Sven M. Ivansson, and Ilkka Karasalo (Underwater Res., Swedish Defence Res. Agency, FOI kista, Stockholm SE-16490, Sweden, erland.sangfelt@foi.se)

Sound propagation under ice in the Baltic is of interest for military as well as civilian purposes. Important questions are, for example, how sonar systems are affected by ice in shallow waters and how marine mammal life under ice is affected by ship traffic. Changing climate and increasing ship traffic are factors of great concern also in the Arctic region. Modeling results for sound propagation under ice in the Baltic Sea are presented for low as well as high frequencies. The sound propagation is influenced by bottom as well as ice-cap interaction in these shallow waters. The low- and high-frequency modeling is performed with wavenumber and ray models, respectively. Both types of models are amended to include a solid ice layer on top of the water column. In addition, the modeling efforts are extended to study the performance of an underwater communication system developed at FOI. The modeling results are evaluated using data from sea trials in the Gulf of Bothnia under ice-covered and ice-free conditions.

1:20

2pUWb2. Propagation of seismic exploration noise in the Marginal Ice Zone. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no) and Hanne Sagen (Nansen Environ. and Remote Sensing Ctr. (NERSC), Bergen, Norway)

This paper presents data from ambient noise recordings in the Marginal Ice Zone (MIZ) in the Fram Strait (deep water) and Barents Sea (shallow water) with focus on noise due to seismic exploration activity. Data were recorded on fields of sonobuoys deployed from P-3C aircraft in June 2011 and May 2012, each covering a 100 km x 100 km area. Recordings from three bouys in each area, from open water to within the solid ice cover, are analyzed for low-frequency (10 Hz–1 kHz) noise levels, with broadband noise due to distant seismic exploration identified. Strong attenuation of this noise component with distance into the MIZ is observed in the shallow-water data, whereas noise levels decrease less with distance in the deep-water data despite under-ice conditions. Propagation loss is modeled using a raytrace model including under-ice reflection loss due to an elastic ice cover, with environment input from the TOPAZ coupled ocean-sea ice model and from satellite images. Properties of the noise field, modeling results, and model-data discrepancies will be discussed. [Data collected under the ACOBAR and WIFAR projects at NERSC.]

1:40

2pUWb3. Modelled and measured sound levels from a seismic survey in the Canadian Beaufort Sea. Marie-Noël R. Matthews and Alexander O. MacGillivray (JASCO Appl. Sci., 202 - 32 Troop Ave., Dartmouth, NS B3B 1Z1, Canada, Marie-Noel.Matthews@jasco.com)

JASCO Applied Sciences performed acoustic modeling and measurements to calculate marine mammal exclusion zones for Chevron Canada Limited's 2012 Sirluaq 3-D seismic program in the Canadian Beaufort Sea. The Sirluaq survey was located in deep water (>650 m), on and beyond the continental slope, and presented unique challenges from both modeling and measurement standpoints. The modeling was performed with JASCO's Marine Operations Noise Model (MONM), which uses a parabolic-equation-based algorithm to accurately predict $N\times2$-D sound propagation in ocean environments. Sound levels were measured with five calibrated Autonomous Multichannel Acoustic Recorder (AMAR) systems at distances of 50–50 000 m from the airgun array, in water depths ranging from 50 to 1500 m. The sensors were laid out to capture sound levels in both the broadside (perpendicular to survey line) and endfire (along the survey line) directions. The high-resolution digital recordings of seismic sounds were analyzed to determine peak and rms sound pressure levels (SPL), and sound exposure levels (SEL) as functions of range from the airgun array. The model estimates were generally conservative; however, the model predictions at the specific depth of the receivers accurately predicted the existence of a shadow zone and the overall transmission loss trend.

2:00

2pUWb4. Arctic ambient noise measurements in support of the Northern Watch Project. Garry J. Heard, Nicos Pelavas, Sean Pecknold (Atlantic, Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, garry.heard@gmail.com), Carmen E. Lucas (Dartmouth, Nova Scotia, Canada), and Bruce Martin (Jasco Res., Dartmouth, NS, Canada)

During August 2012, acoustic recording systems were deployed in Barrow Strait as part of the Defence Research and Development Canada (DRDC) Northern Watch Technology Demonstration Project. Two Starfish Sensor Cubes each with a 1-m cube of seven hydrophones operating in the frequency range of 5–750 Hz, and two single-hydrophone, Autonomous Multichannel Acoustic Recorders (AMAR) providing a 30-kHz signal bandwidth were deployed. The Starfish were deployed for two one-week intervals. One AMAR was deployed for two weeks partially overlapping the Starfish deployment. The second AMAR was deployed for a period of one year with recovery planned for August 2013. The observed underwater noise picture is one of high variability ranging from an extremely quiet to a noisy environment. Noise sources included: A 500-m long iceberg grounded within 500 m of one of the Starfish; a large ice island (4–5 sq-km) that passed within 4 km of the sensors; a small number of motoring vessels; significant wind events that caused rapid and strong variations in the noise field; and a small number of marine mammal detections. After our departure, a large number of Beluga whales were observed visually. The remaining AMAR may detect these late summer visitors.

Contributed Papers

2:20

2pUWb5. Acoustic manifestations of frozen bubbles. Alexey Maksimov (Phys. of the Ocean, Pacific Oceanological Inst. Far Eastern Branch of the Russian Acad. of Sci., 43, Baltic St., Vladivostok 690041, Russian Federation, maksimov@poi.dvo.ru)

The hydrocarbon seeps emitting buoyant bubble plumes from seafloor vents have been actively investigated in different regions of the World Ocean. In winter, rising bubbles, which have reached the sea surface, freeze in ice. These clouds of the frozen bubbles are observed in Arctic seas and represent a common element of an ice cover of lakes. Deposits of gas hydrates were discovered in many marginal seas of the World Ocean. The nature of the relationship between the acoustic and physical properties of gas hydrates is still under debate for both saturated sediments and sediment containing free-gas bubbles. Acoustic manifestations of bubbles frozen in ice and gas hydrate are the subjects of the current study. On the basis of the general solution of the elastic wave scattering problem for the sphere, the scattering cross section for a bubble frozen in an ice has been found. The derived expressions coincide in the limiting cases with known results: a bubble in a rubber-like media, where the longitudinal wave speed is significantly faster than the transverse wave speed and an empty cavity in the elastic media. The structure of low-frequency resonances of bubbles clouds of the simplest geometry has been investigated.

2:40

2pUWb6. Source bearing and range estimation using an ice-mounted triaxial geophone. Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper presents results from Arctic field trials to estimate the bearing and range of an acoustic source in the water column using seismic particle motion measured at a tri-axial geophone on the sea ice surface. Measurements were carried out on smooth, rough, and ridged annual ice, and on a multi-year ice floe. At each site, a hammer seismic study was carried out to selectively excite various ice seismic waves and investigate their propagation properties. Impulsive acoustic sources were deployed in the water at a variety of bearings and ranges from 0.2 to 50 km. Source bearings are estimated by applying polarization filters to suppress shear waves with transverse particle motion and computing the incident power as a function of radial look angle; the inherent 180-degree ambiguity is resolved by requiring prograde particle motion in the vertical-radial plane. Results indicate good bearing estimation (<10-degree average errors) at all ranges with little dependence on ice type. Source range is estimated from the time difference between the water-borne arrival and the critically-refracted longitudinal plate (Lp) wave. Results are limited due to strong attenuation of the Lp wave, with good range estimation to <1 km for smooth annual ice and <0.5 km for other sites.

3:00

2pUWb7. Measurements and modeling of transmission loss variability in Barrow Strait. Sean Pecknold, Nicos Pelavas, and Garry Heard (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca)

During the summer of 2012, a field trial was held in Barrow Strait, south of Devon Island in the Canadian Arctic. The trial included a set of acoustic transmission loss experiments recorded on Starfish Sensor Cubes, which include a 1-m cube of seven hydrophones operating in the frequency range of 5–750 Hz. The transmission loss runs consisted of 10-min and 20-min duration transmissions of 400 and 500 Hz tones made at a discrete set of distances up to 60 km from the recorders. Supporting environmental measurements included sets of CTD (conductivity-temperature-depth) profiles and bathymetric measurements. The effects of the measured environmental properties and variability are investigated via propagation modeling, and compared to the experimental data acquired during these experiments.

3:20

2pUWb8. A parabolic equation for under ice propagation. Adam M. Metzler (Environ. Sci. Lab., Appl. Res. Lab.: The Univ. of Texas at Austin, ARL-Environ. Sci. Group, P.O. Box 8029, Austin, TX 78713, ametzler@arlut.utexas.edu), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, Texas)

Parabolic equation methods are useful to accurately and efficiently model propagation in the ocean for range-dependent environments. Most methods treat environments where the ocean surface is a flat perfect reflector. For some problems, the ocean surface characterized as a random rough scattering water-air boundary produces significant effects on the overall propagation. A rough surface parabolic equation has been developed [A. P. Rosenberg, J. Acoust. Soc. Am. **105**, 144 (1999)] that extends the split-step Padé approach in the RAM implementation, and others have extended split-step Fourier methods. For an upper ice surface, elasticity should be incorporated into the rough scattering boundary. In this paper, a parabolic equation is presented that captures effects of seismo-acoustic interactions and scattering from a rough surface modeled as rigid as opposed to pressure release. Particular attention is given to upward-refracting sound speed profiles, typically found in Arctic environments, which encourage interactions with an ice surface. The scattering technique can be extended to environments with fluid/ice interfaces. [Work supported by ARL:IR&D.]

3:40

2pUWb9. Effective ice model for under-ice propagation using the fluid-fluid parabolic equation. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Richard L. Campbell (OASIS Inc., Seattle, Washington), and Lee Freitag (Woods Hole Oceanogr. Inst., Woods Hole, Washington)

An approach is presented to permit efficient computation of the waterborne acoustic field in the presence of sea ice. The range-dependent wide-angle parabolic equation (PE) is used to model the acoustic field with a hard, lower-density layer (ice) placed above the ocean and seafloor. The ice layer is characterized by its thickness, compressional speed, density, and attenuation. Acoustic loss due to sea ice is primarily driven by conversion to shear waves, and in this model the effect will be approximated by attenuation within the ice layer. Rough interface scattering at the air-ice and ice-water interfaces will be handled by generating range-dependent realizations from a data-derived ice thickness model. An inversion for ice parameters is conducted using the work of Jin *et al*. [J. Acoust. Soc. Am. **96**(5)] and by matching the transmission loss of Diachok [J. Acoust. Soc. Am. **59**(5)]. Model-data comparisons between this three-layer PE and measurements the taken in the Fram Strait in 2010 and in the Canada Basin in 2011 will be presented.

TUESDAY EVENING, 4 JUNE 2013 — 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings beginning at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday are as follows:

Architectural Acoustics	513abc
Engineering Acoustics	512ae
Musical Acoustics	512dh
Physical Acoustics	519b
Psychological and Physiological Acoustics	514abc
Structural Acoustics and Vibration	512cg

Session 3aID

Interdisciplinary: Plenary Lecture: Objective Evaluation of Musical Instrument Quality: A Grand Challenge in Musical Acoustics

Thomas D. Rossing, Chair
Stanford Univ., Stanford, CA 94022

Chair's Introduction—7:55

Invited Paper

8:00

3aID1. Objective evaluation of musical instrument quality: A grand challenge in musical acoustics. D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk)

Over the last few decades, increasingly sophisticated experimental and computational studies have clarified the processes involved in sound production in musical instruments. One of the principal goals of this research effort has, however, remained tantalizingly elusive: the establishment of clear and unambiguous relationships between objectively measured properties of an instrument and judgments of its musical qualities by an experienced player. This is partly because player evaluation is a subtle and highly subjective process in which many different aspects of the instrument's performance may be tested. Early studies concentrated on the steady state spectra of sound recorded in the far field of the instrument. More recently, it has been recognized that transient aspects of an instrument's performance are important in judgments of quality made by performers. These aspects include the ease with which a stable regime of oscillation can be initiated, and the flexibility with which pitch, amplitude, and timbre can be modified during performance. Attempts to define "playability" of an instrument in scientific terms, and to relate these scientific metrics to the vocabulary used by performers in judgments of playability, have been partially successful, but many questions remain unanswered.

WEDNESDAY MORNING, 5 JUNE 2013 513ABC, 8:55 A.M. TO 12:00 NOON

Session 3aAAa

Architectural Acoustics and Musical Acoustics: Virtual Concert Hall Acoustics I

Sungyoung Kim, Cochair
RIT, ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623

Wieslaw Woszczyk, Cochair
Music Res., McGill Univ., Schulich School of Music, 527 Sherbrooke St. West, Montreal, QC H3A1E3, Canada

Chair's Introduction—8:55

Invited Papers

9:00

3aAAa1. Towards the state of the art in virtual acoustics technology. Wieslaw Woszczyk, Doyuen Ko, and Jonathan Hong (Music Res., McGill Univ., Schulich School of Music, 527 Sherbrooke St. West, Montreal, QC H3A1E3, Canada, wieslaw@music.mcgill.ca)

Active acoustics enhancement systems have been applied in room acoustics for decades yet it would be hard to say that their ultimate performance has been achieved, even with the power of today's fast digital processors. There are many challenges ahead including the creation of homogeneous directionless diffuse field using a limited number of discrete loudspeaker sources. There is also a need to create soundfield having some localized directional properties mimicking the phenomena found in acoustics of musical instruments interfacing the acoustics of rooms. Some solutions to these challenges will be presented and discussed by the authors.

9:20

3aAAa2. Active field control using sound field generation technique—Case study of a Live concert at a virtual Renaissance church. Takayuki Watanabe, Masahiro Ikeda (Spatial Audio System, Yamaha Corp., 10-1 Nakazawa-cho, Nakaku, Hamamatsu 430-8650, Japan, takayuki_watanabe@gmx.yamaha.com), and Sungyoung Kim (ECTET, Rochester Inst. of Technol., Rochester, NY)

In the Renaissance area, musical culture was centered at and spread across churches. In order to appreciate Renaissance music, therefore, it is important to account for the influence of acoustics of churches at that era so that audiences today can experience homogeneous musical appreciation. We used an Active Field Control system to create the acoustics using measured the impulse responses (IRs) of a church. The system consists of directional microphones, head amps, a convolution engine, a matrix processor, amplifiers, and loudspeakers. And it picks up the direct response of performance, convolves it with the measured IRs, and reproduces the resulting sound using loudspeakers around the room. Loudspeaker positions in the performance hall are equivalent to the positions where the IRs had been measured at the church. This technique allows us to convincingly recreate not only the reverberation time but also the spatial impressions of the church at the performance hall. In addition, we modified the IRs so that the inherent acoustical characters of the performance space would less influence to recreation of the target church acoustics. We used a 6-channel recording/reproduction system to evaluate the modification to the IRs and the impressions of the recreated acoustics in the room. This paper presents the results of the experiment.

9:40

3aAAa3. Optimizing acoustics for spoken word using active acoustics. Steve Ellison and Pierre Germain (Meyer Sound Labs., Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com)

Teleconferencing, classrooms, lectures, drama, and worship all rely on spoken word to convey a message. The successful receipt of the message is largely dependent on the acoustic of the room, the vessel for the message, both in amplified and unamplified situations. A room that supports teleconferencing well will have minimal early reflections and reverberation, yet the same room may be used in a classroom environment that would benefit from early reflections. Active acoustic systems can be used to provide this acoustic energy. Early Reflection Benefit (ERB) will be revisited, and active acoustic systems utilized for speech in various contexts will be described.

10:00

3aAAa4. Evaluation of stage acoustics preference for a singer using oral-binaural room impulse responses. Luis A. Miranda Jofre, Densil A. Cabrera, Manuj Yadav (Faculty of Architecture, Design and Planning, Univ. of Sydney, 5/27 Fisher St., Petersham, Sydney, NSW 2049, Australia, lmir9852@uni.sydney.edu.au), Anna Sygulska (Faculty of Architecture, Poznan Univ. of Technol., Poznan, Greater Poland, Poland), and William L. Martens (Faculty of Architecture, Design and Planning, Univ. of Sydney, Sydney, NSW, Australia)

There are two main objective measurement methods in current practice that can be used to evaluate the stage acoustic conditions for singers. One is the stage support metrics (ST_{Early} and ST_{Late}, included in the standard ISO 3389-1), and the other is the voice support metrics proposed by Pelegrín-García (room gain (G_{RG}) and voice support (ST_V)). All of these metrics use energy integration from impulse responses to derive the acoustic descriptors. This overlooks two potentially important features of the responses: the temporal distribution of the impulse response within the evaluation period, and the directional distribution for the spatial impulse response within the evaluation period. In this paper, a method to study the effect of these features is proposed and tested. This method is based on the auralization of ones' own voice in rooms using oral-binaural room impulse responses (OBRIRs). The OBRIRs used are created by combining synthesized early reflections with a recorded reverberant tail. The early reflections are manipulated in their arrival time, arrival direction, and strength. Results of a pilot study indicate that a wide range of on-stage acoustic quality can be achieved for responses showing a given ST_{Early} value due to variation in the temporal and spatial distribution of reflected energy.

10:20

3aAAa5. Augmented stage support in ensemble performance using virtual acoustics technology. Doyuen Ko, Wieslaw Woszczyk, and Jonathan Hong (Sound Recording, Music Res., McGill Univ., Schulich School of Music, 555 Sherbrooke west, Montreal, QC H3A 1E3, Canada, doyuen.ko@mcgill.ca)

Perceptual evaluation of electronically varied acoustic conditions has been perfomed with 15 musicians in a chamber orchestra rehearsal. Using Virtual Acoustics Technology (VAT), a room impulse response based active acoustic enhancement system, four different acoustic conditions were presented to the musicians. Condition 1 was the natural room itself without the VAT system. Condition 2 had a moderate level of VAT enhancement with approximately 10% increased EDT value from the condition 1. Condition 3 duplicated the condition 2 except for another 10% rise of EDT, and condition 4 offered enhancement utilizing multiple early reflections IRs without extending the reverb time of the space. The conditions were presented to the ensemble in random order and the loudness of all four conditions was carefully matched within 1 dB variance. Evaluation results indicated that 65% of participating musicians prefered condition 4 and their preference was highly correlated with perceptual attributes such as "feeling of stage support," "feeling of intimacy," and "quality of reverberation." The objective measurements also confirmed the improvements in stage acoustics support parameters (ST1 and ST2) in condition 4.

10:40

3aAAa6. Sound Cask: Music and voice communications system with three-dimensional sound reproduction based on boundary surface control principle. Yusuke Ikeda and Shiro Ise (Faculty of Eng., Kyoto Univ., Yasaka Shijo building 6F, Tateuri nakano-cho 106, shimogyou-ku, Kyoto-shi, Kyoto-fu 6008006, Japan, ae-yusuke-ikeda@archi.kyoto-u.ac.jp)

To reproduce a highly realistic sound field reproduction, we have developed a 3-D sound reproduction system based on the boundary surface control (BoSC) principle. We set up an Internet connection among the systems, which enables distant speakers to carry out voice telecommunication by simultaneously hearing the same sound field and sensing the other speakers' simulated positions as if they were in the same location. In this system, a listener can freely move his head, because this system reproduce the sound not only at points near his ears but also an area around his head. In this paper, we introduce the "Sound Cask," which is a 96-channel sound reproduction system

based on the BoSC principle. The Sound Cask is used to realize musical telecommunication as if the performers were all playing in the same hall. The system provides a space large enough to play a small musical instrument such as a violin, which allows listening and communication accompanied with the natural body movements of the performer. Another specific feature of the system is that loudspeakers are installed in every possible direction except the floor. The system is designed to be portable so that it can be assembled or disassembled when needed.

11:00

3aAAa7. Subjective evaluation of a virtual acoustic system: Trials with three-dimensional sound field reproduced by the "Sound Cask". Maori Kobayashi (Meiji Univ., 4F Bell Shimokitazawa, 2-36-9 Kitazawa, Setagaya, Tokyo 155-0031, Japan, te11001@meiji.ac.jp), Kanako Ueno (Meiji Univ., Kanagawa, Japan), Mai Yamashita, Shiro Ise (Kyoto Univ., Kyoto, Japan), and Seigo Enomoto (National Inst. of Information and Commun. Technol., Kyoto, Japan)

It has been necessary to establish subjective measures for the performance of the virtual acoustic systems. In this paper, we report our trials to evaluate the performance of a three-dimensional sound field reproduction system based on the boundary surface control principle, the "Sound Cask." First, we introduce our investigations for the experts of audio engineering in order to clarify the difference of auditory impression between the Sound Cask and conventional audio systems. Second, we report psychological and physiological experiments focusing on the advantageous points of the Sound Cask, localization performance, and a clear sense of reality, which were pointed out in the investigations for the experts. Finally, we discuss the issues to be considered for subjective evaluation of virtual acoustic systems for future studies.

Contributed Papers

11:20

3aAAa8. Modeling and simulation of Gamelan Bali concert hall based on objective acoustic parameters. Ni Putu Amanda Nitidara, I G. Nyoman Merthayasa, and Joko Sarwono (Eng. Phys., Bandung Inst. of Technol., Program Studi Teknik Fisika Institut Teknologi Bandung Jalan Ganesha 10 Gedung Lab. Teknologi VI Lantai 2, Bandung, West Java 40132, Indonesia, amandanitidara@yahoo.co.id)

Gamelan Bali music performances require special place to highlight the quality of the acoustic performances. A concert hall dedicated for Gamelan Bali was proposed to perform a better acoustic quality. Studies on Gamelan Bali has been done to achieved an optimum value of sound fields in Gamelan Bali concert hall. This paper shows a geometrical model designed to fulfill the suitable sound fields of Gamelan Bali Concert Hall. The model was modified from shoebox shaped with rear and side balconies. The acoustic performances of the model at 1 kHz summarized as follows: Tsub = 1.41 s, LL = 80.2 dBA, $\Delta t1 = 39.43$ ms, and IACC = 0.33. Those values meet the optimum values from the result of previous studies. Auralization of sound in the room was also done for the purpose of subjective judgment.

11:40

3aAAa9. Electronic architecture—Recent developments in the design, implementation, and performance of time variant acoustic enhancement systems. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

When an electro-acoustic enhancement system is integrated with architectural surfaces in an enclosed volume, the acoustical conditions experienced throughout the venue result from the interaction between the two. Over two decades, we have encountered performance venues with a variety of sizes, shapes, and configurations. Each venue presents unique challenges in configuring architectural elements, as well as the makeup of the electro-acoustic enhancement system, in order to meet the expectations of the users and the audience. This paper describes important elements germane to a successful outcome—from the essential qualities of system components, to the physics and physiology that enable humans to perceive sonic quality. Examples of performance spaces with unique physics and system integration will be discussed.

WEDNESDAY MORNING, 5 JUNE 2013 513DEF, 9:00 A.M. TO 11:40 A.M.

Session 3aAAb

Architectural Acoustics: Architectural Acoustics Potpourri

Roger M. Ellingson, Chair
RM Ellingson Design & Development, LLC, 8515 SW Barnes Rd., Portland, OR 97225

Contributed Papers

9:00

3aAAb1. Assessing the acoustics of offices: A case of standardization. Marc Asselineau (Acoustics, PEUTZ & Assoc., 10 B rue des Messageries, Paris F75010, France, m.asselineau@peutz.fr)

Standards are supposed to help promote better understanding between the various interested parties. As such they can help an end user understand what the acoustic performances of his future premises will be… as long as such a standard exists. Over the years, various standards have been developed by several countries to characterize the acoustics of non partitioned offices. The tactics developed by ISO follow the Scandinavian standards trend with emphasis on the spatial sound level decay in the furnished space while the north American standards are more concerned with speech discretion. Are those tactics really opposed, or are they complementary? This paper intends to submit a comparison of a few standards (ISO, NF, CAN) applied on a couple of representative cases. It turns out that such standards represent the habits and practice of those who wrote it.

9:20

3aAAb2. Subjective experiment on suitable speech-rate of public address announcement in public spaces. Sakae Yokoyama (I.I.S., The Univ. of Tokyo, Komaba 4-6-1, Meguro-ku 153-8505, Japan, sakae@iis.u-tokyo.ac.jp) and Hideki Tachibana (Chiba Inst. of Technol., Narashino, Japan)

In such public spaces as railway stations, airport terminal buildings, shopping arcades, etc., it is often the case that information provided through public address (P.A.) system is deteriorated by the influences of reverberation and background noise. This problem is serious especially for the case of providing various information and emergency evacuation alarms in the case of such disasters as earthquake and fire. Regarding this acoustic problem in public spaces, the authors performed field surveys in various public spaces, in which actual P.A. announcements were recorded and reproduced in an anechoic room by applying the 6-channel recording/reproduction technique to realize 3D aural impression. As a result of the subjective experiments on speech intelligibility (ease in listening) performed in the simulated sound field, it has been found that the speed of speech is an essential condition as well as reverberation and noise level. To investigate the way of improving speech intelligibility in such spaces, subjective experiment was performed by changing the reverberation time, background noise level, and the speech rate in Japanese announcements at several steps. In the experiment, a Text-To-Speech (TTS) software was used to change the speech rate. As the test subjects, students from abroad participated in the experiment and the difference of the effects of the adverse conditions between natives and non-natives was also investigated.

9:40

3aAAb3. Design of the new public address system for the cathedral of Münster, Germany. Gottfried Behler and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustraße 50, Aachen D-52056, Germany, gkb@akustik.rwth-aachen.de)

One of the most renowned cathedrals in Germany, the Dom St. Paul in Münster was completely closed for renovation for almost one year. During this time, the entire electro acoustical sound reinforcement system has been renewed. As for most buildings of this type, the acoustical situation is far away from optimal. This mainly is due to a huge reverberation time, which makes the understanding of spoken words almost impossible. Moreover, the old concept of sound reinforcement by using distributed loudspeaker systems all over the church is not satisfying anymore with respect to nowadays demands for quality and speech intelligibility. Due to the situation that the number of people in the cathedral during service times is varying from only some hundreds to over 2500 a more exible PA system is required, that takes into account that only occupied areas inside of the cathedral should be supplied with amplified sound. To achieve the target, an entirely new concept for the sound reinforcement based on digital signal distribution and modern digitally operated array loudspeakers was developed. The requirement for the speech intelligibility was to reach at least an STI of more than 0.5. The system will be discussed and results will be shown.

10:00

3aAAb4. Acoustical design of Turkish Religious Affairs Mosque. Zühre Sü Gül (R&D, MEZZO Studyo LTD, METU Technopolis KOSGEB-TEKMER No112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com) and Mehmet Čalışkan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

The new Turkish Religious Affairs (DIB) Mosque with its prayer capacity and outstanding volume is the largest classical-contemporary mosque project of the past decade built in Ankara, Turkey. The Mosque is also one of unique examples of its scale for which the room acoustic design is applied in its design phase. Acoustical design of DIB Mosque is critical considering speech and music related activity patterns held in such religious spaces. Interior surface forms and materials of walls and pendentives, floor finishing alternatives, geometry, and finishing of the dome are studied simultaneously with the architectural design as early as in the concept phase. Impedance tube is used for testing alternative materials for specifying sound absorption characteristics of reliefs and perforations. Computer simulation is applied as an acoustical design tool and estimations are held by ODEON v.11.23. Objective acoustical parameters including reverberation time, speech transmission index, and A-weighted sound levels are assessed with and without sound reinforcement systems for fully and partially occupied mosque conditions. Auralizations are held for imam and muezzin in different forms of religious call out to prayers. Evaluation of the space indicates that the optimized acoustical field is proper for intended functions of use in a mosque and satisfies desired tranquil environment.

10:20–10:40 Break

10:40

3aAAb5. Design and experience with subterranean installation of a fully anechoic acoustical testing chamber. Roger M. Ellingson (RM Ellingson Design & Development, LLC, 8515 SW Barnes Rd., Portland, OR 97225, Rogere@Rmegen.net) and Patrick V. Helt (National Ctr. for Rehabilitative Auditory Res., Portland, OR)

One of the purposes of a fully anechoic chamber is to provide a very quiet, near-echo-free environment simulating free-field acoustical conditions. From design specification to completion, installation of a fully anechoic chamber can be an enormous undertaking as compared to installation of conventional sound-attenuated acoustical test rooms, which generally are smaller in physical size and have less stringent sound attenuation requirements. The authors had the opportunity to specify the requirements and oversee the installation of a fully anechoic chamber designed to support near-full-frequency, human-hearing range acoustical experiments at the VA National Center for Rehabilitative Auditory Research (NCRAR), located in Portland, OR, USA. Design and installation of NCRAR's chamber to support entry at laboratory floor level was complicated by the job site location, a subterranean area beneath a structure that accommodates clinical and research offices on the upper floors, and serves as a parking garage on the lower floors. In addition to ambient noise considerations, existing architectural restrictions included parking traffic patterns, below-grade earthquake beams, limited overhead clearance, ground water seepage, and storm water flow patterns. The purpose of this paper is to share pictorial-illustrated experiential results on specification, design, installation, and chamber operation as well as architectural considerations, site preparation, and construction detail.

11:00

3aAAb6. A six sensor method for measuring acoustic properties in ducts. Timothy J. Newman, Anurag Agarwal, Ann P. Dowling (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, United Kingdom, tjn25@cam.ac.uk), Ludovic Desvard, and Ryan Stimpson (Aeroacoustic Res. Team, Dyson Ltd., Malmesbury, Wiltshire, United Kingdom)

An accurate description of sound propagation in a duct is important to obtain the sound power radiating from a source in both near and far fields. A technique has been developed and applied to decompose higher-order modes of sound emitted into a duct. Traditional experiments and theory based on two-sensor methods are limited to the plane-wave contribution to the sound field at low frequency. Due to the increase in independent measurements required, a computational method has been developed to simulate sensitivities of real measurements (e.g., noise) and optimize the set-up. An experimental rig has been constructed to decompose the first two modes using six independent measurements from surface, flush-mounted microphones. Experiments were initially performed using a loudspeaker as the source for validation. Subsequently, the sound emitted by a mixed-flow fan has been investigated and compared to measurements made in accordance with the internationally standardized in-duct fan measurement method. This method utilizes large anechoic terminations and a procedure involving averaging over measurements in space and time to account for the contribution from higher-order modes. The new method does not require either of these added complications and gives detail about the underlying modal content of the emitted sound.

11:20

3aAAb7. Acoustics evolution of the Moscow Conservatory Great Hall after renovation in 2011. Nikolay Kanev and Anatoly Livshits (Acoust. Group, 4, Svernika st., Moscow 117036, Russian Federation, nikolay.kanev@mail.ru)

The Great Hall of the Moscow P.I. Tchaikovsky Conservatory is one of the best concert halls in Russia. Its acoustics is appreciated very much by both musicians and audience. In 2011, the Great Hall was renovated, after renovation its acoustics conditions remained at very high level that was confirmed by means of objective impulse response measurements and subjective estimations. In order to control how acoustics conditions are changing with time the observation of main acoustics parameters is carrying out with half year gaps. In this work we present the results of four measurements fulfilled in June 2011 (just after the renovation), December 2011, June 2012, and December 2012. The reverberation time decreases at low frequencies, whereas it is stable at middle frequencies. Periodic variations of the reverberation time take place at high frequencies. These variations are probably connected with seasonal changes of temperature and humidity conditions in the hall. Changes of other acoustics parameters correlate to reverberation time.

WEDNESDAY MORNING, 5 JUNE 2013 510B, 8:55 A.M. TO 12:00 NOON

Session 3aAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Perceiving Objects I

Caroline M. DeLong, Cochair
Psych., Rochester Inst. of Tech., 18 Lomb Memorial Dr., Rochester, NY 14623

Eduardo Mercado, Cochair
Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260

Chair's Introduction—8:55

Invited Papers

9:00

3aAB1. Human performance in aural classification of sonar echoes. Nancy Allen and Paul C. Hines (Defence R&D Canada - Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, nancy.allen@drdc-rddc.gc.ca)

Active sonar echoes from man-made objects and those from naturally occurring features in certain coastal environments can be difficult to distinguish when using traditional sonar processing techniques and visual displays. An approach being investigated at Defence R&D Canada-Atlantic for addressing this challenge is to exploit the capability of human hearing for discriminating between these two classes of echoes. Part of the work consisted in producing a human-performance baseline. Two human listening tests were designed and carried out. Both used stimuli that were generated from a sample of echoes recorded during actual sonar experiments at sea. In the second test, a 500-Hz high-pass filter was applied to the stimuli. Quantitative data from the rating-exercise portion of the tests were used to produce receiver-operating-characteristic (ROC) curves for modeling how well the participants could distinguish the two classes of echoes. For both tests, the results show that the listeners could hear differences, but performance was significantly better in the first test. Qualitative data collected from the questionnaire portion of the tests helped to interpret some of the performance results.

9:20

3aAB2. Recognizing objects from multiple orientations using dolphin echoes. Caroline M. DeLong, Amanda Heberle, Kayla Mata (Psychology, Rochester Inst. of Tech., 18 Lomb Memorial Dr., Rochester, NY 14623, cmdgsh@rit.edu), Heidi E. Harley (Psychology, New College of Florida, Sarasota, FL), and Whitlow W. Au (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Kaneohe, HI)

Object constancy, the ability to recognize objects despite changes in orientation, has not been well studied in the auditory modality. Dolphins use echolocation for object recognition, and objects ensonified by dolphins produce echoes that can vary significantly as a function of orientation. In four experiments, human listeners had to classify echoes from objects ensonified with dolphin signals. Participants were trained to discriminate among the objects using an 18-echo stimulus from a 10 degree range of aspect angles, then tested with novel aspect angles across a 60 degree range. In the first two experiments, the three objects varied in material, size, and shape. Participants were typically successful recognizing the objects at all angles (M = 78%). In experiment 3, the three objects had the same material but different shapes. Participants were often unsuccessful recognizing the objects at all angles (M = 46%). In experiment 4, participants had to classify echoes from four fish species across a wider range of angles (330 degrees). Preliminary results show overall poor performance (M = 45%). These results suggest that object characteristics play a role in whether performance is more view-dependent or view-invariant. These studies can provide insight into the process dolphins use to identify objects.

9:40

3aAB3. Exploring the capacity of neural networks to recognize objects from dolphin echoes across multiple orientations. Matthew G. Wisniewski (Psychology, Univ. at Buffalo, The State Univ. of New York, 260 Callodine Ave., Amherst, NY 14226, mgw@buffalo.edu), Caroline M. DeLong, Amanda L. Heberle (Psychology, Rochester Inst. of Technol., Rochester, NY), and Eduardo Mercado (Psychology, Univ. at Buffalo, The State Univ. of New York, Buffalo, NY)

Dolphins naturally recognize objects from multiple angles using echolocation. With training, humans can also learn to accurately classify objects based on their echoic features. In this study, we used neural networks to identify acoustic cues that enable objects to be recognized from varying aspects. In simulation 1, a self-organizing map was able to differentiate a subset of objects using only amplitude and frequency cues, but it classified some echoes from different objects as being from the same object. In simulation 2, a multilayer

perceptron was trained through error correction to identify objects based on echoes from a single aspect, and then tested on its ability to recognize those objects using echoes from different orientations. Overall, perceptrons performed similarly to trained undergraduates. Analysis of network connection weights revealed that both the amplitude and frequency of echoes, as well as the temporal dynamics of these features over the course of an echo train, enabled perceptrons to accurately identify objects when presented with novel orientations. These findings suggest that learning may strongly impact an organism's ability to echoically recognize an object from any viewpoint.

10:00

3aAB4. Object selection by head aim and acoustic gaze in the big brown bat. Jason Gaudette (NUWC, Providence, RI), Laura Kloepper, and James Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02906, laura_kloepper@brown.edu)

Echolocating bats use their active sonar to locate, discriminate, and capture flying prey. A special challenge is tracking and pursuing a discrete moving object, sometimes in cluttered surroundings. Bats rely on head aim to follow their prey's location throughout an entire capture sequence. By directing the sound emission organ (mouth or noseleaf) and thus acoustic gaze toward the target, the prey is kept on the main broadcast axis with the flattest incident spectrum. Although bats can perform head-aim tracking with an accuracy of a few degrees, we have explored the dynamics of the transmitted beam during tracking and its impact on angular precision. We measured head aim and acoustic gaze in the big brown bat (*Eptesicus fuscus*) with a 224-element microphone array. This array allows for fine scale, independent measurements of the beam across many frequencies with a high signal-to-noise ratio. Bats were trained to track moving real and electronic targets, and the head aim and acoustic gaze were recorded. We specifically examined the possibility that bats move different frequencies of their beam at different angular rates, with particular attention to the harmonics of the broadcasts. [Work supported by ONR, NSF, and NUWCDIVNPT.]

10:20

3aAB5. Temporal signal processing of dolphin biosonar echoes from fish prey. Whitlow Au and Hui Ou (Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu)

Dolphins that emit whistle signals (except for sperm whales) project short broad- band biosonar clicks containing about 5 to 7 cycles with exponential decaying envelope and Q (center frequency over bandwidth) between 2 and 3. The broadband nature of the biosonar clicks allow for good temporal resolution of echo highlights, which in turn allows for the discriminations of different targets including fish prey. Most of the echoes from fish originate from signals reflecting off the swim bladder of fish. Different species of fish have swim bladders of different shape, size, and orientation so that echoes from these species can often be discriminated from temporal cues. The echoes contain many highlights as the signals reflect off different surfaces and parts of the fish body and swim bladder. This presentation will discuss the temporal characteristics of echoes from fish prey, which are highly aspect dependent and will discuss the six temporal parameters that were used in a support vector machine (SVM) to discriminate between species. Results suggest how dolphins can classify fish based on their echoes and provide some insight as to which features might enable the classification.

10:40

3aAB6. Dolphin strategies for long-range object detection and change detection. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil)

Dolphin strategies for detecting objects and changes in objects were investigated by having three trained bottlenose dolphins perform long-range echolocation tasks. The tasks featured the use of "phantom" echoes produced by capturing the dolphin's outgoing echolocation clicks, convolving the clicks with the impulse response of a physical target to create an echo waveform, then broadcasting the delayed, scaled echo waveform back to the dolphin. Dolphins were trained to report the presence of phantom echoes or a change in phantom echoes. Target simulated ranges varied from 25 to 800 m. At ranges below 75 m, all dolphins followed a single click-echo paradigm, where inter-click intervals exceeded the two-transit time (i.e., the dolphins waited to receive the echo from a click before emitting the next click). As the range increased beyond 75 m, two of the three dolphins increasingly produced bursts, or "packets," of several clicks, then waited for the packet of echoes to return before emitting another packet of clicks. The third dolphin instead utilized very high click repetition rates. The use of click packets may be a response to a limitation in the dolphin's ability to employ multi-echo processing with large inter-echo delays.

11:00

3aAB7. Dolphin echolocation is not seeing with sound. Heidi E. Harley (Psychology, New College of Florida, Div. of Social Sci., 5800 Bay Shore Rd., Sarasota, FL 34243, harley@ncf.edu), Wendi Fellner, and Barbara Losch (The Seas, Epcot®, Walt Disney World® Resorts, Kissimmee, FL)

Dolphin echolocation is often described as "seeing with sound;" however, vision and audition vary substantially in terms of direct access to spatial information. This study investigated spatial representation based on echoic information by having a dolphin match objects that varied only in shape. Stimulus sets (3 objects each) included unfamiliar objects made from (1) the same PVC parts (controlled sets), (2) different PVC parts (uncontrolled sets), or (3) non-PVC junk objects (junk sets). Sets were presented for 5 18-trial sessions in each of two conditions: an echoic condition (objects underwater with dolphin eyecups occluding vision) and a visual condition (objects in air). Performance accuracy varied across set type and condition. Worst was echoic controlled (12 sets, M = 45%) followed by echoic uncontrolled (12 sets, M = 50%). Visual performance was significantly better: controlled (M = 67%), uncontrolled (M = 76%). Echoic performance ranged from 32% to 86% across sets; visual performance from 50% to 100%. In contrast, performance accuracy with junk sets was higher echoically (2 sets, M = 81%) than visually (2 sets, M = 58%). Shape does not appear to be easily accessible to dolphins via echolocation, although it is accessible through vision. Dolphins integrate information about objects across modalities; they likely gain most shape information through vision.

11:20

3aAB8. Auditory object formation in Cope's gray treefrogs (*Hyla chrysoscelis*). Mark A. Bee (Ecology, Evolution and Behavior, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108, mbee@umn.edu) and Katrina M. Schrode (Grad. Program in Neurosci., Univ. of Minnesota, St. Paul, MN)

Hearing and acoustic communication in "real world," multi-source environments require animals to group sound elements produced by the same source into perceptually coherent "auditory objects." However, research on nonhuman animal communication rarely investigates perceptual processes involved in forming auditory objects of communication sounds. We tested the hypotheses that spectral and spatial proximity promote the sequential integration of temporally separated sounds produced by the same source into coherent auditory objects of acoustic signals. Male gray treefrogs produce a pulsatile advertisement call; females prefer longer calls (= more pulses) to shorter calls and discriminate against calls missing pulses. We gave females a choice between a short but spectrally and spatially coherent call (25 pulses) and a longer call (35 pulses) in which alternating groups of 5 pulses had different frequencies (ΔF, 0–12 semitones) and came from different locations ($\Delta\theta$, 0° or 90°). Females generally preferred the longer call at smaller values of ΔF and $\Delta\theta$, indicating a role for spectral and spatial proximity in sequential integration. Under some conditions, however, subjects showed a surprising willingness to integrate pulses despite large ΔFs. Together, these data shed light on the perceptual cues that receivers exploit to form coherent auditory objects of communication sounds.

11:40

3aAB9. Auditory scene analysis in budgerigars (*Melopsittacus undulatus*) and zebra finches (*Taeniopygia guttata*). Micheal L. Dent, Erikson G. Neilans, Mary M. Flaherty, and Amanda K. Martin (Psychology, Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260, mdent@buffalo.edu)

Deciphering the auditory scene is a problem faced by humans and animals alike. However, when faced with overlapping sounds from multiple locations, listeners are still able to attribute the individual sound objects to their individual sound-producing sources. Here, we determined which characteristics of sounds are important for streaming versus segregating in birds. Budgerigars and zebra finches were trained using operant conditioning procedures on an identification task to peck one key when they heard a whole zebra finch song and to peck another when they heard a zebra finch song missing a middle syllable. Once the birds were trained to a criterion performance level on those endpoint stimuli, probe trials were introduced on a small proportion of all trials. The probe songs contained modifications of the incomplete training song's missing syllable. When the bird responded as if the probe was a whole song, it suggests they streamed together the altered syllable and the rest of the song. When the bird responded non-whole song, it suggests they segregated the altered probe from the rest of the song. Results show that some features, such as spectrotemporal similarity and location, are more important for streaming than other features, such as timing.

WEDNESDAY MORNING, 5 JUNE 2013 510D, 10:55 A.M. TO 11:40 A.M.

Session 3aAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

John A. Colosi, Chair
Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943

Chair's Introduction—10:55

Invited Paper

11:00

3aAO1. Acoustical tomography in the shallow water ocean: Dream or reality? Philippe Roux (ISTerre, CNRS - Université Grenoble 1, Maison des Géosciences, Rue de la piscine, Grenoble 38041, France, philippe.roux@obs.ujf-grenoble.fr)

Acoustical tomography in shallow waters relies on the identification and tracking of stable ray arrivals. The variation of the arrival time of these rays is used to solve the inverse problem and to estimate the physical properties as sound speed variations associated to internal waves or currents. In practice, however, technical difficulties appear (1) when the number of resolved eigenrays in this multipath environment is too small using a set of point-to-point recordings or (2) when the travel-time fluctuations are dominated by the fast-evolving surface that blur the slower internal-wave driven perturbations around the thermoclyne. Recent experimental studies based on laboratory-scaled demonstrators showed the efficiency of array processing algorithms in combination with source-receiver arrays to tackle issue (1) above. Depending on the waveguide geometry, the acquisition procedure can also be adjusted to provide a separation between the slow and fast travel-time fluctuations. Inversion results based on laboratory experiments are presented that focus on the dynamics of (1) a gravity wave at the air-water interface and (2) a thermal plume in the water column. Finally, these results are discussed in the framework of ocean acoustic research.

Session 3aBAa

Biomedical Acoustics: Delivery of Nucleic Acids (DNA, siRNA, antisense oligos)

Tyrone M. Porter, Cochair
Boston Univ., 110 Cummington St., Boston, MA 02215

Raffi Karshafian, Cochair
Dept. of Phys., Ryerson Univ., 350 Victoria St., KHE 329E, Toronto, ON M5B 2K3, Canada

Invited Papers

9:20

3aBAa1. Ultrasound-mediated gene delivery – Cardiovascular applications for Chronic ischemia, heart failure, and ischemia-reperfusion injury. Howard Leong-Poi (Cardiology/Medicine, St. Michael's Hospital, 6-044 Queen Wing, 30 Bond St., Toronto, ON M5B1W8, Canada, leong-poih@smh.ca)

Ultrasound-mediated gene delivery (UMGD) is a non-invasive gene transfer technique, utilizing high power ultrasound and DNA-bearing microbubbles. Despite modest transfection efficiency, its high organ, tissue specificity, and repeatability make it an attractive therapeutic option. UMGD has been used in a variety of *in vivo* applications, including cardiac and skeletal muscle, kidney, liver, cerebral, and even lung, and have been studied using many gene vectors, including plasmid, viral, and small interfering RNA. This presentation will focus specifically on cardiac applications using plasmid DNA, including (1) introduction of UMGD in the heart, including optimization of parameters and protocols, (2) UMGD for therapeutic angiogenesis in chronic ischemia, including multi pro-angiogenic gene therapy and combination gene- and progenitor cell-based therapies for chronic hindlimb ischemia, and (3) applications for anti-apoptotic therapy in heart failure and ischemia-reperfusion injury.

9:40

3aBAa2. Effective ultrasound-targeted microbubble destruction-mediated gene transfer into the livers of small and large animals. Carol H. Miao, Misty L. Noble, Shuxian Song, Ryan R. Sun (Pediatrics, Seattle Children's Res. Inst. and Univ. of Washington, 1900 Ninth Ave. C9S-7, Seattle, WA 98125, miao@uw.edu), Christian S. Kuhr (Benaroya Res. Inst., Seattle, WA), Scott S. Graves (Fred Hutchinson Cancer Res. Ctr., Seattle, WA), George W. Keilman, Kyle P. Morrison (Sonic Concepts Inc., Seattle, WA), Keith R. Loeb (Fred Hutchinson Cancer Res. Ctr., Seattle, WA), Andrew A. Brayman, Marla Paun (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Rainer F. Storb (Fred Hutchinson Cancer Res. Ctr., Seattle, WA), and Samuel S. Sun (Pediatrics, Seattle Children's Res. Inst. and Univ. of Washington, Seattle, WA)

Ultrasound (US)-targeted microbubble (MB) destruction (UTMD) can significantly enhance gene delivery in mouse livers when pDNA/MBs were injected into the portal vein (PV) with simultaneous US exposure using a focused transducer. However, this transducer was ineffective in enhancing gene transfer into rats. A 13-mm diameter unfocused transducer was designed and the delivery route of pDNA/MBs was modified into a specific liver lobe, resulting in >100-fold increase in luciferase expression in rats. To facilitate the translation into human application, many technical issues were explored in large animal models. We applied 1.1 MHz US to a targeted canine liver lobe with simultaneous injection of pDNA/MBs into a PV segmental branch and occlusion of the inferior vena cava. For more effective treatment of large tissue volumes, a 52-mm planar unfocused transducer was specifically constructed. Its apodized dual element configuration greatly reduced the near field transaxial pressure variations, producing a uniform field of US exposure for the treated tissues. Together with a 15 kW-capacity US amplifier, a 692-fold increase of gene expression in canines was achieved at 2.7-MPa. Transaminase levels and histology analysis indicated minimal tissue damage. These results demonstrated that UTMD is highly promising for safe and efficient gene delivery into the liver.

10:00

3aBAa3. Non-invasive delivery of small interfering ribonucleic acid for reduction of *Huntingtin* expression in the brain is achieved using focused ultrasound to disrupt the blood-brain barrier. Alison Burgess, Yuexi Huang (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), William Querbes, Dinah W. Sah (Alnylam Pharmaceuticals Inc., Cambridge, MA), and Kullervo Hynynen (Med. Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca)

Huntington's disease is caused by a mutation in the *Huntingtin* (Htt) gene, which leads to neuronal dysfunction and cell death. Silencing of the Htt gene can halt or reverse the progression of the disease indicating that RNA interference is the most effective strategy for disease treatment. However, small interfering RNA (siRNA) does not cross the blood-brain barrier and therefore delivery to the brain is limited. Here, we demonstrate that focused ultrasound (FUS), combined with intravascular delivery of microbubble contrast agent, was used to locally and transiently disrupt the BBB in the right striatum of adult rats. 48 hours following treatment with siRNA, the right (treated) and left (control) striatum was dissected and analyzed for Htt mRNA levels. We demonstrate that FUS can non-invasively deliver siRNA-Htt directly to the striatum leading to a significant reduction of Htt expression in a dose dependent manner. Furthermore, we show that reduction of Htt with siRNA-Htt was greater when the extent of BBB disruption was increased. This study demonstrates that siRNA treatment for knockdown of mutant Htt is feasible without the surgical intervention previously required for direct delivery to the brain. Non-invasive delivery of siRNA through the blood-brain barrier (BBB) would be a significant advantage for translating this therapy to HD patients.

10:20

3aBAa4. Polyplex-microbubbles for improved ultrasound-mediated gene therapy. Mark Borden, Shashank Sirsi (Mech. Eng., Univ. of Colorado, 1111 Eng. Dr., Campus Box 427, Boulder, CO 80309, mark.borden@colorado.edu), Sonia Hernandez, Shunichi Homma (Oncology, Columbia Univ., New York, NY), Jessica Kandel (Surgery, Columbia Univ., New York, NY), and Darrell Yamashiro (Oncology, Columbia Univ., New York, NY)

Sonoporation is an established method whereby acoustically stimulated microbubbles create pores in the endothelium to promote the delivery of nucleic acids and other macromolecules through the vasculature. In this presentation, we describe the development of polyplex-microbubbles to enhance delivery and intracellular trafficking to target cells beyond the endothelium following sonoporation. Our design combines the tissue targeting capability of microbubbles with the cellular targeting capability of polyplexes, which are self-assembled particles comprising nucleic acids and a cationic polymer. The first purpose of the cationic polymer (polyethylenimine, PEI) is to condense and protect the nucleic acids on the microbubble surface as it travels in circulation from the site of injection to the target tissue. Polyethylene glycol (PEG) is conjugated to the PEI to reduce immunogenicity. Ultrasound is applied to the target tissue to fragment the microbubbles and release the polyplexes and to porate the endothelium, allowing the polyplexes to extravasate. The second purpose of PEI is to promote endocytotic uptake and subsequent endosomal escape of the nucleic acids into the cytoplasm by the so-called "proton-sponge" effect. Here, we will describe polyplex-microbubble fabrication and characterization, as well as *in vivo* testing. Our results indicate that this is a promising design for cancer gene therapy.

10:40

3aBAa5. Small interfering ribonucleic acid delivery with phase-shift nanoemulsions. Mark T. Burgess and Tyrone M. Porter (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, marktb@bu.edu)

Acoustic cavitation offers a unique approach to small interfering RNA (siRNA) delivery compared to current methods. Typically, preformed microbubbles are used as cavitation nuclei to permeabilize cells and facilitate siRNA entry into the cytoplasm. However, microbubbles are restricted to the vasculature space and suffer from stability issues that limit their applicability. Alternatively, phase-shift nanoemulsions (PSNE) possess the long circulation and extravasation properties of nanoparticles, while also serving as cavitation nuclei in tissue upon acoustic droplet vaporization. Here we report the use of PSNE for delivery of siRNA engineered to knockdown green fluorescent protein (GFP) expression. A cell suspension (5×10^6 cells/mL) of GFP expressing breast adenocarcinoma cells was exposed to 5 MHz pulsed ultrasound (4 MPa peak negative pressure, 3 cycles, 250 Hz, 100 s exposure duration) in the presence of PSNE ($\sim 10^9$/mL) and free siRNA (1.8 μM). Flow cytometry was used to quantify GFP expression and cell viability. There was 20% ($p < 0.05$, $n = 6$) reduction in GFP fluorescence for cells treated with GFP siRNA and 80% (+/− 6.4%) cell survival. This work highlights the potential for PSNE to serve as interstitial cavitation nuclei for siRNA delivery in tissue. [Work supported in part by NIH grants R25CA153955 and R03EB015089.]

11:00

3aBAa6. Physical mechanisms responsible for bubble translation near an interface. Daniel R. Tengelsen, Todd Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, danieltengelsen@gmail.com)

Previous models and experiments have shown that direction of bubble translation near a viscoelastic layer depends on both the standoff distance of the bubble and the elastic properties of the layer. Here the individual forces due to the incident sound field and the field reflected from the viscoelastic layer are shown to compete with one another and ultimately determine the direction of bubble translation. In addition, many other factors pertinent to the direction of bubble translation such as the incident acoustic waveform, the phase and propagation direction of the incident field, and the radial bubble dynamics are considered. The force due to the viscoelastic layer is calculated using a Green's function, which takes into account elastic waves and viscosity in the layer and the viscous boundary layer at the solid-liquid interface. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and NIH DK070618.]

Contributed Paper

11:20

3aBAa7. Ultrasonic bubbles in microfluidics for red blood cells, bacteria, and yeast lysis. Siew-Wan Ohl, Tandiono Tandiono (Inst. of High Performance Computing, 1 Fusionopolis Way, #16-16 Connexis North, Singapore 138632, Singapore, ohlsw@ihpc.a-star.edu.sg), Dave Siak-Wei Ow (Bioprocessing Technol. Inst., Singapore, Singapore), Evert Klaseboer (Inst. of High Performance Computing, Singapore, Singapore), Andre Boon-Hwa Choo (Bioprocessing Technol. Inst., Singapore, Singapore), and Claus-Dieter Ohl (School of Physical and Mathematical Sci., Nanyang Technolog. Univ., Singapore, Singapore)

A custom microfluidic ultrasound device has been designed and implemented to efficiently create oscillating ultrasonic bubbles. These bubbles interact with the red blood cells, bacteria (*Escherichia coli*), and yeast (*Pichia pastoris*). Observations using high speed photography show that the red blood cells were strongly stretched. A numerical model using the Boundary Element Method is used to simulate the oscillating bubble's interaction with a nearby elastic pocket (a red blood cell model). Complex dynamics are discussed. Our ultrasonic microfluidic device is efficient in lysing *E. coli* and yeast cells for the harvesting of active intracellular content. Complete lysis of *E. coli* takes only 0.4 s while it takes about 1 s for the yeast cell. Temperature rise is minimal. We perform fluorescent intensity measurement and qRT-PCR (real-time polymerase chain reaction) to show that the functional integrity of the proteins and DNA is preserved.

Session 3aBAb

Biomedical Acoustics: Generation and Detection of High Intensity Focused Ultrasound Lesions

Ashwinkumar Sampathkumar, Chair
Biomedical Eng., Riverside Res. Inst., 156 William St., #9, New York, NY 10038

Contributed Papers

9:20

3aBAb1. Variations of temperature distribution and lesion formation induced by tissue inhomogeneity for therapeutic ultrasound. Dong Zhang (Inst. of Acoust., Hankou 22, Nanjing 210093, China, dzhang@nju.edu.cn)

High intensity focused ultrasound (HIFU) has shown potential applications in therapeutic medical fields. This work reported both theoretical and experimental investigations on the influence of tissue inhomogeneity on the temperature distribution and tissue lesion formation using the phase screen model. The inhomogeneous tissue is considered as a combination of a homogeneous medium and a phase aberration screen. Numerical simulations were performed by using the nonlinear acoustic equation and bio-heat transfer equation. Four polyethylene (PE) plates with various standard deviations were made to mimic the inhomogeneity induced by human body abdominal. Temperature rise and lesion formation induced by HIFU were measured and compared with the numerical calculations. Results indicate that the standard deviation is associated with the focusing capability and temperature field observably. This study provides a theoretical and experimental basis for the development of precise HIFU treatment plan.

9:40

3aBAb2. Thermal ablation by high-intensity-focused ultrasound using a toroidal transducer for the treatment of colorectal liver metastases during an open procedure. Clinical results. David Melodelima, Jeremy Vincenot (LabTAU - U1032, INSERM, 151 cours Albert Thomas, Lyon 69003, France, David.Melodelima@inserm.fr), Yao Chen, Aurelien Dupre, Michel Rivoire (Dept. of Surgery, Ctr. Leon Berard, Lyon, France), and Jean-Yves Chapelon (LabTAU - U1032, INSERM, Lyon, France)

The objective of this clinical study was to validate the effectiveness, accuracy, tolerance, and safety of a HIFU treatment developed for the treatment of liver metastases. Fifteen patients were included. The transducer has a toroidal shape (diameter: 70 mm, radius of curvature: 70 mm) and was divided into 256 ultrasound emitters operating at 3 MHz. A 7.5 MHz ultrasound imaging probe was placed in the center of the device. All HIFU ablations were obtained in 40 s. The demarcation between ablated and non-ablated tissue was clearly apparent in ultrasound images and histology. In phase I (6 patients), we demonstrated that the dimensions of HIFU ablations measured on ultrasound imaging were correlated ($r = 0.88$) with dimensions measured during histological analysis. The average dimensions obtained from each HIFU ablation were a diameter of 21.0 ± 3.9 mm and a depth of 27.5 ± 6.0 mm. In phase II (9 patients), the HIFU ablations were centered on a target previously identified with a precision of 1–2 mm. It was demonstrated that HIFU ablations can be precisely located at 7.0 ± 2.3 mm from the target (expected distance 7.5 mm). This toroidal HIFU transducer achieved fast, selective, safe, and well-tolerated large volume liver ablation.

10:00

3aBAb3. Non-invasive toroidal high intensity focused ultrasound transducer for increasing the coagulated volume in depth. Jeremy Vincenot, David Melodelima, Francoise Chavrier, and Jean-Yves Chapelon (LabTAU - U1032, INSERM, 151 cours Albert Thomas, Lyon, France, jeremy.vincenot@inserm.fr)

A device composed of 32 elements (78 mm^2 each) arranged on a toroidal transducer (operating frequency: 2.5 MHz) was developed to increase the coagulated volume. To date, our previous work on toroidal transducers used the outer envelope of a torus as a reference surface. Here, the transducer geometry was based on the interior part of a torus. This produces a focus that is ring-shaped but the ultrasound beams also intersect between the principal focal ring and the transducer surface to form a secondary focal zone, which contributes to increase the size of the lesion. The radius of curvature was 70 mm with a diameter of 67 mm. A 7.5 MHz ultrasound imaging probe was placed in the center of the device. Twenty ablations were produced *in vitro* by using electronic beam steering, each ablation was created in 55 s. The average depth of intervening tissues (skin-fat-muscle) was 11 ± 1 mm and the average depth of liver tissues that have been spared was 21 ± 4 mm. No significant temperature rise in intervening tissues was measured (maximal temperature: 41°C). The dimensions of these ablations were an average diameter of 10 ± 1 mm and an average depth of 27 ± 3 mm.

10:20

3aBAb4. Improving the acousto-optic detection of high-intensity focused ultrasound lesions. Matthew T. Adams, Qi Wang (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, adamsm2@bu.edu), Robin O. Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Ronald A. Roy (Mech. Eng., Boston Univ., Boston, MA)

Real-time acousto-optic (AO) sensing has been shown to non-invasively detect changes in *ex vivo* tissue optical properties during high-intensity focused ultrasound (HIFU) exposures. Although proof-of-concept experiments have been successful, the underlying parameters and mechanisms affecting the AO detectability of HIFU lesion formation are not well understood. In this work, a modeling based approach is used to improve the AO sensing of lesion formation during HIFU therapy. The angular spectrum method is used to model the acoustic field from the HIFU source and the temperature field, due to the absorption of ultrasound, is modeled using a finite-difference time-domain solution to the Pennes bioheat equation. Wavelength specific changes in tissue optical properties are calculated using a thermal dose model, calibrated by experimental data. The diffuse optical field is modeled using an open-source graphics processing unit accelerated Monte Carlo algorithm. The Monte Carlo algorithm is modified to account for light-sound interactions, using the acoustic field from the angular spectrum method, and to account for AO signal detection. Results will

demonstrate the important roles of optical wavelength selection, and illumination and detection configurations on the detectability of HIFU lesions by optical and AO sensing methods. [Work supported in part by NSF.]

10:40

3aBAb5. Determination of tissue injury thresholds from ultrasound in a porcine kidney model. Yak-Nam Wang, Julianna C. Simon, Bryan W. Cunitz, Frank L. Starr, Marla Paun (APL, CIMU, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, ynwang@u.washington.edu), Liggitt Denny (Dept. of Comparative Medicine, Univ. of Washington, Seattle, WA), Andrew P. Evan, James A. McAteer, James C. Williams (Dept. of Anatomy and Cell Biology, Indiana Univ., Indianapolis, IN), Ziyue Liu (Dept. of Biostatistics, Indiana Univ., Indianapolis, IN), Peter J. Kaczkowski (APL, CIMU, Univ. of Washington, Seattle, WA), Ryan S. Hsi, Mathew D. Sorensen, Jonathan D. Harper (Dept. of Urology, Univ. of Washington, Seattle, WA), and Michael R. Bailey (APL, CIMU, Univ. of Washington, Seattle, WA)

Therapeutic ultrasound has an increasing number of applications in urology, including shockwave lithotripsy, stone propulsion, tissue ablation, and hemostasis. However, the threshold of renal injury using ultrasound is unknown. The goal of this study was to determine kidney injury thresholds for a range of intensities between diagnostic and ablative therapeutic ultrasound. A 2 MHz annular array generating spatial peak pulse average intensities (I_{SPPA}) up to 30,000 W/cm^2 in water was placed on the surface of *in vivo* porcine kidneys and focused on the adjacent parenchyma. Treatments consisted of pulses of 100 μs duration triggered every 3 ms for 10 min at various intensities. The perfusion-fixed tissue was scored by three blinded independent experts. Above a threshold of 20,000 W/cm^2, the majority of injury observed included emulsification, necrosis, and hemorrhage. Below this threshold, almost all injury presented as focal cell and tubular swelling and/or degeneration. These findings provide evidence for a wide range of potentially therapeutic ultrasound intensities that has a low probability of causing injury. While this study did not examine all combinations of treatment parameters of therapeutic ultrasound, tissue injury appears dose-dependent. [Work supported by NIH DK43881, DK092197, and NSBRI through NASA NCC 9-58.]

11:00

3aBAb6. The origins of nonlinear enhancement in *ex vivo* tissue during high intensity focused ultrasound ablation. Edward Jackson, Robin O. Cleveland, and Constantin C. Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Old Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, edward.jackson@magd.ox.ac.uk)

Thermal ablation by high intensity focused ultrasound (HIFU) is an emerging technique for non-invasive treatment of tumors. One barrier to its wider use is that current imaging modalities are not able to directly monitor changes in the tissue during treatment. Nonlinear imaging has been suggested as a possible mode, but it is unclear whether the nonlinear enhancement is due to changes in the properties of treated tissue or cavitation. This study uses a finite-amplitude insertion technique to measure B/A of *ex-vivo* bovine liver as a function of temperature. The technique creates quasi-plane wave conditions by measuring the waveform in the near field of a 10 cm diameter 1 MHz unfocused source. The linear acoustic properties are measured in the same location and then B/A determined by fitting data to a numerical solution of the Burgers equation. Measurements are taken during heating in a water bath (at a slow rate) and HIFU exposure (at a fast rate). Cavitation is monitored during the HIFU exposures. Previous data suggested B/A doubles after heating. However, in these experiments, the changes were less than 20%. These data suggest that cavitation effects dominate changes in B/A.

WEDNESDAY MORNING, 5 JUNE 2013 512AE, 8:55 A.M. TO 12:00 NOON

Session 3aEA

Engineering Acoustics: Computational Methods in Transducer Design, Modeling, Simulation, and Optimization II

Daniel M. Warren, Chair
Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60134

Chair's Introduction—8:55

Invited Papers

9:00

3aEA1. Electret microphone modeling and optimization by combined finite element analysis and lumped-element techniques. David Schafer (Knowles Electron., 1151 Maplewood Dr., Itasca, IL 60143, david.schafer@knowles.com)

Electret microphone performance modeling must represent detailed coupled mechanical, electrostatic, and fluid dynamics of the moving diaphragm, stationary back electrode and intervening air layer (the "motor") while also evaluating the net performance characteristics which result. We model electret microphone response and noise by combined FEA and lumped-element methods. In detail, the diaphragm is treated as a tensioned membrane. Electrostatics are evaluated in locally parallel-plate fashion. Air layer dynamics are treated by reducing the exact thermoacoustic parallel-plate 3D squeeze solution to a 2D "transmission sheet" differential equation. Analyses (COMSOL FEA 2D PDE-mode) include initial diaphragm deflection upon back electrode polarization, then the harmonic responses to changes in frontside pressure, backside pressure, and polarization. Motor responses (frontside/backside volume flows and output current) are reduced to impedance and transfer characteristics of a three-port (front and back acoustic, electrical) element, which is then embedded in a traditional host lumped-element equivalent circuit (including amplifier). FEA-based motor characteristics over an array of frequencies lead to microphone frequency response and noise spectrum. Performance optimization is done by scripting motor geometry generation and FEA analysis as a subroutine, defining a space of adjusted parameters (motor dimensions, etc.) and running recurring FEA analyses generated by a downhill simplex algorithm.

9:20

3aEA2. Transducer modeling for optimal *in-situ* performance in a hearing instrument context. Martin Kuster (Sci. & Technol., Phonak AG, Laubisrütistr. 28, Stäfa 8712, Switzerland, kuster_martin@hotmail.com)

In a hearing instrument context, the transducers design has to be optimized for optimal performance in the installed configuration. This is typically behind or inside the outer ear of the instrument's wearer. Several examples will be shown where the *in-situ* acoustics is taken into account in the design or selection process of transducers. In this process, acoustic modeling is used extensively. Transducers are modeled by network analogs and they are coupled under defined impedance conditions to 3D numerical acoustics models, which incorporate the *in-situ* influence of, e.g., the wearer's ear anatomy. Moreover, in some cases, the modeling also includes part of the signal processing present in the hearing instrument.

Contributed Papers

9:40

3aEA3. Investigation of vibrations of piezoelectric spherical shells with axisymmetric holes. David A. Brown and Colton T. Brown (BTech Acoust., Adv. Tech. & Manuf. Cntr./ECE, UMass Dartmouth, 151 Martine St., Fall River, MA 02723, DBrown@BTechAcoustics.com)

An experimental investigation of the vibration of radially polarized thin-walled piezoelectric ceramic spherical shells with axisymmetric holes is presented. Piezoelectric spherical electroacoustic transducers having holes at their poles to permit passage of wires, cables, or structural members is of particular interest in underwater acoustics. The coupled vibrations follow three resonance branches corresponding to (0) azimuthal extensional modes, (1) bending flexural modes, and (2) meridional extensional modes. The resonances and effective electromechanical coupling coefficient for each mode as a function of the hole-to-sphere diameter ratio have been determined from admittance measurements. In the limit that the hole-to-sphere diameter ratio approaches zero, the (0) mode is dominant corresponding to the spherically symmetric breathing mode having a measured coupling coefficient of 0.546, which is consistent with the planar coupling coefficient for PZT-4 (Type I) material. In the limit that ratio approaches unity, the element in nearly cylindrical and the lowest (0) mode corresponds to the breathing mode of a ring having a coupling coefficient of about 0.33, which is consistent with the transverse (31) coupling coefficient for the material. The focus of the study is on determining the resonances and electromechanical coupling in the intermediate region and obtaining corresponding vibration mode shapes with a non-contact optical fiber displacement sensor.

10:00

3aEA4. Design of resonant frequencies of the piezoelectric actuator with integrated components. Jun Kuroda, Yasuharu Onishi, Motoyoshi Komoda (Common Platform Development Div., NEC CASIO Mobile Commun., LTD., 1753, Shimonumabe, Nakahara-Ku, Kawasaki, Kanagawa, Kanagawa 211-8666, Japan, j-kuroda@bq.jp.nec.com), Yasuhiro Oikawa, and Yoshio Yamasaki (Dept. of Intermedia Art and Sci., Waseda Univ., Tokyo, Japan)

The piezoelectric actuators have been applied to various electrical devices such as piezoelectric speakers, buzzers, haptics, and ultrasonic transducers. The improvement of the electromechanical conversion efficiency is the most important issue of piezoelectric actuator systems in mobile devices. The electrical power consumption of actuators must be suppressed as possible, due to mobile devices having small batteries. The frequency response around the mechanical resonance should be carefully designed for low power consumption driving. The resonant frequencies of the piezoelectric actuators consist of integrated components, for example, metal corns of ultrasonic speakers, are decided by the energy dispersion of the total system. Therefore, design factors such as sizes and physical properties of each component, are necessary to optimize the resonant frequencies for practical applications. The total energy of the piezoelectric system is described by Lagrange-Maxwell equation. Even though it is not easy to solve the differential equations written in the Lagrangian coordinate system by the exact calculation, useful information for design of the system can be derived by the approximate calculation. In this paper, we will indicate the design guideline to optimize resonant frequencies of the piezoelectric actuators with integrated components, based on the analysis in the Lagrangian coordinate system.

10:20

3aEA5. Calculation of characteristics of nonlinear normal waves in plates of lithium niobate for the designing of acousto-electronic devices. Alina Kuslyva and Valery Storozhev (Donetsk National Univ., bul. Kramatorskij 21/33, Kramatorsk, Donetsk 84331, Ukraine, kuslivaya@gmail.com)

The research of anharmonic effects is essential for the design of nonlinear acousto-electronic devices. Such effects involve the generation of nonlinear second harmonics in propagation of normal electroelastic waves in crystal plates. Thereby the analytical and numerical technique of the analysis of small nonlinear anharmonic effects in distribution of normal electroelastic waves in the layer of a trigonal piezocrystal of lithium niobate with thin short-circuited electroconductive coverings of sides has been developed. The research is based on the model of physically and geometrically nonlinear electroelastic deformation with finite deformations and Gibbs's function that includes quadratic and cubic components on deformations and characteristics of intensity of quasistatic electric field. The analysis of nonlinear wave effects is build on the representation of characteristics of a normal electroelastic wave in the form of the sum of summands, which are proportional to the powers of the small parameter. The analytical form has been received for the representations of functions of the elastic displacements, intensity, induction of quasistatic electric field in nonlinear second harmonics for the studied waves from the different modes of the dispersive spectrum. Quantitative estimates have been researched for the amplitude levels of second harmonics for normal electroelastic waves with variable frequencies.

10:40

3aEA6. A numerical study of non-collinear mixing of three-dimensional nonlinear waves in an elastic half-space. Zhenghao Sun, Hongguang Li, and Fucai Li (State Key Lab. of Mech. System and Vib., School of Mech. Eng.,Shanghai Jiao Tong Univ., Rm. B322, 800 Dongchuan Rd., Shanghai 200240, China, sunzhh@163.com)

Interactions of two non-collinear nonlinear ultrasonic waves in an elastic half-space with quadratic and hysteretic nonlinearity are investigated in this paper. The numerical problem is solved by a semi-discrete central scheme as well as a finite element method in two and three dimensions, respectively. Features and intensity distribution of the resonant wave are analyzed both in time and frequency domains, and the method of non-collinear wave mixing is proved to be a promising method, which is both effective and sensitive in material characterization and structure damage detection.

11:00

3aEA7. Simulative measures for structure borne sound radiation of composites. Matthias Klaerner (Inst. of Lightweight Structures, Chemnitz Univ. of Technol., Reichenhainer Str. 70, Chemnitz 09126, Germany, matthias.klaerner@mb.tu-chemnitz.de), Steffen Marburg (Insitute of Mech., Universtitaet der Bundeswehr München, Neubiberg, Germany), and Lothar Kroll (Inst. of Lightweight Structures, Chemnitz Univ. of Technol., Chemnitz, Germany)

Due to the high stiffness-to-weight ratio, composite structures tend to be acoustic sensitive. The sound emission of such radiating components is commonly measured by the sound power requiring the determination of the sound intensity in normal direction and—in numerical simulations—the sound pressure on the radiating surface. Assuming a unit radiation efficiency all-over the surface and neglecting local effects, the equivalent radiated power (ERP) is a common approach for an upper bound of structure borne noise. Therein, the sound power finally results from the squared velocity integrated over the radiating surface and the fluid impedance. As ERP usually requires extra post processing to consider the velocity in normal surface direction, the kinetic energy is essential in common FEA results including all velocity components apart from the normal direction, too. Moreover, ERP necessitates the knowledge of the radiating surfaces increasing the effort especially for complex geometries. Thus, the possibilities and limits of estimating the emitted sound power by the kinetic energy as well as using the ERP method will be shown. Test cases are a rectangular plate and a thin-walled bonded part with linear anisotropic material properties.

11:20

3aEA8. Transduction as energy conversion: Harvesting of acoustic energy in hydraulic systems. Ellen A. Skow, Kenneth Cunefare, and Alper Erturk (School of Mech. Eng., Georgia Inst. of Technol., Georgia Tech, Atlanta, GA 30332-0405, eskow3@gatech.edu)

Energy harvesting from acoustic energy sources is a form of transduction. While energy densities in typical airborne acoustic noise fields are extremely low, those in hydraulic systems may be orders of magnitude greater and represent an opportunity for direct energy conversion from piezoelectric materials for powering sensor and communication nodes. Hydraulic systems are challenging from a design perspective in that the device must be capable of withstanding static pressures up to and exceeding 35 MPa, while being simultaneously exposed to dynamic pressures on the order of 3.5 MPa. Hydraulic pressure energy harvester devices have been developed to exploit the high energy densities of dynamic pressures in hydraulic systems. There is an immediate application for this technology in that state-of-the-art hydraulic hose and piping systems employ integral sensor nodes for structural health monitoring for early detection of incipient failures. This paper presents the acoustic and electromechanical modeling of the piezoelectric power output from dynamic pressure in terms of the force transmitted into an energy harvester designed for hydraulic systems.

11:40

3aEA9. Accurate determination of piezoelectric ceramic constants using a broadband approach. Nicolas Perez (Centro Universitario de Paysandu, Universidad de la Republica, Montevideo, Uruguay), Marco Aurelio B. Andrade, Ronny C. Carbonari (Biomedical Eng., Federal Univ. of ABC, Sao Paulo, Brazil), Julio C. Adamowski (Mechatronics Eng., Univ. of Sao Paulo, Sao Paulo, Brazil), and Flavio Buiochi (Mechatronics Eng., Univ. of Sao Paulo, Av. Prof. Mello Moraes, 2231, Butanta, Sao Paulo, SP 05508-030, Brazil, fbuiochi@usp.br)

Piezoceramic property values are required for modeling piezoelectric transducers. Most datasheets present large variations in such values. For precise simulations, adjustments are necessary. Recently, the authors presented a methodology to obtain the real part of ten material constants of piezoelectric disks. It comprises four steps: experimental measurements, identification of vibration modes and their sensitivity to material constants, preliminary identification algorithm, and final refinement of the constants using an optimization algorithm. Given an experimental electrical impedance curve of a piezoceramic and a first estimate for the material constants, the objective is to find the constants that minimize the difference between the experimental and numerical curves. Using a new finite element method routine implemented in MATLAB, the original methodology was extended to obtain the corresponding imaginary part of all the material constants. Results of sensitivity analysis for the imaginary part and the guidelines to construct an algorithm are presented. This complex model allows adjusting the amplitude over a wide frequency range, as opposed to the models described in the literature. It is applied to 1-MHz APC850 disks with diameters of 10 and 20 mm. The methodology was validated by comparing the numerical displacement profiles with the displacements measured by a laser Doppler vibrometer.

WEDNESDAY MORNING, 5 JUNE 2013 — 510C, 8:55 A.M. TO 12:20 P.M.

Session 3aED

Education in Acoustics and Psychological and Physiological Acoustics: Learning by Listening: Education in Acoustics Based on Listening

Akira Nishimura, Cochair
Media and Cultural Studies, Tokyo Univ. of Information Sci., 4-1, Onaridai, Wakaba-ku, Chiba 2658501, Japan

Kaoru Ashihara, Cochair
AIST, AIST Tsukuba Central, 1-1-1 Higashi, Tsukuba 3058566, Japan

Chair's Introduction—8:55

Invited Papers

9:00

3aED1. Technical ear training: Tools and practical methods. Jason Corey (School of Music, Theatre & Dance, Univ. of Michigan, 1100 Baits Dr., Ann Arbor, MI 48109-2085, coreyja@umich.edu)

Broadly defined, technical ear training seeks to make associations between aural impressions of sound quality and quantifiable characteristics of audio signal processing and acoustical measurements. Technical ear training typically focuses on attributes of sound such as spectral balance (e.g., filtering and parametric equalization); dynamic range of musical signals (including artifacts produced by dynamics processing); reverberation, delay, and early reflections (from real acoustic spaces or generated artificially); and spatial extent (width and depth). These elements of recorded sound can be broken down into graduated levels of audibility for the development of critical listening skills. With repeated and regular practice of carefully chosen exercises, listeners can gain increased sensitivity to subtle details of sound, as well as efficiency and accuracy in identifying specific parameters of signal processing by ear. With applications

primarily in sound recording and production, technical ear training is also highly relevant to the evaluation of acoustical spaces as a complement to objective measurements. This presentation will review a selection of software modules developed by the author to teach critical listening skills to undergraduate students. The author will also discuss some practical methods and exercises used for teaching technical ear training and critical listening.

9:20

3aED2. Learning acoustic phonetics by listening, seeing, and touching. Takayuki Arai (Dept. of Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

There is a huge volume of written textbooks available in virtually every modern field, including acoustic phonetics. However, in areas dealing with acoustics, learners often face problems and limitations when they deal with only written material and no audio or visual information. As one response to this problem, we have developed several sets of physical models of the human vocal tract and have shown that they are extremely useful for intuitive understanding. In addition, we also developed a tool called "Digital Pattern Playback." Another solution is an online version featuring demonstrations. We are currently collecting materials, mainly in the form of sounds, for educational purposes in acoustics and phonetics and are releasing them as "Acoustic-Phonetics Demonstrations" through our Web site. These demonstrations are designed for students in linguistics, phonetics and phonology, speech pathology, audiology, psychoacoustics, speech engineering, and others. However, potential users are not limited to these groups, as we feel that a wide range of learners can obtain tremendous benefits from the demonstrations, including those who are studying foreign languages or patients undergoing speech articulation therapy. [Work partially supported by a Grant-in-Aid for Scientific Research (24501063) from the Japan Society for the Promotion of Science.]

9:40

3aED3. Enriching the aural experience in audio education. Alex Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Audio education's essential outcome is aural. Lectures and readings on aesthetics, techniques, and technologies can never communicate audio concepts effectively without critical elucidation through sound. Quality audio education has always made frequent use of laboratories, recording sessions, and critical listening classrooms to keep sound at the center of student learning. Recently authored and published web-based multimedia and digital audio workstation self-study experiences are discussed and demonstrated. The sonic illustrations, visual reinforcement, and associated interactivity are found to provide meaningful pedagogical advancements in audio education.

10:00

3aED4. Sonority in British English. Yoshitaka Nakajima, Kazuo Ueda (Dept. of Human Sci., Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka, Fukuoka 815-8540, Japan, nakajima@design.kyushu-u.ac.jp), Shota Fujimaru (Human Sci. Course, Kyushu Univ., Fukuoka, Japan), and Yuki Ohsaka (The 21st Century Program, Kyushu Univ., Fukuoka, Japan)

Our previous study on British English speech [Nakajima *et al.*; Fechner Day (2012), Ottawa] was extended into the domain of phonology. Factor analyses were performed on the power fluctuations of the outputs of 19 critical-band filters, which separated British English sentences uttered by two female speakers and one male speaker into narrow-band signals. A database of British English spoken sentences [The ATR British English speech database (Campbell, 1993)] was used for the present analysis, because each identifiable speech sound was indicated in the speech waveform with a label of a phonetic symbol. About 80% of 31,663 labels were considered to represent English phonemes. Three factors appeared as in our previous research [Ueda *et al.*; Fechner Day (2010), Padova], and one of them corresponding to a frequency range of about 600–1800 Hz was closely related to sonority or aperture described in linguistics literature. The acoustic sonority could be related to a few phonological phenomena: (1) A sonorant consonant immediately after an obstruent can be a syllable nucleus, (2) a consonant cluster at the beginning of a word mostly begins with an obstruent, and (3) a short schwa cannot be a nucleus of a stressed syllable.

Contributed Papers

10:20

3aED5. Toward the development of objective difficulty measure in technical ear training tasks. Atsushi Marui and Toru Kamekawa (Faculty of Music, Tokyo Univ. of the Arts, 1-25-1 Senju, Adachi, Tokyo 120-0034, Japan, marui@ms.geidai.ac.jp)

Technical Ear Training is a method to improve the ability to focus on a specific sound attribute. It is also used to be able to communicate using the common language used in the industry such as Hz and dB. Although it is essential to gradually harden the task difficulty for successful technical ear training, the objective measure of the difficulty is still not known. Therefore, the tasks are decided by the teacher's own ears and experiences, leading to inefficiency when students want to train themselves in the teacher's absence. As the first step toward understanding this tacit ability of knowing the task difficulty, the authors investigated the correlation between the students' subjective ratings of the task difficulty and the physical measures calculated from the sound materials used in the training. A linear regression model ($R^2 = 0.629$) which predicts the subjective task difficulty from residual of the linear fit through the spectra of the sound material was created. This model may provide a firm step toward the goal of developing objective difficulty measure in technical ear training.

10:40

3aED6. Human echolocation system using a miniature dummy head. Shunsuke Uchibori, Masataka Kinoshita (Faculty of Life and Med. Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, dmm1028@mail4.doshisha.ac.jp), Kaoru Ashihara (National Inst. of Adv. Industrial Sci. and Technol., Ibaraki, Japan), Tetsuo Ohta, and Shizuko Hiryu (Faculty of Life and Med. Sci., Doshisha Univ., Kyotanabe, Japan)

To promote understanding of sonar mechanisms in bats, we propose a novel tool that makes echolocation available for humans. In this method, ultrasonic echoes are captured by a miniature dummy head so that they can be converted to binaural audible sounds using time expansion.In order to examine the effectiveness of this technique, perceptual listening tests were conducted on human listeners with normal hearing.The sounds (white noise with frequencies between 5 and 90 kHz with 0.7-s duration, including 0.05-s rise/fall time) were recorded at a distance of 1 m from a loudspeaker

using two condenser microphones that were placed in the ear canals of a 1/7 size miniature dummy head. The recorded ultrasounds were 1/7-times pitch converted, and then were presented to the listener through headphones. As a result, the listeners perceived correct directions of the pitch converted sounds, which were recorded using the miniature dummy head, although front-back error was occasionally observed. When the miniature dummy head was rotated during the recording, the listeners perceived the movement of the sounds as out-of-head sound localization. The miniature dummy head may provide humans with a tool to understand biosonar mechanisms.

11:00

3aED7. Capturing spatial audio information by using a miniature head simulator. Kousuke Taki (Tokyo City Univ., 105, 2-4-13, Noge, Setagaya-ku, Tokyo 158-0092, Japan, g0922040@tcu.ac.jp), Kaoru Ashihara (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Ibaraki, Japan), and Shogo Kiryu (Tokyo City Univ., Tokyo, Tokyo, Japan)

A conventional dummy head or a microphone array with a number of microphones has been used to record spatial audio information. We propose an audio capture system called "Miniature head simulator" that consists of a microphone system and a signal processor. Instead of using a conventional dummy head, a small microphone system that consists of three omni-directional microphones is used to capture acoustic signals. The signals are then encoded to a custom data format and transmitted online. The transmitted data can be decoded and reproduced as binaural signals. Because of its size and weight, the conventional dummy head has been used exclusively for the research purpose. A miniature head simulator can provide much more convenient tools to record spatial audio information. Since it deals with only three channels of audio stream, data can be processed with relatively low computational cost. By using a miniature head simulator, spatial audio information can be streamed to the browser or even to the smartphones of the end-users. It can be used in a remote acoustic sensor and remote control system, sound detection system, video conference, and various other fields of industry.

11:20

3aED8. Effectiveness of technical listening training in Department of Acoustic Design of Kyushu University. Kazuhiko Kawahara, Masayuki Takada, and Shin-ichiro Iwamiya (Faculty of Design, Kyushu Univ., 4-9-1 shiobaru, Minami-ku, Fukuoka 815-8540, Japan, kawahara@design.kyushu)

What is the professional listening? The listening ability of Sound/Acoustic Professionals listening is categorized into three phases: the ability to discriminate between different sounds, the ability to correlate the auditory difference with the physical properties of sounds, and the ability to imagine the proper sounds when given the acoustic properties of the sounds. These kinds of ability can be trained through listening training. Furthermore, through trainings, trainees can share same auditory experiences. The shared experiences improve ability of trainees to express their auditory impression with appropriate words and this ability contributes to smooth communication on auditory imagery in their group. In this paper, as a listening training, the overview and effectiveness of Technical Listening Training in Kyushu University is described. To evaluate the effectiveness of training, the average correct answer ratios of trainees were examined through the training. The improvement of correct answer ratios was observed. We could show the effectiveness of Technical Listening Training.

11:40

3aED9. Effect of speaking rate variation on the perception of singleton and geminate consonants in Japanese by native and Korean listeners. Mee Sonu, Takayuki Arai (Faculty of Sci. and Technol., Sophia Univ., 7-1 Kioi-Cho, Chiyoda-ku, Tokyo 102-8554, Japan, sonumeephonetic@gmail.com), Hiroaki Kato (National Inst. of Information and Commun. Technol., Kyoto, Japan), and Keiichi Tajima (Dept. of Psych., Hosei Univ., Tokyo, Japan)

Perception of phonemic length contrasts in Japanese is difficult for non-native listeners. To better understand the source of this difficulty, the present study investigated native Korean listeners' perception of consonant length contrasts at different speaking rates. Stimuli were created by modifying the duration of the second consonant of a non-word /ereC:e/ along a continuum to /ereCe/, where C was /k/ or /s/. The base words were spoken by a professionally trained native Japanese speaker with a carrier sentence at three rates, fast, normal, slow. Twenty-seven native Korean and eleven native Japanese listeners participated in a perception test. They listened to one of the created stimuli and identified whether the second consonant was singleton or geminate. Results show that even though Korean listeners' perceptual boundary location between singleton and geminate consonants shifted according to speaking rate in a similar manner as the natives, their boundary location was more variable than native listeners at all speaking rates. Korean listeners also showed greater perceptual boundary width than Japanese listeners. These results suggest that Korean listeners have ambiguous criteria for phonemic length contrasts. Results are discussed in terms of the perceptual similarity between Korean and Japanese consonants. [Work supported by JSPS.]

12:00–12:20 Demonstrations

Session 3aMU

Musical Acoustics and Signal Processing in Acoustics: Aeroacoustics of Wind Instruments and Human Voice II

Shigeru Yoshikawa, Cochair
Grad. School of Design, Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan

Xavier Pelorson, Cochair
CNRS, 11 rue des mathematiques, Saint Martin d'Heres 38330, France

Invited Papers

9:00

3aMU1. Adaptive stabilized finite element framework for simulation of vocal fold turbulent fluid-structure interaction and towards aeroacoustics. Johan Jansson (KTH Royal Inst. of Technol. and Basque Ctr. for Appl. Mathematics, CSC KTH, Stockholm SE-10044, Sweden, jjan@kth.se)

As a step toward building a more complete model of voice production mechanics, we assess the feasibility of a fluid-structure simulation of the vocal fold mechanics in the Unicorn incompressible Unified Continuum framework. The Unicorn framework consists of conservation equations for mass and momentum, a phase function selecting solid or fluid constitutive laws, a convection equation for the phase function and moving mesh methods for tracking the interface, and discretization through an adaptive stabilized finite element method. The framework has been validated for turbulent flow for both low and high Reynolds numbers and has the following features: implicit turbulence modeling (turbulent dissipation only occurs through numerical stabilization), goal-oriented mesh adaptivity, strong, implicit fluid-structure coupling, and good scaling on massively parallel computers. We have applied the framework for turbulent fluid-structure interaction simulation of vocal folds, and present recent results. Initial steps toward aeroacoustics have been carried out in the framework (exhaust system and landing gear applications), and will be presented as well as the current state of aeroacoustic modeling for the human voice in the framework.

9:20

3aMU2. Synergistic interactions underlying the production of voice. Tokihiko Kaburagi (Grad. School of Design, Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8540, Japan, kabu@design.kyushu-u.ac.jp)

This paper presents recent progress in research examining the mechanisms of voice generation and a method for physiologically based speech synthesis. The overarching goal of this research is the precise modeling of interactions among physical systems involved in the processes underlying voice generation. In the basic voice generation system, a flow-structure interaction between glottal flow and the vocal folds causes self-oscillations of the folds, where flow separation, a nonlinear aerodynamic phenomenon, plays an important role. The fluid dynamic theory implies that a thin boundary layer formed near the glottal wall characterizes the flow behavior, including flow separation, jet formation, and pressure loss across the channel. We therefore use the interactive boundary layer method to analyze glottal flow and show how the flow-structure interaction is effective in maintaining vocal fold oscillations. In addition, the interaction between the voice generation system and the vocal-tract filter, i.e., the source-filter coupling, has been found to involve nonlinear factors in speech, such as skewing of glottal flow, unsteadiness in vocal fold oscillations, and transitions in voice register.

9:40

3aMU3. Computational analysis of the dynamic flow in single-reed woodwind instruments. Andrey R. da Silva (Structures and Civil Eng., Federal Univ. of Santa Maria, Av. Roraima 1000, Santa Maria, Rio Grande do Sul 97050421, Brazil, andrey@eac.ufsm.br), Shi Yong, and Gary Scavone (Music, McGill Univ., Montreal, QC, Canada)

The dynamics of the air flow within the mouthpiece of single-reed wind instruments makes an important contribution to the acoustic behavior of this type of system, particularly during transient regimes. In this work, a two-dimensional numerical model of the mouthpiece-reed system is used to evaluate the behavior of the flow within the mouthpiece at three different frequencies within the playing range of the clarinet. The relationship between volume flow and blowing pressure, as well as the behavior of the vena contracta during one duty cycle are compared with the available analytical models and with recent experimental observations.

10:00

3aMU4. Theoretical and experimental study of glottal geometry in phonation. Xavier Pelorson, Annemie Van Hirtum, Bo Wu, and Fabrice Silva (Depertement Parole et Cognition, Gipsa-Lab, 11 rue des mathematiques, Saint Martin d'Heres 38330, France, xavier.pelorson@gipsa-lab.grenoble-inp.fr)

Most existing theoretical models of phonation assume that the vocal folds are parallel and that the glottis forms a two-dimensional channel for the flow. However, during phonation the vocal folds can be very accurately abducted or adducted using intrinsic muscles acting on the arytenoid cartilages. The resulting shape of the glottis can then vary between an almost uniform slit (when the vocal folds are parallel) to a V shape

(with an angle between the vocal folds up to 20 °). Further, the vocal folds surface can present some irregularity which can be severe, in some pathological cases such as cysts or nodules. In this paper, we present a theoretical and experimental study performed in order to evaluate and to predict these geometrical effects. Several theoretical models to predict the pressure losses in a non-uniform glottis will be presented and tested against an *in-vitro* experimental set-up using a self-oscillating latex replica of the vocal folds. The angle between the artificial folds could be controlled using micrometers while surface irregularities could be simulated by inserting small spheres with various masses and diameters.

10:20

3aMU5. Influence of epithelium and fiber locations on glottal closure and sound production at soft-phonation conditions. Zhaoyan Zhang and Yue Xuan (UCLA School of Med., 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Previous studies showed that isotropic vocal fold models often vibrated with incomplete glottal closure at onset despite the vocal folds were in contact at rest. This contrasts with human phonation in which complete glottal closure is observed even during soft phonation with minimal or low laryngeal muscle contraction. Based on previous experimental studies, we hypothesize that this difference in glottal closure patterns is due to the relatively large stiffness in the anterior-posterior direction or the presence of the epithelium layer. These hypotheses were tested in self-oscillating physical vocal-fold models, with anisotropic stiffness conditions simulated by fibers loosely imbedded at different locations in otherwise isotropic vocal folds. The results showed that, compared to isotropic one-layer models, the presence of a stiff epithelium layer led to complete glottal closure along the anterior-posterior direction, increased maximum glottal opening, strong excitation of high-order harmonics in the resulting voice spectra and reduced noise production. Similar improvement in glottal closure and high-order harmonics excitation was observed with fibers in the cover layer, but to a less degree. Presence of fibers in the body-layer led to reduced maximum glottal opening but did not yield noticeable improvement in glottal closure and harmonic excitation. [Work supported by NIH.]

Contributed Papers

10:40

3aMU6. Synchronous visualization of multimodal measurements on lips and glottis: Comparison between brass instruments and the human voice production system. Thomas Hézard (IRCAM - CNRS UMR 9912 - UPMC, 1, place Igor Stravinsky, Paris 75004, France, thomas.hezard@ircam.fr), Vincent Fréour (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montréal, QC, Canada), René Caussé, Thomas Hélie (IRCAM - CNRS UMR 9912 - UPMC, Paris, France), and Gary P. Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montréal, QC, Canada)

Brass instruments and the human voice production system are both composed of a vibrating "human valve" (constriction in a pipe) coupled to an acoustic resonator: lips coupled to the brass instrument or vocal folds coupled to the vocal tract. In both cases, the aeroacoustic coupling is responsible for the self-oscillations and the large variety of regimes. Additionally, brass instruments and voice share difficulties for the *in-vivo* measurement of the exciter activity. Hence, the development of a common tool is relevant. It is also relevant to explore the effect of some known differences between these systems, namely, the strength of the coupling and the physiological characteristics. This paper introduces components for the development of such a tool. First, two corpuses of multimodal measurements are presented: one for a singer's larynx during sustained vowels, one for a musician's lips during sustained notes. They include high-speed video (HSV) recordings, electrical impedance (EI) measurements (electrolabiograph/electroglottograph), and audio recordings (AR). Then, we introduce two estimation algorithms: (AW) one of the opening area waveforms from videos, (LF) one of the LF-model parameters on these waveforms. Moreover, we build a video tool displaying, synchronously, the signals (HSV), (AW), (EI), and (AR) in time and frequency domains. Finally, this tool is exploited to exhibit common behaviors and relevant differences between brass instruments and human voice.

11:00

3aMU7. Physical modeling of bilabial plosives production. Louis Delebecque, Xavier Pelorson, Denis Beautemps, and Xavier Laval (Speech and Cognition, GIPSA-lab, 11 rue des Mathématiques, Grenoble Campus BP46, Saint-martin d'hères F - 38402, France, louis.delebecque@gipsa-lab.grenoble-inp.fr)

The context of this study is the physical modeling of speech production. The first step of our approach is to realize *in vivo* measurements during the production of the vowel-consonant-vowel sequence /apa/. This measurements concerns intra oral pressure, acoustic pressure radiated at the lips and labial parameters (aperture and width of the lips) derived from a high-speed video recording of the subject's face. In a second time, theoretical models from speech production literature are under investigation to describe air flow in the lips. Their prediction are compared with measurements obtained using an experimental set-up including a replica of vocal folds, which are able to self-oscillate and a rigid replica of lips. Finally, this validation allows us to achieve numerical simulations of the sequence /apa/. The comparison between the measured intra oral pressure and the simulated one leads us to take into account for the cheeks expansion in the physical modeling of bilabial plosives.

11:20

3aMU8. Influence of cross section shape on the outcome of a two-mass model. Bo Wu, Annemie Van Hirtum (Gipsa-Lab, Grenoble Univ., 11 rue des Mathematiques, Grenoble 38402, France, annemie.vanhirtum@gipsa-lab.grenoble-inp.fr), and Xiaoyu Luo (Dep. of Mathematics, Glasgow Univ., Glasgow, United Kingdom)

For the last decades, the two-mass model has shown its value to model the fluid-structure interaction during voiced speech production. Its main interest lies in its simplicity since it allows a quasi-analytical solution for a complex phenomenon using only a limited amount of physical meaningful parameters. Nevertheless, the use of the two mass model with respect to model speech pathologies can be questioned due to the lack of detail in the used mechanical and/or flow model. In the current paper, we focus on the influence of the cross section shape taken into account. Indeed, varying the cross section shape is likely to alter the pressure distribution due to viscous effects. The influence of the cross section shape on the flow outcome is modeled as well as experimentally assessed. Next, the influence of varying the cross section shape on phonation parameters such of the threshold pressure and fundamental frequency are addressed by considering a linear stability analysis of the two mass model.

11:40

3aMU9. The spectral and acoustical impact of vowel changing in choral tuning. Bertrand Delvaux and David Howard (Dept. of Electron., Univ. of York, Heslington, York, York YO10 5DD, United Kingdom, bertrand.delvaux@gmail.com)

Mixed soft/solid models of the vocal tract were moulded with a 3D rapid prototyping technique based on MRI data obtained from two male singers during the phonation of five English vowels as in hard, stern, neap, port, and food. The replicas are used to assess the interaction of several vocal tracts in different settings: twice the same singer or two different singers, singing on the same vowel or on different vowels, on a consonant or a dissonant interval. The spectral output is analyzed and the acoustical output is submitted to a listening test to evaluate the spectral and acoustical grounds for an interval/chord to perceptually sound "in tune."

Session 3aNSa

Noise, ASA Committee on Standards, Engineering Acoustics, and Structural Acoustics and Vibration: Wind Turbine Noise I

Nancy Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118

Paul Schomer, Cochair
Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821

Sheryl Grace, Cochair
Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215

Invited Papers

9:00

3aNSa1. Activities of the Acoustical Society of America's subcommittee on wind turbine noise and some studies being done. Nancy Timmerman (Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118, nstpe@hotmail.com)

This paper will document the activities of the Acoustical Society of America's (ASA's) subcommittee (of the Panel on Public Policy) on Wind Turbine Noise, including what technical committees are represented, what special sessions will be held in the future, and the goal to generate a policy statement on the topic. The author, who is Chair of this subcommittee, will also describe what other current studies are or have been done in Massachusetts (in the United States) and, if applicable, elsewhere.

9:20

3aNSa2. Development of a real time compliance system for wind farms regulated by ambient-relative noise standards. Michael Hankard (Hankard Environ., 211 East Verona Ave., Verona, WI 53593, mhankard@hankardinc.com)

Some noise level regulations in the United States require wind turbine farms to not exceed the ambient sound levels at nearby residences by more than a fixed amount. For the project discussed herein, compliance with such a regulation requires curtailment of turbine operations to some degree at times when turbine operations are at or near maximum, atmospheric conditions are conducive to sound propagation, and sound from other sources including vegetation rustle are at a minimum. Based on the analysis of months of time-synchronized sound, meteorological, and operations data, a system was developed to assess compliance on a real-time, ongoing basis. The primary element in the determination of compliance is the shape of the one-third octave band spectrum at the residence, augmented by ground and hub-height meteorological conditions, and wind farm operations information. Without the spectral filter, curtailment would need to take place under a broader array of meteorological conditions to ensure compliance, which would result in loss of power generation revenue. This paper will describe the data collection and analysis methods, the development of the spectral filter, and the results of field testing including both the partial and entire shut down of the wind farm.

9:40

3aNSa3. Criteria for wind-turbine noise immissions. George Hessler (Hessler Assoc., Inc., 3862 Clifton Manor Place, Ste. B, Haymarket, VA 20169, George@HesslerAssociates.com) and Paul Schomer (Schomer and Assoc., Inc., Champaign, IL)

Each of the two authors has developed recommended single, 24-h, constant wind turbine noise criterion; the criteria are constants because wind turbine noise is basically not adjustable. Hessler develops his criterion from his knowledge of how wind turbine noise is being regulated at the local, state, and national levels, from regulations in other countries, and from his extensive experience with numerous wind turbine projects. Schomer develops his recommended criterion on the basis of existing national and international standards; notably ISO 1996-Part 1 and ANSI/ASA S12.9 parts 4 and 5. Ultimately, Hessler comes up with a single, 24-h A-weighted average criterion of 40 dB, and Schomer comes up with a 24-h A-weighted average criterion of 39 dB. These two researchers have decidedly different backgrounds, different experience, and a slight difference in orientation towards the industry. Thus, it is remarkable that these two criteria, derived in such different ways result in nearly identical 24-h A-weighted criteria levels. Although there is essential agreement in immissions criterion, there are variables debated herein for both modeling wind turbine emissions and certifying such emissions at far-off receptors that could result in a 10 dBA difference in the actual immissions level.

10:00

3aNSa4. Prevalence of complaints related to wind turbine noise in northern New England. Kenneth Kaliski (RSG Inc., 55 Railroad Row, White River Junction, VT 05001, ken.kaliski@rsginc.com)

As of September 2012, there were a dozen large operating wind projects with a total capacity of approximately 565 MW in northern New England, with more coming online by the end of the year. This paper evaluates the prevalence of noise complaints to regulatory authorities from those wind projects. Where possible, the exposure of residences to wind turbine sound is calculated. Exposure is

estimated through standard ISO 9613-2 modeling procedures. A comparison of the exposure of complainants and non-complainants is made with the goal of assessing the prevalence of complaints at various modeled sound levels.

10:20–10:40 Break

10:40

3aNSa5. Can wind-turbine sound that is below the threshold of hearing be heard? Paul Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

This paper is geared toward wind-turbine sound, but it is really a simple variation on the basic concepts that this author used in the development of loudness-level-weighted sound exposure (Schomer *et al*., J. Acoust. Soc. Am. **110**(5, Pt. 1), 2390–2397 (2001)] and of Rating Noise Curves (RNC) [Schomer, Noise Control Eng. J. **48**(3), 85–96 (2000)], which are used in our Standard, ANSI/ASA S12.2 Criteria for evaluating room noise. The fundamental issue is: Can we hear slowly surging or pulsating sounds for which the LEQ spectrum is below the threshold of hearing, where "slowly" means that the pulses come at a rate that is no faster than about 4 pulses per second? The short answer is yes, and the longer answer is that this effect is a function of the spectral content and becomes more-and-more prominent as the spectral content goes lower-and-lower in the audible frequency range. So surging or pulsing sound that is primarily in the 16 or 31 Hz octave bands will show the greatest effect. This paper shows the applicability of these results to wind-turbine sound. Variation in the threshold of hearing at low frequencies is an additional factor that also is discussed in this paper.

11:00

3aNSa6. Amplitude modulation of audible sounds by non-audible sounds: Understanding the effects of wind turbine noise. Jeffery Lichtenhan and Alec Salt (Otolaryngology, Washington Univ. in St. Louis, 660 South Euclid, St. Louis, MO 63110, LichtenhanJ@ent.wustl.edu)

Our research has suggested a number of mechanisms by which low-frequency noise could bother individuals living near wind turbines: causing endolymphatic hydrops, exciting subconscious pathways, and amplitude modulation of audible sounds. Here we focus on the latter mechanism, amplitude modulation. We measured single-auditory-nerve fiber responses to probe tones at their characteristic frequency in cats. A 50 Hz tone, which did not cause an increase in spontaneous firing rate (i.e., was not audible to the fiber when presented alone) was used to amplitude modulate responses to the probe tone. We found that as probe frequency decreased, a lower level of the low-frequency non-audible tone was needed to achieve criterion amplitude modulation. In other words, low-frequencies that are coded in the cochlear apex require less low-frequency sound pressure level to be amplitude modulated as compared to higher-frequencies that are coded in the cochlear base. This finding was validated, and extended to lower frequencies, by amplitude modulating gross measures of onset-synchronous (compound action potentials) and phase-synchronous (auditory nerve overlapped waveforms) in guinea pigs. Our results suggest that that infrasound generated by wind turbines may cause amplitude modulation of audible sounds, which is often the basis for complaints from those living near wind turbines.

11:20

3aNSa7. Generation of wind turbine noise signature for use in lab environment. Aleks Zosuls (Biomedical Eng., Boston Univ., Boston, MA), R. Morgan Kelley (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, sgrace@bu.edu), David Mountain (Biomedical Eng., Boston Univ., Boston, MA), and Sheryl Grace (Mech. Eng., Boston Univ., Boston, MA)

The fact that wind turbines produce infrasound continues to draw attention and discussion. Some argue that while the infrasound level produced by wind turbines is quite low, it still may be affecting the vestibular system or the hearing system, particularly via activation of the outer hair cells. Others hypothesize that the infrasound may be inducing whole body, chest cavity, or other human organ resonance. In order to study these hypotheses, it is first necessary to be able to recreate the turbine noise signature in a lab environment. Thus, the goal of this work is to create an acoustic system that can produce low-level infrasound. The system requirements include low cost, high fidelity, and imperceptible structural coupling to the lab. In addition, the system must be able to produce a broadband spectrum as well as a single tone. Progress toward the design of this audio system is discussed in this paper.

Contributed Papers

11:40

3aNSa8. Wind turbine sound prediction–The consequence of getting it wrong. William K. Palmer (TRI-LEA-EM, 76 Sideroad 33-34 Saugeen, RR 5, Paisley, ON N0G2N0, Canada, trileaem@bmts.com)

The application to permit a wind turbine power development usually involves submission of a prediction for the sound level that will occur at residences, schools, places of worship, and elsewhere people gather for restorative rest. This paper uses the example of a wind power development, and follows iterations taken to finalize the sound level prediction. The paper provides quantitative information collected since the start up of the wind power development on measured sound levels and octave band distribution; and qualitative observations on the special characteristics of the sound. Actual observations are compared to the predictions. More importantly, the paper reviews the consequences self-reported in qualitative interviews by citizens living with the changed environment after four years of operation of the wind power development. Reported impacts included difficulty sleeping, loss of jobs, and changes to social relationships, caregiving, pursuit of hobbies, leisure, learning, and overall health. Changes in measured health outcomes are identified. Both the quantitative and qualitative findings justify revision of the permitting process.

12:00

3aNSa9. Predicting underwater radiated noise levels due to the first offshore wind turbine installation in the United States. Huikwan Kim, James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Road, Narragansett, RI 02882, hkkim524@my.uri.edu)

Noise generated by offshore impact pile driving radiates into the air, water, and sediment. Predicting noise levels around the support structures at sea is required to estimate the effects of the noise on marine life. Based on high demands developing renewable energy source, the United States will

begin the first pile driving within one to two years. It is necessary to investigate acoustic impact using our previously verified coupled Finite Element (Commercial FE code Abaqus) and Monterey Miami Parabolic Equation (2D MMPE) models [J. Acoust. Soc. Am. **131**(4), 3392 (2012)]. In the present study, we developed a new coupled FE-MMPE model for the identification of zone of injury due to offshore impact pile driving. FE analysis produced acoustic pressure outputs on the surface of the pile, which are used as a starting field for a long range 2D MMPE propagation model. It calculates transmission loss for N different azimuthal directions as function of distance from the location of piling with the inputs of corresponding bathymetry and sediment properties. We will present predicted zone of injury by connecting N different distances of equivalent level fishes may get permanent injury due to the first offshore wind farm installation in the United States.

WEDNESDAY MORNING, 5 JUNE 2013 — 511CF, 9:00 A.M. TO 11:40 A.M.

Session 3aNSb

Noise: Aviation, Aviation Engines, and Flow Noise

Victor Sparrow, Chair
Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802

Contributed Papers

9:00

3aNSb1. A robust numerical approach for prediction of turbofan engine noise. Kaveh Habibi, Hao Gong, and Luc Mongeau (McGill Univ., Apt. 713, 3555 Berri St., Montreal, QC H2L4G4, Canada, kaveh.habibi@mail.mcgill.ca)

Noise from the aircraft jet engines is still the dominant source in takeoff condition. Government regulations of community noise are getting more stringent, creating significant challenges for aircraft manufacturers to meet these requirements. Expensive nature of experiments as well as accessing to powerful computers have provided new impetus for computational studies. In this paper, an efficient and robust numerical scheme, namely the Lattice Boltzmann Method was used to simulate the sound created by typical internal mixing nozzles with forced mixers. The simulation includes capturing the time-resolved flow characteristics and large scale turbulent structures. The sub grid scales were modeled using the renormalization group (RNG) forms of the standard k-ε equations. Several test cases including cold and hot core flow experiments conducted by NASA were selected for computational setup and validation purposes. The far field sound was predicted using a surface integral method. The near-field simulation results such as the jet centerline velocity decay and turbulence intensities as well as far-field sound were qualitatively in agreement with experimental results. The far-field sound analysis suggested significant low-frequency noise reduction for the lobed mixers, as well as significant reduction in overall sound pressure level (OASPL) in comparison with the simple confluent nozzle configurations.

9:20

3aNSb2. Airfoil flow and noise computation using monotonically integrated large eddy simulation and acoustic analogy: Effect of the grid resolution. Vasily A. Semiletov (Queen Mary, Univ. of London, Mile End Rd., London E1 4NS, United Kingdom, v.semiletov@qmul.ac.uk) and Sergey A. Karabasov (Queen Mary, Univ. of London, Cambridge, United Kingdom)

A new scalable Monotonically Integrated Large Eddy Simulation (MILES) method based on the Compact Accurately Boundary-Adjusting high-REsolution Technique (CABARET) has been applied for the simulation of unsteady flow around NACA0012 airfoil at $Re = 400{,}000$ and $M = 0.058$. The flow solution is coupled with the Ffowcs Williams–Hawkings formulation for far-field noise prediction. The computational modeling results are presented for several computational grid resolutions: 8, 16, and 32 million grid cells and compared with the experimental data available.

9:40

3aNSb3. Detached Eddy Simulation modeling and far-field trailing-edge noise estimation of a sharp-edged symmetric strut. Patrick G. Marshallsay, Laura A. Brooks, Alex Cederholm (Deep Blue Tech, Mersey Rd., Osborne, SA 5017, Australia, alex.cederholm@deepbluetech.com.au), Con J. Doolan, Danielle J. Moreau, and Cristobal Albarracin (School of Mech. Eng., The Univ. of Adelaide, Adelaide, SA, Australia)

This paper presents results of a Computational Fluid Dynamic (CFD) study of a sharp-edged symmetric flat strut at Reynolds number 500,000 based on chord at zero degrees angle of attack, and the subsequent estimation of far-field noise generated at the trailing-edge. Flow field results obtained using Detached Eddy Simulation (DES) modeling and Reynolds-averaged Navier Stokes (RANS) modeling techniques are compared with empirical wind-tunnel data. The flow is observed to be physically complex in nature, exhibiting numerical solutions that are sensitive to the mesh grid and freestream turbulence intensity. Although originally developed for use specifically with RANS-generated flow data, the RANS-based Statistical Noise Model (RSNM) technique, which estimates far-field noise from mean turbulence data via an acoustic Green's function and a statistical turbulence correlation model, is used here to estimate far-field noise spectra from both RANS and DES flow data. Far-field noise is also estimated from the DES model using the permeable surface form of the Ffowcs Williams and Hawkings (FWH) solver. The FWH estimate gives the closest match to experimental data, while the RSNM-generated noise estimate from the DES data appears to be more successful at capturing the large turbulent structures within the flow than the RANS data.

10:00

3aNSb4. Comparison of supersonic full-scale and laboratory-scale jet data and the similarity spectra for turbulent mixing noise. Tracianne B. Neilsen, Kent L. Gee, Alan T. Wall (Dept. of Phys. and Astronomy, Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, TN), and Anthony A. Atchley (Grad. Program in Acoust., Penn State Univ. Univ., State College, PA)

The broadband, partially correlated noise radiated from supersonic jets has characteristics that scale with nozzle size and flow properties. In particular, the spectral content of jet noise and the variation with angle in many cases agree with empirically derived similarity spectra for large and fine-scale components of turbulent mixing noise [Tam *et al.*, AIAA Paper 96-1716]. In previous studies, measurements made near the F-22 Raptor agreed remarkably well with the similarity spectra, with two exceptions. First, the

high-frequency slopes seen in the data were shallower than the similarity spectra at many angles. Second, the data exhibit a double frequency peak, which is absent from the similarity spectra [Neilsen *et al*., J. Acoust. Soc. Am. **132**, 1993 (2012)]. These observations are explored further by examining the spectral characteristics of noise from a different military jet and a laboratory-scale, unheated jet. In both cases, there is evidence that for supersonic cases the measured spectra are shallower than the similarity spectra due to nonlinear propagation effects. In addition, the military data support the observation that the double spectral peak is a feature of full-scale jet noise. Recommendations are made for applying the similarity spectra to predict spectral levels for full-scale jets. [Work supported by ONR.]

10:20–10:40 Break

10:40

3aNSb5. Autocorrelation analysis of military jet aircraft noise. Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astron., Brigham Young Univ., 562 N 200 E # 17, Provo, UT 84606, blaineharker@byu.net), Sally A. McInerny (Dept. of Mech. Eng., Univ. of Louisiana, Lafayette, LA), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

Jet noise research has seen increased use of autocorrelation analyses to glean physical insight about the source and its radiation properties. Length scales and other features have been identified in support of models incorporating large-scale (LSS) and fine-scale (FSS) turbulent structures. In this paper, the meaningful use of autocorrelation in jet noise analysis is further examined. A key finding is that the effect of the peak frequency on autocorrelation width needs to be removed prior to making conclusions about the relative LSS and FSS contributions. In addition, the Hilbert transform is applied to create an envelope of the autocorrelation function to more consistently define a characteristic time scale. These methods are first applied to the analytical LSS and FSS similarity spectra, previously developed by Tam *et al*. [AIAA 96-1716, 1996]. It is found that the envelope of the FSS similarity autocorrelation function is more similar to that of a delta function than the LSS envelope. These curves are used to more effectively quantify FSS and LSS features in noise spectra from the F-22A Raptor. [Work supported by ONR.]

11:00

3aNSb6. A multi-objective evolutionary optimization approach to procedural flight-noise mitigation. Andrew Christian and Victor Sparrow (Grad. Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, azc144@psu.edu)

Exposure to noise is a significant problem for communities that exist near airports. The distribution of noise exposure can be positively affected by changes in the procedures that aircraft follow in the vicinity of an airport (e.g., rate of ascent, ground track, etc.). When considering such changes, a decision maker often has to weigh the objective of lower noise impact against “more practical” considerations such as fuel consumption and time-of-flight. This study presents a method of numerical optimization which seeks to find the optimal-tradeoff set (Pareto front) of flight procedures given information about an airport and the surrounding population and geography. This front will only include procedures such that an aggregate noise metric cannot be improved without detriment to a more practical objective. A contemporary multi-objective evolutionary algorithm is used as the basis of the optimization effort. Results from a simulated military airfield near Asheville, NC are shown. Ways in which decision makers are empowered by having access to a Pareto front are discussed.

11:20

3aNSb7. Preliminary analysis of acoustic intensity in a military jet noise field. Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen, Alan T. Wall, David W. Krueger (Phys., Brigham Young Univ., 688 north 500 East, Provo, UT 84606, titorep@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Provo, Utah)

Acoustic intensity measurements of the F-22A Raptor are analyzed as part of ongoing efforts to characterize the noise radiation from military jet aircraft. Data were recorded from a rig of microphones and an attached tetrahedral intensity probe at various locations to the sideline and aft of the aircraft. Numerical analysis of the intensity at one-third octave band center frequencies along various measurement planes and at a 23 m radius arc reveals the magnitude and directionality of the vector acoustic intensity. Differences in the trends for low-frequency and high-frequency data are discussed and, via a simple ray tracing back toward the source, interpreted in terms of source location and extent. [Work supported by ONR.]

Session 3aPA

Physical Acoustics: Borehole Acoustics Logging for Hydrocarbon Reservoir Characterization I

Said Assous, Cochair
Geoscience, Weatherford, East Leake, Loughborough LE126JX, United Kingdom

Weichang Li, Cochair
ExxonMobil Res. & Eng., 1545 Rte. 22 East, Annandale, NJ 08801

Chair's Introduction—8:55

Invited Papers

9:00

3aPA1. Complexity penalized hydraulic fracture localization and moment tensor estimation under limited model information. Gregory Ely (ECE, Tufts Univ., 8 Richard Ave., Cambridge, MA 02140, gregory.ely@tufts.edu) and Shuchin Aeron (ECE, Tufts Univ., Medford, MA)

In this paper, we present a novel technique for micro-seismic localization using a group sparse penalization that is robust to the focal mechanism of the source and requires only a velocity model of the stratigraphy rather than a full Green's function model of the earth's response. In this technique, we construct a set of perfect delta detector responses, one for each detector in the array, to a seismic event at a given location and impose a group sparsity across the array. This scheme is independent of the moment tensor and exploits the time compactness of the incident seismic signal. Furthermore, we present a method for improving the inversion of the moment tensor and Green's function when the geometry of seismic array is limited. In particular, we demonstrate that both Tikhonov regularization and truncated SVD can improve the recovery of the moment tensor and be robust to noise. We evaluate our algorithm on synthetic data and present error bounds for both estimation of the moment tensor as well as localization. Furthermore we discuss the estimated moment tensor accuracy as a function of both array geometry and fault orientation.

9:20

3aPA2. Comparison of dispersive relation of the oil well with that of the fluid cylinder and the hollow in solid. Hailan Zhang, Hanyin Cui, Weijun Lin, and Xiuming Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., 21 Beisihuanxilu, Beijing 100190, China, zhanghl@mail.ioa.ac.cn)

Acoustical well logging is a widely used technique in oil fields to investigate the formation outside wells and the quality of wells. The knowledge of the acoustical behavior of the wells is important to the technique and has been widely studied. The wells are usually modeled as a fluid filled cylindrical borehole in an infinite solid medium. This structure with an infinite section, sometimes called as open waveguide, is more difficult to study than the typical acoustical waveguide with finite section and free or rigid boundaries. In this presentation, the well is studied as a coupled vibration system consisted of the fluid cylinder inside the well and the solid formation outside the well. The dispersive relations of three systems, i.e., the circular fluid cylinder with rigid boundary, the circular hollow in solid medium, and the fluid filled borehole in solid, are numerically calculated and compared. The properties of the dispersive relations, such as the cutoff frequencies, normal and abnormal dispersion, are compared and discussed. [Work supported by National Natural Sciences of China, Grant No. 11134011.]

9:40

3aPA3. Space-time methods for robust slowness estimation for monopole logging while drilling. Shuchin Aeron (Dept. of ECE, Tufts Univ., 161 College Ave., Medford, MA 02155, shuchin@ece.tufts.edu), Sandip Bose (Math and Modeling, Schlumberger Doll Res., Cambridge, MA), and Henri-Pierre Valero (Acoust.-Sonic, Schlumberger K.K., Kanagawa-Ken, Japan)

In this paper, we present methods for interference cancelation for robust slowness estimation from noisy monopole logging while drilling (LWD) data. The main contributions are two fold. First, we show via tests on real data sets presence of systematic propagative interferences in monopole LWD data, which is the primary reason for loss of compression and shear semblance in the slowness time coherence (STC) processing of the LWD data. This interference in turn is mostly dominated by Stoneley type propagative component, which, unlike the main Stoneley mode, is time persistent over the entire acquisition interval. In addition, we also show that in fast formations the shear wave can significantly interfere with the compressional wave making the compressional slowness estimates quite bad. Second, based on these observations we propose a Successive Interference Cancellation (SIC) algorithm to estimate and cancel these interferences leading to STC enhancement and improved slowness estimation of the head waves. The algorithm exploits a novel representation of borehole acoustic signals using a dictionary of space-time propagators and is more robust compared to traditional slowness filtering methods, especially for the low aperture borehole acoustic array. We show the superior performance of the proposed algorithm on synthetic and real data sets.

10:00

3aPA4. Evaluation on fracturing effects in a low-permeability reservoir using acoustic logging data. Huang m. Baohua, Chen Hao, Han Jianqiang, and He Xiao (Ultrosonic Phys. and Exploration Lab, Inst. of Acoust., Chinese Acad. of Sci., No. 21, 4th Norwestern Ring RD, Haidian District, Beijing, Beijing 100190, China, huangbh@mail.ioa.ac.cn)

Low-permeability reservoirs are frequently discovered in worldwide petroleum exploration. More than 50% of oil and gas reservoirs are of low permeability. Formation fracturing technique is the most common way to develop oil production in this type of reservoirs. The fracturing effect, however, is hard to be evaluated in practice. And thus arguments always exist between constructors and geologists. We developed a favorable method to evaluate the effect from the reservoir anisotropy analysis results provided by cross dipole logging technique. The data will be measured in an open hole or a borehole when formation is before and after fractured, respectively. The formation anisotropy can be estimated from the logging data. The fracturing effects can thus be evaluated by comparing the results of perforation intervals. Small differences of anisotropy estimation results indicate failure fracturing; while good fracturing effect can be confirmed if the anisotropy of a fracturing reservoir is stronger than before. Fracturing intervals can also be predicted by the anisotropy curves of a fracturing reservoir as well as the new oil production. This approach has been applied for the evaluation of deep tight reservoirs in Daqing Oilfield and low-permeability reservoirs in Hailar. Efficient evaluation results have been obtained, which provided useful information to geologists for further explorations.

Contributed Papers

10:20

3aPA5. Wave propagation in a fluid-filled shell excited by a dipole source. Xiumei Zhang, Xiuming Wang, Hailan Zhang, and Dehua Chen (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 Northwest Ring Rd., Haidian District, Beijing 100190, China, zhangxiumei@mail.ioa.ac.cn)

Wave propagation in a fluid filled cylindrical shell excited by a dipole source is investigated for the design of a new kind of calibration pit for dipole acoustic logging tools. Based on classical elasticity wave equations, phase and group velocity dispersion curves of each mode, excitation spectra and mode contributions to wave field in the shell are presented. The dispersion curves show that the lowest mode exists in the entire frequency range with the phase velocity smaller than shear wave velocity of the shell, higher-order modes have cutoff frequencies, below the cutoff frequencies these modes are non-propagating. Analysis on mode excitation spectra and their contributions suggest that the lowest mode has potential to be used in calibrating the velocity measurement accuracy of dipole acoustic logging tools quantitatively, provided that the contributions of this mode to the wave field dominating the first arrivals gathered by a specific dipole acoustic logging tool. [Work supported by National Natural Sciences of China, Grant No. 11134011.]

10:40

3aPA6. The simulation of the seismoelectric logging while drilling based on elastic model. Xiaobo Zheng (Dept. of Astronautics and Mech., Harbin Inst. of Technol., P.O. Box 344 92# West Dazhi St., Harbin 0086150001, China, zxb3710@163.com), Xien Liu (China Oilfield Services Ltd., Sanhe, China), Hengshan Hu, and Wei Guan (Dept. of Astronautics and Mech., Harbin Inst. of Technol., Harbin, China)

In recent years, the acoustic LWD technology which is drilling and logging at the same time has been developing rapidly. However, the collar wave could cover or interfere with signals from the formation to affect the extraction of P and S wave velocities in the sonic logging. In order to solve this problem, this paper makes a research about the seismoelectric LWD response. In this study, we use the decoupling algorithm to calculate seismoelectric field in the borehole. We can obtain the elastic sound field by solving the wave equation first, and then calculate the electromagnetic field excited by the sound field by using the Pride control equations. Although using the elastic model in the calculation, we simulate the attenuation effect of pore formation by introducing quality factor and gain the pressure of the fluid within the pore by introducing the Skempton factor. Finally, this paper shows the full waveforms of the electromagnetic and acoustic fields excited by multipole source. We find that the collar wave in the electric field is significantly weakened compared with that in the acoustic pressure, in terms of its amplitude relative to the other wave groups in the full waveforms.

11:00

3aPA7. Simulation study on seismic monitoring of aquifers. Timo Lähivaara (Appl. Phys., Univ. of Eastern Finland, Kuopio, Finland), Jari P. Kaipio (Mathematics, Univ. of Auckland, Auckland, New Zealand), Nicholas F. Dudley Ward (Otago Computational Modelling Group Ltd., Kurow, New Zealand), and Tomi Huttunen (Appl. Phys., Univ. of Eastern Finland, P.O. Box 1188, Kuopio, Finland, tomi.huttunen@uef.fi)

This study focuses on developing computational tools to estimate groundwater volume from seismic measurements. The poroelastic signature from an aquifer is simulated and methods to use this signature to estimate porous properties of the permeable rock and the level of the water table are investigated. In this work, the spectral-element method (SEM) is used for solving the forward model that characterizes propagation of seismic waves. The SEM combines the accuracy of the global pseudospectral method with the flexibility of the classical finite element method. The SPECFEM-2D software is used for calculating seismic wave fields (forward + adjoint) in elastic and poroelastic media. The inverse problem is solved in the Bayesian framework, which makes efficient use of *a priori* information related to modeling and measurement uncertainties of the problem. It this study, preliminary results in the two-dimensional case with simulated data are presented.

11:20

3aPA8. Broad-band acoustic low frequency collimated beam for ultrasonic imaging. Cristian Pantea and Dipen N. Sinha (Mater. Phys. and Applications, MPA-11, Los Alamos National Lab., MS D429, Los Alamos, NM 87545, pantea@lanl.gov)

Ultrasonic and sonic imaging from a borehole is a widely used technique for hydrocarbon reservoir characterization. A typical acoustic transducer in a borehole can produce a narrow beam only at ultrasonic frequency hundreds of kHz or higher. On the other hand, high acoustic frequencies are rapidly attenuated in the formation and result in shallow penetration. To allow deeper penetration, low frequency operation is needed. We report on the development of a lower frequency broad band collimated and steerable acoustic beam source, which is sufficiently compact to fit inside a borehole and is capable of probing for rock information in the near-borehole environment. Some of the main advantages of the source presented in this study are related to: (1) the beam collimation for better spatial resolution, (2) beam steerability (360 degree in azimuth and inclination), (3) the broad-bandwidth (20–120 kHz) and (4) specific pulse shape for simpler signal processing and data analysis. Laboratory experimental data simulating open and cased wellbores will be presented.

11:40–12:00 Panel Discussion

Session 3aPP

Psychological and Physiological Acoustics: Auditory Physiology and Modeling (Poster Session)

Magdalena Wojtczak, Chair
Psychology, Univ. of Minnesota, 1237 Imperial Ln, New Brighton, MN 55112

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

3aPP1. The medial olivary complex reflex strength of children with auditory processing disorders. Sangeeta Kamdar and Su-Hyun Jin (Commun. Sci. and Disord., Univ. of Texas at Austin, 1 University Station, A1100, Austin, TX 78712, kamdarsc@gmail.com)

The present investigation is designed to examine speech understanding in noise and the strength of medial olivary complex (MOC) reflex in children diagnosed with auditory processing disorder (APD). APD is a dysfunction associated with limited auditory processing of sounds. Individuals with APD do not show any peripheral hearing impairment but have difficulty understanding speech especially in the presence of noise. Recent neurophysiological studies suggest that the efferent system also make an important contribution. One of the most peripheral parts of the auditory efferent system is MOC system which projects from the auditory brainstem to the cochlea. The MOC system has been known to play an important role protecting the auditory system from intense noise and affecting tone detection or speech perception in noise [Micheyl and Collet (1996), Kumar and Vanaja (2004)]. The strength of this efferent feedback system can be assessed non-invasively through the contralateral suppression of distortion product otoacoustic emissions. It is hypothesized that children with APD show weaker MOC reflex strength compared to the normal peers. Based on the test results, possible efficient testing protocol and intervention program for this special population will be discussed.

3aPP2. Evaluating the role of efferent inhibition on cochlear responses: Simultaneous psychophysical and otoacoustic emission measurements. Simon Henin and Glenis Long (Ph.D. Program in Speech-Language-Hearing Sci., The Grad. Ctr., The City Univ. of New York, 365 Fifth Ave., New York, NY 10016, shenin@gc.cuny.edu)

The auditory system continuously adapts to changes in the acoustic environment. Behavioral experiments in humans have demonstrated that changes in the acoustic environment produce dynamic changes in perception, for example, increases in thresholds in the presence of background noise. This dynamic change in the auditory system is hypothesized to be mediated by efferent feedback from the olivocochlear system. The effect of efferent inhibition on cochlear mechanics was investigated using a simultaneous psychoacoustics and otoacoustic emissions (OAEs) task using identical stimulus conditions. Cochlear responses to short tone-burst stimuli were analyzed under various masking conditions. Robust modification of cochlear responses to the short tone-burst stimuli was observed during contralateral acoustic stimulation and during long-duration ipsilateral masking. Concomitant changes in perceptual thresholds and OAEs are consistent with the hypothesis that both stem from efferent activation. This novel paradigm provides simultaneous perceptual and physiological estimates of cochlear-based efferent activation in the same human subjects.

3aPP3. Modeling psychophysical gain reduction effects as a function of precursor duration. Elin Roverud and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., 2501 Soldiers Home Rd., apt 16G, West Lafayette, IN 47906, eroverud@purdue.edu)

It is known that a forward masker can make threshold for a signal poorer, but the mechanisms underlying this psychophysical effect are not well-understood. One theory, the temporal window model (TWM), proposes that masker and signal excitation are integrated within a temporal window. An additional mechanism may be cochlear gain reduction by the medial olivocochlear reflex, a sluggish sound-evoked reflex. In our laboratory, we have shown evidence of gain reduction in forward masking results. We measure off-frequency growth of masking to estimate the cochlear input/output (I/O) function. A precursor is introduced which reduces the gain of the I/O function. In the present study, we examine this gain reduction effect as a function of precursor duration for on- and off-frequency precursors. From a gain reduction perspective, a long on-frequency precursor may reduce gain for itself, while the off-frequency precursor, assumed to be linear at the signal frequency, would not. The TWM, however, does not predict differences in trends with duration for the two precursor frequencies. In our modeling, we have incorporated a gain reduction module into the TWM. We will compare predictions of the results with the standard TWM and the TWM with gain reduction. [Research supported by NIH(NIDCD)R01 DC008327.]

3aPP4. The effect of the medial olivocochlear reflex on click-evoked otoacoustic emissions during psychoacoustic forward-masking tasks. Jordan A. Beim, Magdalena Wojtczak, and Andrew J. Oxenham (Psychology, Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, beimx004@umn.edu)

Measurements of otoacoustic emissions in animals have shown that the effects of efferent activation are greater in attentive than in anesthetized animals suggesting that the medial olivocochlear reflex (MOCR) effects can be modulated by attention. In this study, the effect of efferent activation was measured in humans using click-evoked otoacoustic emissions while listeners were performing a psychoacoustic forward-masking task. Each trial within a block started with a sequence of 50-dB pSPL clicks presented at a rate of 40 Hz that were followed by a 200-ms harmonic-complex masker. The masker was immediately followed by a 10-ms tonal probe and another click train. The listeners' task was to detect the probe. A constant stimuli method was used to measure performance in the forward-masking task, with the probe presented at seven randomized levels around the predetermined masked threshold. Catch trials were dispersed randomly throughout the block. Click trains before and after the masker-signal segments were recorded from the ear canal and analyzed to extract effects of efferent activation at different levels of difficulty of the psychoacoustic task. The results will be discussed with respect to the role of attention and the role of the MOCR in forward masking. [Work supported by NIH grant R01DC010374.]

3aPP5. Human otoacoustic emissions generated by active outer hair cells. Reinhart Frosch (ETH and PSI (retired), Sommerhaldenstrasse 5B, Brugg 5200, Switzerland, reinifrosch@bluewin.ch)

In the present study, human otoacoustic emissions (OAEs) documented in the literature are shown to agree with predictions based on the hypothesis that the main OAE sources are active cochlear outer hair cells (OHCs) of three different functional categories, namely (1) OHCs enabled by oscillating internal organ-of-Corti resonators (IOCRs) to feed mechanical energy into forward-traveling cochlear waves generated by the acoustic stimuli used, (2) OHCs driving spontaneous localized feedback-generated cochlear-partition vibrations involving standing evanescent sound-pressure waves in the liquids above and below the partition, and (3) OHCs causing nonlinear restoring forces enabling pairs of stationary tones to generate distortion products (DPs). The corresponding predictions of OAE properties are based on cochlear maps, i.e., on certain functions x(f), where f is the frequency of a tone and x is a related distance from the cochlear base, measured along the cochlear channel.

3aPP6. Total and component distortion product otoacoustic emission analysis in persons with induced negative middle ear pressure. Suzanne Thompson, Glenis Long, and Simon Henin (Ph.D. Program in Speech-Language-Hearing Sci., The Grad. Ctr. of the City Univ. of New York, 40 Hathaway Dr., Garden City, NY 11530, sthompson1@gc.cuny.edu)

Distortion product otoacoustic emissions (DPOAEs) are generated when two primary tones (f_1, f_2 with $f_2 > f_1$) are presented simultaneously to the ear. Inter-modulation between primary tones produces distortion products at predictable frequencies not present in the original signal (e.g., $2f_1 - f_2$). In persons with negative middle ear pressure (NMEP), the tympanic membrane is retracted and pulled inward, compressing structures in the middle ear. NMEP is expected to modify DPOAE level and phase. Performing the Toynbee maneuver can artificially induce NMEP. Changes in DPOAE primary f_1 and f_2 level and phase and energy reflectance measures were used to confirm that 8 subjects had artificially induced NMEP using the Toynbee maneuver. DPOAEs were obtained using 1 s/octave duration logarithmic frequency sweeping primaries, $f_2/f_1 = 1.22$ producing $2f_1 - f_2$ from 320–2560 Hz, L1 = L2 = 65, 70, 75 dB SPL. There was a significant effect of condition, primary level, and frequency on total DPOAE and component amplitude. Separation of DPOAE components allowed improved detection of NMEP effects on DPOAE amplitude.

3aPP7. Effect of percussion impulse sounds on hearing. David Pazen, Dirk Beutner, and Martin Walger (Dept. of Otorhinolaryngol., Head and Neck Surgery, Univ. Hospital of Cologne, Kerpener Str. 62, Cologne 50937, Germany, david.pazen@uk-koeln.de)

Impulse-like sounds with high peak pressure levels emitted by percussion instruments often occur in music. The potentially harmful effect to the inner ear caused by these sounds is not fully understood. Solely the physical description of impulses like peak pressure level, etc. is not considered meaningful enough to assess their hazard. Therefore, the Auditory Hazard Assessment Algorithm for Humans (AHAAH), which is based on a physiological model, seems to be more appropriate. Another quantity to assess auditory hazard is the change of otoacoustic emissions (OAEs) after a sound exposure. It allows the individual detection of small physiological changes of the outer hair cells before any serious damages and threshold shifts may occur. In a pilot experiment the sounds of several percussion instruments have been measured at the players' ears in a regular rehearsal situation. To assess the auditory hazard of the measured sounds the players' OAEs have been measured before and after playing on the instruments. Despite peak levels of about 140 dB, the OAEs did not change significantly and the AHAAH rated nearly all sounds as harmless. It can be concluded that impulse sounds in music with high peak levels are not necessarily hazardous.

3aPP8. Efficient estimates of cochlear hearing loss parameters in individual listeners. Michal Fereczkowski (Elektro, DTU, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, mfer@elektro.dtu.dk), Morten L. Jepsen (R&D, Widex A/S, Lynge, Denmark), and Torsten Dau (Elektro, DTU, Kgs Lyngby, Denmark)

It has been suggested that the level corresponding to the knee-point of the basilar membrane (BM) input/output (I/O) function can be used to estimate the amount of inner- and outer hair-cell loss in listeners with a moderate cochlear hearing impairment [e.g., Plack *et al.*, (2004)]. In the present study, results from forward masking experiments based on temporal masking curves [TMC; Nelson *et al.* (2001)] are presented and used to estimate the knee-point level and the compression ratio of the I/O function. A time-efficient paradigm based on the single-interval-up-down method [SIUD; Lecluyse and Meddis (2009)] was used instead of an alternative forced-choice paradigm. In contrast with previous studies, the present study used only on-frequency TMCs to derive estimates of the knee-point level. Further, it is explored whether it is possible to estimate the compression ratio using only on-frequency TMCs. Ten normal-hearing and 10 hearing-impaired listeners (with mild-to-moderate sensorineural hearing loss) were tested at 1, 2, and 4 kHz. The results showed a reasonable reliability and an increased time-efficiency compared to AFC-based results and may be suitable for individualized hearing-aid fitting.

3aPP9. Relationship of distortion product otoacoustic emission components to psychoacoustic measures of noise induced hearing loss. Gavin Coad (Section of Audiol., The Univ. of Auckland, Morrin Rd., Glen Innes, Auckland 2012, New Zealand, g.coad@auckland.ac.nz), Glenis R. Long (Speech-Lang.-Hearing Program, CUNY, New York, NY), David Welch, and Peter R. Thorne (Section of Audiol., The Univ. of Auckland, Auckland, New Zealand)

Distortion product otoacoustic emissions (DPOAEs) have promise as a tool to detect and monitor cochlear hair cell damage in noise induced hearing loss (NIHL). However, variability in OAE amplitudes and thresholds across individuals limits their potential for detecting the early impact of NIHL. The DPOAE has two components, one generated at the point of interaction between the primaries (generator), the second reflected from the characteristic place of the DPOAE frequency (reflected). There has been increasing interest in looking at the components of the DPOAE and how they correlate with injury. A total of 109 men, including 57 with varying degrees of occupational noise exposure, were tested. We recorded DPOAEs using logarithmically swept pure tones and extracted the components of the DPOAE using a least-squares fit approach. Pure-tone audiometry was conducted at 1.5 and 4 kHz with 1dB resolution. Compressive nonlinearity was evaluated psychoacoustically with 1.5 and 4 kHz pure-tones using Schroeder phase maskers. Overall there are associations between the generator component and the psychoacoustic measures: stronger DPOAE amplitudes were associated with better audiometric thresholds and greater difference in Schroeder masking threshold. Associations were stronger at 4 than 1.5 kHz. Similar, but weaker effects were observed for the reflected component. Findings suggest that separation of DPOAE components enhances the assessment of cochlear noise-induced injury.

3aPP10. Correlations between noninvasive and direct physiological metrics of auditory function in chinchillas with noise-induced hearing loss. Kenneth S. Henry (Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, kshenry@purdue.edu), Sandra F. Snyder (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN), and Michael G. Heinz (Speech, Lang., and Hearing Sci., Purdue Univ., West Lafayette, IN)

Noninvasive physiological tools for assessing auditory function in humans can provide valuable information when behavioral tests are not possible. Furthermore, these tools hold promise to provide greater insight into underlying cochlear pathologies. In this study, we used noninvasive distortion product otoacoustic emissions (DPOAEs) and auditory brainstem responses (ABRs) to estimate changes in auditory function in chinchillas with noise-induced hearing loss. Two aspects of cochlear function, sensitivity (threshold) and frequency selectivity (tuning), were measured directly using neurophysiological recordings from auditory-nerve (AN) fibers to assess the predictive value of DPOAEs and ABRs. Both DPOAE amplitude and ABR threshold were well correlated (R-square ~0.5) with AN fiber threshold near stimulus frequency. For DPOAEs, the correlation was strongest for cochlear function near F2. Correlations of both noninvasive metrics with AN tuning were weaker but statistically significant. The relatively weak correlation between DPOAE amplitude and AN tuning was unexpected because both measures are tied to the integrity of outer hair cells. Alternatively, previous results suggest ABR latency may be more predictive of AN tuning. Ultimately, DPOAEs may be a more practical clinical and research screening tool for hearing loss than ABRs due to shorter recording time. [Research supported by NIH (NIDCD) F32-DC012236 and R01-DC009838.]

3aPP11. Effects of inner hair cell damage on temporal coding. David R. Axe and Michael G. Heinz (Biomedical Eng., Purdue Univ., 320 Brown St., Apt. 713, West Lafayette, IN 47906, davidraxe@gmail.com)

It is widely believed that the neural patterns of temporal coding within the auditory periphery and CNS change following cochlear hearing loss, and a number of recent studies have aimed to more fully understand and characterize these changes. In these studies, noise exposure has been a common method for inducing hearing loss in animal models. Unfortunately, its effects are nonspecific, affecting both inner and outer hair cells as well as the surrounding tissues. Because of this mixed hair-cell damage, it has been difficult to tease apart the specific effects that each of these pathologies has on temporal coding. In the present study, we used the chemotherapy drug carboplatin to induce inner hair cell (IHC) specific lesions in the cochleae of chinchillas. Using acoustically evoked potentials and acute single fiber recordings from the auditory nerve, in parallel with computational modeling, we have investigated the effects of IHC damage on temporal coding. Preliminary findings in carboplatin exposed chinchillas, which show near-normal ABR thresholds, showed a decrease in ABR amplitudes at high sound levels, as well as a decrease in the strength of both envelope and fine-structure coding in frequency following responses. [Research supported by NIH (NIDCD) R01-DC009838 and T32-DC00030.]

3aPP12. Prolonged low-grade noise exposure induces aging-like functional and structural changes in cortical auditory pathways. Brishna Kamal, Lydia Ouellet, and Etienne de Villers-Sidani (Neurology, McGill Univ., 3801 University St., Rm. 736, Montreal, QC, Canada, brishnak@gmail.com)

Age-related impairments in the primary auditory cortex (A1) include poor tuning selectivity, neural desynchronization, and degraded responses to low-probability or "oddball" sounds. These changes have been for the most part attributed to reduced inhibition in the aged brain. Since many of these changes can be partially reversed with auditory training, it has been speculated that they might not be purely degenerative but might rather represent negative plastic adjustments to noisy or distorted auditory inputs. To test this hypothesis, we exposed young adult rats to low-grade broadband noise for 6 weeks and then compared the effect of this exposure on several aspects of A1 function and structure. We found that the impact of noise exposure on A1 tuning selectivity and responses to oddball tones was almost undistinguishable from the effect of natural aging. These changes were paralleled by alterations in A1 inhibitory interneuron populations in the exposed group. Moreover we found that noise exposure reduced the anatomical and functional connectivity of A1 to downstream cortical fields. Most of these changes reversed after returning to a non-noisy environment. These results support the hypothesis that age-related changes in cortical auditory pathways might have a strong activity-dependent component, making them potentially preventable and reversible.

3aPP13. How broadband speech may avoid neural firing rate saturation at high intensities and maintain intelligibility. Richard Warren, James Bashford, and Peter Lenz (Psychology, Univ. of Wisconsin-Milwaukee, P.O. Box 413, Milwaukee, WI 53201, rmwarren@uwm.edu)

While broadband speech may remain perfectly intelligible at levels exceeding 90 dB, narrowband speech intelligibility (e.g., 2/3-octave passband centered at 1.5 kHz) may decline by 25% or more at moderate intensities (e.g., 75 dB). This "rollover" effect is substantially reduced, however, when a speech band is accompanied by flanking bands of white noise [Bashford *et al*., J. Acoust. Soc. Am. **117**, 365–369 (2005)], suggesting that lateral suppression helps preserve broadband speech intelligibility at high levels. The present study found that when noise flankers were presented individually at a low spectrum level (−30 dB relative to the speech) only the higher-frequency flanker produced a significant intelligibility increase. However, the lower-frequency flanking noise did produce an equivalent increase when its spectrum level was raised 10 dB. This asymmetrical intensity requirement for noise flankers links the effective dynamic range of speech intelligibility to reported characteristics of both lateral (two-tone) suppression of auditory nerve (AN) fiber activity and lateral inhibition of secondary cells of the cochlear nucleus. These and other observations will be discussed in the broader context of how various auditory mechanisms help preserve speech intelligibility at high intensities by reducing firing rate saturation. [Work supported by NIH.]

3aPP14. The computational prediction of masking thresholds for ecologically valid interference scenarios. Khan Baykaner, Christopher Hummersone, Russell Mason (Inst. of Sound Recording, Univ. of Surrey, 26 Shepherds Hill, Guildford GU2 9RY, United Kingdom, ee51kb@surrey.ac.uk), and Søren Bech (Bang & Olufsen, Struer, Denmark)

Auditory interference scenarios, where a listener wishes to attend to some target audio while being presented with interfering audio, are prevalent in daily life. The goal of developing an accurate computational model which can predict masking thresholds for such scenarios is still incomplete. While some sophisticated, physiologically inspired, masking prediction models exist, they are rarely tested with ecologically valid programs (such as music and speech). In order to test the accuracy of model predictions human listener data is required. To that end a masking threshold experiment was conducted for a variety of target and interferer programs. The results were analyzed alongside predictions made by the computational auditory signal processing and prediction model described by Jepsen *et al*. (2008). Masking thresholds were predicted to within 3 dB root mean squared error with the greatest prediction inaccuracies occurring in the presence of speech. These results are comparable to those of the model by Glasberg and Moore (2005) for predicting the audibility of time-varying sounds in the presence of background sounds, which otherwise represent the most accurate predictions of this type in the literature.

3aPP15. A computational model of spatial tuning in the auditory cortex in response to competing sound sources. Junzi Dong, Steven Colburn, and Kamal Sen (Boston Univ., 18 Medfield St., Apt. 1, Boston, MA 02215, junzid@bu.edu)

Single neurons in the auditory midbrain are sharply tuned to preferred directions, while cortical neurons show broader tuning [King *et al*., Hearing Research (2007)] Recent experiments on cortical responses in birds revealed the emergence of spatial tuning to multiple competing sounds in the cortex [Maddox *et al*., PLoS Biol. (2012)]. In this situation, cortical neurons show broad tuning to single-location stimuli, but develop sharper spatial preference in response to a second competing noise source. We have developed a computational model to match these physiological data. The model takes binaural inputs containing sounds from two locations, and outputs cortical spike trains that emphasize one pre-defined location. In the model, sharply tuned midbrain neurons synapse onto their corresponding interneurons, which then innervate cortical neurons generating the final output. In the presence of stimulus from a pre-defined preferred location, corresponding interneurons actively suppress sources from a pre-defined non-preferred location. The model achieves spatial tuning by performing robustly when the target and masker locations match the pre-defined preferred and non-preferred directions. Looking across cortical neurons tuned to different target and masker combinations, the model provides a mechanism by which binaural inputs from a noisy environment can be separated into independent "streams" based on locations.

3aPP16. Combining the outputs of functional models of organs responsible for binaural cue decoding. Marko O. Takanen (Dept. of Signal Processing and Acoust., Aalto Univ. School of Elec. Eng., Otakaari 5A, Espoo 02015, Finland, marko.takanen@aalto.fi), Olli Santala (Dept. of Signal Processing and Acoust., Aalto Univ. School of Elec. Eng., Helsinki, Finland), and Ville Pulkki (Dept. of Signal Processing and Acoust., Aalto Univ. School of Elec. Eng., Espoo, Finland)

The binaural cue decoding in the human auditory pathway occurs in the medial superior olive (MSO) and the lateral superior olive (LSO). The MSO is sensitive to interaural time difference (ITD), whereas the LSO is sensitive to interaural level difference as well as to ITD at low frequencies. Functional models of such organs have been presented previously [Pulkki and Hirvonen, Acta Acoustica united with Acoustica **95**, 883–900 (2009)]. However, the outputs of those models are not as such applicable as the level of the output varies for a point-like broadband stimulus depending on the frequency and between the models. Here, a method is presented to combine the outputs of such models and an additional model of the MSO designed to decode directional cues based on broadband envelope time-shifts between the ear canal signals. Applicable cues are obtained by mapping the outputs into azimuth direction values following the idea of self-calibration, and by favoring the cue values suggesting more lateral directions. Furthermore, it is shown how the resulting directional cues can be applied to form a binaural activity map, and that the activity map corresponds to the human perception in several scenarios of psychoacoustical experiments.

3aPP17. Interaction between resonant scaling and fundamental frequency on codings in auditory system. Toshie Matsui (Dept. of Otorhinolaryngol. - Head and Neck Surgery, Nara Med. Univ., Shijo-cho 840, Kashihara City 634-8522, Japan, tomatsui@naramed-u.ac.jp) and Minoru Tsuzaki (Faculty of Music, Kyoto City Univ. of Arts, Kyoto city, Japan)

It has been modeled that the auditory system encodes acoustic signals into two independent informations. One is tonotopic information reflecting the frequency response characteristics of the basilar membrane, and another is periodicity information reflecting the temporal patterns of phase-locked auditory nerve firing. Based on the previous study about size perception of

sound source, a hypothesis can be proposed that the tonotopic and periodicity information are combined into an internal "two-dimensional representational plane." The current study tested that this hypothesis by conducting an experiment to recognize the transition pattern of vowel-like sounds on the plane. Listeners were able to recognize the transition patterns on the axes of tonotopic and periodicity information exclusively. However, they were unable to follow the trajectories of sounds in the two-dimensional representational plane. The vowel-like sounds were simulated by a computational model of auditory system, AIM. While the "excitation pattern" displayed transition patterns corresponding to the manipulation of resonant scaling (RS) only, "Summary SAI" displayed both of patterns of fundamental frequency (F0) and RS. It suggests the need to confirm whether RS represented in periodicity information is used for size perception.

3aPP18. A simulation of neural coding and auditory frequency analysis. Robert A. Houde (Ctr. for Commun. Res., 35 Rensselaer Dr., Rochester, NY 14618, rahoude@gmail.com), James M. Hillenbrand (Speech Pathol. and Audiol., Western Michigan Univ., Kalamazoo, MI), Robert T. Gayvert (Ctr. for Commun. Res., Rochester, NY), and John F. Houde (Dept. of Otolaryngol., Univ. of California at San Francisco, San Francisco, CA)

Our understanding of the neural mechanisms underlying the very fine auditory frequency discrimination exhibited by listeners remains far from complete. To investigate this question we developed a functional model of the cochlear process in sufficient detail to allow the simulation of the principal characteristics of the cochlea's response to multi-tone and noise stimuli over a wide range of input levels. The model simulates level-dependent changes in frequency selectivity, combination-tone distortion, tone-on-tone suppression and masking, adaptation, and critical-band masking. The model is structured as 3000 channels, each consisting of a basilar membrane bandpass filter and inner-hair cell assembly. Input to each channel is the stapes displacement signal, and the output consists of ten independent stochastic point processes that are transmitted to the CNS on auditory-nerve fibers (ANFs). Our main purpose is to address these questions: (1) What narrowband spectrum information is available in the cochlea output? (2) How is this information encoded on the ANFs? (3) How might it be decoded in the CNS? An analysis of ensemble coding of the cochlear output showed that the precision (signal-to-noise ratio) of the information decoded by the CNS frequency analysis is directly related to the bandwidth of the basilar membrane filters.

3aPP19. The effect of compression on tuning estimates in a simple nonlinear auditory filter model. Márton Marschall, Ewen MacDonald, and Torsten Dau (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, mm@elektro.dtu.dk)

Behavioral experiments using auditory masking have been used to characterize frequency selectivity, one of the basic properties of the auditory system. However, due to the nonlinear response of the basilar membrane, the interpretation of these experiments may not be straightforward. Specifically, there is evidence that human frequency-selectivity estimates depend on whether an iso-input or an iso-response measurement paradigm is used [Eustaquio-Martin *et al.* (2011)]. This study presents simulated tuning estimates using a simple compressive auditory filter model, the bandpass nonlinearity (BPNL), which consists of a compressor between two bandpass filters. The BPNL forms the basis of the dual-resonance nonlinear (DRNL) filter that has been used in a number of modeling studies. The location of the nonlinear element and its effect on the estimated tuning in the two measurement paradigms was investigated. The results show that compression leads to (i) a narrower tuning estimate in the iso-response paradigm when a compressor precedes a filter, and (ii) a wider tuning estimate in the iso-input paradigm when a compressor follows a filter. The results imply that if the DRNL presents a valid model of the basilar membrane, then compression alone may explain a large part of the behaviorally observed differences in tuning between simultaneous and forward-masking conditions.

3aPP20. Frequency-domain analysis of cochlear gain reduction due to disruptions in the outer hair cell feedback loop. Yi-Wen Liu, Kuang-Yi Lin, and Yong-Zing Chen (Elec. Eng., National Tsing Hua Univ., 101 Kuang-Fu Rd. Sec 2, Delta Bldg. Rm. 828, Hsinchu 30013, Taiwan, ywliu@ee.nthu.edu.tw)

A frequency domain equivalent model was implemented to match the small-signal responses produced by a time-domain cochlear mechanics model [Liu and Neely (2010)]. In the present model, the outer hair cell feedback is characterized by physical parameters, including a hair-bundle transduction ratio (HBTR) and a prestin-associated capacitance (PAC). When either of them is reduced, the model predicts lowered magnitude responses along the cochlea; however, HBTR and PAC seem to play different roles in facilitating cochlear amplification. It appears that the gain drops more drastically with respect to reduction in the HBTR, whereas the degradation is more graceful with respect to lack of prestin. Simulation also suggests that HBTR is more crucial for high-frequency hearing whereas PAC boosts up low frequency responses. Finally, we present attempts on modifying the frequency-domain model iteratively to simulate nonlinear responses. The saturating cochlear responses to high-intensity tones can be predicted by the frequency domain model as long as HBTRs are adjusted appropriately. Currently, high precision is achieved (as compared against time-domain simulation) by usage of Fourier series analysis in determining the HBTR adjustment factor. This numerical approach has a potential to accelerate simulation by orders of magnitude. Its physical meaning will also be discussed.

3aPP21. A physiology-based auditory model elucidating the function of the cochlear amplifier and related phenomena. Part I: Model structure and computational method. Herbert Hudde and Sebastian Becker (Inst. of Commun. Acoust., Ruhr-Univ. Bochum, Bochum D-44780, Germany, herbert.hudde@rub.de)

An auditory model "PhyBAM" is presented, which in the long term aims at reproducing the human auditory perception. In recent years, the awareness has grown that many perceptive features have their origin in the peripheral ear, above all in the cochlea. In the present stage, PhyBAM is actually just a model of the peripheral ear. To simulate the perception of arbitrary sound signals, the signal processing occurring in the cochlea has to be formulated close to the physiological basis. Even so the model must be kept as simple as possible for the given aim. As a compromise, PhyBAM is set up as an ordinary circuit model. In this first of two associated papers, the model structure and the computational methods are presented. The model covers ear canal, middle ear, and cochlea. The cochlea model is by far the most sophisticated part. To include the unsymmetrical conditions at both cochlear windows and the resulting common and differential modes, a two-canal circuit is used. The main challenge is the implementation of the cochlear amplifier on the basis of measured tuning curves and otoacoustic emissions. Finding an appropriate model structure and proper parameters turns PhyBAM into an instrument of cochlear research.

3aPP22. A physiology-based auditory model elucidating the function of the cochlear amplifier and related phenomena. Part II: Model parameters and simulations. Sebastian Becker and Herbert Hudde (Inst. of Commun. Acoust., Ruhr Univ. Bochum, Bldg. ID 2/261 Ruhr-Universität Bochum, Bochum 44780, Germany, sebastian.becker-2@rub.de)

In this second of two associated papers, the properties of the physiology-based auditory model are investigated. This includes finding of appropriate parameters and simulating various responses. In the end the model is intended to reproduce the human ear, hence human data is used for fitting. Only the trend of active tuning curves is based on chinchilla measurements, as human data is not available. To achieve such tuning curves the cochlea amplifier feeds energy into the system basal to the characteristic place, resulting in a locally restricted negative real part of the basilar membrane impedance. Realistic level dependent tuning curves show a reasonable input-output function and a maximum cochlear gain of 55 dB. The growth of distortion product otoacoustic emissions is consistent with measurements and shows a slope of 0.5 dB/dB. The physiology-based model approach shows the origin of the distortion products within the overlap region of the stimuli and elucidates the propagation within cochlea. As reflections are a dominant factor in the generation of transient evoked otoacoustic emissions, parameters need a certain degree of roughness to achieve results corresponding to measurements. In spite of its simplicity the model is able to reproduce a variety of cochlea results with one parameter set.

3aPP23. The coda of the transient response in a sensitive cochlea. Yizeng Li and Karl Grosh (Mech. Eng., Univ. of Michigan - Ann Arbor, 2350 Hayward Ave., Ann Arbor, MI 48109, yizengli@umich.edu)

In a sensitive cochlea, the basilar membrane (BM) velocity response due to transient external acoustic excitation or to localized transient internal bipolar electrical excitation gives rise not only to a primary impulse response, but also to a coda of delayed secondary responses (sometimes called echoes or

ringing) with varying amplitudes but similar spectral content around the best frequency of the measurement location. The coda is physiologically vulnerable, disappearing when the cochlea is compromised even slightly. The multi-component sensitive response is not yet completely understood. We use a mathematical model to describe how the response at the point of excitation can be traced back to three sources. Surprisingly, the first BM response is due to a fast wave emergent from the point of excitation, reflected by the stapes and then repropagated (in amplified fashion) as a traveling wave back to the point of excitation; the second is due to a reverse, slow, traveling wave, which is likewise reflected at the stapes back to the measurement location by the stapes. The coda is also due to systematic (not random) perturbations of the organ of Corti properties. Implications for normal hearing and for the interpretation of otoacoustic emissions are discussed.

3aPP24. Wave finite element analysis of an active cochlear model. Guangjian Ni and Stephen J. Elliott (Inst. of Sound and Vib. Res., Univ. of Southampton, SPCG, ISVR, Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, S.J.Elliott@soton.ac.uk)

The wave finite element method has previously been used to understand the various types of wave in a passive cochlear model. In this paper, the model is extended to give an initial representation of an active cochlea by making the real and imaginary components of the Young's modulus, that defines the local dynamics of the plates representing the basilar membrane, position and frequency dependent. At a given excitation frequency, the distribution of the Young's modulus is chosen so that the mechanical impedance of the plate elements correspond to that obtained from a lumped parameter model of the active cochlea. The types of wave predicted in this representation of the active cochlea are similar to those observed for the passive cochlea model and consist of a fast wave, a slow wave and a large number of higher-order fluid modes, which are evanescent. Although the results of the full finite element analysis for this active model are very different from the passive one, the decomposition into wave components still shows that the slow wave dominates the response along most of the cochlear length, until the response peaks, when a number of higher-order fluid modes are locally excited.

3aPP25. Can cochlear mechanics contribute to amplitude modulation perception? Jungmee Lee and Sumitrajit Dhar (Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, jmlee6@msn.com)

Amplitude modulation (AM) detection has been successfully used as a psychophysical measure of auditory temporal processing. Our understanding of the role of the auditory periphery in processing AM signals is emerging through physiological and psychophysical studies. Unfortunately, direct physiological estimates of the cochlea's mechanical response to AM signals are not obtainable in humans. This study tries to fill this critical gap in knowledge by exploring the relationship between perception (through psychophysical AM detection) and mechanics (through otoacoustic emissions). Psychometric function for AM perception was measured for a 2-kHz carrier frequency and 10-Hz modulation frequency (f_m). Distortion product otoacoustic emissions (DPOAEs) were recorded with amplitude-modulated f_1 with $f_m = 10$ Hz and steady-state f_2. The frequencies of f_1 and f_2 were chosen to yield a $2f_1 - f_2$ DPOAE around 2 kHz near a peak in the fine structure. The ratio between the DPOAE pressure at $2f_1 - f_2$ and that of the sidebands separated by f_m (AMOAE depth) was calculated as a function of different modulation depths. Results indicate that there might be a correlation between AM perception performance and AMOAE magnitude, suggesting that cochlear mechanics might play a role for AM perception. [Work supported by the Knowles Hearing Center and Northwestern University.]

3aPP26. Effects of spontaneous otoacoustic emissions on frequency discrimination. Rói Hansen, Sébastien Santurette (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Ørsteds Plads, DTU Bygning 352, Kgs. Lyngby 2800, Denmark, ses@elektro.dtu.dk), and Sarah Verhulst (Auditory Neurosci. Lab., Boston Univ., Boston, MA)

When an external tone is presented in proximity to the frequency of a spontaneous otoacoustic emission (SOAE), the SOAE typically synchronizes to the external tone, a phenomenon known as "entrainment". As the tone moves further away from the SOAE frequency, beating patterns between the SOAE and the pure tone occur [Long, Hear. Res. **119** (1998)]. This study investigated perceptual consequences of SOAE beating and entrainment on the difference limen for frequency (DLF), which has been found to improve near an SOAE. SOAE entrainment patterns were obtained for six subjects with a strong SOAE in the ipsilateral ear and no SOAE in the corresponding frequency range of the contralateral ear. Hearing thresholds and DLFs were measured ipsi- and contralaterally for nine frequencies covering the entrainment and beating regions of the SOAE. DLFs systematically improved in the entrainment region, worsened when beating occurred, and improved again for frequencies further away from the SOAE. No improvement in DLF was found in any of the contralateral ears tested, suggesting that the effect is of peripheral, rather than of central, origin. The results contradict an earlier hypothesis stating that DLF performance near SOAE frequencies is governed by a central oversensitivity to the SOAE frequency.

3aPP27. Temporal integration near threshold fine structure–The role of cochlear processing. Bastian Epp (Ctr. for Hearing and Speech Sci., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Rm. 118, Lyngby 2800, Denmark, bepp@elektro.dtu.dk), Manfred Mauermann (Med. Phys., Univ. of Oldenburg, Oldenburg, Germany), and Jesko L. Verhey (Dept. of Experimental Audiol., Otto-von-Guericke Univ., Magdeburg, Germany)

The hearing thresholds of normal hearing listeners often show quasi-periodic variations when measured with a high frequency resolution. This hearing threshold fine structure is related to other frequency specific variations in the perception of sound such as loudness and amplitude modulated tones at low intensities. The detection threshold of a pulsed tone also depends not only on the pulse duration, but also on the position of its frequency within threshold fine structure. The present study investigates if psychoacoustical data on detection of a pulsed tone can be explained with a nonlinear and active transmission line cochlea model. The model was successfully applied to other psychoacoustical data at low intensities, various types of otoacoustic emissions and physiological data. The simulations show differences in detection thresholds for tones placed in a minimum or a maximum of the fine structure, but lack a decrease of thresholds with increased pulse duration. The model was extended by including a temporal integrator which introduces a low-pass behavior of the data with different slopes of the predicted threshold curves, producing good agreement with the data. On the basis of the model simulations, it will be discussed to which extent temporal and spectral aspects contribute to the data.

3aPP28. Accuracy of synchrony judgment between two pulses: Effects of variations in cochlear delay amount. Eriko Aiba (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol., 1-8-31 Midorigaoka, Ikeda 563-8577, Japan, aiba.eriko@aist.go.jp), Minoru Tsuzaki (Faculty of Music, Kyoto City Univ. of Arts, Kyoto, Japan), Noriko Nagata (School of Sci. and Technolog., Kwansei Gakuin Univ., Sanda, Japan), and Seiji Nakagawa (Health Res. Inst., National Inst. of Adv. Industrial Sci. and Technol., Ikeda, Japan)

The cochlear delay shifts the arrival of lower-frequency components of an auditory signal slightly but systematically behind that of higher-frequency components. Therefore, even if all of the components of a complex tone physically begin simultaneously, their temporal relation is not preserved at the cochlear level. In our previous study, the accuracy of synchrony judgment was measured using two types of chirps (compensated and enhanced chirps) and a pulse. The compensated chirp had an increasing frequency pattern to cancel out the cochlear delay. An enhanced chirp had a delay pattern that enhances the assumed cochlear delay. The pulse had a usual cochlear delay at the auditory peripheral. As a result, the accuracy of synchrony judgment was the highest in the pulse and higher in the enhanced chirp than the compensated chirp, implying that there is an asymmetric aspect. The purpose of this study is to investigate how our auditory system processes this asymmetric aspect, and what amount of temporal collapse was tolerated. We also measured the accuracy of synchrony judgment using stimuli that reverse the cochlear delay (the higher-frequency components arrive behind the lower-frequency components), or enhance the delay of lower-frequency components up to 8 times.

3aPP29. Do cochlear mechanisms explain the noise-disruption of the auditory brainstem response to speech? Helen E. Nuttall, Antje Heinrich, David R. Moore, and Jessica de Boer (MRC Inst. of Hearing Res., Science Rd., University Park, Nottingham NG7 2RD, United Kingdom, helen@ihr.mrc.ac.uk)

In background noise, the timing precision of the auditory brainstem response to speech (speech-ABR) is disrupted and the response latency increases. The severity of the disruption has been correlated with listeners'

ability to understand speech-in-noise. To date, although a central mechanism is assumed, the locus of the speech-ABR timing disruption is not clear. The present study aimed to investigate the contribution of different cochlear mechanisms to noise-induced latency increases. A first experiment examined the "cochlear place" mechanism, by which the latency of the response increases as cochlear origin moves towards lower frequency regions. The results showed that the speech-ABR reflects an average over responses from a broad range of cochlear regions, which respond with substantial relative delays. This implies that cochlear place can potentially have large effects on masked speech-ABR latency. Another mechanism that is known to be involved in noise-induced ABR latency increases is neural adaptation. This is presumed to occur at the inner hair cell-nerve junction and is thought to reflect cochlear masking. Thus, if this mechanism contributes to speech-ABR latency increases in noise, we would expect this contribution to depend on cochlear frequency selectivity and amplification gain. This hypothesis is tested in the second experiment.

3aPP30. Modeling human auditory evoked brainstem responses to speech syllables. James Harte (Inst. of Digital Healthcare, Univ. of Warwick, International Digital Lab., Coventry CV47AL, United Kingdom, harte_j@wmg.warwick.ac.uk), Filip M. Roenne (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Copenhagen, Denmark), and Torsten Dau (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Lyngby, Denmark)

Auditory evoked brainstem responses (ABR) are routinely used to assess the neural encoding of sound. Various types of stimuli have been historically considered, and there is a current increasing trend towards the use of syllables, speech and complex (non-speech) sounds. Despite the peripheral origin of ABRs, the nonlinear processing within the cochlea and brainstem makes interpreting responses to these new stimuli a challenge. A recent model was developed [Rønne *et al*. (2012)] to simulate ABRs to transient sounds such as tone pulses, clicks and rising chirps as a function of stimulus level. The present study extends this model to simulate synthetic /ba/, /da/ and /ga/ syllable-evoked ABRs, where the stimuli only differ in their spectral energy of the second formant, f2, within their first 60 ms. The model takes the convolution of the instantaneous discharge rates using a "humanized" nonlinear auditory-nerve model of Zilany *et al*. (2009) and an empirically derived unitary response function, assumed to reflect contributions from different cell populations within the auditory brainstem. The ABR model was used to explore the physiological basis and spectro-temporal characteristics of key features observed in syllable-evoked ABRs in the literature [Skoe *et al*. (2011), Hornickel *et al*. (2009)].

3aPP31. Sensitivity to stimulus polarity in speech-evoked frequency-following responses. Steve J. Aiken (School of Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Box 15000, Halifax, NS B3H 4R2, Canada, steve.aiken@dal.ca) and David W. Purcell (National Ctr. for Audiol., Western Univ., London, ON, Canada)

It has been suggested that frequency-following responses recorded to speech sounds presented in opposite polarities can be added to emphasize responses related to the periodicity envelope (e.g., at the fundamental frequency, f0) or subtracted to emphasize responses related to the stimulus fine-structure (e.g., harmonics near formant peaks) because inverting stimulus polarity has little effect on the stimulus envelope. This hypothesis was tested by comparing frequency-following responses to several tokens of two vowels (/a/ and /i/) presented twice in one polarity and once in the opposite polarity, from nine normal-hearing listeners. At harmonics near formant peaks, most listeners displayed frequency-following responses closely related to stimulus fine-structure (i.e., responses that followed stimulus polarity). However, response amplitude and phase at f0 varied greatly across polarities for many listeners, suggesting that the f0 response does not reflect a simple encoding of the periodicity envelope or the fine-structure of the first harmonic. Most listeners had similar responses when identical stimuli were presented in the same polarity, so these varied polarity-sensitive responses were not likely related to temporal encoding difficulties or background electrophysiological noise. The summed alternating polarity frequency-following response at f0 might reflect encoding of both speech fine-structure and the periodicity envelope.

3aPP32. Auditory evoked responses to a frequency glide following a static pure tone. Wen-Jie Wang (Speech-Language-Hearing Sci., Grad. Ctr., CUNY, 365 Fifth Ave., New York, NY 10016, wwang2@gc.cuny.edu), Chin-Tuan Tan (Dept. of Otolaryngol., School of Medicine, New York Univ., New York, NY), and Brett A. Martin (Speech-Lang.-Hearing Sci., Grad. Ctr., CUNY, New York, NY)

In this study, we look at the auditory evoked response to a frequency glide following a static pure tone. A frequency glide is a frequency ramp with specific frequency change range (Δf) and duration (Δt). Frequency change rate (Δf/ Δt) and direction (increasing or decreasing frequency) of a glide are important cues for speech perception. P1-N1-P2 acoustic change complex (ACC) responses to increasing or decreasing frequency glides were observed in the recordings of normal hearing subjects. Subjects were also asked to behaviorally discriminate similar stimuli with a fixed Δt at 50 ms or 200 ms and a varying Δt in a separate experiment. Similar findings were obtained with glides at both 500 and 1 kHz base frequency. In these preliminary data, we observed larger N1-P2 responses with the glides of fixed Δt 50 ms at both 500 Hz and 1000 Hz base frequency. However, larger N1-P2 responses for increasing glides than for decreasing glides were only observed with glides at 500 Hz base frequency. Larger N1-P2 response at shorter Δt seems to tally with the smaller behavioral threshold of Δt difference between stimulus with a fixed Δt at 50 ms and stimulus with varying Δt.

3aPP33. Evidence for modulation rate specific adaptation in the frequency following response? Hedwig E. Gockel, Alexandra Krugliak (MRC-Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, hedwig.gockel@mrc-cbu.cam.ac.uk), Christopher J. Plack (Audiol. and Deafness Res. Group, Univ. of Manchester, Manchester, United Kingdom), and Robert P. Carlyon (MRC-Cognition and Brain Sci. Unit, Cambridge, United Kingdom)

We used an adaptation paradigm to investigate whether the frequency following response (FFR) would show evidence for neurons tuned to modulation rate in humans, as has been previously shown in the inferior colliculus of the macaque using fMRI [Baumann *et al*. Nat. Neurosci. **14**, 423–425 (2011)]. The FFR to a 100-ms, 75-dB SPL, target complex tone with an envelope rate of 213 Hz was measured for ten subjects. The target was preceded by a 200-ms, 75-dB SPL, adaptor complex with an envelope rate of 90, 213, or 504 Hz. All complexes contained alternating-phase harmonics from approximately 3.9 to 5.4 kHz. A "vertical" montage (+ Fz, - C7, ground = mid-forehead) was used, for which the FFR is assumed to reflect phase-locked neural activity from generators in the rostral brainstem. The results showed significant adaptation effects in the spectral magnitude of the 213-Hz envelope-related component of the FFR. However, the identical-rate adaptor did not generally produce more adaptation than the different-rate adaptors. Hence, the present results do not provide evidence for neurons tuned to modulation rate in the human brainstem. [Work supported by Wellcome Trust Grant 088263.]

3aPP34. Simultaneously evoked auditory potentials: A novel paradigm for measuring auditory-evoked electroencephalographic activity at successive levels of the auditory neuraxis. Christopher Slugocki, Dan J. Bosnyak, and Laurel J. Trainor (Psychology, Neurosci., & Behaviour, McMaster Univ., 1280 Main St. West, Hamilton, ON L8S4L8, Canada, slugocc@gmail.com)

Recent electrophysiological work has evinced a capacity for plasticity in subcortical auditory nuclei in adults [Skoe and Kraus (2010)]. Similar plastic effects have been measured in cortically generated auditory potentials [e.g., Bosnyak *et al*. (2007); Näätänen (2008)], but it is unclear how the two interact. Here we present simultaneously evoked auditory potentials (SEAP), a novel paradigm designed to concurrently elicit electrophysiological brain potentials (EEG) from inferior colliculus (IC), thalamus, and primary and secondary auditory cortices. We use a specially designed stimulus consisting of a carrier frequency (500 Hz), amplitude-modulated at the sum of 37 and 81 Hz (depth 100%). We have shown that it elicits a 500 Hz frequency-following response (FFR; generated in IC), 81 (subcortical), and 37 (primary auditory cortex) Hz steady state responses, and mismatch negativity (when there is an occasional change in carrier frequency; secondary auditory

cortex). Furthermore, cortical and subcortical processes are linked as the amplitude of the FFR predicts the amplitudes of the 37 and 81 Hz responses. SEAP offers a new window into the dynamics of encoding along the auditory pathway as well as a new and inexpensive tool with which to measure plasticity at multiple levels of the auditory neuraxis in individuals.

3aPP35. Measuring subcortical and cortical neural activities for music perception: A multilevel electroencephalography study. Inyong Choi, Scott Bressler, and Barbara G. Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, iychoi@bu.edu)

Music perception requires the activation and coordination of many neuronal centers and pathways of the peripheral and central auditory pathway, which are highly overlapped with those of speech communication. Previous experiments have shown influences of musical training on brainstem encoding in the auditory periphery and long-term plasticity in the cortex. By testing listeners with different musical experience we hope to better to understand differences in central auditory processing across individual listeners. Here we explore methods for measuring subcortical and cortical neural activity in response to musically relevant stimuli by electroencephalography (EEG). A passive mismatch negativity (MMN) paradigm using familiar musical intervals was presented to subjects to measure late evoked potentials in response to deviations in absolute musical interval and musical consonance. Brainstem frequency following responses (FFRs) for carrier frequencies and phase locking values to the beat-related envelopes were simultaneously measured. Results from these experiments can provide a means to objectively quantify individual differences in central auditory processing related to musical ability through non-invasive, electrophysiological methods.

WEDNESDAY MORNING, 5 JUNE 2013 — 512CG, 9:00 A.M. TO 11:00 A.M.

Session 3aSA

Structural Acoustics and Vibration, Noise, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials I

Yun Jing, Cochair
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Dean Capone, Cochair
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Invited Papers

9:00

3aSA1. Interactive behavior of internal resonators in acoustic metamaterials under impact pulse loading. Kwek Tze Tan and C. T. Sun (Aeronautics and Astronautics, Purdue Univ., 701 West Stadium Ave., West Lafayette, IN 47907, kttan@purdue.edu)

Acoustic metamaterials exhibit negative effective mass density when the lattice system consists of mass-in-mass microstructural units. It is found out that the effective mass density becomes frequency dependent and displays negativity for frequencies near the resonant frequency of the internal resonators. The effect of a negative mass property implies that stress wave propagation is prohibited; leading to structural applications like vibration control, impact protection, and shock wave mitigation. Under impact loading, internal resonators are revealed to effectively reduce the displacement/velocity of the overall structure, and attenuate a specifically designed range of frequency where the negative effective mass density is exhibited. However, researchers have yet to study the mutual interaction between the internal resonators. Knowing how adjacent resonators interact and response to dynamic profile of preceding resonators may lead to more efficient design for stress wave attenuation. In this paper, we performed detailed investigation on the interactive behavior of internal resonators in acoustic metamaterials under an impact pulse load. Finite element analysis results show that when internal resonators are adjacently placed, they produce a coupled resonance effect, resulting in a leakage of frequency just below the resonant frequency of the resonators. This frequency leakage can lead to energy storage and harvesting applications.

9:20

3aSA2. Acoustic supercoupling and enhancement of nonlinearities in density-near-zero metamaterial channels. Caleb F. Sieck (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Romain Fleury (Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu), and Andrea Alù (Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

Recent theoretical and experimental work has demonstrated that acoustic wave tunneling and energy squeezing can be achieved using density-near-zero (DNZ) metamaterial channels [Fleury *et al*, J. Acoust. Soc. Am. **132**(3), 1956 (2012)]. These channels are directly analogous to supercoupling of electromagnetic waves in near-zero permittivity channels. In optics, the field enhancement and uniformity of response within a near-zero permittivity channel can be employed to produce switching behavior, harmonic generation, and wave mixing even with low amplitude input intensities. These optical channels have been already shown to significantly outperform enhancement of nonlinearity in conventional Fabry-Pérot resonant gratings [Argyropoulos *et al*., Phys. Rev. B **85**, 045129 (2012)]. The analogous properties of velocity field within a DNZ metamaterial channel can result in significant and uniform amplification that may be employed to enhance material or structural nonlinearities in the channel for applications like transmission switches. This work presents recent analytical and finite element modeling of the use of DNZ channels to enhance acoustic nonlinearities. It also explores and discusses metamaterial mechanisms for attaining a tailored and enhanced nonlinear response.

9:40

3aSA3. Acoustic double negativity with coupled-membrane metamaterial. Guancong Ma, Min Yang, Zhiyu Yang, and Ping Sheng (Dept. of Phys., Hong Kong Univ. of Sci. and Technol., HKUST, Clear Water Bay, Kowloon 123456, Hong Kong, phmgc@ust.hk)

Over the past decade, the emergence of acoustic metamaterials has considerably broadened the possibility of acoustic wave manipulations. Within this area, exotic effective constitutive parameters (mass density and bulk modulus) are a most hotly pursued topic, at the core of which is the realization of acoustic double negativity. Here, we show with experiments, simulations and homogenization that a single resonant structure can achieve acoustic double negativity. The metamaterial is comprised of two decorated elastic membranes, which are connected together by a rigid ring. Impedance measurement reveals that the system's transport behavior is governed by the three eigenmode resonances in the sub-kHz regime, which are separately tunable via system parameters. Measured displacement profiles at the sample surfaces using laser vibrometer show that the system's eigenmodes are, respectively, dipolar or monopolar in their nature. The simplicity of the metamaterial also enables us to retrieve its effective mass density and effective bulk modulus by performing homogenization. The results help explaining the physics behind the transport properties, and confirm that a double-negative passband is realized in a frequency range (around 500–800 Hz). Excellent agreement between experiments, simulations, and theory is achieved.

10:00

3aSA4. Conceal an acoustic sensor by using single-negative acoustic materials. Jianchun Cheng, Bin Liang, and Xuefeng Zhu (Inst. of Acoust., Dept. of Phys., Nanjing Univ., 22 Hankou Rd., Nanjing, Jiangsu, Nanjing 210093, China, jccheng@nju.edu.cn)

A coordinate transformation scheme is proposed to make a sensor acoustically undetectable while allowing it to receive external information. The designed structure only comprises complementary media (CM) whose acoustic parameters are single-negative rather than double-negative, and are totally independent of those of the sensor and the background medium. The numerical results show that the incident wave can pass across the CM changelessly, and it is therefore reasonable to identify such a structure as an acoustic cloak which enables one to "hear without being heard." Further, a nonlinear transformation scheme is developed, in an attempt to simplify the practical realization to the fullest extent. The designed multi-layered structure is formed by alternately arranging one single pair of CM with homogeneous isotropic single-negative parameters which are also totally independent of the sensor and background medium. The numerical results show that acoustic scattering from the sensor is suppressed considerably when the number of bilayers is large enough. The feasibility of designing acoustic cloaks by single-negative media and the flexibility in choosing the acoustical parameters of cloak may significantly facilitate the experimental realization of acoustic cloaks and promise new possibilities in a wide range of acoustics, optics, and engineering applications.

Contributed Papers

10:20

3aSA5. Experimental validation of the band-gap and dispersive bulk modulus behavior of locally resonant acoustic metamaterials. Matthew Reynolds, Yan Gao, and Steve Daley (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, mjr304@soton.ac.uk)

Over the last decade there has been significant interest in the design and production of acoustic metamaterials with physical qualities not seen in naturally occurring media. Progress in this area has been stimulated by the desire to create materials that exhibit novel behavior such as negative refraction due to negative material parameters, and band gaps in the frequency response of the material. An acoustic metamaterial is presented that consists of an acoustically transparent mesh with an array of split hollow spheres. Split hollow spheres are analogous to the split ring resonators found in many electromagnetic metamaterials and act as Helmholtz resonators providing a resonant band gap at low frequencies where achieving a Bragg gap would be impractical, and providing a dispersive effective bulk modulus that can become negative. Since an eventual goal of the work is to produce such materials on a micro-scale, the metamaterial is designed for, and produced using, 3D printing techniques (additive layer manufacturing). Results are presented for material comparing theory and experiment, and methods for increasing the bandwidth of the behavior in question are proposed, including a mixed resonator solution and the integration of active components into the material.

10:40

3aSA6. A meta-mass mechanical energy valve and battery—A key to vibration energy harvesting. Joh J. McCoy (The Catholic Univ. of America, Michigan Ave., Washington, DC 20064, mccoy@cua.edu)

A meta-mass comprised of a rigid housing element densely filled by a multiplicity of oscillators, is designed to act, when attached to a vibrating structure, as a one-way energy valve and mechanical battery. When the oscillators are joined with mechanical/electrical converters, the meta-mass becomes a vibration energy harvester that has a capacity that far exceeds harvesters comprised of a limited number of Newtonian mass elements. When viewed from the structure side of its attachment, the meta-mass device action as a one-way valve appears as an energy sink, such that when sufficiently strong initiates a transport process drawing energy from remote locations, thereby solving a fundamental aperture problem to energy harvesting, occasioned by the spatial diffuseness of the energy source. When viewed from the device side of the attachment, the meta-mass device action as a battery provides for the accumulation of mechanical energy, thereby solving a second fundamental problem to energy harvesting, occasioned by the arrival of the vibration energy as a sequence of weak packets. A meta-mass vibration energy harvester (MMVEH) is contrasted with the present art of vibration energy harvesters (VEH).

Session 3aSCa

Speech Communication: Imitation, Accommodation, and Convergence in Speech Communication

Molly E. Babel, Cochair
Linguistics, Univ. of British Columbia, 2613 West Mall, Totem Field Studios, Vancouver, BC V6T 1Z4, Canada

Kuniko Nielsen, Cochair
Linguistics, Oakland Univ., 320 O'Dowd Hall, Rochester, MI 48309-4401

Chair's Introduction—8:55

Invited Papers

9:00

3aSCa1. The cognitive basis of spontaneous imitation: Evidence from the visual world. Stephen D. Goldinger (Psychology, Arizona State Univ., 950 S. McAllister, Box 871104, Tempe, AZ 95287-1104, goldinger@asu.edu)

It is well-established that, when people are asked to identify and quickly repeat spoken words, they show a strong tendency to spontaneously imitate the vocal and/or phonetic characteristics of the stimulus tokens. There is mixed evidence, however, regarding the underlying basis of such imitation: Does it only reflect gestural attunement (as in Direct Realism), or does it also reflect cognitive principles of word perception and memory? The gestural attunement view has face validity, as imitation seems to require tacit knowledge of other peoples' articulatory actions. The role of memory is less obvious, although people can certainly imitate others from memory. In this talk, I will present evidence from three new experiments, pairing procedures from the "visual world" paradigm with a speech production task. Across studies, there is clear evidence that degrees of speech imitation are tightly connected to attention and memory processes that were engaged during initial exposure to spoken words. The results show clear imitation (in naming depicted objects), both with and without spoken words prompting responses, and show strong effects of competition among visual objects: Imitation increases when other potential objects have similar names, or even similar appearances. Spontaneous imitation is both a gestural and a cognitive behavior.

9:20

3aSCa2. Phonetic convergence is not a consequence of stimulus-response compatibility. Holger Mitterer (Dept. of Cognit. Sci., Univ. of Malta, University Ring Rd., Msida MSD2080, Malta, holgermitterer@yahoo.co.uk)

Moving to a different region within a country/language area tends to have the same consequences all over the globe. Phonetic patterns change over time, and they change in the direction of the pattern in the environment [Harrington *et al*. (2000), Sancier and Fowler (1997)]. This has often been attributed to a basic stimulus-response compatibility on a gestural level [Galantucci *et al*. (2009)]. The experimental evidence for a gestural stimulus-response compatibility, however, invariably confounds gestural with phonological compatibility, that is, gesturally incompatible distractor stimuli are also from a different phonological category [Mitterer and Ernestus (2008)]. In this presentation, I will present three lines of evidence that question a gestural stimulus-response compatibility driving phonetic convergence. A first experiment shows that phonetic convergence does not necessarily target concrete phonetic detail, but rather more global and abstract parameters. Second, I will show that there is no relationship between the amount of phonetic stimulus-response overlap and response latency, the typical measure of stimulus-response-compatibility effects. Finally, I will present data that show that two participants in a conversation can maintain a marked phonetic difference. The combined evidence indicates that phonetic convergence is not driven by a gestural stimulus-response compatibility.

Contributed Papers

9:40

3aSCa3. Evidence for an articulatory component of phonetic convergence from dual electromagnetic articulometer observation of interacting talkers. Mark Tiede (Haskins Labs., 300 George St., New Haven, CT 06511, tiede@haskins.yale.edu) and Christine Mooshammer (U.S.C. & Haskins Labs., Los Angeles, CA)

Speech audio and articulatory movements of age- and gender-matched speaker pairs have been recorded in face-to-face interaction using two electromagnetic articulometer (EMA) systems simultaneously. Speakers matched for language (AE:AE) were compared with mixed language backgrounds (AE:Spanish, AE:German). Tasks included synchronized, imitated, and spontaneous speech interspersed with repeated baseline utterances for evaluation of mutual accommodation. Euclidean distances (EDs) between vowel midpoint formant frequencies from the initial and final baseline tasks showed symmetric reductions associated with convergence which were larger for the speakers with the same language background. To test whether the observed reduction might be due to fatigue or repetition effects EDs were also calculated between AE speakers who did not participate in the same experiment. Because reduction was minimal for this comparison the possibility that convergence can be attributed to simple optimization is excluded. Kinematic and dynamic convergence was examined on coda velar gestures from the baseline tasks. EDs in peak velocity, sensor path distance, gesture amplitude, and stiffness computed between speakers all showed

decreases consistent with convergence. Associated within-speaker differences showed that accommodation was effectively symmetric, and because the speakers moved towards one another the difference was not due to practice or fatigue effects. [Work supported by NIH.]

10:00

3aSCa4. Visual enhancement of alignment to noisy speech. James W. Dias and Lawrence D. Rosenblum (Psychology, Univ. of California, Riverside, 900 University Ave., Riverside, CA 92521, jdias001@ucr.edu)

Talkers imitate aspects of perceived speech; a phenomenon known as speech alignment. Previous studies have found that talkers will align not only to auditory speech, but also visual speech information. Furthermore, talkers who interact in full view of a conversational partner will align more than talkers who interact with a partner they can only hear [Dias and Rosenblum, Perception **40**, 1457–1466 (2011)]. The purpose of the current investigation is to evaluate whether visual speech information increases speech alignment when the auditory speech is noisy. Participants shadowed (said out-loud) the utterances of a pre-recorded model presented under audio-alone or audiovisual conditions, either in noise (+10 db SNR) or in the clear. These shadowing tokens were then rated for their similarity (alignment) to the model's tokens using a matching task performed by naïve raters. Despite word identification performance being the same across all conditions, participants aligned more to the model when listening to speech in the clear than speech in noise. Further, the addition of visual information enhanced alignment for words presented in noise, but not for words presented in the clear. These data suggest the availability of visual speech information can enhance speech alignment when shadowing in noise.

10:20–10:40 Break

Invited Papers

10:40

3aSCa5. Reconciling diverse findings in studies of phonetic convergence. Jennifer Pardo (Psychology, Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, pardo@optonline.net)

Phonetic convergence occurs both when individuals interact in conversation and when listeners rapidly repeat words presented over headphones. Results from multiple studies examining phonetic convergence offer an array of often confusing and disparate findings. Reconciling such diverse findings is difficult without a clear rationale for engaging in one acoustic measure over another. The current paper proposes a paradigm that models perceptual and acoustic measures together. Measures of multiple acoustic-phonetic attributes were compared with a perceptual measure of phonetic convergence in a shadowing study. Although convergence was not significant in any acoustic measure alone, the combination of acoustic attributes predicted perceived phonetic convergence on an item-by-item basis. Because perceptual measures integrate across multiple acoustic-phonetic dimensions, future studies of phonetic convergence should use perceptual tasks to calibrate the relative contribution of individual acoustic-phonetic parameters.

11:00

3aSCa6. Phonetic convergence, communicative efficiency, and language distance. Ann Bradlow, Midam Kim (Linguistics, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu), and Minyoung Kim (Management and Int. Business, The Univ. of Kansas, Lawrence, KS)

Many English conversations across the globe today involve talkers with different language experiences. Here we show that, while language barriers challenge communicative efficiency, the detrimental effect of language distance may be mitigated by phonetic convergence. We analyzed a corpus of 42 conversations in which talker pairs solved a spot-the-difference puzzle by verbally comparing two scenes only one of which was visible to each talker ("diapix" task). Language distance was varied by pairing talkers who either matched or mismatched in language background and in native/nonnative status relative to the target language. Communicative efficiency was measured by task-completion-time and word type-to-token ratio. Phonetic convergence was assessed by perceptual similarity tests in which listeners ($n = 161$) compared samples from one talker's speech to samples from his/her partner's speech from either early or late portions of the conversation. In this test, greater similarity for late than early samples indicates convergence. Results showed a negative correlation between language distance and communicative efficiency, a negative correlation between language distance and phonetic convergence, and a mitigating effect of phonetic convergence on the negative correlation between language distance and communicative efficiency. This suggests that convergence may be an effective mechanism for overcoming the detrimental effects of a language barrier.

Contributed Paper

11:20

3aSCa7. Phonetic imitation by individuals with Autism Spectrum Disorders: Investigating the role of procedural and declarative memory. Jeff Mielke (Dept. of English, North Carolina State Univ., Tompkins 286, Raleigh, NC 27695-8105, jimielke@ncsu.edu), Kuniko Nielsen (Linguist Dept., Oakland Univ., Rochester, MI), and Lyra Magloughlin (Dept. of Linguist, Univ. of Ottawa, Ottawa, ON, Canada)

This study investigates the role of procedural and declarative memory in phonetic imitation, by examining the word- and phoneme-specificity of imitation produced by individuals with autism spectrum disorders (ASD). Previous research has shown that individuals with ASD process language differently from the Neurotypical population [e.g., Ullman (2004), Walenski *et al.* (2006)], with Autistic individuals relying more on declarative memory. Previous work with the general population has shown a robust effect of phonetic convergence [e.g., Pardo (2006)], as well as generalization and weak word-specificity effects [Nielsen (2011)]. To test whether individuals with ASD exhibit increased specificity, we used Nielsen's (2011) experimental paradigm, which has been shown to elicit generalized phonetic imitation in the general population. A linear mixed effects regression analysis revealed that increased VOT on the modeled phoneme /p/ was imitated by both ASD and control groups [$p < 0.05$]. However, different patterns emerged in phoneme-level specificity: the control group exhibited sub-phonemic generalization (increasing VOT on /p/ and /k/), while the ASD group exhibited a phoneme-specific pattern (increasing VOT only on /p/) [$p < 0.05$]. Furthermore, a stronger trend toward word-specificity was observed within the ASD group. Taken together, these results confirm the earlier finding that ASD individuals exhibit greater reliance on declarative memory.

11:40–12:10 Panel Discussion

Session 3aSCb

Speech Communication: Components of Informational Masking

Gaston Hilkhuysen, Cochair
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Yoshitaka Nakajima, Cochair
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Chair's Introduction—8:55

Invited Papers

9:00

3aSCb1. Understanding informational masking from a neural perspective. Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

Historically, psychoacousticians have divided the influence of a task-irrelevant sound (a masker) on perception of a task-relevant sound (the target) into components of (1) energetic masking and (2) informational masking. In this apportionment, energetic masking is defined as that masking that can be accounted for by considering how reliably the target is represented in the auditory periphery, and how much the masker disrupts this target representation. In contrast, the term informational masking is a catchall representing any effects of the masker that could not be accounted for by energetic masking. This talk presents a framework for understanding informational masking from a neural perspective, building on both behavioral results and neuro-imaging data. In this account, informational masking is a result of bottlenecks in the neural processing of acoustic information, a problem that the brain mediates by engaging auditory attention. Auditory attention operates by modulating the representation of different auditory objects making up a particular acoustic scene, resulting in a relative enhancement of whatever object is in the attentional foreground at the expense of the representation of competing sources. Thus, most informational masking arises from failures of target formation and/or failures of target selection.

9:20

3aSCb2. The segregation of target speech from competing speech when listening in a second language. Bruce A. Schneider (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, ON L5L 1C6, Canada, bruce.schneider@utoronto.ca)

Introducing a perceived spatial separation (via the precedence effect) between target speech and competing speech reduces the signal-to-noise ratio (SNR) required for the recognition of key target words by 3–7 dB in syntactically correct but semantically anomalous target sentences such a "A rose could paint a fish" (target words: rose, paint, fish). A recent study of monolingual Mandarin participants listening to anomalous sentences in Mandarin Chinese suggests that perceived spatial separation releases listeners from informational masking (IM) by facilitating access to the lexicon at the morphemic level, presumably by enhancing the segregation of the target speech from the competing speech background. This raises the interesting question as to whether perceived spatial separation facilitates lexical access in the same way for people listening in the second language (L2) as it does when listening in their first language (L1) since access to the L2 lexicon is likely to be less robust than access to the L1 lexicon. A second interesting question is the extent to which operating in a L2 environment affects release from IM when the competing speech is in L1. These issues will be addressed in L2 English listeners whose L1 is either Chinese or Korean.

9:40

3aSCb3. Exploring the factors predictive of informational masking in a speech recognition task. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Med. Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov) and Anna C. Diedesch (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

The effects of informational masking (IM) can be recast as a question of which cues to sound source identity (auditory object formation) are most useful for overcoming IM. We hypothesize that individual differences are related to specific interactions of stimulus and listener-specific variables that determine the effectiveness of the auditory object formation process. Results from our laboratory generally support the well-established relationship between performance and stimulus variables such as spectrotemporal cues (in this case, voice differences) and spatial cues (talker locations). In addition, the listener-specific variables of age and hearing loss were found to interact with the stimulus variables and to be correlated with potential mediating variables such as interaural time sensitivity and minimum levels at which speech identification was possible. Future work will involve developing predictive models that focus on identifying the mediating variables responsible for increased susceptibility to IM and efficient tests to reveal these relationships in individual listeners. The clinical relevance of the ability to identify factors predictive of IM susceptibility will be discussed, including the potential for improved fitting of hearing aids and cochlear implants.

10:00

3aSCb4. Interactions between listening effort and masker type on the energetic and informational masking of speech stimuli. Douglas Brungart (Audiol. and Speech Ctr., Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil), Nandini Iyer, Eric Thompson, Brian D. Simpson (711th HPW, Air Force Res. Lab., Wright-Patterson AFB, OH), Sandra Gordon-Salant, Jaclyn Shurman, Chelsea Vogel (Dept. of Hearing & Speech Sci., Univ. of Maryland, College Park, MD), and Kenneth W. Grant (Audiol. and Speech Ctr., Walter Reed NMMC, Bethesda, MD)

In most cases, normal-hearing listeners perform better when a target speech signal is masked by a single irrelevant speech masker than they do with a noise masker at an equivalent signal-to-noise ratio (SNR). However, this relative advantage for segregating target speech from a speech masker versus a noise masker may not come without a cost: segregating speech from speech may require the allocation of additional cognitive resources that are not required to segregate speech from noise. The cognitive resources required to extract a target speech signal from different backgrounds can be assessed by varying the complexity of the listening task. Examples include: (1) contrasting the difference between the detection of a speech signal and the correct identification of its contents; (2) contrasting the difference between single-task diotic and dual-task dichotic listening tasks; and (3) contrasting the difference between standard listening tasks and one-back tasks where listeners must keep one response in memory during each stimulus presentation. By examining performance with different kinds of maskers in tasks with different levels of complexity, we can start to determine the impact that the informational and energetic components of masking have on the listening effort required to understand speech in complex environments.

WEDNESDAY MORNING, 5 JUNE 2013 — 510A, 8:55 A.M. TO 12:00 NOON

Session 3aSP

Signal Processing in Acoustics and ASA Committee on Standards: Methods and Applications of Time-Frequency Analysis

Leon Cohen, Cochair
City Univ. of New York, Hunter-Physics, 695 Park. Ave., New York, NY 10065

Patrick J. Loughlin, Cochair
Bioengineering, Univ. of Pittsburgh, 302 Benedum Hall, Pittsburgh, PA 15261

Chair's Introduction—8:55

Invited Papers

9:00

3aSP1. Speaker identification made easy with pruned reassigned spectrograms. Sean A. Fulop (Linguistics, California State Univ. Fresno, 5245 N Backer Ave., PB92, Fresno, CA 93740-8001, sfulop@csufresno.edu) and Youngwook Kim (Elec. and Comput. Eng., California State Univ. Fresno, Fresno, CA)

One common scenario for speaker identification presents the task of identifying samples of speech from members of a previously enrolled group. One recent (and typical) set of results used 36 s of speech from each speaker to train Gaussian models by expectation-maximization during enrollment, and 20 s of speech for the test samples. Three major problems with this procedure are (1) sensitivity to noise; (2) impractical amounts of speech are required; (3) computationally expensive training is required. In our study, the reassigned spectrogram is pruned using phase-derivative indicator functions to provide a sparse time-frequency matrix of very small (40 ms) samples of speech. The pruning eliminates Gaussian noise up to ~1 dB SNR. Principle components analysis provided a set of 30 features from each spectrogram. Using an enrolled group of 24 speakers recorded under low-fidelity conditions, 83% identification accuracy (comparable to state of the art results with 6 dB SNR) was achieved from real-time classification methods (e.g., support vector machines) without need for extensive training. Moreover, these results extend to 1 dB SNR where standard techniques break down. The three main problems of speaker identification can thus be better addressed by our methodology.

9:20

3aSP2. Phase-space equation for wave equations. Leon Cohen (City Univ. of New York, Hunter-Physics, 695 Park. Ave., New York, NY 10065, leon.cohen@hunter.cuny.edu)

Transforming a space-time function into the phase space of position and wave number offers considerable insight into the nature of the function, and also has many practical applications. If the function is governed by a wave equation then the common procedure is to solve the wave equation and then calculate the phase space distribution function for the solution. We show that significant advantages ensue if one transforms the original differential equation into a phase space differential equation. We give a number of examples and show that phase space equations are often more revealing than the original equation and lead to new approximation methods. [Work supported by ONR.]

9:40

3aSP3. Nonstationary vibration analysis in the smoothed Wigner domain. Lorenzo Galleani (Politecnico di Torino, Corso Duca degli Abruzzi 24, Torino 10129, Italy, galleani@polito.it)

Dynamical systems represent fundamental models for vibration analysis. When the input of such dynamical systems is nonstationary also the output is nonstationary, and its frequency content changes with time. Time-frequency analysis provides an effective representation of this time-varying spectrum. Even more effective is the direct time-frequency representation of dynamical systems. We first show how to transform a dynamical system in the domain of the smoothed Wigner distribution. The result is a time-frequency dynamical system whose input and output are the smoothed Wigner distributions of the input and output signals in time, respectively. Then, we illustrate how to compute the smoothed Wigner output when the input to the dynamical system in time belongs to a class of common nonstationary inputs, including a delta function and a short duration sinusoid. Finally, we show how to obtain the smoothed Wigner output when the input is a linear combination of these nonstationary signals. We provide a series of examples that show how the time-frequency representation of dynamical systems can unveil the spectral structure of nonstationary vibrations.

10:00

3aSP4. Time-frequency analysis with Bayesian filtering methods for dispersion tracking and geoacoustic inversion in the ocean. Nattapol Aunsri and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd, Newark, NJ 07102, michalop@njit.edu)

Iterative and sequential Bayesian filtering methods have been previously used in dispersion tracking for long range sound propagation in oceanic environments. The methods rely on accurate modeling of (i) the acoustic field in the frequency domain as a function of time and (ii) the statistical model governing the behavior of errors in the measured data. Normal distributions are typically used for the latter but may not necessarily be the most accurate models for the description of the data. We investigate alternative methods for describing the statistical errors in power spectra, which we then use for linking the extracted time-frequency information to geoacoustic inversion. Probability density functions computed for frequencies and arrival times via filtering are propagated "backwards" through sound propagation models for quantification of the uncertainty in the estimation process. [Work supported by ONR].

10:20

3aSP5. Single snapshot spatial array processing using time-frequency distributions. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Several signal processing applications, such as spatial beamforming, rely on data collected by an array of sensor to estimate to enhance a weak signal in the presence of noise (e.g., ambient noise or clutter). The commonly applied eigenstructure subspace methods to this signal denoising problem assume stationary signals and require multiple snapshots to correctly estimate the covariance matrix of the array data. However, these multiple snapshots and stationarity requirements can be hard to meet in practical scenarios involving among others a rapidly moving source (which causes differential Doppler effect among sensors) or a single snapshot of the aspect-dependent scattering of an unknown target as measured by a monostatic sonar system (such as side-scan sonar). To handle these scenarios, we propose to form a generalized space-time-frequency covariance matrix from the single-snapshot data by computing Cohen's class time-frequency distributions between all sensor data pairs. The eigenstructure of this space-time-frequency covariance matrix allows to enhance the localization of the signal of interest while spreading the noise power in the time-frequency domain. Hence, this approach is especially suited to handle nonstationary echoes of underwater target resonances that are highly localized in the time-frequency domain as demonstrated using numerical and at-sea data.

10:40

3aSP6. Time-frequency analysis of transient high-frequency dispersive guided waves on tilted cylindrical shells: Review. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Scot F. Morse (Comput. Sci. Div., Western Oregon Univ., Monmouth, OR)

Measurements of the back scattering by bluntly truncated tilted cylindrical shells in water reveal a dependence on the aspect angle, which can be interpreted using geometrically described coupling mechanisms [Morse *et al*., J. Acoust. Soc. Am. **103**, 785–794 (1998)]. By exciting a shell with a suitable acoustic pulse localized in time, the recorded backscattering response reveals a significant evolution of the spectrum as a function of time and tilt angle. This evolution was interpreted using the dispersion relations of the relevant high-frequency shell guided waves [Morse and Marston, J. Acoust. Soc. Am. **111**, 1289–1294 (2002)]. The coupling conditions are affected by the mode properties. This interpretation was facilitated by also computing the scattering properties of an infinitely long titled shell [Morse and Marston, J. Acoust. Soc. Am. **106**, 2597–2600 (1999)] and by measuring and modeling the contributions of helical and meridional rays [Morse and Marston **112**, 1318–1326 (2002); Blonigon and Marston, **112**, 528–536 (2002)]. One of the shells investigated was also convenient for a quasi-holographic imaging of the bistatic scattering of short pulses [Baik *et al*., 130, 3838–3851 (2011)]. [Work supported by ONR.]

Contributed Papers

11:00

3aSP7. Time frequency analysis techniques for long range sediment tomography. Gopu R. Potty and James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu)

Long range sediment tomography inversion technique requires accurate estimation of the arrival times of acoustic normal modes. The inversion technique is based on minimizing the difference between the forward model predictions and the data using a global optimization scheme. Predictions are computed using a trial parameter set which is iteratively modified until the algorithm converges. The modal arrival times are calculated from the time-frequency analysis of broadband acoustic data collected on a single hydrophone. During the initial stages of the development of the inversion scheme, Fourier-based spectrograms and wavelet-based scalograms were used. Taking advantage of some of the recent developments, dispersion based short time Fourier transform (DSTFT) and warping transform techniques were used for the time-frequency analysis, in recent times. This paper will

summarize and compare the performance of these time frequency techniques in the context of long range sediment tomography. Data from some of the recent field tests (Shelfbreak Primer and Shallow Water-06 experiments) will be analyzed for this study. Finally, time-frequency analysis performance of another new technique, Modified S transform, will be examined using the field data. [Work supported by Office of Naval Research].

11:20

3aSP8. Time-frequency analysis of wood anomalies in acoustics. Jingfei Liu and Nico F. Declercq (George W Woodruff School of Mech. Eng., Georgia Inst. of Technol., 2, rue Marconi, Metz 57070, France, benjamin.jf.liu@gatech.edu)

When performing a Fourier transform on the reflection from a periodically corrugated surface for a normally incident beam the reflection spectrum is obtained. Steep dips have been observed from such reflection spectrum and are commonly called Wood anomaly. Because the frequencies at which Wood anomalies appears coincide with the frequencies where stimulation of Scholte-Stoneley waves is expected, Wood anomalies are traditionally explained as being caused by Scholte-Stoneley waves and as drain of energy. In this work, time-frequency domain analysis is made on the reflection from corrugated surfaces. The analysis suggests that Wood anomalies only occur when the entire reflection is processed by a Fourier transform, whereas for short time Fourier analysis this phenomenon does not seem to occur. The analysis also shows that energy is delayed in the time domain at the generation frequency of Scholte-Stoneley waves, and is therefore not necessarily drained. From the time-frequency point of view the source of Wood anomalies is re-examined and the actual relationship between the surface wave generation and Wood anomalies is also investigated.

11:40

3aSP9. Precise order tracking analysis of time-varying vibro-acoustic signature from rotating machines. Jin-Ho Bae (Mech. Eng., KAIST, 291 Daehak-ra, Yuseong-gu, Daejeon 305-701, South Korea, supurpower@kaist.ac.kr), Jeong-Guon Ih (Mech. Eng., KAIST, Daejeon, Chungcheongnam-da, South Korea), and Sang-Ryeol Kim (System Dynam., KIMM, Daejeon, Chungcheongnam-do, South Korea)

Transient operation of rotating machines usually exhibit a large variation in rotating speed and frequencies of interest often fluctuate due to RPM variations. Order tracking is useful for the transient analysis of vibro-acoustic signal from a rotating machine. A fine resolution is required both in time and frequency domains to trace the spectral magnitude of the interested frequency in precision. In this study, characteristics of popular order tracking methods using short time Fourier transform (STFT) and Vold-Kalman filter are studied in the viewpoint of calculation time and resolution. They are compared with a new tracking method using variable frequency resolution STFT (VFR-STFT), being suggested for the sound quality analysis. For a test signal, vehicle interior noise measured in an idle run-up condition is used. Under the same signal processing conditions, time and frequency domain resolutions produced by three methods are compared. It is observed that VFR-STFT detects bandwidth of high order components of engine firing frequency precisely and it produced the finest resolution in time and frequency domain among three methods. Although VFR-STFT needs a bit more calculation time compared to the other methods, it is useful for the cases that do not require a real time analysis.

WEDNESDAY MORNING, 5 JUNE 2013 — 511AD, 8:55 A.M. TO 9:40 A.M.

Session 3aUWa

Underwater Acoustics and Engineering Acoustics: Acoustic Sensing Via Fiber Optics

Jeremy Brown, Cochair
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Andrew R. McNeese, Cochair
ARLUT, 10000 Burnet Rd., Austin, TX 78758

Chair's Introduction—8:55

Invited Paper

9:00

3aUWa1. Tracking a human walker with a fiber optic distributed acoustic sensor. Emery M. Ku and Gregory L. Duckworth (OptaSense, 27 Moulton St., Cambridge, MA 02138, emery.ku@optasense.com)

Traditional ground sensors can be used as security systems to detect third party intrusion, but are costly and cumbersome to scale up to cover a large area. Fiber optic distributed acoustic sensing (DAS) offers a solution that economically and instantaneously monitors up to 40 linear km of a border or perimeter with a single system. The output of the OptaSense third generation fiber interrogation unit is spatially dense, coherent, and exhibits characteristics expected of traditional strain sensors. In this paper, DAS data are shown to indicate a walker's directionality and support beamforming algorithms for localization.

Contributed Paper

9:20

3aUWa2. Measurements of Brillouin gain spectra in erbium-doped optical fibers for long-distance distributed strain and temperature sensing. Mingjie Ding, Yosuke Mizuno, Neisei Hayashi, and Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., R2-26, 4259 Nagatsuta-cho, Midori-ku, Yokohama, Kanagawa, Tokyo 226-8503, Japan, ding.m.aa@m.titech.ac.jp)

Brillouin scattering, one of the most important acousto-optical phenomena, occurs when lightwave interacts with periodical refractive-index change caused by the acoustic wave. The frequency downshift from the incident light to the backscattered Stokes light is called the Brillouin frequency shift (BFS), which depends on the strain/temperature applied to the optical fiber. Fiber-optic distributed Brillouin sensors have already been widely utilized to monitor strain/temperature conditions of structures including buildings, bridges and cables. The measurement range of the sensing system is partially limited by the optical propagation loss inside the sensing fiber. One solution is to employ special fibers capable of amplifying Brillouin signal. In this work, we employ erbium-doped fibers (EDFs) as sensing fibers, which are commonly used as 1550 nm optical amplifiers. As the first step, we investigated the strain/temperature dependence of Brillouin gain spectra (BGS) in EDFs with three different erbium concentrations. For all the samples, with increasing strain/temperature, the BGS shifted toward higher frequency, which was the same as that of silica fibers. Moreover, as the erbium concentration increased, the strain coefficient of BFS was increased, while its temperature coefficient was reduced. These results will definitely contribute to the future research of EDF-based distributed strain/temperature sensing systems.

WEDNESDAY MORNING, 5 JUNE 2013 — 511AD, 10:00 A.M. TO 11:40 A.M.

Session 3aUWb

Underwater Acoustics: Experimental Sensors and Methods

Jeremy Brown, Cochair
Biomedical Eng., Dalhousie Univ., 1276 South park St., DCson Bldg. rm 3191, Halifax, NS B3H2Y9, Canada

Andrew R. McNeese, Cochair
ARLUT, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

10:00

3aUWb1. A towable combustive sound source for ocean surveys and ocean acoustics experiments. Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and David K. Wareham (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The Combustive Sound Source (CSS) is a broadband impulsive sound source that generates a wide bandwidth underwater acoustic signal, similar to explosives and airguns, yet allows for a reduced and controllable acoustic output, suitable for meeting modern environmental regulations. The source consists of a submersible combustion chamber filled with electrolytically generated hydrogen and oxygen ignited via spark. Upon ignition, the combustive mixture is converted into high temperature water vapor and the ensuing bubble activity radiates broadband acoustic energy. CSS has previously been used in the water column from stationary platforms, and has been deployed on the bottom to generate seismic interface waves. We now report the successful implementation of a self-contained CSS system deployed in a tow body. No hazardous gas is ever stored on board the ship, as it is produced *in situ* while at depth. The system can produce high amplitude acoustic pulses while being stably towed behind a ship with an electromechanical cable. Discussion will focus on the functionality, capability, and expandability of the system. [Work supported by ONR.]

10:20

3aUWb2. A single-crystal acoustic hydrophone for increased sensitivity. Jeremy Brown (Biomedical Eng., Dalhousie Univ., 1276 South Park St., Dickson Bldg. rm 3191, Halifax, NS B3H2Y9, Canada, j.brown@dal.ca), Kevin Dunphy, and Olivier Beslin (Res. and Development, Ultra Electron. Maritime Systems, Dartmouth, NS, Canada)

This study describes the development of a underwater surveillance hydrophone based on next generation $PbMg_{1/3}Nb_{2/3}O_3$-$PbTiO_3$ (PMN-PT) single-crystal piezoelectric as the hydrophone substrate. Although PMN-PT can possess much higher piezoelectric sensitivity than traditional PZT piezoelectrics, it is highly anisotropic and therefore there is a large gain in sensitivity only when the crystal structure is oriented in a specific direction. Because of this, simply replacing the PZT substrate with a PMN-PT cylinder is not an optimal solution because the crystal orientation does not uniformly align with the circumferential axis of the hydrophone. Therefore, we have developed a novel composite hydrophone that maintains the optimal crystal axis around the hydrophone circumference. An 11.3 mm diameter composite hydrophone cylinder was fabricated from a single <110> cut PMN-PT rectangular plate. Solid end caps were applied to the cylinder and the sensitivity was directly compared with a solid PZT-5A cylindrical hydrophone of equal dimensions in a hydrophone test tank. The charge sensitivity showed a 7.5 dB improvement over the PZT hydrophone and the voltage sensitivity showed a 2.7 dB improvement. This was in relatively good agreement with the theoretical improvements of 12.88 dB and 5.44 dB, respectively.

10:40

3aUWb3. The shallow sea experiment with usage of linear hydrophone array. Wojciech Szymczak (Hydroacoustic Inst., Polish Naval Acad., str. Smidowicza 69, Gdynia 81-103, Poland, ws2@o2.pl), Eugeniusz Kozaczka (Gdansk Univ. of Technol., Gdynia, Poland), Grazyna Grelowska, Ignacy Gloza, and Sławomir Kozaczka (Hydroacoustic Inst., Polish Naval Acad., Gdynia, Poland)

Purpose of this article is to present designed and made linear hydrophone array and the results obtained during *in situ* trails on Gulf of Gdańsk. The measuring system allowed to localize hydrophones in the selected points and perform measurements in both the horizontal antenna positioning and vertical. Made in this way recordings allow creating accurate 3D imaging of sound intensity/propagation. During research three floating objects were measured: small ship (18 m long), yacht (12 m long) and 5 m pontoon with engine and paddles used to drive. In the article, accurately will be described the entire measurement system and complementary devices (navigation system, sound speed profiler, online underwater monitoring to control linear antenna position) and procedures used during *in situ* measurement circuit check and calibration using Lubell Underwater Speaker with amplifier and connected generator with set of reference signals. Characteristic arrangement of sensors allows use of hyperbolic navigation algorithms which results will be presented with an emphasis on measurements when the unit performed circulation around the measurement system. Furthermore, some spectrograms, cross correlations and frequency classification dependence of the speed using a prepared script in MATLAB programing environment will be discussed and presented.

11:00

3aUWb4. Laboratory investigation with subbottom parametric echosounder SES-2000 standard with an emphasis on reflected pure signals analysis. Wojciech Szymczak, Grazyna Grelowska (Hydroacoustic Inst., Polish Naval Acad., str. Smidowicza 69, Gdynia 81-103, Poland, ws2@o2.pl), Eugeniusz Kozaczka (Gdansk Univ. of Technol., Gdynia, Poland), and Sławomir Kozaczka (Hydroacoustic Inst., Polish Naval Acad., Gdynia, Poland)

The main goal of the paper is to describe correlations between measurements results of trials taken on Guls of Gdańsk bottom sounded with parametric echosounder SES-2000 Standard and laboratory research where collected during survey sediments were measured. Stationary tests took place at Gdańsk University of Technology where 30 meters long 1.8 meter deep and 3 meters wide water tank is located. Main lobe of antenna was directed parallel to the longest dimension. Hydrophones used during experiment were fixed to the 3D positioning system—ISEL, which gave the opportunity to place sensor with high precision in the middle of main lobe or other specified places. Using prepared to this experiment frames different sea bottom layers configurations corresponding to the natural structure were sounded. Data obtained during laboratory measurements and trials *in situ* were combined to draw conclusions about proper interpretation of echograms and begin the process of sediments classification. Analyses were done with MATLAB programing software were data were imported and used to the simulations and comparisons.

11:20

3aUWb5. Environmental acoustic parameters of the Sea of Japan shelf (Peter the Great Gulf). Alexandr N. Samchenko, Igor O. Yaroshchuk, and Alexandra V. Kosheleva (V.I.Il`ichev Pacific Oceanological Inst., 43, Baltiyskaya Str, Vladivostok, 690041, Russia, samchenko.alexandr@yandex.ru)

The paper describes general geoacoustic model of the Peter the Great Gulf (the largest gulf of the Sea of Japan) and detailed model of its part of 400 square km in size. The general geoacoustic model is formed on the basis of geological, seismic, and bathymetric data. It includes distribution of P-, S-wave, density, and attenuation of friable sediments at the bottom surface and averaged characteristics for different types of rocks in the gulf. Bathymetric data processing was carried out by means of two-dimensional singular spectrum analysis, based on the ratio between the size of relief structures and the energy expended under the development of tectonic processes. Detailed geoacoustic model is based on processing of hydroacoustic and seismic authors' studies (3.5 kHz sonar, 50–100 kHz sonar and air-gun). Three sedimentary layers and the location of upper edge of the granite stratum are marked in the observable shelf area. The model demonstrates significant changes in P-, S-wave in the sediments of the lateral (1500–1750 m/s—P-wave and 120–600 m/s—S-wave) and vertical (1500–5400 m/s—P-wave and 120–3300 m/s—S-wave). Some of the most important oceanographic mechanisms determining both large- and small-scale spatio-temporal variations of sound speed field in the same area are presented. These data were obtained by the authors within last several years. Qualitative analysis of the propagation of low-frequency signals is presented as an example of application of the model.

WEDNESDAY AFTERNOON, 5 JUNE 2013 513ABC, 1:00 P.M. TO 2:00 P.M.

Session 3pAAa

Architectural Acoustics and Musical Acoustics: Virtual Concert Hall Acoustics II

Sungyoung Kim, Cochair
RIT, ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623

Wieslaw Woszczyk, Cochair
Music Res., McGill Univ., Schulich School of Music, 527 Sherbrooke St. West, Montreal, QC H3A1E3, Canada

Contributed Papers

1:00

3pAAa1. Is there really a whispering gallery at the Great Ballcourt at Chichen Itza, Mexico? David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

A "whispering gallery" (WG) at the Great Ballcourt (GBC) was first reported during its excavation in the 1920s by American archaeologist Silvanus Morley (1883–1948), Director of the Carnegie Institution's Chichen Itza project. In his 1925 National Geographic article Morley wrote: "Standing in this temple one can speak in a low voice & be heard distinctly at the other end of the court, 500 ft away." Around 2000–2001, queries on AZTLAN, a semi-official Mesoamerican archaeology Internet discussion group, found little or no belief in a WG by mesoamericanists. Some opined that any WG would surely be a design accident or an artifact of ballcourt ageing or reconstruction. They stiffened at the suggestion that the ancient Maya might have possessed the requisite knowledge for intentional design.

Was Morley mistaken? Or are modern mesoamericanists missing something? During a tour of Chichen Itza following the fall 2002 joint acoustical meeting in Cancun, Mexico, the author and two of his colleagues convincingly demonstrated a GBC WG to about 100 acousticians and their companions. This paper describes WG phenomena observed at the Great Ballcourt and suggests physical models to explain them. He also presents evidence for intentional design.

1:20

3pAAa2. Three-dimensional sound spatialization at Auditorio400 in Madrid designed by Jean Nouvel. Emiliano del Cerro and Silvia M[a] Ortiz (TIC, Universidad Alfonso X el Sabio, Avenida Universidad 1, Madrid 28691, Spain, ecerresc@uax.es)

The auditorium 400 was designed by the team of Jean Nouvel, the French architect, Pritzker Prize winner in 2008. It belongs to the organization of the National Museum and Reina Sofia Art Centre in Madrid, and is incorporated in a special room within the cultural life of Madrid (Spain). The center collaborates with the Spanish Ministry of Culture, and organizes a series of concerts of contemporary music and electronic and computer music. To achieve the sound projection equipment, the direction of the audience chose the system Acousmonium, designed by the GRM inside the ORTF in Paris. This paper will explain the involvement of the group LIEM (laboratory for Computer and Electronic Music), from Reina Sofia Museum with space in music: The musical relationships and implications of this choice, as well as the technical, architectural, and signal processing techniques used for the design of algorithms for spacialization of sound. After giving a very general overview of specific algorithms for spacialization, we explain some musicals examples designed specifically for this space, and the impact of its implementation and diffusion in the auditorium400 that have very special technical and artistic features.

1:40

3pAAa3. Effect of acoustic and visual stimuli on preference for different seating positions in a concert hall and an opera theater. Shin-ichi Sato, Adrián Saavedra, Alejandro Bidondo (Ingeniería de Sonido, Universidad Nacional de Tres de Febrero, Varentín Gómez 4752, Caseros, Buenos Aires 1678, Argentina, ssato@untref.edu.ar), Shuo Wang, Yuezhe Zhao, Shuoxian Wu (State Key Lab. of Subtropical Bldg. Sci., South China Univ. of Technol., Guangzhou, China), Nicola Prodi, and Roberto Pompoli (Dipartimento di Ingegneria, Università degli Studi di Ferrara, Ferrara, Italy)

The sound fields and the views of several positions in a concert hall and an opera theater were simulated and the subjective preference for different seating positions was investigated. First, the seat preference with and without visual stimuli under the conditions of (1) the original sound level (the sound pressure level at each position was maintained as the impulse response measurements in the auditoria) and (2) the equalized sound level were compared since the subjective scale of seat preference showed the highest correlation with sound level in the previous study investigating the opera theater [Sato *et al*., Acustica united with Acta Acustica **98**, 749–759 (2012)]. Some positions were judged acoustically preferred but visually less preferred or vice versa. Thus, another preference test was conducted by using the combinations of the acoustic and the visual stimuli of different positions to further investigate the audio-visual interaction.

Demonstration

The organizers of "Virtual Concert Hall Acoustics" have arranged a demonstration and a live music concert using virtual acoustics technology of McGill University. The demo and concert will take place Wednesday evening in the Multimedia Room (MMR) at the Schulich School of Music of McGill University. For information, please contact Sungyoung Kim (sxkiee@rit.edu) or Wieslaw Woszczyk (wieslaw@music.mcgill.ca).

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 513DEF, 1:00 P.M. TO 3:00 P.M.

Session 3pAAb

Architectural Acoustics: Balancing Risk and Innovation in Acoustical Consulting

Eric L. Reuter, Chair
Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801

Invited Papers

1:00

3pAAb1. Risk and innovation—Following on from the 2012 Knudsen Lecture, recent experience with calculated risk for the purpose of creating remarkable spaces is reviewed. Scott D. Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd., Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

Owner expectations, architectural vision, and creation of the intended aural environment often come together at a critical point in the design process. Frequently the answer that satisfies all of the requirements stretches the comfort level of all parties. When balanced properly, this element of risk frees the entire project team and reaches unexpected, but welcome, outcomes. The author will illustrate these critical decision points in several project examples, outlining the calculation in the calculated risk, and the opportunity to deepen the confidence in the consulting process through further research.

1:20

3pAAb2. Letters from the edge: Less conventional acoustical solutions. Jack B. Evans (JEAcoustics, Engineered Vib. Acoust. & Noise Solutions, 1705 West Koenig Ln., Austin, TX 78756, Evans@JEAcoustics.com)

The essence of acoustical analysis and problem solving still is source–path–receiver; yet, as standards, criteria and products evolve, solution precedents should be revalidated or invalidated, so that new, innovative approaches may be considered. While weighing the best interests of the client, should one rigidly follow procedures developed long ago for average or anticipated conditions, or might the conventional wisdom not always be correct? This paper presents a series of short case-studies where noise or vibration were treated at the source and/or along the path, but with some "twist" or variation, either from typical solution applications or owner/client response. Relevant standards, ordinances, or criteria are referenced. Where available, on-site acoustical measurements, observations, photos, or receiver experiences illustrate concepts. For the case studies presented: indicate (i) importance of the problem, (ii) method of the development used for problem solving, (iii) original contribution of the work, and (iv) conclusions. Case studies may include any of the following: standby power generators for data center, university cafeteria, and dining hall, coffee grinding and packaging shop within a grocery market, exercise/therapy floor impact above offices, air-cooled refrigeration chillers near residences, high-rise elevator equipment room adjacent to residential units, and/or high-rise domestic water booster/circulation pump room adjacent to residential unit.

1:40

3pAAb3. Design and implementation of a tuned low-frequency absorber in a residential music listening/practice room. Aaron M. Farbo and Christopher A. Storch (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Subury, MA 01776, afarbo@cavtocci.com)

The historic Bradley Mansion in Boston's Back Bay neighborhood was recently completely renovated and subdivided into multiple luxury-level condominium units built to suit to the new homeowners requirements. The historic preservation requirements imposed on the project presented challenges for designing and implementing sufficient acoustic absorption and sound isolation for the design of a new music listening and piano practice room in the lowest level residence. A custom-designed low-frequency absorber was developed to fit within an existing architectural niche at one end of the elliptical music room, and additional absorptive treatment was added to the walls to blend in with the existing finishes. Sound isolation to the neighboring condominium above the music room was improved via floor/ceiling modifications—a task made more challenging by the need to retain the terracotta structure and other historic details. Construction details and measurement results will be discussed in this case study presentation.

2:00

3pAAb4. A statistical analysis of acoustical measurement uncertainties for assemblies within multi-family dwellings in the United States. David W. Dong and John LoVerde (Veneklasen Assoc., 1711 16th St, Santa Monica, CA 90404, wdong@veneklasen.com)

Over the last several years, the authors have demonstrated that the uncertainties in the methods of acoustical testing are much larger than realized by professionals and lay people [LoVerde and Dong, J. Acoust. Soc. Am. **125**, 2629 (2009); J. Acoust. Soc. Am. **126**, 2171 (2009); J. Acoust. Soc. Am. **130**, 2355 (2011)]. Acceptance of the large uncertainty immediately raises many practical questions. How much of the uncertainty is inherent in the test procedure and how much is due to differences between laboratories, installation methods, contractors, materials, etc.? Is the data hopelessly chaotic, or is there "true value" that can be obtained by suitable data processing? How many tests are required to feel confident in the characterization of an assembly? Some rules of thumb have been developed based on our experience [LoVerde and Dong, J. Acoust. Soc. Am. **131**, 3319 (2012)], but these questions have not yet been systematically addressed before. A statistical analysis has been performed using a database of thousands of laboratory and field noise isolation tests. Results are presented that address these questions.

2:20

3pAAb5. Coping with curves in room design. Timothy Foulkes (Cavanaugh Tocci Assoc., 327 Boston Post Rd., Sudbury, MA 01776, tfoulkes@cavtocci.com)

Concave curves are pleasing to the eye, but they cause a number of different acoustic anomalies. Depending on the radius of curvature, finish material, included angle, and position relative to source and receiver, one may hear a strong return echo, a noticeable coloration of the frequency balance, a dramatic shift in the acoustic image, or extended reverberation at low frequencies. The effects of standard curved forms such as the Capital rotunda are well known. These anomalies are of little consequence in a transient space such as a lobby, but are problematic in rooms for speech and music presentation. Covering the entirety of the concave surface with sound absorbing material is not always the best solution. In most cases, the client will want to know the minimum acoustic treatment to avoid complaints. The author will present a series of case studies showing room designs with concave curves and the acoustic solutions.

Contributed Paper

2:40

3pAAb6. Building information modeling and the consultant: Managing roles and risk in an evolving design and construction process. Norman H. Philipp (Geiler & Assoc., 1840 E. 153rd Circle, Olathe, KS 66062, nphilipp@geileracoustics.com)

Paramount changes are occurring within the building and construction industries, fueled by the ever expanding abilities of computer modeling technologies. This revolution not only impacts our approach to and execution of the physical design of a building, but also the construction and the day to day management of the facilities. Current technologies have allowed the Building Information Modeling (BIM) process to replace many of the time tested design methods of the past. With this shift to new technologies also come new risks which require recognition by the acoustical consultant to ensure our evolution to meet the new paradigm of the current design environment. In this paper the importance of understanding the purpose and role of a BIM implementation/execution plan will be covered inclusive of defining the role, responsibilities, and risks associated with the acoustical consultant.

Session 3pAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Perceiving Objects II

Caroline M. DeLong, Cochair
Psychology, Rochester Inst. of Tech., 18 Lomb Memorial Dr, Rochester, NY 14623

Eduardo Mercado, Cochair
Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260

Chair's Introduction—12:55

Invited Papers

1:00

3pAB1. Categories, concepts, and calls: Auditory perceptual mechanisms and cognitive abilities across different types of birds. Allison H. Hahn, Lauren M. Guillette, Marisa Hoeschele (Psychology, Univ. of Alberta, P217 Biological Sci. Bldg., Edmonton, AB T6G2E9, Canada, ahhahn@ualberta.ca), Robert G. Cook (Psychology, Tufts Univ., Medford, MA), and Christopher B. Sturdy (Psychology, Univ. of Alberta, Edmonton, AB, Canada)

Although involving different animals, preparations, and objectives, our laboratories (Sturdy's and Cook's) are mutually interested in category perception and concept formation. The Sturdy laboratory has a history of studying perceptual categories in songbirds, while Cook laboratory has a history of studying abstract concept formation in pigeons. Recently, we undertook a suite of collaborative projects to combine our investigations to examine abstract concept formation in songbirds, and perception of songbird vocalizations in pigeons. This talk will include our recent findings of songbird category perception, songbird abstract concept formation (same/different task), and early results from pigeons' processing of songbird vocalizations in a same/different task. Our findings indicate that (1) categorization in birds seems to be most heavily influenced by acoustic, rather than genetic or experiential factors (2) songbirds treat their vocalizations as perceptual categories, both at the level of the note and species/whole call, (3) chickadees, like pigeons, can perceive abstract, same-different relations, and (4) pigeons are not as good at discriminating chickadee vocalizations as songbirds (chickadees and finches). Our findings suggest that although there are commonalities in complex auditory processing among birds, there are potentially important comparative differences between songbirds and non-songbirds in their treatment of certain types of auditory objects.

1:20

3pAB2. Reverberlocation in chickadees? Eduardo Mercado, Matthew W. Wisniewski, Brittany E. McIntosh (Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260, emiii@buffalo.edu), Lauren M. Guillette, and Christopher B. Sturdy (Dept. of Psych., Univ. of Alberta, Edmonton, AB, Canada)

Chickadee songs provide conspecifics with information about the locations of singers. Song amplitude, frequency, and reverberation all vary with distance, and it is thought that chickadees use such cues to estimate distance. The current study examined transmission of chickadee songs in an open field to assess whether other cues such as relative changes in inter-note timing or relative differences in spectral energy might also provide useful information about a singer's location. Surprisingly, the difference between direct signal energy and reverberant spectral energy provided clear indications of how far a song had traveled. Preliminary analyses suggest that this cue may be robust to variations in source level, note duration, note frequency, and transmission loss. If chickadees use this cue to judge auditory distance, then this may explain why they maintain specific spectral ratios between the notes within their songs. Specifically, the spectral spacing of notes within songs appears to be directly related to chickadee auditory filter bandwidth. We describe ranging of a singing chickadee based on the spectral profile of its songs as reverberlocation (construed as an instance of passive echolocation) because it involves comparisons between a direct signal and echoes of a signal.

Contributed Papers

1:40

3pAB3. Localization of flying insects by echolocation. Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@cs.tohoku-gakuin.ac.jp) and Takuma Takanashi (Forestry and Forest Products Res. Inst., Tsukuba, Japan)

Using the echolocation, bats can capture insects in real 3D space. Bats can accurately localize these objects from echoes by emitting the frequency modulation sound. The object's range could be estimated from delay times between the emitted sound and echoes from objects. In the case of flying insects, the echoes were influenced by Doppler shift, that is, the wing beats and flight speed. In the case of the linear frequency modulated (LFM) sound, this range accuracy was dependent on not only the frequency width of emitted sound but also the Doppler shift. It has been shown that the previous proposed model could accurately estimate each range of static objects by using the frequency modulation sound. However, it was unknown whether this model could estimate locations and movements of the flying insect. In this study, the echoes were measured from the flying insect by emitting intermittently the LFM sounds. At the same time, the movements of the insects were measured by the camera. The time-frequency pattern

were computed by using the convolution of the chirplet filters. It was examined that the the insect's positions were estimated by extracting the onset from the time-frequency pattern. [Research supported by JST, CREST.]

2:00

3pAB4. Analysis of Northern bottlenose whale pulses and associated reflections recorded from the Gully Marine Protected Area. Bruce Martin (Halifax, JASCO Appl. Sci., 32 Troop Ave., Ste. 202', Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com) and Hilary Moors-Murphy (Bedford Inst. Of Oceanogr., Dept. of Fisheries and Oceans, Dartmouth, NS, Canada)

The Gully Marine Protected Area (MPA) is a large submarine canyon at the edge of the Scotian Shelf, south of Nova Scotia. A resident population of northern bottlenose whales are known to occur in the Gully throughout the year, and the canyon provides important foraging grounds for the population. Bottom-mounted Autonomous Multichannel Acoustic Recorders (AMAR) were deployed in the Gully for ten days in in March 2010 (sampling rate = 375 ksps) and two days in October 2011 (sampling rate = 128 ksps). Biosonar pulses produced northern bottlenose whales (likely used to echolocate prey) were recorded consistently throughout these AMAR deployments. The swept FM characteristics of the northern bottlenose whale pulses recorded were consistent over both years, and both data sets contained clear pulse reflections from bottom clutter or prey targets. In this paper, we provide a description of the northern bottlenose whale pulses recorded in the Gully and make recommendations on short-time Fourier transform parameters for analysis of the pulses. A description of the pulse reflections is also provided, based on analysis of the reflection patterns using short-time Fourier transforms and by matched filtering with the direct arrival from the whales.

2:20

3pAB5. Preliminary results from collaborative referring to impulsive sonar sounds. Charles F. Gaumond (Acoust. Div., Naval Res. Lab., Code 7162, 4555 Overlook Ave. SW, Washington, DC 20375, charles.gaumond@nrl.navy.mil), Derek Brock, and Christina Wasylyshyn (Information Technol. Div., Naval Res. Lab., Washington, DC)

A recent experiment is described wherein pairs of listeners (a "director" and a "matcher") collaboratively refer to eight-element sets of impulsive sonar sounds, which are the same, but ordered differently for each listener. The sounds in a given set are privately displayed on each listener's computer as a line of blank cards that play a corresponding sound when clicked and can be rearranged from left to right. The listeners' task is to move the matcher's sounds into the same order as the director's. Through conversation, the listeners work out how to verbally characterize the sounds and develop a shared vocabulary. This vocabulary is presented for selected participants and is shown to generally consist of names, actions, and properties of familiar, everyday auditory events. In general, these references function as classes and descriptors. Classes correspond to causal categories that are aurally analogous to (i.e., homophonic with) the acoustic origins of the impulsive sonar sounds. Similarly, descriptors, distinguish between the properties of signal processing features that are appropriate to impulsive sounds within a given category. [Research funded by the Office of Naval Research.]

2:40

3pAB6. An auditory perception of changes in the intensity of pulses, presented in complicated sound complex. Liudmila K. Rimskaya-Korsakova (Lab. of Bioacoustics, N.N. Andreev Acoust. Inst., Shvernika. st 4, Moscow 117036, Russian Federation, lkrk@mail.ru)

The auditory system of humans and animals is able to detect and discriminate high frequency pulses in a complicated sound complex. The purpose of the work was to find new examples of a facilitation the discrimination of intensity (or level, defined by a peak amplitude) of a test pulses, presented under composite masking conditions, and to find the possible mechanisms underlying the facilitation. The discrimination tended to deteriorate if the test pulse was presented through 50 ms after a pulse's masker. However, if the test pulse was mixed with stationary noise, the beginning of which coincided with the end of the pulse's masker, discrimination became better. The noise levels, at which facilitation occurred, depended on amplitudes of both the test pulses and the pulse's maskers. When the duration of the noise was less than 50 ms, an auditory adaptation could not influence on the discrimination. The reason of the facilitation could be in the temporal redistribution of the auditory nerve fibers activities, which occurred at coding of the complicated sound complex "pulse's masker - test pulse - stationary noise."

WEDNESDAY AFTERNOON, 5 JUNE 2013 510D, 1:20 P.M. TO 2:40 P.M.

Session 3pAO

Acoustical Oceanography: Ocean Acoustical Tomography

Lora J. Van Uffelen, Chair
Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96815

Contributed Papers

1:20

3pAO1. Influence on sound spread considering the flow velocity in a horizontal layer media. Yang Song and Zhenqi Zhao (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin 150001, China, syang@163.com)

The marine currents could often influence sound propagation underwater. The traveling time of sound ray is distinctly effected by the velocity of current in some conditions. So it will be made certain errors in seeking eigen rays and inversing sound speed profile if the velocity of the current is ignored. In order to improve the computation accuracy of sound ray model, a sound ray model of horizontal layer is induced in which the media flow is considered. Eigen rays are searched and their traveling time is calculated by the ray model. It is also discussed that the velocity of flow media influences ray trace and traveling time. An average sound speed profile measured under a shallow water is cited to calculate the eigen rays. The differences of sound ray are given under two kinds of condition which the velocity of current is considered and not considered. The computation results show that the sound ray trace is changed indistinctively under small Mach number condition, but the traveling time of eigen rays is fluctuated obviously. The fluctuation of ray traveling time is bigger according to the larger Mach number and the longer spread distance. The results of study will provide some help in the inversion of ocean acoustic tomography.

1:40

3pAO2. Time-angle ocean acoustic tomography using sensitivity kernels: The forward problem. Florian Aulanier, Barbara Nicolas (Gipsa lab, Institut Polytechnique de Grenoble, 11 rue des Mathématiques, Grenoble Campus BP46 F, SAINT MARTIN D'HERES Cedex 38402, France, Florian.Aulanier@gipsa-lab.grenoble-inp.fr), Philippe Roux (ISTerre, Observatoire des Sci. de l'Univers, Université de Grenoble, Grenoble, France), and Jérôme I. Mars (Gipsa Lab., Institut Polytechnique de Grenoble, Grenoble, France)

Broadband acoustic signals around 1 kHz propagate through shallow water oceanic waveguides of ~100 m in depth and ~2 km in range as multiple ray-like wavefronts. These acoustic arrivals can be characterized by the following observables: travel-time (TT), direction-of-arrival (DOA), and direction-of-departure (DOD). By applying double-beamforming on the point-to-point signals recorded between two source-receiver arrays, the acoustic contribution of each arrival can be separated from the multi-reverberated data and the TT, DOA, and DOD observable variations are accurately measured. This study deals with the use of time-angle sensitivity kernels (TASK) to estimate the observable variations induced by sound speed perturbations in the waveguide. This approach is based on the first order Born approximation and takes into account the finite-frequency effects associated with wave propagation. The robustness the TASK approach is analyzed and compared to numerical parabolic equation simulations involving different sound speed perturbations. For example, parameters such as the perturbation location, the value and shape of the perturbation in the waveguide are modified. The combination of several perturbations and the influence of the source-receiver array apertures on the TT, DOA, and DOD estimates are also studied.

2:00

3pAO3. Time-angle ocean acoustic tomography using sensitivity kernels: Numerical and experimental inversion results. Florian Aulanier, Barbara Nicolas (Gipsa Lab, Institut Polytechnique de Grenoble, 11 rue des Mathématiques, Grenoble Campus BP46 F, SAINT MARTIN D'HERES Cedex 38402, France, Florian.Aulanier@gipsa-lab.grenoble-inp.fr), Philippe Roux (ISTerre, Observatoire des Sci. de l'Univers, Université de Grenoble, Grenoble, France), Romain Brossier (ISTerre, Observatoire des Sci. de l'Univers, Université de Grenoble, Saint Martin D'Heres, France), and Jérôme I. Mars (Gipsa Lab, Institut Polytechnique de Grenoble, Grenoble, France)

In shallow water acoustic tomography, broadband mid-frequency acoustic waves (1 to 5 kHz) follow multiple ray-like paths to travel through the ocean. Travel-time (TT) variations associated to these raypaths are classically used to estimate sound speed perturbations of the water column using the ray theory. In this shallow water environment, source and receiver arrays, combined with adapted array processing, provide the measurement of directions-of-arrival (DOA) and directions-of-departure (DOD) of each acoustic path as new additional observables to perform ocean acoustic tomography. To this aim, the double-beamforming technique is used to extract the TT, DOA, and DOD variations from the array-to-array acoustic records. Besides, based on the first order Born approximation, we introduce the time-angle sensitivity kernels to link sound speed perturbations to the three observable variations. This forward problem is then inversed by the maximum *a posteriori* method using both the extracted-observable variations and the proposed sensitivity kernels. Inversion results obtained on numerical data, simulated with a parabolic equation code, are presented. The inversion algorithm is performed with the three observables separately, namely TT, DOA, and DOD. The three observables are then used jointly in the inversion process. The results are discussed in the context on ocean acoustic tomography.

2:20

3pAO4. Toward subsurface positioning of gliders using fixed acoustic tomography sources. Lora J. Van Uffelen, Eva-Marie Nosal, Bruce M. Howe, Glenn S. Carter (School of Ocean and Earth Sci. and Technol., Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96815, loravu@hawaii.edu), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Kevin D. Heaney, Richard L. Campbell (OASIS, Inc., Fairfax, VA), and Patrick S. Cross (OASIS, Inc., Honolulu, Hawaii)

Acoustic Seagliders can be positioned precisely using GPS at the surface, but are underwater and unable to utilize GPS for up to 9 h at a time as they dive to depths of up to 1000 m. During this time, a kinematic model estimates the position of the glider. Four acoustic Seagliders were deployed in the Philippine Sea November 2010–April 2011, and received transmissions from five broadband acoustic tomography sources moored in the region. Over 2000 acoustic receptions were recorded at ranges up to 700 km from the moored sources. Measured acoustic arrival peaks were unambiguously associated with ray arrivals predicted using the model-estimated glider position at the time of reception and a mean sound-speed profile. Estimates of source-receiver range uncertainty were calculated from statistics of travel-time offsets between the measured arrivals and the eigenray dispersion patterns. The uncertainty in range between the source and the modeled glider position during a dive is estimated to be 639 m (426 ms) rms disregarding the effects of ocean sound-speed variability, which are anticipated to be on the order of 70 ms rms.

Session 3pBA

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Award Poster Session

Kevin J. Haworth, Chair
Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD$500 for first prize, USD$300 for second prize, and USD$200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee. Below is a list of students competing, with their abstract numbers and titles listed. Full abstracts can be found in the oral session associated with the abstract numbers.

All entries will be on display and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

1aBA3. A contrast source inversion method for breast cancer detection. Student author: N. Ozmen-Eryilmaz

1aBA7. Electromagnetic hydrophone for high-intensity focused ultrasound measurement. Student author: Pol Grasland-Mongrain

1pBAa4. Sound speed estimation in single cells using the ultrasound backscatter power spectrum. Student author: Eric M. Strohm

1pBAa6. An analysis of the acoustic properties of the cell cycle and apoptosis in MCF-7 cells. Student author: Maurice M. Pasternak

1pBAb2. Acoustic and optical characterization of targeted ultrasound contrast agents. Student author: Camilo Perez

1pBAb4. Radiation for bubble contrast agents in inhomogeneous media. Student author: Chrisna Nguon

1pBAb5. Temporal evolution of subharmonic emissions from a lipid-encapsulated contrast agent. Student author: Himanshu Shekhar

1pBAb6. Simulations of transcranial passive acoustic mapping with hemispherical sparse arrays using computed tomography-based aberration corrections. Student author: Ryan Jones

1pBAb11. Passive acoustic mapping of magnetic microbubbles in an *in vitro* flow model. Student author: Calum Crake

1pBAb12. A two-component speckle model for detection of microbubble signals in linear contrast-enhanced ultrasonography. Student author: Matthew R. Lowerison

2aBA9. Ultrasonic atomization: A mechanism of tissue fractionation. Student author: Julianna C. Simon

2pBAa7. Investigating the sensitivity of microbubble acoustic response for biosensing applications. Student author: Caroline J. Harfield

2pBAa8. Modeling of microbubbles pushed through clots via acoustic radiation force. Student author: Ascanio Guarini

2pBAa11. Effect of shell thickness on sound propagation through encapsulated bubbles: A resonator approach. Student author: Craig N. Dolder

2pBAb5. Validation of three-dimensional strain tracking by volumetric ultrasound image correlation in a pubovisceral muscle model. Student author: Anna S. Nagle

2pBAb6. Measurement of surface acoustic wave in soft material using swept-source optical coherence tomography. Student author: Yukako Kato

3aBAa5. Small interfering ribonucleic acid delivery with phase-shift nanoemulsions. Student author: Mark T. Burgess

3aBAb4. Improving the acousto-optic detection of high-intensity focused ultrasound lesions. Student author: Matthew T. Adams

3aBAb6. The origins of nonlinear enhancement in *ex vivo* tissue during high intensity focused ultrasound ablation. Student author: Edward Jackson

4aBA5. Investigation on the inertial cavitation threshold of micro-bubbles. Student author: Xiasheng Guo

5aBAa3. Ultrasonic assessment of the *in vitro* biomechanical stability of a dental implant. Student author: Romain Vayron

5aBAa6. Developement and validation of resonant ultrasound spectroscopy for the measurement of cortical bone elasticity on small cylindrical samples. Student author: Simon Bernard

5aBAa8. Computational simulations of time of flight and attenuation of first arriving signal from healing process of diaphyseal femur fractures. Student author: Paulo Tadeu Rosa

5aBAb1. The effect of boundary proximity on the fundamental and subharmonic emissions from individual microbubbles at higher frequencies. Student author: Brandon Helfield

5aBAb2. Bifurcation structure of the ultrasonically excited microbubbles undergoing buckling and rupture. Student author: Amin Jafari Sojahrood

5aBAb5. Temporal and spatial characteristics of nonlinear acoustic field generated by an extracorporeal shockwave therapy device: Modeling and measurements. Student author: Maria Karzova

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 512AE, 12:55 P.M. TO 3:00 P.M.

Session 3pEA

Engineering Acoustics: Computational Methods in Transducer Design, Modeling, Simulation, and Optimization III

Daniel M. Warren, Chair
Knowles Electronics, 1151 Maplewood Dr, Itasca, IL 60134

Chair's Introduction—12:55

Contributed Papers

1:00

3pEA1. Coupling elastic-poroelastic material in structure-borne sound modeling. Katherina Rurkowska and Sabine Langer (Institut für Angewandte Mechanik, Technische Universität Braunschweig, Spielmannstraße 11, Braunschweig, Niedersachsen 38106, Germany, infaminfo@tu-braunschweig.de)

Porous materials are widely used in noise reduction applications. To minimize the external noise produced by aircraft propeller drives, porous materials are implemented. As a part of the project *Sonderforschungsbereich 880* "Fundamentals of High Lift for Future Civil Aircraft," porous surfaces are used in the High-lift configuration to mitigate the flow noise and to influence the structure-borne sound. In order to model the performance of the applied poroelastic material, an approach coupling a poroelastic material with an elastic structure using Finite Element Method is presented. The Biot's theory is used to model the poroelastic material. The aim of this work is to simulate the effect of the entry and transmission of the structure-borne sound into the poroelastic medium. An example of the implemented model shows the plausibility of presented approach.

1:20

3pEA2. Numerical investigation of the functionally graded materials by the interaction of the plate guided waves with discontinuities and cracks. Farouk Benmeddour, Emmanuel Moulin, and Jamal Assaad (OAE Dept., IEMN, CNRS UMR 8520, Univ. of Valenciennes and Hainaut Cambrésis, Campus Mont Houy, Valenciennes 59313, France, farouk.benmeddour@univ-valenciennes.fr)

This work intends to give a better comprehension of the guided wave interactions with damage in a functionally graded material (FGM). The propagation and interactions of plate guided waves with discontinuities in a FGM composed of ceramic and metal mixture are investigated. For this purpose, a two dimensional finite element (FE) method is used to analyze the near field surrounding the damage. Then, expansion of the solution into sums of guided modes enables the determination of the reflection and transmission coefficients of each existent mode. The determination of the modal features is ensured by the way of the so called semi-analytical finite element (SAFE) method applied to the one dimensional inlet and outlet cross-sections. The latter has the benefit to study an arbitrary shape-like damage in an infinite structure having the same shape by translation in the propagation direction in a fast and efficient way. Results are obtained by solving the global system of the 2D hybrid FE-SAFE method. Different symmetrical and asymmetrical notches are studied and so for cracks. Results are achieved and discussed for a FGM and compared to those obtained for an isotropic material.

1:40

3pEA3. Generalized Debye series expansion to improve the non-destructive testing and health monitoring of cylindrical structures by guided waves. Slah Yaacoubi (Institut de Soudure, Yutz, France), Marc Deschamps, Eric Ducasse (I2M, Bordeaux, France), Laurent Laguerre (IFSTTAR, Bouguenais, France), Weina Ke Yaacoubi (Institut de Soudure, Yutz, France), Peter McKeon (Georgia Insitute of Technol., GTL, Metz, France, peter.mckeon@gatech.edu), Salah Ramadan (Institut de Soudure, Yutz, France), and Nico F. Declercq (Georgia Insitute of Technol., Metz, France)

Many structures in civil engineering notably bridges and nuclear power plants must be regularly, strictly, and carefully tested to avoid any human or environmental catastrophe. Among the NDT techniques, which can be

applied, ultrasonic guided waves are a good candidate to monitor bars and cables. However, its multimodal and dispersive behaviors can limit its performances. Theoretical modeling is sometimes needed to understand the behavior of the traveling waves to improve the testing/monitoring and made a right *in-situ* decision. The aim of this paper is to derive the space-time velocity field in a cylindrical waveguide perfectly embedded in an infinite solid matrix and generated by an inside bounded beam. This beam is generated by an off-axis source. Vector Hankel transform and Fourier series are combined to decompose the inside field into infinity of elementary cylindrical waves propagating in radial direction and planar waves propagating in axial direction. Global resolution method and Generalized Debye series expansion are used both to calculate the 3D global cylindrical reflection/transmission coefficients. The method is demonstrated through a steel bar embedded in cement matrix. Simulated frequency–wavenumber diagrams show that the embedding material acts like filter for different frequency ranges. Other results will be presented.

2:00

3pEA4. The effect of a middle layer on ultrasonic wave propagating in a three-layer structure. Raymond B. Mabuza and Ngeletshedzo Netshidavhini (NDT and Phys., Vaal Univ. of Technol., Private Bag X021, Vanderbijlpark, Gauteng 1900, South Africa, raymondm@vut.ac.za)

In this paper, the focus of attention is on the effect of an elastic middle layer on the propagation behavior of ultrasonic waves. Systematic parametric studies are conducted to quantify the effects of the middle layer upon the ultrasonic wave propagation, including its thickness and acoustic impedance. We treat this problem analytically and numerically. The three-layer structure is also used to investigate the influence of the imperfect interfaces between two outer layers and a middle layer on the ultrasonic wave propagation. The theoretical analysis considers successive reflections of waves radiated by the transmitting transducer. The output signal is a superposition of successive reflections. Our results demonstrate clearly that there is significant influence of the middle layer in our three-layer problem. Various aspects of our approach are discussed and numerical examples are used to illustrate the suitability of our approach. Some details about the numerical methods employed are also given. The results are presented and discussed.

2:20

3pEA5. Comparison of finite element models simulating the interaction of ultrasonic guided waves with sites of disbonding in composites. Peter McKeon (Mech. Eng., Georgia Insitute of Technol., 2 rue Marconi, Metz 57070, France, peter.mckeon@gatech.edu), Slah Yaacoubi (Institut de Soudure, Yutz, France), Nico F. Declercq (Mech. Eng., Georgia Insitute of Technol., Metz, France), and Salah Ramadan (Institut de Soudure, Yutz, France)

Disbonding in composite structures is a serious defect which can dramatically reduce the structures' life, and can lead to catastrophic failure. To avoid this, non-destructive testing or structural health monitoring techniques are needed. One such technique involves ultrasonic guided waves, which has recently found use in this field thanks to its ability to inspect in non-accessible areas and over long distances. Numerical models are often used because they can help explain experimental results, and offer the ability to simulate different damage scenarios, predicting results with less cost than experiments. In this work, sites of disbonding between an orthotropic composite plate and an isotropic polyamide plate were modeled via the finite element method. Several methods for modeling the damage site are employed, and results are compared with experimental ones. Model types range from the introduction of a geometrical void at the interface boundary to addressing boundary conditions between adjacent layers. The amount of mode conversion after interaction with the damage site is used to evaluate the validity of each model type. Results are discussed in terms of computational effort and accuracy in predicting true physical behavior.

2:40

3pEA6. Energy flux streamlines versus acoustic rays for modeling interaction with rigid boundaries: near field of sound from a circular loudspeaker. Cleon E. Dean (Physics, Georgia Southern Univ., P.O. Box 8031, Math/Phys. Bldg., Statesboro, GA 30461-8031, cdean@georgiasouthern.edu) and James P. Braselton (Mathematical Sci., Georgia Southern Univ., Statesboro, GA)

Sound emitted by a circular loudspeaker can be treated as equivalent to a plane wave diffracted by a circular aperture in a rigid, sound absorbing screen. Axial symmetry leads one to expect constructive interference along the symmetry axis in the near field (the Poisson-Arago spot). An energy flux streamline model was developed to help visualize this and other features of the near sound field. The model is used to draw out similarities and differences between energy flux streamlines and acoustic rays, particularly in the transition to the far field.

WEDNESDAY AFTERNOON, 5 JUNE 2013 510C, 1:20 P.M. TO 2:20 P.M.

Session 3pED

Education in Acoustics: Take 5's

Andrew Morrison, Chair
Natural Sci. Dept. Joliet Junior College, 1215 Houbolt Rd, Joliet, IL 60431

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign-up for two consecutive slots.

Session 3pMU

Musical Acoustics and Psychological and Physiological Acoustics: Perception and Orchestration Practice

Stephen McAdams, Cochair
Music Res., McGill Univ., 555 Sherbrooke St. W., Montreal, QC H2W 1S2, Canada

Punita G. Singh, Cochair
Sound Sense, 16 Gauri Apartments, 3 Rajesh Pilot Ln., New Delhi 110011, India

Invited Papers

1:00

3pMU1. Timbre as a structuring force in music. Stephen McAdams (Schulich School of Music, McGill Univ., 555 Sherbrooke St. W., Montreal, QC H2W 1S2, Canada, smc@music.mcgill.ca)

Most of the music we enjoy uses the musical qualities of different instruments to create specific perceptual and emotional effects that composers sculpt over time. Timbre is the auditory attribute that distinguishes different instruments. Research on timbre perception has demonstrated that it is multifaceted and contributes in many ways to the perceptual organization of musical structures. The art of structuring music with timbre is orchestration. A survey of orchestration treatises reveals the dearth of underlying theory, in sharp contrast to other traditional areas such as harmony and counterpoint, which have long theoretical traditions. We seek to develop a theoretical ground for orchestration practice starting with the structuring role that timbre can play in music. Many aspects of musical structuring are achieved by auditory scene analysis, the perceptual processes that result in unified events, integrated streams of events, groups of events segmented into phrases and sections, and larger-scale units extended over time that we call orchestral gestures. The roles that timbre plays in the manifestation of these principles in orchestration practice will be considered as potential elements of a theory of orchestration. How such principles might be incorporated into computer-aided orchestration systems and computer-aided orchestral rendering systems will also be examined.

1:20

3pMU2. Acoustic and musical features of emotional response to orchestral gestures. Meghan Goodchild (CIRMMT and McGill Univ., 4515 rue Drolet, Unit 6, Montreal, QC H2T 2G1, Canada, meghan.goodchild@mail.mcgill.ca)

Recent empirical research indicates the impact of prominent changes in instrumentation on the listening experience: several studies suggest that timbral changes evoke music-induced emotions. However, orchestration remains an underdeveloped area of music theory. A model of orchestral gestures defined by changes in instrumentation in terms of time course (gradual or sudden) and direction (addition or reduction) is presented. An exploratory behavioral study that tested the perceptual relevance of orchestral gestures on listeners' continuous ratings of emotional intensity was conducted. We demonstrate a new type of visualization that illustrates the relative textural density of each instrument family over time combined with other time-varying parameters extracted from the signal (loudness, spectral centroid, tempo, and roughness) and calculated from the score (instrumental texture, melodic range, and attack density). In addition to quantitative and qualitative comparison of similar orchestral gestures across pieces, we use this method to study the interaction of specific instrumentation changes and other musical parameters. Through discussion of the visualizations, we highlight relationships between the perceptual and musical/acoustical dimensions and quantify elements of the temporality of these experiences.

1:40

3pMU3. Perception and orchestration of melody, harmony, and rhythm on instruments with "chikari" strings. Punita G. Singh (Sound Sense, 16 Gauri Apartments, 3 Rajesh Pilot Ln., New Delhi 110011, India, punita@gmail.com)

The use of "chikari" strings on instruments such as the sitar and sarod manifests principles of Auditory Scene Analysis in creating a harmonic reference, melodic contrast, and rhythmic accompaniment. Unlike the principal "baj" strings on which the main melody is played, or resonant "tarb" strings that reinforce volume, the "chikari" strings are sounded at strategic points in performance to provide a drone, add texture, outline chords, mark rhythmic positions, and keep tempo. Listening and transcription experiments conducted with recordings of interleaved notes played on "chikari" and "baj" strings validate how differences in their timbre and tuning help to keep them perceptually apart while forming more coherent patterns based on timbre similarity and pitch proximity. Such grouping and segregation can affect the perception of temporal order, maintain the illusion of melodic continuity and in some cases of virtual polyphony. These observations add to the growing body of evidence supporting the role of timbre as a structural dimension of music and illustrate how a single instrument can bring about orchestral effects via the strategic use of devices such as "chikari" strings.

Contributed Papers

2:00

3pMU4. Predicting blend between orchestral timbres using generalized spectral-envelope descriptions. Sven-Amin Lembke (Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., 555 Sherbrooke St. West, Montreal, QC H3A 1E3, Canada, sven-amin.lembke@mail.mcgill.ca), Eugene Narmour (Dept. of Music, Univ. of Pennsylvania, Philadelphia, PA), and Stephen McAdams (Ctr. for Interdisciplinary Res. in Music Media and Technol. (CIRMMT), Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Composers rely on implicit knowledge of instrument timbres to achieve certain effects in orchestration. In the context of perceptual blending between orchestral timbres, holistic acoustical descriptions of instrument-specific traits can assist in the selection of suitable instrument combinations. The chosen mode of description utilizes spectral-envelope estimates that are acquired as pitch-invariant descriptions of instruments at different dynamic markings. Prominent local spectral-envelope traits, such as spectral maxima or formants, have been shown to influence timbre blending, involving frequency relationships between local spectral features, their prominence as formants, and constraints imposed by the human auditory system. We present computational approaches to predict timbre blend that are based on these factors and explain around 85% of the variance in behavioral timbre-blend data. Multiple linear regression is employed in modeling a range of behavioral data acquired in different experimental investigations. These include parametric investigations of formant frequency and magnitude relationships as well as arbitrary combinations of recorded instrument audio samples in dyads or triads. The cataloguing of generalized acoustical descriptions of instruments and associated timbre-blend predictions for various instrument combinations could serve as a valuable aid to orchestration practice in the future.

2:20

3pMU5. Timbre saliency vs. timbre dissimilarity – What is the relationship? Song Hui Chon and Stephen McAdams (CIRMMT, McGill Univ., 3484 Rue Durocher #401, Montreal, QC H2X 2E4, Canada, songhui.chon@mail.mcgill.ca)

We have proposed the notion of timbre saliency as the attention-capturing quality of timbre. The definition of saliency requires an object to stand out with respect to its surroundings, implying dissimilarity between the object and its neighbors. What then might be the relationship between timbre saliency and timbre dissimilarity? A classic timbre dissimilarity experiment and a timbre saliency experiment were carried out with 20 participants on the same set of stimuli. Multidimensional scaling revealed a two-dimensional dissimilarity space. Using the features obtained from the Timbre Toolbox [Peeters *et al*., J. Acoust. Soc. Am. **130**, 2902–2916 (2011)], the first dimension shows a high correlation with spectral centroid [$r(13) = 0.845$, $p < 0.0001$] and spectral spread [$r(13) = 0.855$, $p < 0.0001$], both based on the ERB-FFT model spectrum, and the second with the attack time [$r(13) = -0.692$, $p = 0.004$] and power spectral crest [$r(13) = 0.732$, $p < .005$]. This confirms spectral centroid and attack time as two major acoustic correlates of timbre dissimilarity. The saliency dimension shows a moderate correlation with the second dimension [$r(13) = 0.578$, $p = 0.0241$] but not with the first dimension [$r(13) = 0.182$, $p = 0.517$], suggesting that the saliency might be more related to the temporal characteristics of timbre.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 511BE, 1:00 P.M. TO 2:20 P.M.

Session 3pNSa

Noise, ASA Committee on Standards, Engineering Acoustics, and Structural Acoustics and Vibration: Wind Turbine Noise II

Nancy Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118

Paul Schomer, Cochair
Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821

Sheryl Grace, Cochair
Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215

Contributed Papers

1:00

3pNSa1. Wind farm—Long term noise and vibration measurements. Martin Meunier (Environment, SNC-Lavalin, 2271, boul. Fernand-Lafontaine, Longueuil, QC J4G2R7, Canada, martin.meunier@snclavalin.com)

Most of the energy produced in Quebec comes from renewable sources. The concept of wind energy emerged in the late 1990's and has since become a complementary source of energy alongside hydroelectricity. Wind farms are generally seen as a good sustainable way to produce energy. However, they are not implemented without some impact on the environment. SNC-Lavalin Environment has performed many surveys in recent years for wind farm projects, including noise measurements both before and after their commissioning. This presentation will give an overview of one such project where long term noise and vibration measurements where conducted. Vibration measurements, as well as outdoor, indoor, and low frequencies noise measurements were completed both with and without the wind turbines in operation. Data will be presented showing different problems encountered in the analysis phase. For example, multiple intermittent and non-steady noise sources were present during measurement (wind turbines, car pass-bys, wind in the trees, human activities). Methods used to overcome these obstacles will be discussed (use of statistical parameters, linear regression), and the effect of the wind turbine operation on the noise level (including low frequencies) and vibration level will be presented.

1:20

3pNSa2. RoBin - A one-man measurement system for standard acoustic emission measurement according to IEC 61400-11. Daniel Vaucher de la Croix (ACOEM, 200 Chemin des Ormeaux, Limonest 69578, France, daniel.vaucherdelacroix@acoemgroup.com) and Timo Klaas (WOLFEL MESSSYSTEME, Höchberg, Germany)

Wind turbines are built at more and more locations—which makes their noise emission an important subject. The international standard IEC 61400-11 and the German Technische Richtlinie fur Windenergieanlagen, Teil of the FGW were set up in order to unify the evaluation of noise emission. Measurement of noise emission according to these standards is linked to formidable challenges, especially for the installation of testing equipment and evaluation of data. After a short reminder on the ISO 61400 standard, the proposed paper will discuss the details of operational constraints linked with on-site measurements and how modern communication technologies help in an easy system deployment and most efficient operation for the benefits of its users.

1:40

3pNSa3. Building integrated wind turbines—A pilot study. Ben Dymock (The Acoust. Group, Dept. of Urban Eng., London South Bank Univ., 12 Deans Close, Amersham HP6 6LW, United Kingdom, dymockb@lsbu.ac.uk) and Stephen Dance (The Acoust. Group, Dept. of Urban Eng., London South Bank Univ., London, United Kingdom)

The current planning guidance in London requires that all new or refurnished large buildings should include 20% renewables. As part of a study on urban wind a pilot investigation based on the building integrated wind turbines on the skyscraper Strata Tower in London will be monitored for acoustics, vibration, anemometry and electrical generation. Strata Tower is a 150 m building in an urban location with three 19 kWe turbines in a specially designed venturi housing. The effect of the wind turbines on residents, the local community, and the building structure will be assessed and reported.

2:00

3pNSa4. Assessment of annoyance due to wind turbine noise. Malgorzata Pawlaczyk-Luszczynska, Adam Dudarewicz, Kamil Zaborowski, Malgorzata Zamojska, and Malgorzata Waszkowska (Dept. of Physical Hazards, Nofer Inst. of Occupational Medicine, 8, Sw. Teresy str., Lodz 91-348, Poland, mpawlusz@imp.lodz.pl)

The overall aim of this study was to evaluate the perception and annoyance of noise from wind turbines in populated areas of Poland. The study group comprised 378 subjects. All subjects were interviewed using a questionnaire developed to enable evaluation of their living conditions, including prevalence of annoyance due to noise from wind turbines, and the self-assessment of physical health and well-being. In addition, current mental health status of respondents was assessed using Goldberg General Health Questionnaire GHQ-12. For areas where respondents lived, A-weighted sound pressure levels (SPLs) were calculated as the sum of the contributions from the wind power plants in the specific area. It has been shown that the wind turbine noise at the calculated A-weighted SPL of 30−50 dB was perceived as annoying outdoors by about one third of respondents, while indoors by one fifth of them. The proportions of the respondents annoyed by the wind turbine noise increased with increasing A-weighted sound pressure level. Subjects' attitude to wind turbines in general and sensitivity to landscape littering was found to have significant impact on the perceived annoyance. Further studies are needed, including a larger number of respondents, before firm conclusions can be drawn.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 511CF, 1:00 P.M. TO 2:40 P.M.

Session 3pNSb

Noise: Noise Barriers

Murray Hodgson, Chair
UBC, 2206 East Mall, Vancouver, BC V6T1Z3, Canada

Contributed Papers

1:00

3pNSb1. Effect of scaling laws for noise reduction optimization of wind fences. JohnPaul R. Abbott, Richard Raspet, and Jeremy Webster (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., Rm. 1044, Oxford, MS 38677, johnpaul.abbott@gmail.com)

This paper will report on an investigation to increase noise reductions at low wavenumbers for a large windscreen enclosure described in two earlier papers [J. Acoust. Soc. Am. **129**, 2445 (2011); J. Acoust. Soc. Am. **132**, 2048 (2012)] by first doubling its height and then doubling its diameter. According to the scaling laws developed for small windscreens, windscreens of similar shape but differing size will have nearly identical reductions for scaled wavenumbers; therefore the wavenumbers at which noise reduction for a windscreen occurs is dependent on its size and by increasing either its height or diameter, or both, reductions should shift to lower wavenumbers. Such a shift was observed when the screen's height was doubled. Also, when scaled to height, the measured reductions for the single and double height windscreens were found to match closely, with 6 dB of reduction and greater for wave numbers between 5 and 30 1/m and max reductions of 10–13 dB.

1:20

3pNSb2. Designing canopies to improve downwind shielding at various barrier configurations at short and long distance. Timothy Van Renterghem and Dick Botteldooren (Information Technol., Ghent Univ., Sint-Pietersnieuwstraat 41, Gent 9000, Belgium, tvrenter@intec.ugent.be)

The positive effect of a row of trees to improve downwind shielding in the acoustic shadow zone behind a noise wall has been shown before by means of a wind tunnel experiment, a field study and by numerical simulations. This research focused at a rather short distance in downwind direction, where important recovery of the shielding lost by screen-induced wind refraction was observed. However, opposite effects are possible at longer distance. This can be explained by shifts in the zones with strong (positive) gradients in the horizontal component of the wind speed. Leaving a gap between the barrier top and canopy bottom helps reducing these negative effects at longer distance, and results in a generally optimized performance downwind. Trees behind noise walls at either side of the source lead to a full canceling of wind effects at short distance, but to strong negative effects at longer distances downwind. Trees as windbreaks seem to be especially useful near single, vertically erected noise walls. Near steep berms, no net effect of trees is predicted. The design rules presented in this paper are derived based on numerical calculations with a previously validated CFD-FDTD-PE model.

1:40

3pNSb3. A review of road traffic barriers for low frequency noise. Samaneh M. Fard, Nicole Kessissoglou, Stephen Samuels (School of Mech. and Manufacturing Eng., The Univ. of New South Wales, 11/127A, Barker St., Sydney, NSW 2032, Australia, fardsmb@gmail.com), and Marion Burgess (School of Eng. and Information Technol., The Univ. of New South Wales, Canberra, NSW, Australia)

Australia relies heavily on road transport due to its large area and low population density in many parts of the country. Trucks and heavy vehicles are commonly used for road freight. In addition to their normal vehicle brakes, heavy vehicles are typically fitted with release engine brakes which operate by causing the engine to act as a compressor when braking. Compression braking generates a distinct low frequency rumble that can heard at large distances and is a major source of community annoyance reactions against the heavy vehicle industry. Noise from compression brakes is an ongoing cause of complaint from many Australian residents, particularly in rural areas and at night-time. Noise barriers can be used to reduce the spread of general traffic noise and their effectiveness is determined by many factors. This paper presents a review of barriers optimized for road traffic noise and the frequency ranges at which the various barrier designs are most efficient, with a view to selecting the barriers that may be more effective at reducing the low frequency noise from compression brakes.

2:00

3pNSb4. *In-situ* measurements of sound reflection and sound insulation of noise barriers: Validation by means of signal-to-noise ratio calculations. Massimo Garai and Paolo Guidorzi (DIN, Univ. of Bologna, Viale Risorgimento 2, Bologna 40136, Italy, massimo.garai@unibo.it)

After some years from its first release, the CEN/TS 1793-5 European standard for *in-situ* measurement of sound reflection and airborne sound insulation characteristics of noise barriers has been significantly enhanced and validated in the frame of the EU funded QUIESST project. The procedure, based on impulse response measurements near the noise barrier and in the free field, is robust and easily applicable but much attention must be paid when: (i) applying the signal subtraction technique to get the reflected signal component and (ii) extracting the transmitted component, especially measuring highly insulating noise barriers. In both cases, it is essential to avoid a poor signal-to-noise ratio of the critical part of the impulse response. In the frame of the QUIESST project specific quality criteria, applicable on site, have been introduced in order to check and validate the result. These criteria are rigorously described here for the first time and illustrative examples are presented.

2:20

3pNSb5. Compliance and vegetated-barrier acoustical testing in a purpose-built sound-transmission suite. Murray Hodgson, Shira Daltrop (Acoust. and Noise Res. Group, Univ. of British Columbia, 2206 East Mall, Vancouver, BC V6T1Z3, Canada, murray.hodgson@ubc.ca), Rick Peterson, and Paul Benedict (Retaining Walls Northwest, Inc., Bellevue, WA)

Random-incidence transmission losses and absorption coefficients of a vegetated noise barrier of Criblock^{TM} construction were measured without and with plants in a sound-transmission suite built specifically for the purpose, constructed from concrete noise barriers, with the vegetated barrier separating source and receiver rooms. The suite was tested for compliance with ASTM E90-09, and found to be substantially but not completely in compliance with respect to uniformity of steady-state levels and surface absorption. It was found that the transmission loss of the vegetated barrier ranged from 42 dB at low frequencies to 66 dB at 1000 Hz; above 1000 Hz only a lower limit of the TL could be determined—values of 57–62 dB were found. These values are at least 25 dB higher than recommended by BC Ministry of Transportation guidelines. The absorption coefficients of the unplanted and planted barriers were measured; the plants decreased the absorption slightly, from NRC 0.42 to 0.37.

WEDNESDAY AFTERNOON, 5 JUNE 2013 516, 1:00 P.M. TO 3:00 P.M.

Session 3pNSc

Noise and Architectural Acoustics: Joint Poster Session on Noise and Architectural Acoustics (Poster Session)

Hideki Tachibana, Chair
Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino, Chiba 275-0016, Japan

Contributed Papers

All posters will be on display from 1:00 p.m. to 3:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:00 p.m. and contributors of even-numbered papers will be at their posters from 2:00 p.m. to 3:00 p.m.

3pNSc1. Remote keyless entry honking, convenience horn honking, and audible car alarms: Redundancies and quieter options. Jeanine Botta (Epidemiology and Biostatistics, CUNY School of Public Health at Hunter College, 1594 Metropolitan Ave., Apartment 7D, Bronx, NY 10462, jbotta@hunter.cuny.edu)

Vehicle sound emissions, car alarms, and horn honking are the subject of many noise complaints. Auto manufacturers spent years engineering quieter vehicles, and have created cars whose approach is so subtle that they pose a danger to blind pedestrians. But while engine noise has decreased and car alarms are less reactive, horn honking that is linked with remote keyless entry (RKE) technology increasingly contributes to community noise. RKE horn noise has never been the subject of public health inquiry. In scientific literature, discussion of road noise and health does not distinguish noise among separate sources, and tends to measure aggregate ambient noise levels rather than impulsive noise. RKE horn noise violates state traffic laws and some local noise ordinances regarding horn use, but there have been no legislative attempts to address the technology. This raises questions about whether political leaders and policy setters are not exposed to RKE noise or do not discern RKE sounds from traffic noise, and are therefore unaware of it. Using available auto industry data and case studies, this paper will introduce key facts about RKE horn use in the United States and Canada, reviewing new technologies that render noisy counterparts still in use as redundant.

3pNSc2. Determination of noise emission data of construction sites. Ilya E. Tsukernikov, Igor L. Shubin (Acoust. Lab., Res. Inst. of Bldg. Phys., Odoevskogo proezd, h.7, korp.2, fl. 179, 21 Lokomotovny pr., Moscow 117574, Russian Federation, 3342488@mail.ru), Nikolay I. Ivanov (Dept. of Health and Safety, Baltic State Tech. Univ., Sankt-Petersburg, Russian Federation), Tatiana O. Nevenchannaya (Dept. of Phys., Moscow State Univ. of Printing Arts, Moscow, Russian Federation), and Igor A. Nekrasov (Stock Co. Algoritm-Acoustics, Moscow, Russian Federation)

The reasons for development and substantive provisions of Russian standard GOST R 53695-2009 "Noise. Method for determination of noise emission data of construction sites" are presented. The concept of the noise emission characteristics of a construction site and the method of their determination are entered. Various kinds of civil work, construction site location, environment acoustic conditions, and feature of a landscape are taken into account. Instead of noise of separate sources operating inside a construction site noise of the construction site as a whole sound source is considered. The mean time averaged and maximum values of A-weighted sound pressure levels along the sides of a construction site are taken as its noise emission data. Sound reflection from the barriers located near to a construction site is considered by means of environmental correction for which determination the special method was developed. Procedures of determination of measurement uncertainty, the noise characteristics declared values which are brought in construction site specifications and accuracy degree of the method to be applied are considered.

3pNSc3. Engine sounds of small boats at night transmitted to room in apartment built along canal. Kenji Muto and Toru Akahira (Commun. Eng., Shibaura Inst. of Technol., Toyosu 3-7-5, Koto-ku, Tokyo 1358543, Japan, k-muto@shibaura-it.ac.jp)

In this paper, we show the measurement results of the engine sounds of small boats that cruise in a canal. The canal that is called an Toyosu canal is in the residential area in Tokyo in Japan. It is a canal with the role of the waterway traffic in Tokyo. The engine sounds were measured there from October to November in 2011. There were a lot of tugboats or tugboats pulling a freighter in daytime. There were a lot of fishing boats day and night, and there were a lot of houseboats at night. Many of engine sounds were sound exposure level around Lae = 80 dB. The level of the the greatest was more than sound exposure level Lae = 90 dB. Especially, the most of boats were passed in the morning and evening. They passed in the early morning when the background noise was 50 dB. They passed while obstructing the conversation or waking up. It was a sound with the influence for the inhabitant by the canal. It was shown the result of the engine sound transmitted to the room. The engine noise of the boat was transmitted to the canal side room with high level. These results was described in this paper.

3pNSc4. Aerodynamic noise reduction of a gangway in a high-speed train. Hee-Min Noh and Hyo-In Koh (Korea Railroad Res. Inst., #176, Cheldo bangmulgwan-ro, Uiwang 437-757, South Korea, hmnoh@krri.re.kr)

Excessive interior noise of high-speed trains causes annoyance, fatigue, and stress to passengers. Moreover, the noise occurred in gangway is greater than other noise in the room. Therefore, a research for gangway noise reduction was carried out. At first, cavity noise which causes major noise between car-sections was simulated with FLUENT 6.0 (computer fluid dynamics program). From the simulation result, the flow feedback loop phenomena in the cavity were observed. Then, noise measurements at internal and external positions in between-cars sections were conducted during the driving of a high-speed train. From the measurement results, noise characteristics of gangway and between-cars section were identified. Finally, noise mitigation methods were suggested in this paper.

3pNSc5. Improvement of the acoustic environment inside the high-speed train stations depending on the increase of the speed. Chan Hoon Haan and Chan Jae Park (Architectural Eng., Chungbuk National Univ., 52 NaeSudong-Ro, HeungDeok-gu, Chungbuk National Univ., CheongJu, Chungbuk 361763, South Korea, chhaan@chungbuk.ac.kr)

The speed of trains has been increased due to the development of railway technologies. Recently, operation speed up to 400 km/h is come to effect in Korea. But, it can be easily predicted that noise and vibration could be increased depending on the speed of trains. Especially, train stations are exposed to much noises for 24-h at the nearest place when high-speed trains stop or pass the terminals. In the present study, noise levels of the passing high-speed trains were measured in four different stations and noise levels at the speed up to 400 km/h were calculated. Also, the predicted noises were analyzed and compared with the interior noise criteria (NC-curve). As a result, it was found that the noise levels exceed 10 dB higher than the noise standards in average when train speed was 350 km/h. Based on the results, some design proposals are suggested to satisfy with the noise standards including reinforcement of walls and ceilings, change of finishing materials, which can improve the sound insulation of rooms in the train stations.

3pNSc6. Position optimization of Helmholtz resonator in ducts using a genetic algorithm. Maria A. Nunes and Gabriela Silva (Faculdade UnB Gama - Automotive Eng., Universidade de Brasília, Área Especial de Indústria Projeção A – UnB Setor Leste, Gama, DF 72.444-240, Brazil, maanunes@unb.br)

The Helmholtz resonators (HR) are classic reactive mufflers used to attenuate noise at low frequencies mainly pure tones propagating in ducts from venting systems. In industrial environments the equipments layout, the maintenance and operation purposes limit the installation of this kind of device in terms of space and location. As part of the muffler design it is necessary to considerate these restrictions and an optimization process may necessary in order to increase its acoustic performance. Keep in mind that in real application the downstream radiating end of the duct must be modeled as an open end radiating into free space, the insertion loss (IL) parameter is more proper for evaluating the HR's performance than the transmission loss. Using the IL to estimate the effectiveness of the acoustic filter, the main purpose of this paper is to numerically analyze and maximize this parameter in the maximum attenuation frequency considering position restrictions (bounds constraints) in a duct. An evolutionary search algorithm (GA) has been applied in order to solve the best position for a fixed shape HR in a duct. The finite element method was used to model the acoustic system HR/duct. The pressure data and the optimization step were processed in MATLAB®. For optimal positions the results reveal an increase of 19 dB in the IL parameter at the desired frequency. To verify the sensibility of the methodology simulations were performed varying some GA parameters.

3pNSc7. Possibility of sound environmental design by introducing wave sound into the indoor space. Takane Terashima (Architecture, Mie Univ., 1577 Kurimamachiya-cho, Tsu 5148507, Japan, tera@arch.mie-u.ac.jp) and Kazuhiro Shimahashi (School of Design & Architecture, Nagoya City Univ., Nagoya, Japan)

The purpose of this study is to develop the means of designing sound environment of waterfront area. As one of the means of improving sound environment in the campus space of our university, introducing wave sound from the adjoining seashore into campus area has been proposed and its feasibility have been studied. But wave sound reaches 300 m inland at most and cannot be listened in the most of campus area outdoor. So we plan to introduce wave sound by picking up through microphone and steaming over the campus LAN. In this report, if wave sound is streamed and broadcasted to indoor spaces, the influence of wave sound as background/masking noise on the mental state of users in the space is studied. The samples of various wave sound recorded at seashore near the campus are broadcasted in indoor spaces, such as cafeteria, learning spaces, etc. And subjects are asked to answer the questionnaire about preference and subjective impression for indoor environment. The results show that wave sound is almost accepted by users, but to be recognized by uses, output level must be high and could be harmful. The optimum level of wave sound in the space is discussed.

3pNSc8. A design of control signal in reducing discomfort of the dental treatment sound based on auditory masking. Yuko Suhara, Daisuke Ikefuji, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is023083@ed.ritsumei.ac.jp)

In dental treatment, patients feel a strong discomfort feeling by the treatment sounds which arise by a tooth grinding. In order to add comfort to quality of life, we aim to reduce the discomfort feeling with dental treatment sounds. We previously proposed the unpleasantness reduction method based on auditory masking to reduce discomfort feeling of noise. The previously proposed method can reduce discomfort feeling by emitting a control signal to a listener, but we had focused on unpleasantness reduction to noise which has a peak frequency. Meanwhile, dental treatment sounds tend to consist of multiple spectral peaks. Therefore, in the present paper, we propose the design method of control signals for reducing discomfort feeling of dental treatment sounds which have multiple spectral peaks. More specifically, we detect the main spectral peaks, which bring a discomfort feeling, and design the control signal, which can mask these spectral peaks. Also, we employ the sound of running water as a source for the control signal. We carried out subjective evaluation experiments to confirm the effectiveness of the proposed method. As a result of evaluation experiments, we confirmed the effectiveness of the proposed method.

3pNSc9. Noise in hospitals as a strain for the medical staff. Silvester Siegmann and Gert Notbohm (Inst. for Occupational and Social Medicine, Heinrich Heine Univ. Duesseldorf, Universitätsstr. 1, Duesseldorf, NRW D-40225, Germany, siegmann@uni-duesseldorf.de)

Noise research in hospitals focuses mainly on the harmful effects on patients. But at least in intensive care units and operation theaters, also the staff is exposed to high levels of noise during considerable portions of working time. Evidence from literature is summarized here. During operation sessions lasting from 30 min to several hours, reported average Leq values range from 58 to 72 dB(A) with maximum levels above 105 dB(A). Similar noise levels are reported from emergency departments. As concentration, precise communication, and fast decisions are necessary in these situations, the acoustical environment has to be considered an enormous strain for the staff and a potential risk with regard to faults at work. But also during normal day and night shifts in intensive care units, noise is mentioned as an important disturbance by the medical staff. Most disturbing are noises from telephones and other communication tools and the signals and sounds from medical devices. Questionnaire surveys result in 80 to 91% of the staff reporting negative effects of noise in their daily work. A variety of measures for noise reduction and prevention in hospitals is suggested in literature emphasizing that the staff plays a decisive role in such projects.

3pNSc10. A consideration on sound masking system for achieving speech privacy using parametric acoustic array speaker. Takahiro Tamesue and Tetsuro Saeki (Yamaguchi Univ., 2-16-1, Tokiwadai, Ube 755-8611, Japan, tamesue@yamaguchi-u.ac.jp)

Speech privacy in open spaces is becoming increasingly important in various situations. Although measures such as the use of sound partitions are already used in many cases, measures that mask speech by emitting sounds have also been considered. A method of masking meaningful speech with meaningless noise would be valuable. Because of this, previous studies have investigated the ability of meaningless steady noise to mask speech and consequently achieve speech privacy. However, the research to date has focused on evaluating speech privacy when the masking noise is emitted from the normal loud speaker system all over the room. The masking noise emitted to the area where high level of speech privacy is not required, may cause an increased psychological impression of annoyance, leading to a decline in performance. In this study, we used a highly directional sound from modulated ultrasound as a masking noise for achieving speech privacy in the narrow area. Psychological experiments were conducted in which the masking sound was transmitted to participants from frontal or above directions with a parametric acoustic array speaker. Using the experimental data, the relationships between the degree of speech privacy and frequency characteristics and directivity of parametric acoustic array speaker were investigated. The results suggested that it is possible to maintain speech privacy in the narrow area by presenting highly directional masking sound.

3pNSc11. Categorization of street types in urban thermoacoustic analysis. Elcione L. Moraes, Irving M. Franco, Marcelle V. Silva, Isabela A. Rocha, Dorival F. Pinheiro, and Mindiyarauakti P. Freitas (Architecture and Planning, Federal Univ. of Pará, av. Augusto Corres, 01, Belém, Pará 66075900, Brazil, elcione@ufpa.br)

Urban areas suffer several environmental perturbations as a result of human activities and technological developments that contribute to the formation of heat islands and increasing noise contamination. Environmental effects are incorporated by population in urban areas and, especially, in areas near roads with heavy traffic. This paper presents a theoretical-experimental analysis on the relationship between climatological conditions and the propagation of noise in traffic corridors with high, medium, and low intensity. Some variables, such as the width of the streets, the height of the buildings, the distance between buildings, the volume flow of vehicles offer the possibility to make traditional techniques for mitigating the air temperature increase and reduce noise levels in urban zones. The results obtained in this work by measuring temperature, humidity, and noise levels, made during certain periods of time in different parts of the city of Belem/Brazil, were linked to a database georeferenced (GIS) that allowed interpolation of data in a single platform, enabling integration between data allowing to correlate them in order to assess which typological conditions are most favorable to the thermoacoustic comfort.

3pNSc12. Effects of age on feasible sound level of possible warning sounds for quiet vehicles. Katsuya Yamauchi (Faculty of Eng., Nagasaki Univ., Bunkyo 1-14, Nagasaki 852-8521, Japan, yamauchi@cis.nagasaki-u.ac.jp), Takayuki Shiizu, Fumio Tamura, and Yuichiro Takeda (Pioneer Corp., Kawagoe, Japan)

It has been noted that reduced noise can also lead to potentially dangerous situations for pedestrians because electric and hybrid vehicles are quieter than conventional internal combustion engine vehicles. Hence, the use of warning sounds which are radiated by the vehicle to alert pedestrians is being considered by various governments. To design the sound itself or to develop the regulation concerning the sound, it is much important to know the feasible sound level of the warning sounds compared to the background sounds. Pilot studies on this topic were performed by Yamauchi *et al*. in 2011 with young subjects. This present study was aimed to reveal the effect of age on feasible sound level of warning sounds. In the experiment, level of five possible warning sounds was adjusted in three different urban environmental sounds in a laboratory. Thirty subjects aged from 19 to 74 years old participated in the experiment. The subjects were asked to adjust the level of warning sounds so that they are clearly audible or just audible depends on the instruction. Results of the adjustments are presented and compared to current recommendations for sound levels of warning signals in quiet vehicles.

3pNSc13. Green cork-based innovative resilient and insulating materials: Acoustic, thermal, and mechanical characterization. Marco Caniato, Sbaizero Orfeo (Architecture and Eng., Univ. of Trieste, via valerio 6/a, Trieste 34100, Italy, mcaniato@units.it), Jan Kaspar, and Roberta Di Monte (Dept. of Chemistry Sci., Univ. of Trieste, Trieste, Italy)

Nowadays, efficient thermal insulation is a principal requirement for buildings and, accordingly, huge amounts of insulators are applied in the constructions, particularly for external walls, radiant floor, etc. Acoustic insulation is another of the most stringent parameters to be taken into account both in the construction of new buildings or their rejuvenation in order to obtain good internal comfort. On the other hand, the use of bio-derived construction materials is gaining stronger and stronger interest. Cork has a low density (120–240 kg m^3) and can be regarded as a hydrophobic and viscoelastic material, with good thermal and acoustic insulation properties. With respect to wood, it presents good resistance to microbial activity and water. Last but not least is the negative carbon fingerprint of cork-based materials. Here we will describe a new class of polymer—inorganic oxides—cork composites that feature enhanced thermal and acoustic properties with respect to traditional commercial composites and maintain, at the same time, all the favorable properties of conventional cork-base composites.

3pNSc14. Impulse response measurement in public space using musical signal including swept-sine signals. Fumiaki Satoh, Junichi Mori, Tomoya Nishii, and Hideki Tachibana (Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino 275-0016, Japan, fumiaki.satoh@it-chiba.ac.jp)

In design of public spaces, e.g., railway stations, airport terminal buildings, and underground shopping centers, careful attention should be paid from an acoustical viewpoint. It is not only for acoustical comfort but also for safety ensured by a public address system with high intelligibility. As a study for this aim, we have been investigating acoustical characteristics of various public spaces. In these studies, it is strongly desired to measure impulse responses in the spaces under live condition with occupants but the usual measurement method using Swept-Sine signals are not applicable because the signals sound very peculiar to the occupants. To mitigate such a problem, we are trying a method using test music signals in which Swept-Sine signals are inserted. In this paper, the availability of this measurement technique is outlined and some measurement results are introduced.

3pNSc15. Modeling room impulse response via composites of spatial-temporal Gaussian processes. Tatsuya Komatsu (Grad. School of Information Sci., Nagoya Univ., Furo-cho, chikusa-ku, Nagoya, Aichi 464-8601, Japan, komatsu.tatsuya@g.sp.m.is.nagoya-u.ac.jp), Gareth W. Peters (Dept. of Statistical Sci., UCL, London, United Kingdom), Tomoko Matsui (Dept. of Statistical Modeling, Inst. of Statistical Mathematics, Tokyo, Japan), Ido Nevat (Wireless & Networking Lab., CSIRO, Sydney, NSW, Australia), and Kazuya Takeda (Grad. School of Information Sci., Nagoya Univ., Nagoya, Japan)

We develop a novel algorithm to estimate a spatial-temporal transfer function of a time-domain room impulse response for reverberation in closed environments. This novel approach involves developing two non-parametric models, one for the early phase and the other for the late phase for reverberation. These models are based on a composite of two Gaussian Process (GP) structures. We also investigate the impact of the choice of spatial and temporal kernels on the estimation and prediction performance. The proposed algorithm incorporates as a special case the widely utilized exponentially decaying model and also extends this model structure within the GP setting to more advanced spatial-temporal structures suitable to perform estimation of the reverberant transfer function. The performance of the proposed algorithm is evaluated in a real environment using nine spatially distributed microphones. The microphones collect reverberant response from a directional speaker allowing observation of noisy realizations of the impulse response to reverberate during early and late stages. We compare the performance of our algorithm with a 3D spatial-temporal cubic interpolation algorithm and show that the proposed algorithm provides equal or better performance than the cubic interpolation.

3pNSc16. Large-scale sound field rendering with graphics processing unit cluster for three-dimensional audio with loudspeaker array. Takao Tsuchiya (Dept. of Information Systems Design, Doshisha Univ., 1-3 Tatara-Miyakodani, Kytanabe City, Kyoto 610-0321, Japan, ttsuchiy@mail.doshisha.ac.jp), Yukio Iwaya (Dept. of Elec. Eng. and Information Technol., Tohoku Gakuin Univ., Tagajo, Miyagi, Japan), and Makoto Otani (Dept. of Comput. Sci. and Eng., Shinshu Univ., Nagano, Nagano, Japan)

The sound field rendering is a technique to simulate the sound field from the three-dimensional numerical models constructed in the computer, and it is the same concept as the graphics rendering in the computer graphics. In this paper, a graphics processing unit (GPU) cluster system is applied to the sound field rendering for a large room simulation. The compact explicit finite difference time domain (CE-FDTD) method is implemented on the GPU cluster system. The CE-FDTD method is a kind of the finite difference method in which the wave equation is directly discretized based on the central differences. The developed GPU cluster system consists of eight PC nodes in which four GPUs are mounted respectively. The rendering results are reproduced by a speaker array system in which 157 speakers surround a small room. The sound field renderings are performed for a large room with a volume capacity of about 5000 cubic meters, in which the impulse responses of one-second length with a sampling rate of 40 kHz are calculated at 157 points corresponding to the loudspeaker positions. The impulse responses are then convoluted with dry music sources. The sound field rendering with the 157ch loudspeaker array system provides the realistic sound field reproduction with natural reverberation.

3pNSc17. Acoustic characterization of three archeological sites in the state of Guanajuato, Mexico. Alejandro Ramos-Amezquita (Comput. Sci. Dept., Tecnológico de Monterrey, Calle del Puente 222, Colonia Ejidos de Huipulco Tlalpan, Mexico City, Mexico DF 14380, Mexico, alejandro.ramos.amezquita@itesm.mx) and David I. Ibarra-Zarate (CAEND, Universidad Politécnica de Madrid, Madrid, Spain)

The present work shows the results obtained in collaboration with the government of the state of Guanajuato in Mexico in a project that looked to include the acoustical analysis of Archaeological sites as a tool for gathering information regarding the historical social use of the areas in question. To that end, the acoustical characterization of 3 archaeological sites recently opened to the public in the state was in order: Cañada de la Virgen, Peralta and Plazuelas. Results include the 3D modeling of the areas of interest and the simulation of the acoustic response of them using the software EASE. Specific acoustic parameters were extracted from the simulations and then analyzed in comparison to archeological hypothesis of the use of such spaces as areas of public appearances, performance, ethno-musicological reports on the type and use of musical instruments, and other archaeological findings in the area in order to support or disprove such hypothesis.

3pNSc18. Basic study on discrimination between sound fields of architectural spaces. Maya Katoh and Takane Terashima (Architecture, Mie Univ., 1577 Kurimamatiya-cho, Tsu, Mie 514-8507, Japan, 412m409@m.mie-u.ac.jp)

Many objective criteria by attenuation property of room acoustic energy have been suggested, and relationships with subjective attributes have been elucidated. The difference of synthetic subjective impression between sound fields could be discriminated by acoustical parameters, i.e., objective criteria above, but details of mechanism in discrimination among different sound fields, weightings or grounds of judgment, etc. have not been clarified. The purpose of this study is to clarify the discrimination factor over difference between sound fields. In this report, subjective experiments are carried out by using impulse response data of existing ten concert halls and music data, made from convolution of impulse responses and dry sources, as stimuli. In these experiments, subjects are asked to evaluate the impression of each stimulus, and to judge the difference among stimuli in pared comparison. The results of these experiments are analyzed and it seems that the factor related to the difference of Reverberance is dominant in discrimination by factor analysis, but there are some cases in which Loudness or Clarity is dominant. The weightings of factors, the boundary to switch judgment between factors, etc. will be discussed.

3pNSc19. Effect of visual information on subjective impression for sound field in architectural space. Yuko Wani, Takane Terashima (Architecture, Mie Univ., 1577 Kurimamachiya-cho, Tsu, Mie 514-8507, Japan, 412m421@m.mie-u.ac.jp), and Yasunobu Tokunaga (Civil Eng., Maizuru National College of Technol., Maizuru, Japan)

In architectural and urban space, we are always exposed to multimodal stimuli of visual information and sound fields in various scenes of everyday life. The purpose of this study is to clarify relationship of subjective impression of vision and auditory, and acquire knowledge which contributes to architectural design or acoustic design. In this report, two following experiments are carried out in which subjects are presented with sound fields by real time convolution as auditory stimuli and panoramic VR images of 360 interactive views of interior as visual stimuli. (1) Comparison between subjective responses for single or multi modal presentations of visual and auditory stimuli from various architectural spaces. (2) Comparison between subjective responses for various combinations of multi modal presentations of visual and auditory stimuli from various architectural spaces. Analysis for results of these experiments clarify the influence of visual information upon the subjective impression for sound field and mutual relationship between subjective impression of vision and auditory of interior of buildings. It is already found that visual information significantly effects subjective impression for sound field by experiment 1, and the details of relationship between elements of visual information and parameters of sound field will be clarified by experiment 2.

3pNSc20. Study of the acoustic of Jean Nouvel's Auditorium 400, at the Museum Reina Sofia in Madrid. Emiliano del Cerro and Silvia M[a] Ortiz (TIC, Universidad Alfonso X el Sabio, Avenida Universidad 1, Madrid 28691, Spain, ecerresc@uax.es)

The Auditorio 400 is one of the buildings that make up the National Art Museum Reina Sofia in Madrid. It is the work of renowned French architect Jean Nouvel. This space was designed to accommodate primarily chamber music concerts but now can be considered as a multi-purpose venue. This hall hosts events with different content: acts with the voice as main sound

source as conferences, seminars etc. and concerts with music from diverse styles, classical, contemporary, avant-garde, and electro acoustic music. This versatility assumes that the acoustic conditions required for the different uses of Auditorio 400 must be diverse and special depending on the sound source, in order to achieve the adequate sound quality for the various events that are held there. This paper presents the study of the acoustics of the Auditorium 400, analyzing various parameters for evaluating the sound quality of the room, highlighting the worst areas of listening, the reasons for the existence of such areas and the description of improvements to be made to ensure that the enclosure meets the expectations in a hall of its relevance.

3pNSc21. Influence of visual information on sound evaluation in auditorium. Yasunobu Tokunaga, Daichi Okuie (Maizuru National College of Technol., 234 Shiroya, Maizuru-shi 625-8511, Japan, tokunaga@maizuru-ct.ac.jp), and Takane Terashima (Grad. School of Eng., Mie Univ., Tsu-shi, Japan)

When hearing music in an auditorium, audience is provided with aural information and visual information at the same time. Visual and auditory sense sometimes interact with each other so that the auditory sense is considered to have some influence on sound evaluation made by audience. The purpose of this study is to reveal an influence of visual information obtained in audience seats in a hall on subjective evaluation. We investigated the relationship between sound evaluation and whether a musical performance video as visual stimulus is provided or not, and between the sound evaluation and evaluation concerning the space at certain position in the audience seats. As a result, it was revealed that visual information gave statistically significant influence on sound evaluation.

3pNSc22. Design and positional accuracy of straight-path traversing room acoustic measurement system based upon low-cost servo motor and light-weight multi-field microphone. Roger M. Ellingson (RM Ellingson Design & Development, LLC, 8515 SW Barnes Rd., Portland, OR 97225, Rogere@Rmegen.net) and Guillaume J. Bock (Bruel & Kjaer, North America Inc., Snohomish, WA)

The construction and operation of a straight-path traversing, acoustical microphone-based, measurement system is described. The system was designed to support test room qualification procedures such as prescribed in standards ANSI S12.35-1990 and Annex A of ISO 3745-2003. Major system components include a taut line suspending a sliding microphone carriage drawn by a string attached to a rotating drum. Central to the overall light-weight, low-power, mechanical design is the physically small Bruel & Kjaer 4961 microphone, holder, preamp, and signal cable whose multi-field response characteristic should well support accurate room qualification measurements. The battery-powered drum drive mechanism is built using the servo motor, wheels, and gears from a Lego Mindstorms NXT kit. Precision motor rotation is remotely programmable over the NXT controller's wireless Bluetooth interface. Commonly available sport fishing tackle composes the majority of the suspension assembly. A software interface library is also described which enables PC-based applications to automate microphone positioning in synchrony with source emission, signal acquisition, and analysis. The system has been used to document the free-field characteristic of a perimeter loudspeaker array with centrally located listener in a fully anechoic chamber environment. Together with construction and operational detail, results indicating overall microphone positioning accuracy and reliability are presented here.

WEDNESDAY AFTERNOON, 5 JUNE 2013 519B, 1:00 P.M. TO 2:40 P.M.

Session 3pPAa

Physical Acoustics: Borehole Acoustics Logging for Hydrocarbon Reservoir Characterization II

Said Assous, Cochair
Geoscience, Weatherford, East Leake, Loughborough LE126JX, United Kingdom

Weichang Li, Cochair
ExxonMobil Res. & Eng., 1545 Rte. 22 East, Annandale, NJ 08801

Contributed Papers

1:00

3pPAa1. Simulation of sonic logging for deviated wells in anisotropic formations. Evgeniya Deger (SKK, Schlumberger, 2-2-1 Fuchinobe, Chuo-Ku, Sagamihara, Kanagawa 252-0206, Japan, emyalo@slb.com), Marwan Charara (SMR, Schlumberger, Moscow, Russian Federation), Henri-Pierre Valero (SKK, Schlumberger, Sagamihara, Japan), Denis Sabitov, and Grigory Pekar (SMR, Schlumberger, Moscow, Russian Federation)

Interpretation of sonic logging data acquired in environments with complex anisotropy is a difficult problem attracting attention of researchers and oil industry. In order to better understand physics of wave propagation in highly anisotropic medium and be able explaining observations from field data there is a need for fast and accurate numerical modeling capability. To address such challenge, we developed an efficient and accurate numerical algorithm for the simulation of sonic logging experiments in highly anisotropic formation. The basis of the approach is a heterogeneous spectral element method implemented on multi-GPU applied to acoustic-elastic wave equation. The approach was designed to simulate wave propagation in 3D arbitrary anisotropic elastic media with attenuation for a constant quality factor via standard linear solid using the tau-method. Due to the use of an unstructured grid, the spectral element algorithm enables handling tools in a fluid-filled borehole with surrounding geological models of high complexity. Several examples of log simulations for deviated wells in VTI formations for monopole, dipole, and quadrupole source symmetries and their comparison with real field data will be presented in this paper. Discussion regarding complex wave propagation will be developed in view of these simulations and real data.

1:20

3pPAa2. Numerical simulations of dipole sonic responses in a liquid-filled trough with arc-shaped section. Xiao He, Hao Chen, and Xiuming Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., 21, Northern 4th Ring Rd. West, Haidian District, Beijing 100190, China, hex@mail.ioa.ac.cn)

To test the running performance of a sonic logging tool, it is an effective way to place the tool in a water-filled trough and record the sonic responses for an integrated check of the transducers in laboratory. Hence it is necessary to investigate the wave propagation in such a non-symmetric structure. In this study, we present the numerical modeling results of dipole sonic responses in a trough with arc-shaped section. A 3D cylindrical finite difference code as well as the irregular free-surface conditions is implemented. It is revealed that the flexural mode in a trough is evidently slower than that in a cylindrical pipe with same sizes. The flexural velocity decreases with the increasing gap angle of the trough. Moreover, the trough structure shows strong elastic anisotropy. The in-line responses (XX and YY) of the logging tool are difference in phases and amplitudes. The amplitudes of both cross-line responses (XY and YX), which reflect the level of anisotropy, become greater as the gap angle increases. The waveforms with varying dipole orientations are also illustrated. The XY and YX responses excited by an inclined transmitter are much stronger than those generated by a horizontal or a vertical dipole source.

1:40

3pPAa3. Research on a kind of low-frequency broadband cross-dipole projector. Dai Yuyu, Xin Penglai, Wang Xiuming, and He Hongbin (The Ultrasonic Phys. and Exploration Lab., Inst. of Acoust., Chinese Acad. of Sci., No.21, Bei-Si-huan-Xi Rd., Beijing 100190, China, daiyuyu001@126.com)

The finite element method (FEM) is used to simulate a low-frequency broadband cross-dipole projector based on trilaminar bender bar in this paper. In the simulated model, four long trilaminar bender bars and four short trilaminar bender bars are attached on a skeleton two form two square arrays, and very array excite two different response frequencies. The four response frequencies distribute on the range from 400 Hz to 5 kHz to reach broadband exciting. Finally, a sample is fabricated, and test in an anechoic tank, it is shown that the test result meet the simulated result very well.

2:00

3pPAa4. Phase-based dispersion analysis of borehole acoustic logs. Said Assous (Geoscience, Weatherford, East Leake, Loughborough LE126JX, United Kingdom, said.assous@eu.weatherford.com), Laurie Linnett (Geoscience, Weatherford, Edinbourgh, United Kingdom), and Peter Elkington (Geoscience, Weatherford, East Leake, United Kingdom)

The dispersive behavior of acoustic waves in boreholes is of interest in the evaluation of reservoir rocks, particularly from the point of view of near wellbore stress distribution. It is also used as a quality control on dipole sonic calculations that estimate formation shear slowness from the low frequency asymptote of the flexural wave slowness. Multiple methods are available for dispersion analysis; the paper reviews the most commonly used, including the Prony and the spectral semblance methods, and proposes a new phase-based analysis technique that has the benefit of improved slowness resolution. The methods are applied to synthetic and real data sets, and results compared. The new method is show to have lower slowness uncertainty for any given frequency, and the upper and lower frequency limits for which dispersion can be calculated is also extended.

2:20

3pPAa5. Unrelaxed drained bulk modulus for fluid-saturated rocks on full frequency range. Yong J. Song, Hengshan Hu (Dept. of Astronautics and Mech., Harbin Inst. of Technol., P.O. Box 344 92 West Dazhi St., Harbin, Heilongjiang (+0086)150001, China, songyongjia061220110@126.com), and Changwen Li (Technol. Ctr. of China Petroleum Logging CO., LTD., Xian, China)

Local flowing between cracks and pores is called squirt-flow that usually induce elastic moduli dispersion and waves attenuation. Expression of unrelaxed drained bulk modulus on full frequency range is derivated in this paper when liquid pressure in soft crack is unrelaxed. Unrelaxed drained bulk modulus's real part increases with frequency, and unrelaxed drained bulk modulus's imaginary part is nonzero. This studies also show that liquid pressure in cracks equals to zero at he low frequency limitation, that is to say liquid pressure in cracks have sufficient time to relax and in this case the drained bulk modulus correspondents to Biot's static drained bulk modulus.At the high frequency limitation, the unrelaxed drained bulk modulus approximate to Mavko-Jizba's expression. The expression of drained modulus in this paper also degenerates to Biot theory's drained bulk modulus when crack density equals zero, but the latter is just a modulus on a hypothetical state and is not the true static experiment data when rocks contain soft cracks. Unrelaxed drained bulk modulus is used to calculate fast P-wave and slow P-wave velocities and attenuation instead of static drained or dry bulk modulus in Biot's theory. Squirt-flow generates much more velocity dispersion and attenuation in fast P-wave than Biot flow. The magnitude of attenuation depends on crack density and the relaxation frequency depends on aspect ratio.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 519A, 1:00 P.M. TO 2:40 P.M.

Session 3pPAb

Physical Acoustics: General Physical Acoustics II

Raymond Panneton, Chair

Mech. Eng., Univ. de Sherbrooke, 2500 Universite Blvd., QC J1K 2R1, Canada

Contributed Papers

1:00

3pPAb1. Modeling sound fields from radially symmetric impulsive planar sources using Rayleigh's Integral. Stephen I. Warshaw (LLNL (Ret.), Univ. of CA, 40 West 15 St Loft 1C, New York, New York 10011, siw1939@yahoo.com)

The evaluation of Rayleigh's integral in cases where the integrand contains delta functions with time-dependent arguments provides a highly useful and easily implemented vehicle for modeling and understanding the radiation of impulsive sound waves from planar sources. We present elementary examples of such calculations and show their application in various realistic scenarios, including the radiation of impulsive aeroacoustic sound from buried explosions and potential extensions to lithotripsy. In this paper, we represent the surface motions of a planar radiating source as delta functions of radially expanding or contracting arguments with easily specified geometrical parameters, and present model near-field to far-field calculations of the launched sound waves using analytic and numerical integration. We show that significant features in these traveling impulses can easily be related to the details of the kinematic history of the planar source. We restrict our purview to the time domain and to radial symmetry; frequency analyses and azimuthally asymmetric motions will not be considered.

1:20

3pPAb2. Application of the spectral method for computation of the spectrum of anisotropic waveguides. Timur Zharnikov, Denis Syresin (Schlumberger Moscow Res., Pudovkina St., 13, Moscow 119285, Russian Federation, tzharnikov@slb.com), and Chaur-Jian Hsu (Schlumberger-Doll Res., Cambridge, MA)

Spectral method is formulated in cylindrical coordinates for the general case of waveguide with arbitrary anisotropy with the spatial dependence. According to the idea of this approach, matrix representation of operator in

the right-hand side of governing equations is considered. As a result, the latter are cast into exact infinite set of integro-differential equations. Explicit expressions for their kernels expose coupling between axial and azimuthal harmonics. Coupling of axial harmonics vanishes in important case of waveguide with translational invariance in axial direction. It results in the set of differential equations, which is used to introduce practical approximation procedure. The latter yields generalized eigenvalue problem, which can be solved numerically for the spectrum of the operator. The spectrum is sorted according to eigenmodes' properties. Thus dispersion curves of eigenmodes are constructed. Presented consideration can be adapted for waveguides of different physical nature (elastic, electromagnetic, etc.) and different geometry (rectangular, elliptical, etc.). Developed technique is verified by comparison with results of controlled laboratory measurements on anisotropic sample. Monopole, dipole and quadrupole normal modes for scaled borehole in anisotropic rock sample with TTI symmetry are considered. The comparison of spectral method results with the dispersion analysis of synthetic data is provided as well.

1:40

3pPAb3. Influence of reflecting walls on edge diffraction simulation in geometrical acoustics. Alexander Pohl (HafenCity Universität Hamburg, Hebebrandstrasse 1, Hamburg 22297, Germany, alexander.pohl@hcu-hamburg.de), Dirk Schröder (EPFL Lausanne, Lausanne, Switzerland), Uwe M. Stephenson (HafenCity Universität Hamburg, Hamburg, Germany), and U. Peter Svensson (NTNU Trondheim, Trondheim, Norway)

Edge diffraction can be introduced into geometrical acoustics mainly by three models: detour-based, energetic and wave-based diffraction models. In the past, we thoroughly compared Maekawa's detour law, the uncertainty relation based diffraction method and the secondary source model by the example of edge diffraction of a single wedge. However, the influence of the wedge shape has not yet been analyzed. Therefore, we consequently study in this contribution the influence of the wedge's faces. This is analyzed by varying both the faces' reflection properties and their opening angle. This is extended to the crucial case of approximately parallel faces (inner angle, e.g., 179°), where diffraction is physically neglectable, but computationally problematic for the uncertainty based diffraction method. Additionally, wedges are placed on an infinitely long surface. Therewith, we can analyze the floor reflections' impact on the sound field behind the wedge by varying both their absorption and scattering coefficients. Furthermore, we discuss artifacts which can arise in the uncertainty based diffraction model due to arbitrary positioning of diffraction planes, so called "transparent walls." Finally, we discuss the advantages and disadvantages of the presented methods.

2:00

3pPAb4. Acoustic response of a buried landmine with a low grazing-angle source array, focused on the ground. Benjamin J. Copenhaver, Justin D. Gorhum, Charles M. Slack, Martin Barlett, Thomas G. Muir, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, bcopenhaver@utexas.edu)

An array of 16 loudspeakers, deployed along a segment of the base of a right circular cone, was used to focus intense tone bursts at low audio frequencies on a soil, with and without a buried target, having a compliant lid. The response of the target site was examined as a function of source frequency, intensity level, and excitation signal type, including multi-tone radiations. Nonlinear interaction to produce sum and difference frequencies at the target site was examined and compared with observations of Korman and Sabatier [J. Acoust. Soc. Am. **116**, 3354 (2004)]. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

2:20

3pPAb5. The farfield impulse response for a rectangular piston in viscous media. Pedro C. Nariyoshi and Robert J. McGough (Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48864, mcgough@egr.msu.edu)

Calculations of the transient radiation pattern in the farfield of rectangular transducers often employ the impulse response of the velocity potential. Analytical expressions for the impulse response are known for lossless media, and similar expressions are needed for propagation in viscous media. Solutions are obtained with two different Green's functions for viscous media. One solution, which is causal, is based on an approximate Green's function for the Stokes wave equation, and the other solution, which is noncausal, is based on an approximate Green's function for the Blackstock wave equation. The impulse response is calculated with these Green's functions using the Rayleigh-Sommerfeld integral, and the results of these calculations are compared to analytical expressions for the impulse response derived for viscous media. Numerical results are obtained for a 1 mm by 1 mm square transducer evaluated in the farfield region, where the Rayleigh-Sommerfeld integral provides the reference calculated with 400 abscissas in each direction. The results show that the causal and noncausal solutions are nearly identical in the farfield region, and the analytical impulse response expressions derived from the causal and noncausal Green's functions are consistently within 1% of the Rayleigh-Sommerfeld reference in the farfield. [Work supported in part by NIH Grant R01 EB012079.]

Session 3pPP

Psychological and Physiological Acoustics: Multimodal Influences on Auditory Spatial Perception

William L. Martens, Cochair
Univ. of Sydney, 148 City Rd., Wilkinson Bldg. G04, NSW 2006, Australia

Shuichi Sakamoto, Cochair
Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai 980-8577, Japan

Invited Papers

1:00

3pPP1. Spatial sound and its effect on visual quality perception and task performance within a virtual environment. Brent Cowan (Business and Information Technol., Univ. of Ontario Inst. of Technol., Oshawa, ON, Canada), David Rojas (SickKids Learning Inst., The Hospital for Sick Children, Oshawa, ON, Canada), Bill Kapralos (Business and Information Technol., Univ. of Ontario Inst. of Technol., 2000 Simcoe St. North, Oshawa, ON L1H 7K4, Canada, bill.kapralos@uoit.ca), Karen Collins (The Games Inst., Univ. ofWaterloo, Waterloo, ON, Canada), and Adam Dubrowski (SickKids Learning Inst., The Hospital for Sick Children, Toronto, ON, Canada)

Immersive 3D virtual environments such as simulations and serious games for education and training are typically multimodal, incorporating at the very least both visual and auditory cues, each of which may require considerable computational resources, particularly if high fidelity environments are sought. It is widely accepted that sound can influence the other modalities. Our own previous work has shown that sound cues (both contextual and non-contextual with respect to the visual scene) can either increase or decrease (depending on the sound) visual fidelity (quality) perception in addition to the time required to complete a simple task (task completion time) within a virtual environment. However, despite the importance and benefits of spatial sound (sound that goes far beyond traditional stereo and surround sound techniques, allowing users to perceive the position of a sound source at an arbitrary position in three-dimensional space), our previous work did not consider spatial sound cues. Here we will build upon our previous work by describing the results of an experiment that will be conducted to examine visual fidelity (quality) perception and task performance in the presence of various spatial sound cues including acoustical reverberation and occlusion/diffraction effects, while completing a simple task within a virtual environment.

1:20

3pPP2. Touch the sound: The role of audio-tactile and audio-proprioceptive interaction on the spatial orientation in virtual scenes. M. Ercan Altinsoy and Maik Stamm (Chair of Commun. Acoust., Dresden Univ. of Technol., Helmholtzstr. 18, Dresden 01062, Germany, ercan.altinsoy@tu-dresden.de)

Being able to localize objects in the space close to the body is an important prerequisite for precise object interaction. It is also very important for the spatial orientation in virtual scenes. Since sound is usually produced by the vibrations of a body, sound emitting objects, such as shaver or hair dryer, provide both auditory and haptic information. This study focuses on auditory-haptic localization in the spatial domain. We carried out two experiments to investigate the interaction effects. In the first experiment, the influence of tactile signals on auditory localization task was investigated. Similar to the ventriloquist effect from auditory-visual interaction, the results of the first experiment show that the perceived location of auditory stimuli is influenced by tactile stimulation. The results also indicate some hints that there may be an audiotactile precedence effect. In the second experiment, the influence of auditory signals on proprioception was investigated. The results show that the auditory and proprioceptive information can be combined in such a way that the localization errors in a virtual scene are minimized.

1:40

3pPP3. Compression of perceived auditory space during forward self-motion. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Wataru Teramoto (Grad. School of Arts and Letters, Tohoku Univ., Sendai, Japan), Fumimasa Furune (Grad. School of Information Sci., Tohoku Univ., Sendai, Japan), Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Jiro Gyoba (Grad. School of Arts and Letters, Tohoku Univ., Sendai, Japan)

Humans can perceive a stable auditory environment and appropriately react to a sound source, even when they are moving. This suggests that the inputs are reinterpreted in the brain, while being integrated with information on the movements. Although several studies have shown the influence of the vestibular semicircular canal signals on auditory localization, it is not clear how auditory space representation is modulated by linear accelerations which are obtained from the macular receptors of the otolith system (utricle and saccule). We investigated the effect of the linear acceleration on auditory space representation. During the forward/backward self-motion, a short noise burst was presented from one of the loudspeakers which were aligned parallel to the motion direction when the listener's coronal plane reached the location of one of the speakers (null point). The results showed that the sound position aligned with the subjective coronal plane was displaced ahead of the null point only during forward self-motion. Moreover, all the sounds that were actually located in the traveling direction were perceived as being biased towards the null point. These results suggest a distortion of perceived auditory space in the direction of movement during forward self-motion.

2:00

3pPP4. Dominance of head-motion-coupled directional cues over other cues during walking depends upon source spectrum. William L. Martens (Faculty of Architecture, Design and Planning, Univ. of Sydney, 148 City Rd., Wilkinson Bldg. G04, Univ. of Sydney, NSW 2006, Australia, william.martens@sydney.edu.au), Shuichi Sakamoto (Res. Inst. of Elec. Commun. and Grad. School of Information Sci., Tohoku Univ., Sendai, Japan), Luis Miranda, and Densil Cabrera (Faculty of Architecture, Design and Planning, Univ. of Sydney, Sydney, NSW, Australia)

Listeners who walk past a continuously presented speech sound source emanating from a fixed spatial position will typically experience veridical perception of source location. If, however, walking listeners are fitted with binaural hearing instruments that allows for the signals reaching their ears to be interchanged, left for right and right for left, the sound source is typically reported to be located in a spatial region that is reversed with respect to all three spatial axes: left for right, front for back, and above for below. This result has been taken as evidence for the relative dominance of dynamic interaural directional cues over the spectral directional cues associated with each listener's own pinnae which should support veridical perception. In order to investigate the relative importance of the spectral energy distribution of the source on the illusory reversals of source location, bursts of broadband noise were presented rather than continuous speech. Under these circumstances, with greater energy in higher frequency bands, the reversals did not readily occur. Therefore, it has been concluded that head-motion-coupled directional cues are likely to dominate spectral cues associated with the filtering effects of the listener's pinnae only for sources containing greater energy at lower frequencies.

2:20

3pPP5. Impact of dynamic binaural signal associated with listener's voluntary movement in auditory spatial perception. Tatsuya Hirahara, Daisuke Yoshisaki, and Daisuke Morikawa (Faculty of Eng., Toyama Prefectural Univ., 5180 Kurokawa, Imizu 939-0398, Japan, hirahara@pu-toyama.ac.jp)

The effect of listener's voluntary movement on the horizontal sound localization was investigated using a binaural recording/reproduction system with TeleHead, a steerable dummy head. Stimuli were static binaural signals recorded with a still dummy-head in head-still condition, dynamic binaural signals recorded with a dummy-head that followed precise or modified listener's head rotation, dynamic binaural signals produced by steering-wheel rotation with listener's hands in head-still condition, and dynamic binaural signals produced by an experimenter in head-still condition. For the static binaural signals, some were localized within the head and the front-back errors often occurred. For the dynamic binaural signals, none of them was localized within the head, and the front-back confusions seldom occurred. Sound images of the dynamic binaural stimuli produced by head rotation were localized out-of head, while those produced by the steering-wheel rotation or by an experimenter were moving around the listener's head. Listeners could judge the orientation of each stimulus more correctly with dynamic binaural signals produced by listener's head or steering-wheel rotation than with static binaural signals and with dynamic binaural signals produced by an experimenter. Results suggest that the dynamic binaural signal associated with listener's voluntary movement play crucial role in sound localization.

2:40

3pPP6. Cue weighting and vestibular mediation of temporal dynamics in sound localization via head rotation. Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., Elborn College, 2262, 1201 Western Rd., London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

Our studies have quantified the salience and weighting of dynamic acoustic cues in sound localization via head rotation. Results support three key findings: (1) low-frequency interaural time-difference (ITD) is the dominant dynamic binaural difference cue; (2) when available, high-frequency spectral cues dominate front/rear localization; and (3) the temporal dynamics of dynamic cue processing are particular to auditory-vestibular integration. ITD dominance is shown indirectly in findings that head movements are highly effective for localizing low-frequency targets but not narrow-band high-frequency targets and that only normal low-frequency hearing is required to localize via dynamic cues. Direct evidence comes from manipulation of dynamic binaural cues in spherical-head simulations lacking spectral cues. If the stimulus provides access to dominant high-frequency spectral cues, location illusions involving head-coupled source motion fail. For low-frequency targets, localization performance improves with increasing head-turn angle, but decreases with increasing velocity such that performance depends primarily on stimulus duration; ~100 ms being required for accurate front/back localization. That duration threshold only applies in dynamic localization tasks, and not in auditory-only tasks involving the same stimuli. Correct spatial interpretation of dynamic acoustic cues appears to require vestibular information about head motion, thus the 100-ms temporal threshold is likely a property of vestibular-auditory integration.

Session 3pSAa

Structural Acoustics and Vibration, Noise, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials II

Yun Jing, Cochair
Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695

Dean Capone, Cochair
Penn State, P.O. Box 30, State College, PA PA

Contributed Papers

1:00

3pSAa1. Broadband transparent periodic acoustic structures. Gregory J. Orris (Acoust. Div., U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, gregory.orris@nrl.navy.mil), Christopher N. Layman, Christina J. Naify (National Res. Council, Washington, DC), Theodore P. Martin, and David C. Calvo (Acoust. Div., U.S. Naval Res. Lab., Washington, DC)

The creation of acoustically transparent materials is of interest for enhanced energy focusing in metamaterial lenses, vibration isolation and structure concealment in underwater environments. It has previously been shown that metal pentamode metamaterials may provide water like behavior yet retain enough shear to provide structural stability. This is achieved through a periodic lattice with sub-wavelength cells. The current talk presents a study on the simulated and experimental behavior of other structural materials that operate similar to pentamode composites, which exhibit transparency in underwater conditions. Specifically, we examine a two-dimensional design composed of a honeycomb arrangement of fluid inclusions in a solid polymer background. Materials, designs and simulations are supported through both band structure calculations and transmission modeling of finite slabs of the device. Experiments are performed in a water-filled test tank and broad frequency behavior is examined. [Work supported by ONR.]

1:20

3pSAa2. Evaluation of an acoustic metamaterial leaky-wave antenna. Christina J. Naify, Christopher N. Layman (National Res. Council, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil), Theodore P. Martin, David Calvo, and Gregory J. Orris (Acoustics, Naval Res. Lab., Washington, DC)

An acoustic projector array, which can be steered between ± 90 degrees backfire to endfire directions based solely on input frequency, is presented using a combination of transmission line (TL) analysis and negative index metamaterial ideas. An acoustic version of a leaky wave antenna, this TL structure is composed of acoustically loaded membranes (acoustic masses) and open channels (acoustic shunts). This type of TL structure had been shown previously to have broadband negative index behavior below a cutoff frequency, and positive index behavior above the cutoff frequency. By carefully designing the geometry of the acoustic elements, continuous scanning with no acoustic bandgap was achieved. The fast-wave radiation band of the antenna was determined using a lumped acoustic parameter method. Angle of radiation of the acoustic waves out of the acoustic shunts was continually scanned backfire-to-endfire, including broadside. Applications of this antenna structure include both source and sensing technologies. Finite element analyses and acoustic circuit analysis were used to predict the angle of radiation of the antenna which agreed with experimentally obtained results. [Work supported by the Office of Naval Research.]

1:40

3pSAa3. Underwater sound transmission through thin soft elastomers containing arrays of pancake voids: Measurements and modeling. David C. Calvo, Abel L. Thangawng (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, david.calvo@nrl.navy.mil), and Christopher N. Layman (NRC Postdoctoral Fellow, Naval Res. Lab., Washington, DC)

Measurement of underwater sound transmission through thin (~750 microns) layers of the soft elastomer polydimethylsiloxane (PDMS) containing microfabricated arrays of pancake-shaped cavities is presented. Cavities are 120 microns in diameter and 2.5 microns in height with a nominal lattice spacing of 300 microns. A sound transmission minimum is found at 282 kHz which agrees with predictions of a finite-element model of the array and the value for monopole resonance frequency of an air-filled single pancake cavity in unbounded PDMS. This resonance is a factor of 0.62 lower than the null that would occur for spherical cavities of equivalent volume. The width of the minimum is also significantly broader than that which would be obtained with spherical voids. Modeling results incorporate careful measurements of attenuation for both shear and compression waves in PDMS done in a separate effort. Acoustic transmission variation as a function of lattice spacing and the number of layers is discussed. We also present measurements of transmission through PDMS layers featuring randomly positioned (but not overlapping) pancake cavities to evaluate how the lattice constant (or lack thereof) affects sound transmission near the pancake resonance frequency or in higher acoustic bandwidths. [Work sponsored by the Office of Naval Research.]

2:00

3pSAa4. Broadband acoustic metamaterials with electro-magnetically controlled properties. Dimitri Donskoy (Civil, Environ., and Ocean Eng. Dept., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, ddonskoy@stevens.edu) and Vladimir Malinovsky (Dept. of Phys. and Eng. Phys., Stevens Inst. of Technol., Hoboken, NJ)

The proposed class of acoustic metamaterials utilizes clouds of electrically charged nano or micro particles exposed to an external magnetic field. The particles are also elastically supported or embedded into elastically compliant medium in a way that the designed structure exhibits two resonances: mechanical spring-mass resonance and electro-magnetic cyclotron resonance. It is shown that if the cyclotron frequency is greater than the mechanical resonance frequency, the designed structure could be highly attenuative (40–60 dB) for vibration and sound waves in a very broad frequency range covering low and very low frequencies. The approach opens up wide range of opportunities for design of adaptive acoustic metamaterials by controlling magnetic field and/or electrical charges.

2:20

3pSAa5. Broadband directional ultrasound propagation using sonic crystal and nonlinear medium. Dipen N. Sinha and Cristian Pantea (Mater. Phys. & Applications, Los Alamos National Lab., MPA-11 D429, P.O. Box 1663, Los Alamos, NM 87545, sinha@lanl.gov)

The development of a passive, sonic crystal-based device with unusual properties will be reported. This device combines a 1D sonic crystal, a nonlinear medium, and an acoustic low pass filter to allow broadband ultrasound propagation as a collimated beam for specialized underwater communication. The signal to be transmitted is first amplitude modulated with a high-frequency ultrasonic carrier wave and applied to one side of the device. The device then demodulates this signal and the original low frequency signal appears as a collimated beam on the other side. The sonic crystal provides a band pass acoustic filter through which the high-frequency ultrasonic signal can pass through and the nonlinear medium then demodulates the signal and also generates the low frequency sound beam through the parametric array concept. The low pass filter removes any remaining high frequency components. The device also functions in a unidirectional manner. Design details of the device and experimental data will be presented.

2:40

3pSAa6. Equations for energy characteristics of oscillatory systems with internal (hidden) degrees of freedom and application to acoustic metamaterials. Yuri Bobrovnitskii (Theor. and Appl. Acoust., Blagonravov Mech. Eng. Res. Inst., 4, Griboedov Str., Moscow 101990, Russian Federation, yuri@imash.ac.ru)

General equations are derived for calculating the kinetic and potential energies and other energy characteristics of linear oscillatory NDOF-systems a portion of DOFs of which are internal or inaccessible for measurement and excluded from consideration. The energy characteristics are expressed through parameters pertaining only to the input or accessible DOFs. The equations are based on the certain novel properties of the so-called Shur matrix complement. The theory is applied to calculating the energy characteristics of acoustic metamaterials for which this is still an unsolved problem, especially for those with negative effective density and stiffness. A metamaterial is thought as a medium or periodic structure in which the role of effective inertia and elastic elements is played by sufficiently complex oscillatory systems with internal DOFs. Applying the derived equations to a cell of periodicity of a metamaterial one can obtain the exact values of the needed energy characteristics. The theory is verified in computer simulation and laboratory experiment.

WEDNESDAY AFTERNOON, 5 JUNE 2013 — 512BF, 1:00 P.M. TO 3:00 P.M.

Session 3pSAb

Structural Acoustics and Vibration: Applications in Structural Acoustics and Vibration I

Robert M. Koch, Cochair
Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Eric E. Ungar, Cochair
Acentech, Inc., 33 Moulton St., Cambridge, MA 02138-1118

3p WED. PM

Contributed Papers

1:00

3pSAb1. Squeal noise generated by railway disc brakes: Experiments and stability computations on large industrial models. Olivier Chiello (LTE, IFSTTAR, 25 avenue François Mitterrand, Case 24, Bron Cedex F-69675, France, olivier.chiello@ifsttar.fr), Jean-Jacques Sinou (Laboratoire de Tribologie et de Dynamique des Systèmes, Ecole Centrale de Lyon, Ecully, France), Nicolas Vincent (Vibratec, Ecully, France), Guillaume Vermot des Roches (SDTools, Paris, France), Franck Cocheteux, Selim Bellaj (Agence d'Essai Ferroviaire, SNCF, Vitry-sur-Seine, France), and Xavier Lorang (Innovative and Res. Dept., Phys. of Railway System and Passenger Comfort, SNCF, Paris, France)

The squeal noise generated by railway disk brakes is an everyday source of discomfort for the passengers both inside and outside the trains in stations. The development of silent brake components is needed and requires a better characterization and understanding of the phenomenon. This is the aim of the experimental and numerical investigations performed in the framework of the French AcouFren project and presented in this paper. The first part is concerned with the analysis of experimental data coming from bench tests in a lot of braking configurations including different brake pads. In the second part, the measurements are compared with the results of a large FE model of the brake taking into account the mechanical complexity of each component, especially the brake pads. Components models have been previously updated using experimental modal analysis but the whole model is a direct assembling of it, without updating. The assumption of unilateral contact and Coulomb friction at the pad/disc interface is sufficient to destabilize the sliding equilibrium of the brake and lead to self-sustained vibrations. Complex vibrating modes are computed in order to describe and understand the dynamic instabilities.

1:20

3pSAb2. The modeling of wheel squeal in the time domain and its validation. Xiaogang Liu and Paul A. Meehan (School of Mech. and Mining Eng., Univ. of Queensland, St Lucia, Brisbane, QLD QLD 4072, Australia, xiaogang.liu@uq.edu.au)

Wheel squeal is a tonal noise generated when a train negotiates a curve, whose sound pressure level is normally 30 dB above rolling noise. The sound pressure level of wheel squeal has been shown to increase with the angle of attack and rolling speed in both field and laboratory tests, but the causes behind the manner of increase are still unknown. To investigate this, a model in the time domain was developed by integrating the contact mechanics with the vibration of the wheel to demonstrate how the nonlinear friction creep behavior interacts with the wheel vibration. This model simulated the vibration velocity of the testrig wheel at different rolling speed and angle of attack. The results correlate well with the recorded sound pressure level of wheel squeal. The lateral creepage and lateral force in various situations were also simulated. It was found that due to the interaction of wheel vibration with lateral force and lateral creepage the vibration velocity amplitude of the wheel at a high angle of attack and rolling speed is larger. This explains why the sound pressure level of wheel squeal also increases in the same manner. The phenomenon is explained theoretically using the mechanics based model.

1:40

3pSAb3. The role of pad-modes and nonlinearity in instantaneous mode squeal. Sebastian M. Oberst and Joseph C.S. Lai (Acoust. & Vib. Unit, School of Eng. and Information Technol., UNSW Canberra, UNSW Canberra, Northcott Dr. bld 15/117, Canberra, ACT 2600, Australia, s.oberst@adfa.edu.au)

Disc brake squeal is a major source of customer dissatisfaction and related warranty costs for automobile manufacturers. Although mode coupling is recognized as a mechanism often found in squealing brakes, recent research results show that friction induced pad-mode instabilities could be the cause of instantaneous mode squeal reported in the literature. In this paper, the nonlinear characteristics of instantaneous mode squeal initiated by pad-mode instabilities are studied by analyzing phase space plots of vibrations and sound pressure for a numerical model of a pad-on-plate system as the friction coefficient increases. Results show that as the friction coefficient increases from 0.05 to 0.65, attractors of vibration in the phase space transits from limit cycle to quasi-periodic, showing signs of approaching chaotic behavior. It is shown here that the correlation of the sound pressure behavior in the phase-space with structural vibration is crucial to understanding the role of pad modes and nonlinearity in instantaneous mode squeal.

2:00

3pSAb4. Squeak and rattle noise prediction for trimmed door of a car using hybrid statistical energy—Finite element method analysis. Sajjad Beigmoradi (Automotive Eng. Dept., Iran Univ. of Sci. & Technol., No13, Emmami alley, Golzarand Alley, Safdari St. Navab Safavi St, Tehran, Iran, s.beigmorady@gmail.com), Kambiz Jahani (Mech. Eng. Dept., Sharif Univ. of Technol., Tehran, Iran), and Hassan Hajabdollahi (Mech. Eng. Dept., Iran Univ. of Sci. & Technol., Tehran, Iran)

Squeak and rattle (S&R) noise are important in-cabin sources of annoyance for occupants. Originally, S&R is generated as a result of colliding and slamming of car's trim and body structure, which in turn occurs because of the dynamic displacements of components excited by road and powertrain. Squeak noise is interpreted as periodic stick and slip phenomena in the contact boundary of two neighbor surfaces. While rattle noise is the emitted noise when adjacent surfaces collide and impact. Both squeak and rattle phenomena happen in high-frequency range, even though the excitation sources (road and powertrain line) works in low frequency range. In practice, squeak and rattle noise can be minimized via controlling the gaps through a tolerance analysis, as well as the appropriate choice of materials. In this research, potential S&R sources are investigated for the trimmed door of a car using clearance analysis. Impact statistics and overall force level at potential rattle places are calculated through random vibration excitation analyses and afterwards, acoustic sensitivity, overall acoustic response, and loudness are calculated by the aim of hybrid SEA-FE method. The results of this prediction will be used in noise and vibration control plan of the whole car in design phase.

2:20

3pSAb5. Intrinsic characterization of structure-borne sound sources and isolators from *in-situ* measurements. Andy Moorhouse, Andy Elliott (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg., Salford M20 1JJ, United Kingdom, a.t.moorhouse@salford.ac.uk), and Yong Hwa Heo (Ctr. for Noise and Vib. Control, KAIST, Daejeon, South Korea)

The paper addresses the problem of how to characterize vibration sources and isolators with measurements made *in-situ*, either on a working installation or on a test bench. For example, automotive components are often characterized by test bench measurements, but there is a need to know how they will behave when coupled to components with potentially different properties in a vehicle. Ideally all components should be characterized by intrinsic properties, which can then be transferred to other installations. In the paper, some novel *in-situ* measurement methods for obtaining these properties are presented. First, the active properties of a source are characterized by the blocked force measured *in situ*. Structural dynamic properties are represented by frequency response functions (mobilities) and it is shown how, if necessary, these may be obtained by indirect measurements, for example when access to measurement points is difficult. New results for dynamic stiffness of isolators are then presented obtained using a novel *in-situ* measurement approach. The method allows rotational (moment) as well as translational (force) dynamic stiffness to be obtained over a wider frequency range than many test rigs. Results are validated by measurement on an ideal laboratory structure.

2:40

3pSAb6. A320 flight deck shape effect on turbulent boundary layer auto-spectrum. Olivier Collery, Manuel Etchessahar, and Miloud Alaoui (Acoust. and Environment Dept., AIRBUS OPERATIONS SAS, 316 route de Bayonne, Toulouse 31060, France, olivier.collery@airbus.com)

The turbulent boundary layer excitation is one of the main sources of aircraft interior noise over a large frequency range and in particular in the flight deck where its geometry drives the physics of the turbulences. The present study investigates measured turbulent boundary layer auto-spectrum properties. For that purpose an A320 has been instrumented with 15 flush-mounted microphones on the upper right quarter of the flight deck. The flight test campaign has been performed in cruise conditions between 27,000 and 39,000 ft with Mach numbers from 0.7 up to 0.82. Analysis of measured data shows strong shape effect on the aerodynamic excitation near windows area. This study points out that advanced numerical tools are required to model these complex aerodynamic phenomena.

Session 3pUW

Underwater Acoustics and Signal Processing in Acoustics: Underwater Acoustic Communications

Mohsen Badiey, Chair
School of Marine Sci. and Policy, Univ. of Delaware, 114 Robinson Hall, Newark, DE 19716

Contributed Papers

1:00

3pUW1. Multiband transmissions for underwater acoustic communication. Aijun Song and Mohsen Badiey (School of Marine Sci. and Policy, Univ. of Delaware, 114 Robinson Hall, Newark, DE 19716, ajsong@udel.edu)

Underwater acoustic communication is important to a variety of scientific and commercial missions in the ocean, for example ocean exploration and monitoring. Due to hardware limitations, often limited bandwidth, for example, 6–7 kHz, has been used in underwater communication systems. In addition to high spectral efficiency, large bandwidth can also lead to increased data rates. When utilizing a large frequency band, frequency-division multiplexing, which refers to dividing an available frequency band into smaller sub-bands, is the common practice. We propose to use multiband transmissions for the underwater acoustic channel, where the wide frequency band is divided into multiple separated sub-bands. The sub-band is much wider than the sub-carrier in the orthogonal frequency-division multiplexing (OFDM). The former is several kilohertz in width while the latter is often only tens of hertz. During our experiment in Hawaii in 2011, high data rates were achieved through the use of multiband transmissions, combined with time reversal demodulation. In the meeting, we will present the receiver algorithms for single- and multi-source acoustic communication systems in the multiband transmission framework. Comparison between the multiband transmissions and OFDM schemes will be also discussed. [Work supported by ONR Code 322OA.]

1:20

3pUW2. Application of differential amplitude and phase-shift keying in underwater acoustic communication based on orthogonal frequency division multiplexing. Pan Zhengrong, Wang Chi, Han Xiao (Sci. and Technol. on Underwater Acoust. Lab., Harbin, Heilongjiang, China), and Yin Jingwei (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China, yinjingwei@hrbeu.edu.cn)

With the increase of of marine resource and underwater users, developing a high-bit-rate underwater acoustic communication has become a hot topic. Amplitude and phase-shift keying (APSK) is a modulation technique having high efficiency in spectrum utilization, it attracts more and more attention in high-bit-rate underwater acoustic communication. Differential APSK (DAPSK) is modulated using differential amplitude and phase code in time domain, and it has higher bandwidth efficiency and be easier to realize the system than APSK. A transmission system based on DAPSK modulation and OFDM is presented to solve the problems of effectiveness and reliability in high-bit-rate underwater acoustic communication. Simulation results show that the system using DAPSK modulation has a better performance than those using APSK and QAM modulation.

1:40

3pUW3. Study on Doppler effects estimate in underwater acoustic communication. Zhang Xiao, Han Xiao (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China), Yin Jingwei (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China, yinjingwei@hrbeu.edu.cn), and Sheng Xueli (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China)

Two estimate methods of the Doppler effects in mobile underwater acoustic communication have been proposed. For the first method the Doppler coefficients are obtained by estimating frequency change of CW impulse signal with notch filter. For the other method the Doppler coefficients are obtained by estimating chirp rate change of LFM signal with fractional Fourier transform (FRFT). And the performance of the Doppler effects estimation method based on notch filter or the FRFT are compared by the computer simulation. The advantages, shortcoming and the application occasion of the two methods are elaborated. The effectiveness and robustness of the two methods have been proved. Key Word: Notch Filter, Fractional Fourier transform, Doppler Effects, Underwater Acoustic Communication

2:00

3pUW4. Influence of multipaths on coherent acoustic communication in shallow water channel. Su-Uk Son and Jee Woong Choi (Dept. of Marine Sci. and Convergence Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 426-791, South Korea, suuk2@hanyang.ac.kr)

In shallow water communication channel, acoustic interactions with sea surface and bottom interfaces cause the inter-symbol interference that hinders the efficient and reliable communication. In this case, signal-to-multipath ratio (SMR) rather than signal-to-noise ratio can be used as an indicator to describe the quality of the communication channel. However, it is difficult to estimate precisely the SMR from the measured communication data. In this talk, we propose the energy fraction of the channel impulse response existing within one symbol duration as an alternative to SMR. Communication experiment was conducted on the southern coast of Korea in waters 45 m deep in source-receiver ranges of 100 m to 1 km. The bit-error-rate performance is compared to the energy fraction in one symbol duration. In addition, the correlation between the energy fraction in a symbol and SMR is investigated through a Monte Carlo simulation. [Work supported by ADD (Agency for Defense Development, Korea).]

2:20

3pUW5. Study on time reverse mirror in underwater acoustic communication. Yin Jingwei (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China, yinjingwei@hrbeu.edu.cn), Du Pengyu (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China), Shen Jianwen (Kunming Shipbuilding Electron. Equipment Co. Ltd, Kunming, China), and Guo Longxiang (Sci. and Technol. on Underwater Acoust. Lab., Harbin, China)

Time reversal mirror (TRM) can adaptively match the sound channel without any prior knowledge. In this paper, active TRM, passive TRM and virtual TRM which are all based on a single array element and the application of TRM in the underwater acoustic communication including single-user communication and multi-user communication are studied. Single sensor TRM which has time compression performance lacks array processing space gain, however, it can meet the requirements for underwater acoustic communication nodes being in the pursuit of simple structure and low power consumption. It is verified that TRM could focus multipath signal and achieve real-time adaptive channel equalization through computer simulation and test results, which could suppress the inter-symbol interference(ISI) and improve the signal-to-noise ratio(SNR).

2:40

3pUW6. Study of underwater speech coding technique based on contact conduction transmitter. Xuelli Sheng, Ye Bai, Jia Lu, Jin Han, and Weijia Dong (Harbin Eng. Univ., Shuisheng Bldg., 803#, Harbin, China, shengxueli@yahoo.com.cn)

Low bit rate speech coding of 2.4 kbps is actualized by mixed excitation linear prediction algorithm, which meets the requirement of high data rate of underwater speech communication system and the limit of high communication rate of underwater acoustic communication technique. In the process of speech coding, a great deal of speech data is compressed to high speed underwater acoustic communication, and the main characteristics of speaker are remained perfectly. Meanwhile, the capability of avoiding interference is improved obviously by the speech based on contact conduction transmitter as input signal. This underwater speech coding technique and differential OFDM technique are combined and experimented under 434, 1310, and 2000 m in the lake. The real-time transmission of speech signal is presented in the condition of complicated multi-path in the extremely shallow sea. The results show the synthesized speech has well quality of intelligibility and clarity, which satisfies the demand of underwater speech coding.

Plenary Session and Awards Ceremony

David L. Bradley
President, Acoustical Society of America

Christian Giguère
President, Canadian Acoustical Association

Michael Vorländer
President, International Commission for Acoustics

Acoustical Society of America

Presentation of ASA Fellowship Certificates

Peter F. Assmann – For contributions to vowel perception and the influence of talker variability on speech patterns
Li Cheng – For contributions to vibroacoustic modeling of complex structures
M. Patrick Feeney – For contributions to clinical middle-ear function through wideband reflectance
Eric W. Healy – For contributions of spectral-temporal analysis in speech perception
Philip X. Joris – For contributions to neural encoding in binaural hearing
Michael V. Scanlon – For contributions to the development of systems to detect and localize transient sounds in air
Michael Verslusis – For contributions to high speed imaging of fine scale acoustic phenomena

Presentation of Acoustical Society of America Awards

William and Christine Hartmann Prize in Auditory Neuroscience to Tom C. T. Yin
Medwin Prize in Acoustical Oceanography to Philippe Roux
R. Bruce Lindsay Award to Eleanor P. J. Stride
von Bèkésy Medal to M. Charles Liberman
Helmholtz-Rayleigh Interdisciplinary Silver Medal to Timothy J. Leighton
Gold Medal to Lawrence A. Crum

Canadian Acoustical Association

Announcement of Canadian Acoustical Association Award Recipients

International Commission for Acoustics

Presentation of the ICA Early Career Award to Tapio Lokki

Session 3eED

Education in Acoustics: Women in Acoustics—Listen Up and Get Involved

Tracianne B. Neilsen, Cochair
Brigham Young Univ., N311 ESC, Provo, UT 84602

Marcia J. Isakson, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713

This workshop for Montreal area Pathfinder Girl Guides (ages 12–18) consists of a hands-on tutorial, interactive demonstrations, and a panel discussion about careers in acoustics. The primary goals of this workshop are to expose the girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please email Traci Neilsen (tbn@byu.ed) if you hav time to help with either guiding the girls through the tutorial led by Wendy Adams (5:00 p.m.–6:15 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:45 p.m.–7:30 p.m.). We will provide many demonstrations, but feel free to contact us if you would like to bring your own.

WEDNESDAY EVENING, 5 JUNE 2013 7:00 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees on the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Wednesday are as follows:

7:30 p.m. Biomedical Acoustics	519a
7:00 p.m. Signal Processing in Acoustics	510a

Session 4aID

Interdisciplinary: Plenary Lecture: Sensory Evaluation of Concert Hall Acoustics

Michael Vorländer, Chair
ITA, RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany

Chair's Introduction—7:55

Invited Paper

8:00

4aID1. Sensory evaluation of concert hall acoustics. Tapio Lokki (Media Technol., Aalto Univ., P.O. Box 15500, Aalto FI-00076, Finland, Tapio.Lokki@aalto.fi)

Consumer products can be perceived in many ways, and individual taste influences quality judgments. Sensory evaluation techniques have been developed to reveal detailed information about perception of products; recently, sensory evaluation has also been shown to be very useful in subjective evaluation of concert hall acoustics. In particular, individual vocabulary based methods have helped to disentangle the detailed perceptual differences between the different seats within a hall and between different halls. For simultaneous and accurate comparison of acoustics, a symphony orchestra needs to play identically in each hall. Therefore, a symphony orchestra simulator has been developed. It consists of 34 loudspeakers reproducing synchronized recordings of individual musicians playing parts of symphonies in an anechoic chamber. In addition, an advanced spatial sound recording technique via impulse responses from a 3D microphone array is applied to reproduce the acoustics of a concert hall in laboratory conditions. Analysis of spatial impulse responses also enables spatio-temporal visualization of sound energy distributions at measured seats, thus helping us to link the physical properties of the sound to the perception and architecture of concert halls. Finally, this paper highlights our recent results to explain which perceptual characteristics of acoustics drive preference ratings.

THURSDAY MORNING, 6 JUNE 2013 513ABC, 9:00 A.M. TO 12:00 NOON

Session 4aAAa

Architectural Acoustics: Room Acoustics Computer Simulation I

Diemer de Vries, Cochair
RWTH Aachen Univ., Inst. fuer Technische Akustik, Aachen D-52056, Germany

Lauri Savioja, Cochair
Dept. of Media Technol., Aalto Univ., P.O. Box 15500, Aalto FI-00076, Finland

Invited Papers

9:00

4aAAa1. Toward a full-bandwidth numerical acoustic model. Jonathan Hargreaves and Yiu W. Lam (Acoust. Res. Ctr., School of Computing, Sci. & Eng., Univ. of Salford, Salford M5 4WT, United Kingdom, y.w.lam@salford.ac.uk)

Prediction models are at the heart of modern acoustic engineering. Current commercial room acoustic simulation software almost exclusively approximates the propagation of sound geometrically as rays or beams. These assumptions yield efficient algorithms, but the maximum accuracy they can achieve is limited by how well the geometric assumption represents sound propagation in a given space. This comprises their accuracy at low frequencies in particular. Methods that directly model wave effects are more accurate but they have a computational cost that scales with problem size and frequency, effectively limiting them to small or low frequency scenarios. This paper will report the results of initial research into a new full-bandwidth model which aims to be accurate and efficient for all frequencies; the name proposed for this is the "Wave Matching Method." This builds on the Boundary Element Method with the premise that if an appropriate interpolation scheme is designed then the model will become "geometrically dominated" at high frequencies. Other propagation modes may then be removed without significant error, yielding an algorithm which is accurate and efficient. This paper will present the general concepts of wave matching and the results from some numerical test cases.

9:20

4aAAa2. Fast multipole accelerated indirect boundary elements for the Helmholtz equation. Nail A. Gumerov, Ross Adelman, and Ramani Duraiswami (Inst. for Adv. Comput. Studies, Univ. of Maryland, 115 A.V. Williams Bldg., College Park, MD 20742, gumerov@umiacs.umd.edu)

The indirect boundary element method for the Helmholtz equation in three dimensions is of great interest and practical value for many problems in acoustics as it is capable of treating infinitely thin plates and allows coupling of interior and exterior scattering problems. In the present paper, we provide a new approach for treatment of boundary integrals, including hypersingular, singular, and nearly singular integrals via analytical expressions for generic boundary conditions on the both sides of the surface. The fast multipole accelerated boundary element solver in Gumerov and Duraiswami (2009) is extended to incorporate the indirect formulation. The new formulation is compared with the analytical solution of scattering off a disk. Previous authors have not provided such comparisons for an extended range of frequencies. The performance of the method and its scalability are investigated. It is demonstrated that problems with millions of boundary elements can be solved efficiently on a personal computer using the present method.

9:40

4aAAa3. Modeling binaural receivers in finite difference simulation of room acoustics. Jonathan Sheaffer (Acoust. Res. Ctr., School of Computing, Sci. and Eng., Univ. of Salford, Salford, Salford M5 4WT, United Kingdom, j.sheaffer@edu.salford.ac.uk), Craig Webb (Acoust. Group/Edinburgh Parallel Computing Ctr., Univ. of Edinburgh, Edinburgh, United Kingdom), and Bruno Fazenda (Acoust. Res. Ctr., School of Computing, Sci. and Eng., Univ. of Salford, Salford, United Kingdom)

Binaural room impulse responses are important for auralization as well as for objective research in room acoustics. In geometrical room simulation methods, obtaining such responses is easily achieved by convolving each computed reflection tap with a corresponding pre-measured angle-dependent head-related impulse response. Unfortunately, employing such an approach in wave based methods is challenging due to temporal overlap of room reflections in the calculated response. One alternative is to physically embed a listener geometry in the grid. Whilst this method is straightforward, it requires voxelization of a geometrically complex object. Furthermore, with non-conformal boundary conditions, the voxelized geometry is sample-rate dependent, meaning that numerical consistency is compromised. In this paper, we discuss the merits and drawbacks of embedding different listener geometries in the grid, ranging from a simple rigid sphere to a fully featured laser-scan of a Kemar mannikin. We then introduce a parametric model of a human listener whose head related effects are structurally approximated by digital filters. The model is applied to simulated results in order to extrapolate a binaural response from a single pressure-velocity receiver, without the need to embed any objects in the grid. A comparative analysis of the two methods is presented, and results are discussed in light of room acoustics modeling.

10:00

4aAAa4. Validation of adaptive rectangular decomposition for three-dimensional wave-based acoustic simulation in architectural models. Lakulish Antani (Comput. Sci., Univ. of North Carolina at Chapel Hill, 1100 W NC Highway 54 Bypass Apt 25G, Chapel Hill, NC 27516, lakulish@cs.unc.edu), Anish Chandak (Impulsonic, Inc., Chapel Hill, NC), Matthew Wilkinson (Arup Acoust., Los Angeles, CA), Alban Bassuet (Arup Acoust., New York, NY), and Dinesh Manocha (Comput. Sci., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Computer-based simulation is an increasingly popular way to predict the acoustics of real-world architectural designs. Most commercial acoustic simulation tools are based on geometric techniques and cannot accurately model low-frequency diffraction and other wave phenomena. Numerical wave simulation techniques can model these effects, but are less commonly used, since they are compute- and memory-intensive, and cannot scale to large spaces. Moreover, it is challenging to ensure that numerical methods do not suffer from high dispersion errors. Recent techniques have begun to overcome these limitations. One such method is adaptive rectangular decomposition (ARD), which combines analytical solutions to the wave equation in rectangular subdomains with a finite difference stencil for interface handling between subdomains, resulting in high-performance wave simulation with low dispersion error. ARD, along with high-performance ray tracing, are available as part of Impulsonic's IPL SDK, a software development kit that allows custom acoustic simulation tools to be easily built with state-of-the-art simulation technology. In this paper, we evaluate the performance and accuracy of the IPL SDK and ARD, by analyzing simulation results and comparing them against measurements obtained for real-world architectural designs.

10:20

4aAAa5. Simulation of non-locally reacting boundaries with a single domain boundary element method. Robertus Opdam (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, rob.opdam@akustik.rwth-aachen.de), Diemer de Vries (Inst. of Tech. Acoust., RWTH Aachen Univ., Amsterdam, Netherlands), and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany)

The significance of taking into account non-locally reacting behavior of boundaries compared to the often-used locally reacting assumption in room acoustics has not been extensively investigated. To make this possible a boundary element method is developed, inspired on a seismic simulation method known as the WRW method. The novelty of this method compared to other boundary element methods (BEM) is that the calculation is performed in only one domain. There is no need for a fluid-structure coupling, which in general allows faster simulation times. The theory of the method is presented and some example structures are simulated with both locally and non-locally reacting behavior. The results are shown and discussed.

10:40

4aAAa6. Estimation of absorption coefficients values of surface materials using a diffusion equation model. Juan M. Navarro (Adv. Telecommunications Res. Group, San Antonio's Catholic Univ., Campus de los Jeronimos, s/n, Guadalupe, Murcia 30107, Spain, jmnavarro@ucam.edu), Jose J. Lopez (ITEAM, Universitat Politécnica de Valencia, Valencia, Spain), and Jose Escolano (Multimedia and Multimodal Processing Res. Group, Univ. of Jaen, Linares, Spain)

In the auralization process of an enclosure, the right definition of the acoustic properties of the materials is very important. Sometimes, the absorption coefficients of materials in a real room are not found in the literature and their measurement in the laboratory or *in-situ* are complex. Been the reverberation time of a room and its geometry known, but the absorption coefficient values of the materials that cover the room unknown, it is possible to estimate its values by means of an inverse problem using a room acoustics simulation model. Since the acoustic diffusion equation model is a fast simulation method, it can be used to perform an iterative process to estimate these values. In this paper, we propose a statistical procedure that compares actual measurement values of reverberation time with predictions obtained by the diffusion equation model. This process does an automatically adjustment whose ultimate goal is that the reverberation time predicted values do not differ from those measured in situ by more than 5%. As a preliminary work, this algorithm is tested in a cubic room obtaining satisfactory results, but can be extended to be employed in more complex geometry rooms and even with non homogeneous distribution.

11:00

4aAAa7. A diffusion equation model for investigations on acoustics in coupled-volume systems. Yun Jing (Multimedia and Multimedia Processing Res. Group, Univ. of Jaén, 911 Oval Dr., EB III, Campus box 7910, Raleigh, North Carolina 27695, yjing2@ncsu.edu), Ning Xiang (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), Juan M. Navarro (Polytechnic Sci. Dept., San Antonio's Catholic Univ., Murcia, Murcia, Spain), and Yun Jing (Dept. of Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

Coupled-volume rooms remain as one of the most exciting and challenging research lines in room acoustics. Their benefits lie on their multiple-slope energy decay profiles, being of interest in many current concert halls. However, so far there is no consistent predictive model being able to help architects and acousticians in selecting appropriate design parameters. This work is devoted to studying effects of aperture-size and source/receiver positions on the energy decay characteristics. For this purpose, a diffusion equation model is used to model a coupled-volume system, providing an effective tool for analysis. The diffusion equation model is first validated by experimental investigations using scale models. Bayesian energy decay analysis is applied to the results of both the acoustical scale model and the diffusion-equation model to provide deeper insight in the energy decay characteristics and their dependence on the aperture sizes and the sound source/receiver positions.

11:20

4aAAa8. Time-domain formulation of an edge source integral equation. U. Peter Svensson (Dept. of Electron. and Telecommunication, Norwegian Univ. of Sci. and Technol., O.S. Bragstads pl. 2B, Trondheim NO-7491, Norway, svensson@iet.ntnu.no) and Andreas Asheim (Dept. of Comput. Sci., Katholieke Universiteit, Leuven, Belgium)

In computer simulations of sound in enclosures, diffraction components can be added to geometrical acoustics ones for increased accuracy. A computational problem with diffraction is the large number of higher-order terms that is generated. A recent frequency-domain edge source integral equation (ESIE) efficiently handles the sum of all higher-order diffraction for rigid, external scattering objects, while computing first-order diffraction separately. Here, a time-domain formulation of the same ESIE is presented. An initial version handles higher-order diffraction for separate scattering objects, such as stage ceiling reflectors, and the extension to general geometries is outlined. With this approach, in a first step, an incident transient sound field is computed at discretized edge points, including the outgoing directivity. In a second step, the effects of diffraction of arbitrarily high order is handled by solving the IE iteratively, yielding the complete edge source time signals. In a third step, the edge source signals are propagated to receiver points. Numerical issues will be discussed, including discretization strategies and how to handle shadow zone boundary singularities.

11:40

4aAAa9. The contributions of pairs of parallel surfaces in a simple analytical model of room reverberation. Jean-Jacques Embrechts (Intelsig Res. Group, Univ. of Liege, Campus du Sart-Tilman B28, Institut Montefiore, Liege 4000, Belgium, jjembrechts@ulg.ac.be)

In a recent paper [Embrechts, "Searching for a theoretical relation between reverberation and the scattering coefficients of surfaces in a room," in *Proceedings of the Acoustics 2012 Nantes Conference* (2012), 2397–2402], we derived an analytical model of the sound energy decay in a room from the acoustic radiative transfer equation. This model includes the surfaces' scattering properties and it is presently valid for rooms in which the cloud of image sources is approximately isotropic and constant for all receptor's positions. Its validity is extended in this paper by the inclusion of a pair of parallel surfaces. Indeed, it is known that parallel surfaces can introduce significant anisotropy in the cloud of image sources. We show that the room reverberation can be represented by a sum of exponential decays (except in its early part), each decay having its specific slope and amplitude depending on the surfaces' absorption and scattering properties. It is also shown how this simple model can be applied to speed up geometrical acoustics computer simulations.

Session 4aAAb

Architectural Acoustics and Psychological and Physiological Acoustics: Methods and Materials That Improve Speech Intelligibility for the Elderly and Hearing Impaired

Bonnie Schnitta, Chair
SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937

Chair's Introduction—8:55

Invited Papers

9:00

4aAAb1. Achieving optimal reverberation time in a room, using newly patented tuning tubes. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

The aging population has several acoustic requirements in order to optimize their ability to hear better, as well as feel better, in a room. First, there is the requirement that the reverberation time of the room must be reduced in order to assist in hearing. Secondly, there needs to be a lowering of the NC in order to increase the SNR. Increasing the SNR not only helps to assist in hearing, but also reduces some hearing aid problems. In addition to these two standard room requirements, there is also a need for the reduction of lower frequency sounds within a room, such as sounds typical of mechanical equipment. Recent data support the fact that there is a correlation between certain diseases and low frequency intolerance. Since standard products used to absorb sound have a greater absorption in speech frequencies there is a need for products that have greater absorption in lower frequencies. Ideally, these products should also be washable. This paper presents detail on each of this requirements, as well as recommendations on methods to achieve each requirement.

9:20

4aAAb2. Optimizing the signal to noise ratio in speech rooms using passive acoustics. Peter D'Antonio (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Malboro, MD, pdantonio@rpginc.com)

Adults with normal hearing require roughly a 0 dB signal-to-noise ratio for good speech intelligibility. However, significantly higher values may be needed to compensate for neurological immaturity, sensorineural and conductive hearing losses, language proficiency and excessive reverberation. ANSI 12.60 addresses ways to lower the noise interference due to background levels and reverberation time. However, it is also possible to increase the signal, by reflecting or diffusing early reflections. Speech intelligibility is delivered in the consonants, which occur in the 2–6 kHz frequency range. Therefore, intelligibility can be enhanced by incorporating scattering surfaces, rather than solely surfaces that absorb sound in the 2–6 kHz region, on the front wall, lower side walls, and central ceiling areas, to increase the speech signal by temporal fusion. The decay time can be controlled with broadband absorption on the perimeter of the ceiling and upper wall surfaces. Since ceiling diffusion is an important design ingredient and the ceiling plane is coveted by many trades, including lighting, HVAC, speakers, sprinklers, etc., we will describe a 24 VDC combined LED lighting and sound diffusor, with a 24 VDC emergency lighting central battery system, dynamic lighting capability, and the ability to incorporate sonic actuators for announcements.

9:40

4aAAb3. A holistic approach to room design for hearing impaired populations. Jennifer Levins (Independent, 2669 E Thompson St., Philadelphia, PA 19125, jenlevins@gmail.com)

Acoustic design considerations for hearing impaired populations are widely misunderstood outside of the acoustics community. Some clients have even expressed the sentiment that room acoustics are not important because their patrons are hard of hearing. Contrary to this widely held belief, acoustical design is more critical for these populations. To properly design spaces for these communities, it is imperative to take a holistic approach, which considers not just architectural acoustics, but incorporates an understanding of the biological and psychological components of hearing impairment. It is also important to consider how room systems can be integrated with modern hearing technology. Addressing room acoustics, background sound levels, and audio technology should all be considered in the strategy of designing for hearing impaired persons. This is important not only for their comfort, but also for their health. Strategies and implications for a holistic approach will be discussed.

10:00

4aAAb4. Reduction in reverberation time, resulting from acoustic treatment behind the final surface layer of plywood or drywall. Steve Mittendorf (Mittendorf Quality Construction, 2552 5th Ave. West, Seattle, WA 98119, steve@mittqc.com)

When sound energy generated in room strikes a surface, it is partially reflected, partially transmitted, and partially absorbed. This is true for each layer of material in a wall, ceiling, or floor. The wave interaction with the surface depends on many factors, but the main factors that are typically involved in calculations of reverberation time are the frequencies of concern, the rigidity, and density of the surfaces, and the absorption of various objects in the room. For the case of an empty room, the estimation of the reverberation time is simplified down to the absorptive properties of the surfaces. This paper presents results that show the importance of considering the composition of the surface (wall, floor, or ceiling) including materials located behind the exposed surfaces. Specifically, it will be demonstrated that a properly installed layer of a loaded vinyl sheeting under the final wall surface layer of drywall will produce a significant reduction in the room reverberation time. With this technique, the preferred reverberation time can be achieved more naturally, while accommodating design constraints such as washable surfaces and minimizing the amount of additional surface treatments required. An additional benefit in the use of the loaded vinyl product behind the surface is a significant improvement in the STC of the wall or ceiling in which it was installed.

Contributed Papers

10:20

4aAAb5. The sensitivity of hearing-impaired adults to acoustic attributes in simulated rooms. William M. Whitmer, David McShefferty, and Michael A. Akeroyd (Inst. of Hearing Res. (Scottish Section), Med. Res. Council, Glasgow Royal Infirmary, Glasgow G42 9UA, United Kingdom, bill@ihr.gla.ac.uk)

In previous studies, we have shown that older hearing-impaired individuals are relatively insensitive to changes in the apparent width of broadband noises when those width changes were based on differences in interaural coherence [Whitmer *et al*., J. Acoust. Soc. Am. **132**, 369–379 (2012)]. This insensitivity has been linked to senescent difficulties in resolving binaural fine-structure differences. It is therefore possible that interaural coherence, despite its widespread use, may not be the best acoustic surrogate of spatial perception for the aged and impaired. To test this, we simulated the room impulse responses for various acoustic scenarios with differing coherence and lateral (energy) fraction attributes using room modeling software (ODEON). Bilaterally impaired adult participants were asked to sketch the perceived size of speech tokens and musical excerpts that were convolved with these impulse responses and presented to them in a sound-dampened enclosure through a 24-loudspeaker array. Participants' binaural acuity was also measured using an interaural phase discrimination task. Corroborating our previous findings, the results showed less sensitivity to interaural coherence in the auditory source width judgments of older hearing-impaired individuals, indicating that alternate acoustic measurements in the design of spaces for the elderly may be necessary.

10:40

4aAAb6. Still able. Trent Still and Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, William.T.Still-1@ou.edu)

In a world where most students are habitually connected to headphones, one student is harnessing power outside the sense of hearing to unite acoustics and craft into particular listening environments. Trent Still is a student, a craftsman, an avid fan of acoustics, and to my surprise legally deaf in one ear. What initially could be viewed as a hindrance within the study of acoustics, has developed into an avenue of expressive talent and determination. As a student of architectural design, Still focuses on materials, connections, and overall aesthetics of the listening environment. For example: in a recent gallery exhibit of handcrafted furniture, one of Trent's entries was a pair of handmade loudspeaker enclosures that were French cleated to the wall. They were not merely wall mounted; they were wall dependent. The wall cavity between framing members and the wall finish was part of the installation; thereby actively integrating acoustics into architecture. This paper does not focus solely on one student; it is about unequivocal enthusiasm for acoustical craft within inhabitable space. No matter what seems like a disadvantage or disability, students and educators can work together to ascertain visual and auditory beauty. Sight and sound are uniquely codependent.

THURSDAY MORNING, 6 JUNE 2013 — 517B, 9:00 A.M. TO 12:00 NOON

Session 4aAAc

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition

Norman H. Philipp, Cochair
Pittsburg State Univ., 1701 S. Broadway, Pittsburg, KS 66762

Andy Miller, Cochair
BAi, LLC, 4006 Speedway, Austin, TX 78751

David Woolworth, Cochair
Oxford Acoustics, Inc., 356 CR102, Oxford, MS 38655

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund and the National Council of Acoustical Consultants are sponsoring the 2013 Student Design Competition that will be professionally judged at this meeting. The 2013 design competition involves the design of a college performance hall and related facilities primarily for a school's strong opera program. The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of USD$1,250 will be made to the submitter(s) of the design judged "first honors." Four awards of USD$700 each will be made to the submitters of four entries judged "commendation."

Session 4aAB

Animal Bioacoustics and Noise: Modeling and Measurement of Anthropogenic Noise in Marine Environments

Bruce Martin, Chair
JASCO Appl. Sci., 32 Troop Ave., Ste. 202', Dartmouth, NS B3B 1Z1, Canada

Invited Papers

10:00

4aAB1. Computing cumulative sound exposure levels from anthropogenic sources in large data sets. Bruce Martin (Halifax, JASCO Appl. Sci., 32 Troop Ave., Ste.202', Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com)

The goal of many underwater acoustic environmental assessments is to characterize the soundscape in an area before, during, or after an anthropogenic activity. The assessment determines the range of baseline noise levels from natural and anthropogenic sources and the contribution of the new anthropogenic activity. The noise levels are considered in aggregate for possible effects on the environment. It is accepted that the effects of anthropogenic noise on marine life depend on the intensity and duration of exposure, the frequency content of the sound relative to the hearing abilities of the species, and the behavior context of the species exposed to the sounds. A growing body of scientific evidence is being analyzed to establish threshold sound levels and dose-response curves for injury or behavioral disturbance effects to marine life. Recent research is also raising new questions about the most appropriate ways to compute ambient sound levels and exposure metrics. In this paper, we present our methods for quantifying ambient sound levels and anthropogenic sound levels from shipping and seismic survey activities in large data sets. We also make recommendations on how to estimate background sound levels in the presence of these sound sources.

10:20

4aAB2. Spectral probability density as a tool for marine ambient noise analysis. Nathan D. Merchant (Dept. of Phys., Univ. of Bath, Claverton Down, Bath BA2 7AY, United Kingdom, n.d.merchant@bath.ac.uk), Tim R. Barton, Paul M. Thompson, Enrico Pirotta (Univ. of Aberdeen, Lighthouse Field Station, Cromarty, United Kingdom), D. Tom Dakin, and John Dorocicz (Ocean Networks Canada, Univ. of Victoria, Victoria, BC, Canada)

The empirical probability density of the power spectral density has been successfully applied as tool to assess signal variability and sensor system performance in the seismic literature. This paper presents the application of this analysis method to underwater ambient noise measurements, and demonstrates its utility in assessing the field performance of passive acoustic monitoring systems and the statistical distribution of noise levels across the frequency spectrum. Using example datasets from an autonomous passive acoustic recorder in the Moray Firth, Scotland, UK, and a cabled subsea observatory in the Strait of Georgia, British Columbia, we show how this method can reveal data limitations such as persistent tonal components and insufficient dynamic range, and phenomena such as bimodality and outliers, which may be undetected by standard analysis techniques. We then combine this approach with conventional percentiles and spectral averages, illustrating how the underlying noise level distributions influence these metrics, and propose this technique as a standard, integrative presentation of ambient noise spectra. Finally, the paper presents cumulative probability density as a method for frequency-domain characterization of chronic noise exposure in marine acoustic habitats.

Contributed Papers

10:40

4aAB3. Global ocean soundscapes. Michael B. Porter and Laurel J. Henderson (HLS Res., 3366 N. Torrey Pines Ct., Ste. 310, La Jolla, CA 92037, mikeporter@hlsresearch.com)

There has been increasing interest in understanding the effects of human-induced noise on the marine environment. Under a variety of programs around the world, researchers are modeling "soundscapes" that depict the undersea sound fields in localized areas such as national exclusive economic zones (EEZs). In this work, we develop techniques for modeling soundscapes on a truly global scale and present as an example world maps of ship noise. The resulting soundscapes compose a database for global shipping noise. The noise due to such shipping can travel very long distances producing sort of a background haze for localized modeling in the EEZs.

11:00

4aAB4. The effects of sound in the marine environment workbench: A simulation tool to predict the impact of anthropogenic sound on marine mammals. David C. Mountain (Biomedical Eng., Boston Univ., 44 Cummington St., Boston, MA 02215, dcm@bu.edu), David Anderson, GraHam Voysey, and Andrew Brughera (Hearing Res. Ctr., Boston Univ., Boston, MA)

The Effects of Sound in the Marine Environment (ESME) Workbench is a software tool designed to predict the impact of anthropogenic sounds on marine mammals. The ESME Workbench (http://esme.bu.edu) allows the user to use site-specific environmental data such as bathymetry and sound-speed profiles to predict sound propagation in a wide range of scenarios and to record the sound exposures received by virtual animals. The acoustic propagation models use range-dependent depth profiles and depth dependent

sound speed profiles to compute the received sound level for simulated animal from each simulated source. The propagation models use bottom and sea surface characteristics to account for losses that occur during reflection at these boundaries. Sound sources are specified through parameters such as source location, frequency, intensity, and beam pattern. The animal behavior is simulated using the 3 MB animal movement model. We will provide hands-on demonstrations at the meeting for those interested in learning more about the ESME Workbench. [Funded by ONR.]

11:20

4aAB5. Behavioral responses of humpback whales to seismic air guns.
Douglas H. Cato (Defence Sci. & Technol. Organisation & Univ. of Sydney, P.O. Box 44, Pyrmont, NSW 2009, Australia, doug.cato@sydney.edu.au), Michael J. Noad, Rebecca A. Dunlop (School of Veterinary Sci., Univ. of Queensland, Gatton, QLD, Australia), Robert D. McCauley (Ctr. for Marine Sci. & Technol., Curtin Univ. of Technol., Bentley, New South Wales, Australia), Hendrik Kniest (Univ. of Newcastle, Newcastle, NSW, Australia), David Paton (Blue Planet Marine, Canberra, ACT, Australia), Chandra P. Salgado Kent (Ctr. for Marine Sci. & Technol., Curtin Univ. of Technol., Bentley, WA, Australia), and K. Curt S. Jenner (Ctr. for Whale Res., Fremantle, New South Wales, Australia)

A study of the responses of humpback whales to seismic air guns is being conducted in Australian waters and two of four major experiments have been completed. It aims to assess the impact of seismic surveys on the whales and the effectiveness of ramp-up in mitigation. In separate trials, whales were exposed to a 20 cu in air gun, ramp-up in level from 20 to 440 cu in with an air gun array, and a "hard start" of 140 cu in. Trials exposing whales to air gun treatments were balanced by controls without air guns firing. Whales were tracked from land using theodolites. Behavioral observations were made from these land stations, from three small vessels, and from the source vessel. Vocalizing whales were tracked with an array of hydrophones. Dtags were attached to some of the whales. Observations were made before, during, and after exposure. Characterization of the sound field throughout the area and the exposure at each whale were determined from propagation measurements and recordings on the hydrophone array and several moored acoustic recording systems. Some preliminary results will be discussed. [Work supported by E&P Sound & Marine Life Joint Industry Program and the U.S. Bureau of Ocean Energy Management.]

11:40

4aAB6. Prediction of noise of moored ships. Antonino Di Bella and Francesca Remigi (Dept. of Industrial Eng., Univ. of Padova, Via Venezia 1, Padova, PD 35131, Italy, antonino.dibella@unipd.it)

The European Directive 2002/49/CE suggests to map noise in harbor areas with methods mainly used for industrial noise, in accordance with ISO standards. Nevertheless, in many cases, these methods do not seem suitable to describe the effects of noise due to moored cruise ships. For noise measurements of ships, it is possible to refer to ISO 2922 standard, but it results ineffective for the estimation of noise levels at distances bigger than 25 m from the sound source and to obtain reliable information about sound power level of big vessels. The aim of this study, performed by the University of Padova on behalf of Venice Port Authority, is to improve a procedure for predicting noise of moored ships in the harbor area by the means of measures and reverse analysis with numerical models.

THURSDAY MORNING, 6 JUNE 2013 519A, 9:15 A.M. TO 11:40 A.M.

Session 4aBA

Biomedical Acoustics and Physical Acoustics: Biophysical Mechanisms of Sonoporation

Richard Manasseh, Cochair
Mech. Eng., Swinburne Univ. of Technol., P.O. Box 218, Hawthorn, VIC, Melbourne, VIC 3122, Australia

John S. Allen, Cochair
Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822

Chair's Introduction—9:15

Invited Papers

9:20

4aBA1. Size effect of complexed plasmid DNA to gene transfection efficiency of microbubble-mediated sonoporation. Yoichiro Matsumoto, Yiwei Zhang, Takashi Azuma (Mech. Eng., The Univ. of Tokyo, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, ymats@fel.t.u-tokyo.ac.jp), Kiyoshi Yoshinaka (Mech. Eng., The Univ. of Tokyo, Tsukuba, Japan), Kensuke Osada, Kazunori Kataoka, and Shu Takagi (Mech. Eng., The Univ. of Tokyo, Tokyo, Japan)

Ultrasound-mediated gene transfection in the presence of microbubbles is a recently developed promising nonviral gene delivery method. Detailed dynamics of pore opening on the cell surface has not been clarified. Especially, the pore size is one of the most essential parameters. In this study, we investigated the size effect of the complexed plasmid DNA (pDNA) on the transfection efficiency by packaging within the polyplex micelles. Both naked pDNA and complexed pDNA were transfected into cultured NIH3T3 cells using ultrasound in the presence of microbubble contrast agent, Sonazoid. The both size of the hydrodynamic diameter of naked and complexed pDNA estimated by a dynamic light scattering measurement were 600 and 120 nm, respectively. The transfection rates of the complexed pDNA evaluated by counting the number of cells that exhibited green fluorescent was 1.67%, while that of the naked pDNA was 0.92%. This efficiency enhancement depending on the size reduction showed that the pore sizes were distributed in the range of pDNA diameters. Since complexation changes the structure of pDNA in size and stability, more detailed study will be discussed in the presentation.

9:40

4aBA2. Enhancement effect of ultrasound-induced microbubble cavitation on branched polyethylenimine-mediated vascular endothelial growth factor 165 (VEGF165) transfection. Juan Tu, Qian Li, Chunbing Zhang, and Dong Zhang (Physics, Inst. of Acoust., Nanjing Univ., #22 Hankou Rd., Nanjing 210093, China, juantu@nju.edu.cn)

Angiogenesis is a complex process that is mediated by growth factor. One isoform of the vascular endothelial growth factor, VEGF165, has been reported to be a dominant mediator and regulator of angiogenic process. Branched polyethylenimine (bPEI) has been widely used as a non-viral delivery vector for gene therapy. HEK 293T cells, mixed with bPEI:VEGF165 complexes with different N/P ratios, were exposed to 1-MHz ultrasound (US) pulses. The enhancement effect of microbubble inertial cavitation (IC) on bPEI-mediated VEGF165 transfection was systemically investigated, in an effort to optimize transfection efficiency using low nitrogen:DNA phosphate (N/P) ratios. The results show that: (1) Microbubble IC activity can be quantified as an IC "dose" (ICD) and will be affected by US parameters; (2) DNA transfection efficiency initially increases with the increasing ICD, then tends to saturate instead of achieving a maximum value while ICD keeps going up; (3) the measured ICD, sonopration pore size, and cell viability exhibit high correlation among each other; and (4) microbubble IC activity has less cytotoxicity than bPEI, although a combinatorial effect of IC activity and bPEI can be observed on cell viability. All the results indicated that ICD could be used as an effective tool to monitor and control US-mediated gene/drug delivery effect, and it is possible to optimize bPEI-mediated VEGF transfection efficiency with relatively low N/P ratios by employing appropriate US parameters.

10:00

4aBA3. Time-resolved high-speed fluorescence imaging of bubble-induced sonoporation. Michel Versluis (Phys. of Fluids Group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl)

The uptake of drugs through a cell membrane is enhanced by the use of bubbles and ultrasound. Little is known about the physical mechanisms underlying the uptake at short timescales. Here we study the bubble-assisted uptake of propidium iodide (PI) by endothelial cells at a millisecond timescale using high-speed fluorescence imaging. Single microbubbles were insonified at a driving frequency of 1 MHz and at acoustic pressures varying from 200 to 1200 kPa for a duration of 10 and 100 cycles. At a pressure of 200 kPa and 10 cycles, 50% of the cells showed uptake of PI, and this percentage increased to 90% for a pressure of 400 kPa. At a pressure of 1200 kPa all cells showed uptake of PI. The high-speed fluorescence recordings revealed that a localized pore in the cell membrane is formed right at the position of the bubble. Uptake was observed within several milliseconds after insonation and the size of the induced pore was found to be dependent on the bubble radius. Furthermore, the inflow of PI is diffusion-driven. The pore is formed temporarily and closes within several seconds after the ultrasound exposure.

10:20

4aBA4. Ultrasound-mediated drug delivery with real-time cell permeability measurements. Pavlos Anastasiadis (Molecular Biosciences and Bioengineering, Univ. of Hawaii, Honolulu, HI), Michelle L. Matter (John A. Burns School of Med., Univ. of Hawaii, Honolulu, HI), and John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

Ultrasound-mediated drug and gene delivery offers a variety of novel possibilities for improved localized treatment of vascular- and cancer-related diseases. This therapeutic application benefits from the use of acoustic radiation force, which facilitates the exposure for enhanced binding from ligand-receptor interactions. The unique merits of ultrasound are the transient increase of cell permeability without any detrimental and irreversible side-effects. The related underlying molecular and cellular pathways of ultrasound-induced permeability and the subsequent recovery of cells are topics of on-going research. Real-time studies of cell behavior during and past-ultrasound exposure have been limited by the lack of appropriate techniques. The electric-cell impedance sensing (ECIS) technique is a suitable way of studying cell permeability changes in real-time. Its nanoscale sensitivity and speedy acquisition of data allows for the accurate and timely monitoring of cell behavior. Our preliminary results suggest that cells recover within 24–36 h post-exposure. During this time window the cells undergo drastic changes exhibiting an increased permeability of $2.4 \pm 0.6\ \Omega\bullet\mathrm{cm}^2$ compared to $3.8 \pm 0.5\ \Omega\bullet\mathrm{cm}^2$ that normal untreated cells exhibit.

Contributed Papers

10:40

4aBA5. Investigation on the inertial cavitation threshold of micro-bubbles. Xiasheng Guo, Dong Zhang, and Juan Tu (Dept. of Phys., Inst. of Acoust., No. 22, Hankou Rd., Nanjing 210093, China, guoxs@nju.edu.cn)

Experimental measurements and numerical analyses were performed to investigate the IC thresholds of two commercialized UCAs, albumin-shelled KangRun® and lipid-shelled SonoVue®. The IC thresholds of these two UCAs were measured at varied acoustic pulse lengths and bubble concentrations, according to the IC dose quantifications based on passive cavitation detection (PCD). Then, the shell properties of UCAs were estimated by fitting the measured acoustic attenuation data. Finally, the influences of acoustic pulse length and UCA shell properties on the microbubble nonlinear behaviors were discussed based on numerical simulations, which would give us better understanding of the dependence of microbubble IC threshold on the sonication condition and physical structure properties of the coating shells. The experimental results show that: (1) the IC threshold of UCAs is dependent on the acoustic driving conditions, the shell properties of UCAs and the bubble concentration; (2) for both the lipid- and albumin-shelled UCAs, the IC threshold generally decreases with the increasing UCA volume concentration; (3) IC threshold is observed higher for short-pulse excitation, then its value decreases as the acoustic pulse length increases from 5 to 20 cycles and finally tends to reach a steady state for even longer pulsed exposures.

11:00

4aBA6. Elucidating the effects of low-intensity ultrasound on mesenchymal stem cell proliferation and viability. Nirali Shah, Yosry Morsi (Mech. Eng., Swinburne Univ. of Technol., Melbourne, VIC, Australia), Ursula Manuelpillai (Ctr. for Reproduction and Development, Monash Inst. for Med. Res., Melbourne, VIC, Australia), Tim Barry, and Richard Manasseh (Mech. Eng., Swinburne Univ. of Technol., P.O. Box 218, Hawthorn, VIC, Melbourne, VIC 3122, Australia, rmanasseh@swin.edu.au)

The effects of low-intensity ultrasound (LIUS) on the proliferation, viability and extracellular matrix (ECM) production of mesenchymal stem cells (MSCs) were investigated. Continuous-wave ultrasound was applied at 1 MHz and and 350 mW/cm^2 to microwells, using a LIUS system assembled in the laboratory. Needle hydrophone mapping showed that pressure amplitudes ranged from 0.015 MPa at the well edge to 0.080 MPa at the center. The LIUS group received US for 10, 20, and 30 min/day for one week. Assays were performed daily. Relative to control, 10 and 20 min LIUS very significantly stimulated MSC proliferation and ECM synthesis, while 30 min LIUS had a significant adverse effect. The phenomenon that LIUS accelerates MSC proliferation, but only for appropriate exposures, has been noted previously in the literature. However, the actual relation between the physical forces generated by the LIUS and this phenomenon remains unknown. The fluid flow

pattern created by LIUS was studied by injecting dye in the well and Eckart-streaming-like motions were observed, while thermal effects were negligible. By employing LIUS with appropriate focusing and parameters, it might be possible to exploit MSCs for tissue engineering, independently of biochemical stimuli, and in a highly spatially organized manner.

11:20

4aBA7. "SonoBandage" a transdermal ultrasound drug delivery system for peripheral neuropathy. Matthew Langer, Sabrina Lewis, Shane Fleshman, and George Lewis (ZetrOZ, 421 N. Aurrora St., Ithaca, NY 14850, mlanger@zetroz.com)

Peripheral neuropathy (PN) is a difficult disease to manage. Symptomatic treatment focuses primarily on pain relief, using NSAIDs, opioids, tricyclic antidepressants, and selective serotonin norepinephrine reuptake inhibitors. There is potential for ultrasound transdermal drug delivery to improve the quality of care provided to patients with PN, since it is well-suited to peripheral nerves which are close to the skin. In addition, targeted delivery avoids many of the systemic consequences of taking a drug. We developed a wearable ultrasound drug delivery system called "SonoBandage" that combines low-impedance miniaturization of ultrasound transducer, RF electronics, and battery power supply, with a novel hydrogel coupling bandage loaded with salicylic acid NSAID. The design of the SonoBandage allows the device to be used over a range of ultrasound frequencies (0.1–3 MHz), intensities (0.1–3 W/cm^2), and durations (0.25–4 h) increasing system flexibility for drug delivery protocols. The SonoBandage with NSAID was evaluated on a bench-top model with freshly harvested porcine skin and synthetic biomimetic human skin membrane (Millipore Inc). Across the $n = 40$ samples studied, salicylic acid drug flux was increased by 2–20x as compared to control samples ($p < 0.01$) after 1–4 h of ultrasound treatment. SonoBandage has potential to be used as a practical NSAID delivery platform for peripheral neuropathy.

THURSDAY MORNING, 6 JUNE 2013 512AE, 9:00 A.M. TO 12:00 NOON

Session 4aEAa

Engineering Acoustics: Non-Contact Ultrasonic Methods

Michael R. Haberman, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Nico Declercq, Cochair
Georgia Tech Lorraine, 2 rue Marconi, Metz 57070, France

Invited Papers

9:00

4aEAa1. All-optical nonlinear frequency—Mixing acoustics of cracks. Vitalyi Gusev (IMMM, UMR-CNRS 6283, LUNAM Université, Université du Maine, avenue O. Messiaen, Le Mans 72085, France, vitali.goussev@univ-lemans.fr), Nikolay Chigarev, Sylvain Mézil, and Vincent Tournat (LAUM, UMR-CNRS 6613, LUNAM Université, Université du Maine, Le Mans, France)

Recent advances in all-optical evaluation of nonlinear cracks are reviewed. In experiments, the nonlinear acoustic waves are initiated by the absorption of radiation from a pair of laser beams intensity-modulated at two different frequencies. The detection of the acoustic waves at mixed frequencies, absent in the frequency spectrum of the laser intensity, is achieved by optical interferometry or deflectometry. The high contrast in crack imaging achieved by remote optical monitoring of the nonlinear acoustic processes is due to the strong dependence of the optoacoustic conversion efficiency on the state of the crack. The highest acoustic nonlinearity is observed in the transitional state of the crack, which is intermediate between the open and the closed ones. Several crack parameters can be estimated from the measurements of the dependence of the acoustic spectrum on the pump laser intensity. One-dimensional theory of the nonlinear frequency-mixing photo-acoustic crack imaging is presented. The theory relates experimental observation of the large number of mixed frequencies to strong bi-modular nonlinearity of the crack. The theory provides guidelines for the understanding of the dependence of the spatial resolution of this technique on the laser power and on the choice of a particular mixed-frequency component for the crack imaging. [Work supported by ANR project ANL-MEMS ANR-10-BLAN-092302.]

9:20

4aEAa2. Improving the focal quality of the time reversal acoustic noncontact source using a deconvolution operation. Brian E. Anderson, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, bea@lanl.gov)

The time reversal acoustic noncontact source (TRANS) utilizes time reversal (TR) from several transducers arranged in a cavity to focus energy onto a solid surface to allow inspection of that surface. The advantage of TRANS is that multiple transducers may be used to increase the amplitude of the focused energy onto the surface and potentially allow interrogation of nonlinear surficial features such as cracks and delaminations. TR is known to be a matched signal process and therefore is limited in the temporal fidelity and spatial compression of the focused energy. Fortunately, using a deconvolution operation in conjunction with TR [or inverse filter similar to Tanter *et al*. (J. Acoust. Soc. Am. **108**, (2000), 223–234)] can greatly improve the quality of the spatial focusing as well as increasing the temporal fidelity. Visualizations of the impact of the deconvolution operation will be presented to provide insight into the increased quality of the spatial focusing along with results from using the deconvolution operation in conjunction with TRANS. [Work supported by Institutional Support (LDRD) at the Los Alamos National Laboratory.]

9:40

4aEAa3. Probing of crack breathing by pulsed laser-generated acoustic waves. Vincent Tournat, Chenyin Ni, Nikolay Chigarev (LAUM, CNRS, Université du Maine, Av. O. Messiaen, Le Mans 72085, France, vincent.tournat@univ-lemans.fr), Nicolas Delorme (IMMM, CNRS, Université du Maine, Le Mans, France), Zhonghua Shen (School of Sci., Nanjing Univ. of Sci. and Technol., Nanjing, China), and Vitalyi Gusev (IMMM, CNRS, Université du Maine, Le Mans, France)

Experimental results on all-optical monitoring of the nonlinear motion of a surface-breaking crack are reported. Crack closing is induced by quasi-continuous laser heating, while Rayleigh surface acoustic pulses and bulk longitudinal surface skimming acoustic pulses are also generated and detected by lasers. By exploiting the strong dependence of the acoustic pulses reflection and transmission efficiency on the state—open or closed—of the contacts between the crack faces, the parametric modulation of ultrasonic pulses is achieved. It is observed that bulk acoustic waves, skimming along the surface can be more sensitive to crack motion than Rayleigh surface waves. It has been found that crack closure by thermo-elastic stresses modifies the propagation paths of the acoustic rays from the point source to the point receiver. Consequently, the arrival times of the acoustic waves contain information on the state of crack closure induced by a particular intensity of laser heating. An important dependence of the detected signals on the initial width/state of the crack and on the presence of necks in the crack opening profile is revealed. It is demonstrated that the mode conversion of the skimming longitudinal bulk waves incident on the crack into the transmitted Rayleigh waves is very sensitive to imperfectness of crack closure. The proposed interpretation of the experimental observations is supported by atomic force microscopy measurements.

10:00

4aEAa4. The effects of the transducer beam properties on the ultrasonic geometrical characterization of periodically corrugated surfaces. Jingfei Liu and Nico F. Declercq (George W Woodruff School of Mech. Eng., Georgia Inst. of Technol., 2, rue Marconi, Metz 57070, France, benjamin.jf.liu@gatech.edu)

Periodically corrugated structures are common in many technological applications, and in most cases, the geometry of these corrugated structures are crucial for the designed functionality. As an effective nondestructive characterization method, an ultrasonic imaging technique is investigated in this work for the purpose of accurately characterizing the geometry of periodic corrugations. Among many factors that affect the imaging quality the properties of the transducer beam dominate. The effects of the spatial and spectral properties of transducer beams on the accurate characterization of the characteristic dimensions of corrugations are investigated in details both theoretically and experimentally. The possibility to accurately characterize the corrugation characteristic dimensions, the condition for accurate characterization, and the quantitative relationship between the characterization accuracy and the beam parameters are given. The ways to avoid the diffraction effects and reduce possible errors are also discussed. Experimental results are compared with optical measurements and good agreement is obtained. Both the general principles developed theoretically and the practical techniques proposed can work as a useful guidance for similar work.

10:20–10:40 Break

10:40

4aEAa5. Excitation of Rayleigh and zero-group-velocity Lamb waves using air-borne N-waves focused by an ellipsoidal reflector. Xiaowei Dai (Dept. of Civil, Architectural, and Environ. Eng., The Univ. of Texas at Austin, Austin, TX), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu), Yi-Te Tsai, and Jinying Zhu (Dept. of Civil, Architectural, and Environ. Eng., The Univ. of Texas at Austin, Austin, TX)

Air-coupled ultrasonic non-destructive testing (NDT) of elastic solids is a challenge due to the large impedance contrast between air and most materials used in industrial and structural applications. However, because air-coupled sensing offers many advantages such as rapid scanning of large areas and the elimination of part immersion for inspection, there remains strong incentive to find unique methods for air-coupled excitation of wave motion in elastic solids. This work presents experimental results of an in-air acoustic source that has been shown to excite wave motion in high impedance elastic solids. The source consists of a spark generator and an ellipsoidal reflector. The spark generator radiates a short-duration, high-amplitude acoustic signal as the result of an electrostatic discharge between two electrodes with high potential difference. Analogous to lithotripter, the spark is located at the near focus and generates an outgoing wave that is then focused at the far focus of the reflector which is co-located at the air-solid interface. Measurements of the air-borne acoustic wave in the free-field, the focused acoustic wave, and Rayleigh and zero-group-velocity (ZGV) Lamb waves generated in a concrete slab will be presented and analyzed. [Work supported by NIST Technology Innovation Program (TIP).]

11:00

4aEAa6. Detection and characterization of defects in aerostructures using non-contact ultrasonic transducers. Ngeletshedzo Netshidavhini and Raymond B. Mabuza (NDT and Phys., Vaal Univ. of Technol., Private Bag X021, Andries Potgieter St., Vanderbijlpark, Gauteng 1900, South Africa, ngeletshedzon@vut.ac.za)

This paper describes an investigation into the possibility of using non-contact ultrasonic transducers for detecting and characterizing defects in aerostructures. Our study deals with an ultrasonic method that underscores the recent advances in non-contact and analytical methodologies. Ultrasonic waves are generated by a transducer connected to a Sonatest DryScan 410D using through-transmission mode. In this investigation, ultrasonic through-transmission signals are analyzed for their amplitudes. We treat this problem analytically and experimentally. Various aspects of our approach are presented. The observations reported in this paper deal with defects in aluminum structures. Confirmation of defect geometry is obtained by comparing the results of the non-contact ultrasonic sensors with a conventional ultrasonic testing method. The results obtained are in good agreement with those of the conventional ultrasonic method, indicating that both techniques can be considered as quantitative nondestructive tools for detecting and characterizing defects. Results are presented and discussed.

11:20

4aEAa7. Coupling of finite difference elastodynamic and semi-analytic Rayleigh integral codes for the modeling of ultrasound propagation at the hip. Didier Cassereau, Pierre Nauleau, Quentin Grimal, Jean-Gabriel Minonzio (Laboratoire d'Imagerie Paramétrique, 15 rue de l'Ecole de Médecine, Paris 75006, France, didier.cassereau@upmc.fr), Aniss Bendjoudi, Emmanuel Bossy (Institut Langevin Ondes et Images, Paris, France), and Pascal Laugier (Laboratoire d'Imagerie Paramétrique, Paris, France)

Ultrasonic exploration of the femoral neck is of wide interest as it can provide some information about a potential fracture risk, particularly for osteoporosic patients. *In vivo*, the ultrasonic wave first propagates through soft tissues that can be idealized as a homogeneous fluid. Then, the ultrasonic wave interacts with the bone structure. Transmitted and back-propagated signals are then measured at receivers. A numerical model of this complete chain is useful to understand and control the various parameters involved in this process. The complexity of the bone structure is approached using the elastodynamic finite difference time domain (FDTD) code SimSonic. Due to the small size of the spatial grid needed by FDTD schemes, the propagation between the emitter and the femoral neck may be excessively time and resource consuming. We have developed a coupling between SimSonic and a direct and fast evaluation of the diffraction in homogeneous fluids, based on the numerical discretization of the Rayleigh integral. This approach allows to reduce drastically the total computation time for the complete simulation. Results obtained with this new system are presented, including computation times and computer resources. This approach is particularly useful to simulate experiments with phased arrays, which involve several emissions.

11:40

4aEAa8. Basic examination of noncontact inspection in solid material by using high-intensity aerial ultrasonic waves and optical equipment. Ayumu Osumi (Nihon Univ., 1-8, KandaSurugadai, Chiyoda 101-8308, Japan, oosumi@ele.cst.nihon-u.ac.jp) and Youichi Ito (Nihon Univ., Tokyo, Japan)

Recently, developments have improved methods employing aerial ultrasonic waves for contactless inspection of internal defects in materials such as metals, pipe walls, and fiber-reinforced plastics. Specially, this method is noncontact way differ from conventional ultrasonic inspection that is necessary to contact probe to object. We have developed a new method of aerial ultrasonic inspection that uses high-intensity aerial ultrasonic waves and optical equipments. That is, the object is excited in noncontact way using high-intensity aerial ultrasonic waves and the vibration velocity on the object surface is measured with a laser Doppler galvanometer at same time. We analysis the vibration information and detect defect in materials. We also developed a point-converging acoustic source with a stripe-mode vibration plate to generate the high-intensity aerial ultrasonic waves, an essential component of the method. While the sound source operates at a single resonance frequency, the generated ultrasonic wave has nonlinear acoustic characteristics and generates nonlinear higher harmonics at the focal point because the sound intensity increases by converging a sound wave. Under nonlinear ultrasonic irradiation, the object vibrates at the fundamental frequency and harmonic frequencies corresponding to the ultrasonic waves. Therefore, we also attempted to detect defect in materials for analyzing nonlinear vibration.

THURSDAY MORNING, 6 JUNE 2013 — 512BF, 9:00 A.M. TO 10:20 A.M.

Session 4aEAb

Engineering Acoustics: Acoustics for Navigation

Robert D. White, Chair
Mech. Eng., Tufts Univ., 200 College Ave., Medford, MA 02155

Invited Papers

9:00

4aEAb1. Ultrasonic transducers for navigation. Bernhard E. Boser, Richard J. Przbyla (Berkeley Sensor and Actuator Ctr., Univ. of California, 490A Cory Hall, Berkeley, CA 94720-1770, boser@eecs.berkeley.edu), David A. Horsley, Stefon E. Shelton, and André Guedes (Univ. of California, Davis, CA)

Free space ultrasonic ranging is attractive for applications such as gesture recognition and robotic navigation. Unlike optical ranging technologies, ultrasound based solutions are insensitive to ambient illumination and can therefore be used in- and outdoors. Using time-of-flight, ultrasound rangers work over distances of up to a few meters and achieve sub-mm resolution. Using arrays, objects can be localized in three dimensions. Transducers consist of 400μm aluminum-nitride membranes sandwiched between actuation electrodes batch fabricated on silicon wafers. Unlike capacitive transducers, which require actuation voltages in excess of 100 V, piezoelectric devices are compatible with low-voltage actuation. At the 200 kHz resonance frequency, the wavelength at atmospheric pressure is 2 mm, ideal for compact arrays. The transducers do not dissipate static power and are therefore ideal for battery powered applications. Energy consumption is dominated by the low-noise readout amplifier and is on the order of 1μJ per channel including analog-digital conversion and signal processing, enabling video-rate object tacking at less than 1 mW power dissipation. A prototype system consisting of seven transducers on a 1 mm grid operates up to a 750 mm range and $\pm 35^o$ angle span with $\pm$3.5mm accuracy and $\pm 3^o$ worst case angle error.

9:20

4aEAb2. An infrasound-based avian navigational "map". Jonathan T. Hagstrum (U.S. Geological Survey, 345 Middlefield Rd., MS 937, Menlo Park, CA 94025, jhag@usgs.gov)

The "compasses" (solar, star, geomagnetic) that homing pigeons and other migratory birds use to orient during flight are generally understood, but the "map" sense they need to first determine their homeward direction is not. Atmospheric odor and geomagnetic gradients have been proposed as "map" cues, but are inadequate and remain controversial. Experiments with frosted lenses indicate that sight can also be ruled out. Laboratory tests, however, show that pigeons can detect infrasound (>0.05 Hz), and such signals travel with little attenuation for 1000s of kilometers through the atmosphere. Results from an acoustic ray-tracing program (HARPA) using daily atmospheric profiles are compared with pigeon release data for a number of sites in upstate NY. HARPA runs show that homeward infrasonic cues could have arrived at the sites from directions opposite pigeon departure bearings, especially when these bearings were unusual. Such signals possibly arise from ground-to-air coupling of microseisms or from scattering of microbaroms off terrain features (~0.2 Hz). Pigeons and other birds might use Doppler shifts to determine the directionality of homeward infrasonic cues while flying in circular or other patterns at constant velocity after release; they apparently have built-in airspeed indicators adapted from their olfactory and aural systems.

9:40

4aEAb3. Design, development, and testing of transducers for creating spiral waves for underwater navigation. David A. Brown, Corey Bachand, and Boris Aronov (BTech Acoust. LLC, Adv. Tech. & Manuf. Ctr., Univ. of Massachusetts, 151 Martine St, Fall River, MA 02723, dbAcoustics@cox.net)

The use of spiral waves for underwater acoustic navigation has received much attention in recent years. A spiral wave is characterized as a diverging wavefront that is omnidirectional by magnitude but with a phase that varies linearly by azimuthal angle. Such a signal may be exploited in underwater acoustic navigation when compared with a reference signal of constant phase with respect to

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azimuthal angle. This paper summarizes our sine/cosine spiral wave transducer (SC-SWT) design approach, which creates a spiral wave by generating two orthogonal dipoles driven in phase quadrature using the same cylindrical piezoceramic element. Several acoustic navigation beacon designs that also contain the constant-phase reference transducer have now been fabricated, including variants where spiral and referencing sources have the same effective acoustical origin in the horizontal and vertical planes. Experimental results, test tank calibrations, and problems to overcome are presented.

Contributed Paper

10:00

4aEAb4. Measuring the acoustic response of a compartment fire. Mustafa Z. Abbasi, Preston S. Wilson (Appl. Res. Lab. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E dean Keeton, Austin, TX 78712, mustafa_abbasi@utexas.edu), and Ofodike A. Ezekoye (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Rescue teams have a small window of time to locate a downed firefighter. Their task is made more difficult due to low visibility, smoke, toxic gases, and high temperatures. In the United States, most firefighters are equipped with a Personal Alarm Safety System (PASS) device that emits an alarm sound, when the firefighter becomes incapacitated. Rescue teams can then follow this sound to the source to locate the downed firefighter. While the PASS device has been enormously successful, anecdotal evidence has shown it fails in some interesting scenarios. For example, cases have been recorded where firefighters inside the building were unable to hear the signal, whereas those outside heard it clearly. To explain these cases, and to improve the signal used by the PASS device, it is necessary to understand sound propagation in the fireground environment. This paper will present acoustic transfer measurements inside a laboratory compartment fire, simulating a fire in a residential structure. The research aims to understand how the developing temperature gradient and smoke layer influences sound propagation. A secondary goal is the development and validation of finite element models of fireground acoustics. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

THURSDAY MORNING, 6 JUNE 2013 — 512DH, 9:00 A.M. TO 12:20 P.M.

Session 4aMU

Musical Acoustics: Transient Phenomena in Wind Instruments: Experiments and Time Domain Modeling

Stefan Bilbao, Cochair
Music, Univ. of Edinburgh, Rm. 7306B, JCMB, Kings Bldgs., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom

D. Murray Campbell, Cochair
School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom

Invited Papers

9:00

4aMU1. Modeling pulse-like lip vibrations in brass instruments. Jonathan A. Kemp (Dept. of Music, Univ. of St Andrews, Beethoven Lodge, 65 North St., St Andrews, Fife KY16 9AJ, United Kingdom, jk50@st-andrews.ac.uk) and Richard A. Smith (Smith Watkins Trumpets, Sheriff Hutton, Yorkshire, United Kingdom)

During the starting transient of a note on a brass instrument it can take several cycles of lip vibration before acoustics reflections from the end of the instrument can influence the lip frequency. Under certain conditions, the lip may fail to oscillate at the pitch of the air column resulting in an unwanted pulse-like waveform with relatively low repetition rates (similar to the vocal fry register of phonation in the human voice). This is often observed in the playing of beginners if the lips are insufficiently tense or if the top and bottom lips overlap to a large extent. In this study, the reasons for this behavior will be investigated using modeling techniques with the aim of improving the agreement between physical models and measured transients by including the forces responsible for this effect.

9:20

4aMU2. Transient phenomena in brass instruments. John Chick (School of Eng., Univ. of Edinburgh, Kings Blds., Edinburgh EH9 3JL, United Kingdom, john.chick@ed.ac.uk), Shona Logie (School of Phys., Univ. of Edinburgh, Edinburgh, United Kingdom), Lisa Norman (Reid School of Music, Univ. of Edinburgh, Edinburgh, United Kingdom), and Murray Campbell (School of Phys., Univ. of Edinburgh, Edinburgh, United Kingdom)

The starting transient and the transition between notes are known to be of fundamental musical significance on all instruments. On a brass instrument, the player needs to establish a strongly coupled resonance between the air column and the lips for a note to sound effectively. In the case of a slurred transient, the player must decouple the lips from one resonance before establishing the next. The ease with which this can be achieved depends on several factors including tube length and bore profile, and resonant modes being played. Analysis of measured mouthpiece pressure data and time domain computer modeling have been used to explore transient phenomena in brass instruments, with the aim of identifying desirable playing characteristics of an instrument.

9:40

4aMU3. Transient variation in mechanical action and electric action pipe organs. Alan Woolley (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., King's Bldgs., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, awoolley@staff-mail.ed.ac.uk)

Control of the transients by the player has often been cited as one of the most important characteristics of mechanical pipe organ actions since their reintroduction toward the middle of the last century. Previous research indicates that players do not vary the way that they move the key to a significant extent, except as the result of starting the finger movement from some distance above the key rather than in contact with it. This does not necessarily lead to an audible difference. There are, however, other factors in pipe organ action design, pipe voicing, and the way in which organs are played that may lead to real or apparent transient variation irrespective of the type of action. It is well recorded that it is desirable to stagger the release of a chord starting with the pipes with least wind requirement in order to minimize the effect on the wind chest pressure particularly with traditional pressure regulators remote from the wind chest. This paper investigates some of these mechanisms and compares transient variation on mechanical action organs and electric action organs due to different playing styles.

10:00

4aMU4. Some simulations of the effect of varying excitation parameters on the transients of reed instruments. Fabrice Silva (Laboratoire de Mécanique et d'Acoustique, CNRS-LMA, Marseille, France), Vincent Debut (Appl. Dynam. Lab., Campus Tecnológico e Nuclear, Instituto Superior Técnico/Universidade Técnica de Lisboa, Sacavem, Portugal), Philippe Guillemain, Jean Kergomard, and Christophe Vergez (Laboratoire de Mécanique et d'Acoustique, CNRS-LMA, 31 Chemin Joseph Aiguier, Marseille 13402, France, kergomard@lma.cnrs-mrs.fr)

This paper considers the simulation of self-sustained oscillations in reed and brass instruments, based on a compact continuous-time formulation of the sound production mechanism. The control parameters such as the mouth pressure and the player's embouchure, but also the acoustic resonator and the reed, may vary with respect to time, allowing the analysis of transient and non-stationary phenomena like changes of regime. A particular attention is first given to staccato notes, with comparison of the evolution of the instantaneous frequency in simulations to theoretical and experimental results. This shows the importance of using realistic control parameters on the onset of the oscillations. When the acoustic resonator is modeled using a modal expansion with non-stationary resonance frequencies and damping, it is also possible to simulate and study slurs and musical effects like the wah-wah, gaining some insight on the mechanisms involved.

Contributed Papers

10:20

4aMU5. Modeling articulation techniques in single-reed woodwind instruments. Vasileios Chatziioannou and Alex Hofmann (Inst. of Music Acoust., Univ. of Music and Performing Arts Vienna, Anton-von-Webern-Platz 1, Bldg. M, Vienna 1030, Austria, chatziioannou@mdw.ac.at)

Time-domain simulations of wind instruments can, in principle, deal with non-linear oscillations and are also capable of modeling both the steady-state and the transient behavior of a system. The starting transient is usually an important identifying feature of the instrument that is played. Subtle control of articulation is required from skilled musicians to modulate transients during expressive performance. Focusing on single-reed woodwind instruments, the physical phenomena that underlie different articulation techniques are analyzed. A saxophone player is recorded during portato playing, where articulation is achieved either by the use of the tongue, or by modulating the air flow into the mouthpiece. The bending of the reed and the pressure inside the mouthpiece are measured and a physical model is formulated with the aim to capture the transient effects. Instead of adding new terms (and complexity) to a single mass-spring model, in order to simulate the player's tongue, existing physically meaningful parameters are allowed to vary. In particular, the effect of tonguing is modeled by modulating the equilibrium position of the (lumped) reed and its internal damping, whereas in the case of air-separated tones, only a variation of the blowing pressure is required.

10:40

4aMU6. Measurement setup for articulatory transient differences in woodwind performance. Alex Hofmann, Vasileios Chatziioannou (Music Acoust. (IWK), Univ. of Music and Performance Art Vienna, Anton von Webern Platz 1, Vienna, Select State 1030, Austria, hofmann-alex@mdw.ac.at), Michael Weilguni (Inst. of Sensor and Actuator Systems, Vienna Univ. of Technol., Vienna, Austria), Werner Goebl, and Wilfried Kausel (Music Acoust. (IWK), Univ. of Music and Performance Art Vienna, Vienna, Austria)

To model transient differences caused by varying articulation techniques on single-reed woodwind instruments, human performances have to be measured and analyzed. In a previous study, we investigated differences between tongued and air-separated tones on a saxophone by monitoring inner mouthpiece pressure, mouth pressure, and reed bending during performance. Some of the observed effects (e.g., damping and displacement of the reed) were applied to a physical model. Although tip-opening is an essential parameter of lumped reed models, we were only able to directly compare our measurements to the model by inner mouthpiece air-pressure. In this study, we aim to relate measurable reed-bending to the resulting tip-opening and also determine the sensor-reed's stiffness. A micro-mechanical characterization test system records the static and dynamic mechanical data of the sensor-equipped reeds. Pull and tensile measurements are used to reveal the relation between reed bending and tip opening and will increase our understanding of quasi-static properties of the reed-mouthpiece system. This allows conclusions about the tip opening behavior at transient emergence in expressive single-reed woodwind performance.

11:00

4aMU7. Modes of reed vibration and transient phenomena in free reed instruments. James P. Cottingham (Physics, Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

The motion of air-driven free reeds used in the harmonica, accordion, and reed organ is dominated by the fundamental transverse beam mode, but higher transverse modes and the first torsional mode are usually present during steady oscillation, even at low amplitude. In addition, a lateral mode has sometimes been detected, in which the reed tongue oscillation is perpendicular to the transverse oscillation. Interaction of the reed with a resonance in the instrument can result in unusual effects. In the accordion, resonances of the reed cavity can interfere with the reed self-excitation mechanism. In the harmonica, when the reed is nearly closed, a strong aerodynamic instability can in some cases lead to torsional flutter. A characteristic of some free reed instruments is a slow attack, in which the sound builds gradually and often unevenly, with the effect being greater for the longer, lower-pitched reeds. There is evidence that the first torsional mode and the second transverse mode may be significant in initiating reed oscillation, so that reed design enhancing the torsional mode may be helpful in alleviating the problem of slow attack.

11:20

4aMU8. Direct numerical simulations of the recorder in two and three dimensions. Nicholas Giordano (Dept. of Phys., Purdue Univ., 525 Northwestern Ave., West Lafayette, IN 47923, giordano@purdue.edu)

Direct numerical solutions of the compressible Navier-Stokes equations have been used to study various aspects of sound production in the recorder. A custom algorithm implemented on a parallel computer has enabled us to calculate tones and produce visualizations of the air flow near the labium in both two and three dimensions. In three dimensions, we have observed how the attack portion of the tone and the spectrum at long times depends on the relative alignment of the channel and labium. We also describe subtle differences in the process of vortex shedding in two as compared to three dimensions.

11:40

4aMU9. Numerical reproducibility of time-dependent motions of spatial waves in air-columns of wind instruments. Kin'ya Takahashi, Saya Goya, Kana Goya, and Chisato Susaki (The Phys. Labs., Kyushu Inst. of Technol., Kawazu 680-4, Iizuka, Fukuoka 820-8502, Japan, takahasi@mse.kyutech.ac.jp)

Since Schumacher introduced a time-domain model of single-reed instruments and McIntyre *et al.* gave the general concept of time-domain models of wind instruments, the time-domain models, namely, delayed feedback models, have become an important numerical tool for study of wind instruments due to their simplicity, easiness to handle, and reliability. However, those models only reproduce wave oscillations observed in mouthpieces of wind instruments. In this talk, we will propose a numerical technique, which is able to reproduce time dependent motions of spatial waves in an air-column. It is composed of inversed wave propagator matrices combined with the forward and backward Fourier transformations. The resultant spatial waves in the air-column exhibit very similar time-dependent behavior to those observed by an experiment for the clarinet. Actually, backward and forward rounded-off step waves are observed. We will also discuss difference in wave shapes and their time-dependent behavior depending on the shapes of air-columns, cylindrical one like the clarinet, conical one like the saxophone and horn-shaped one like brass instruments. For the conical and horn-shaped air-columns, Helmholtz-like waves are observed rather than the step waves observed for the cylindrical air-column.

12:00

4aMU10. A thermoviscous tube propagation model suitable for time domain analysis. Stephen C. Thompson, Thomas B. Gabrielson (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16803, sct12@psu.edu), and Daniel M. Warren (Knowles Electron., Itasca, IL)

Modeling acoustic propagation in tubes including the effects of thermoviscous losses at the tube walls is important in thermoacoustics, in hearing aid modeling, and in modeling wind musical instruments. Frequency dependent impedances for a tube transmission line model in terms of the so-called thermal and viscous functions are well established, and form the basis for frequency domain analysis of systems that include tubes. However, frequency domain models cannot be used for systems in which significant nonlinearities are important, as is the case with the pressure-flow relationship through the reed in a woodwind instrument. This paper describes a tube model based on a continued fraction expansion of the thermal and viscous functions. The expansion can be represented as an analog circuit model, which allows its use in time domain system modeling. A simple model of a clarinet-like oscillation will be shown.

THURSDAY MORNING, 6 JUNE 2013 — 511BE, 8:55 A.M. TO 12:00 NOON

Session 4aNSa

Noise, Architectural Acoustics, and Psychological and Physiological Acoustics: Effects of Noise on Human Performance and Comfort I

Lily M. Wang, Cochair
Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

Arianna Astolfi, Cochair
Politecnico di Torino, Corso DC degli Abruzzi, 24, Turin 10124, Italy

Chair's Introduction—8:55

Invited Papers

9:00

4aNSa1. Outline proposal for a good practice guide on the evaluation of human response to vibration from railways in residential environments. Andy Moorhouse, David Waddington, Eulalia Peris, James Woodcock, Calum Sharp, and Gennaro Sica (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg, Salford M20 1JJ, United Kingdom, a.t.moorhouse@salford.ac.uk)

The paper will present outline proposals for a good practice guide for the evaluation of human response to vibration from railways in residential environments. The context is the need to increase the proportion of freight carried by railways in Europe while avoiding additional disturbance from vibration to populations living nearby. Within this context, the European funded project CARGOVIBES is developing the good practice guidelines to assist in the evaluation of potential disturbance. In the guide, it is proposed to include descriptions of the adverse response, primarily annoyance and sleep disturbance. The proposal is to use measured or predicted vibration metrics in conjunction with dose-response relationships to quantify potential adverse impact to residents. To this end, measurement and assessment of vibration will be considered, for example, the equipment, locations, mounting, and a description of the data to be acquired. The latest information on dose-response relationships will then be reviewed. Finally, various national limits for vibration will be evaluated against new scientific evidence on dose-response relationships.

9:20

4aNSa2. Analysis of railway vibration signals using supervised machine learning for the development of exposure-response relationships. Calum Sharp, James Woodcock, Eulalia Peris, Gennaro Sica, Andrew Moorhouse, and David Waddington (Acoust. Res. Ctr., Univ. of Salford, The Crescent, Salford M5 4WT, United Kingdom, c.sharp@edu.salford.ac.uk)

The aim of this work is to investigate the applicability of the use of supervised machine learning methods to classify unknown railway vibration signals within a measurement database. The results of this research will be implemented in the development of exposure-response relationship for annoyance caused by freight and passenger railway vibration, so as to better understand the differences in human response to these two sources of environmental vibration. Data for this research come from case studies comprising face-to-face interviews with respondents and measurements of their vibration exposure collected during the University of Salford study "Human Response to Vibration in Residential Environments." Vibration data from this study are then classified into freight and passenger categories using supervised machine learning methods. Finally, initial estimates of exposure-response relationships are determined using ordinal probit modeling. The results indicate that the annoyance response due to freight railway vibration may be significantly higher than that due to passenger railway vibration, even for equal levels of exposure. The implications of these findings for the potential expansion of freight traffic on rail are discussed. [Work funded by the Department for Environment, Food and Rural Affairs (Defra) UK, and EU FP7 through the CargoVibes project.]

Contributed Papers

9:40

4aNSa3. Study on the annoyance of high frequency noise at industrial workstations. Bozena E. Smagowska (Dept. of Vibroacoustic Hazards, Central Inst. for Labour Protection - National Res. Inst., Czerniakowska str. 16, -, Warsaw, Mazowieckie 00-701, Poland, bosma@ciop.pl)

The aim of the study was subjective assessment of the acoustic environment of persons exposed to high frequency noise at workstations. The study based on the survey regarding the annoyance of this type of noise which was conducted among 52 operators of equipment for the production of platform gratings. The study covered workstations, where acoustic measurements confirmed the presence of high frequency noise (in the frequency range from 10 to 20 kHz). During work at these workstations, the value of the equivalent sound pressure level in one-third octave-bands center frequencies of 10, 12.5, and 16 kHz occurs within the range of 81–103 dB and exceeds the admissible value defined for these frequency bands. The results of the study indicated that 92% of respondents are exposed to noise the whole time of the shift. All respondents wear hearing protection. Most of the employees describe the noise as: buzzing, insistent, high-pitched squeaky, and whistling. Respondents unanimously consider related noise levels as: loud, impeding communication, highly strenuous, and tiring. The highest number of points on a scale corresponding to noise annoyance was achieved by terms: horrible, very, persistently, and firmly.

10:00

4aNSa4. Comparison of discomfort caused by two kind of backup alarm. Laurent Brocolini, Lucie Léger, Etienne Parizet (Laboratoire Vibrations Acoustique, INSA, 25 bis, av. J. Capelle, Villeurbanne F-69621, France, laurent.brocolini@gmail.com), Jean-Marie Verlhac, and Xavier Carniel (Ingénierie Bruit et Vibrations, CETIM, Senlis, France)

Nowadays used on most of construction vehicles, tone backup alarm causes a strong discomfort among resident citizens. To solve this problem, "CETIM" (Mechanical Industries Technical Centre) decided to characterize and test another kind of alarm. This one is called "Cri du Lynx" in french (Lynx scream) because of its particular sound looks like a white noise. A previous study showed that the average sound level of the noise-alarm detectability is 67 dB (A) while it is 64 dB (A) for the tone-alarm (in a 80 dB(A) background noise). However, the noise-alarm could be used if it is less disturbing than the tone-alarm. The present work aims to compare the discomfort caused by exposure to tone and noise back-up alarms. About 50 people rated two times the discomfort caused by 11 sound environments. Each sound environment consisted of a site construction sound environment set at about 65 dB (A) mixed with a sequence of tone or noise alarm set at 56, 59, 62, 65, or 68 dB(A). One of the 11 stimuli had no alarm. Through an ANOVA it can be said if the kind of alarm and/or the sound level has an influence on the discomfort.

10:20

4aNSa5. If a tree falls in a forest, can you hear it? Max P. Weichert (Bionik/Biomimetics in Energy Systems, Carinthia Univ. of Appl. Sci., Emil-von-Behring-Strasse 28, Villach 9500, Austria, max.weichert@edu.fh-kaernten.ac.at)

Anthropogenic noise, from slowly rotating/reciprocating machinery, contributes immensely to environmental low-frequency noise below 100 Hz. This poses an acoustic low-energy problem as our technology offers no effective passive method for remediation. While attention is given to negative impact on human health and ecosystems, we still know and publicly do too little about it. Technical standards in Europe are A-weighted measurements that disregard actual energetic contributions of low frequencies. Different aspects of low-frequency sound have been illuminated by complementing acoustic measurements with findings from technical acoustics, bioacoustics, and human medical science. Sound pressure levels (SPL) were measured from 2.5 Hz to 20 kHz. The measurement setup consisted of SINUS Soundbook quadro (2.3 Hz– 22 kHz; linear@0 dB; ±0.1 dB tolerance; 1 Hz high-pass enabled), 1/2-in. Preamplifier/Free-Field Microphone (3.15 Hz– 20 kHz: ±2.0 dB; 5Hz–10 kHz: ±1.0 dB). Signals were processed with SINUS Driver version 5.1.0.8 and Samurai version 2.0.134. SPL in quiet forest [during 60 s: LZ,eq = 59.3 dB; LA,eq = 29.5 dB(A)] were compared to a quiet university room [during 60 s: LZ,eq = 74.4 dB; LA,eq = 28.2 dB(A)]. Healing properties of felid purrs produced with strong frequencies at an SPL between 30 and 60 dB suggest a general correlation of low-frequency background noise levels and health. Perhaps nature knows better than we do.

4a THU. AM

10:40

4aNSa6. Prediction of the effectiveness of a sound-masking system in an open-plan office including the Lombard Effect. Yizhong Lei and Murray Hodgson (Mech. Eng., Univ. of British Columbia, -Ste. 2137, 6335 Thunderbird Cres., UBC, Vancouver, BC V6T2G9, Canada, light.lei@hotmail.com)

Sound masking can improve speech privacy in rooms by increasing background-noise levels that mask distracting speech sounds. The Lombard Effect indicates that an increase in background-noise level can increase talker voice levels, reducing speech privacy and the benefit of a sound-masking system. To investigate this, a model of an existing open-plan office was created in CATT-Acoustics and validated. The model was used to predict speech-transmission index (STI) and the effectiveness of a sound-masking system, without and with the Lombard Effect, described by the existing Lombard Voice Model. Predictions were made for ambient-noise levels of 30, 40, and 45 dBA, at various distances from a primary talker, and for 0–4 secondary talkers. With 30-dBA ambient noise, STIs at 1 and 4 m varied with the number of talkers from 0.67 to 0.91 and 0.23 to 0.62 without the sound-masking system, from 0.58 to 0.70 and 0.13 to 0.25 with it but ignoring the Lombard Effect, and from 0.64 to 0.73 and 0.18 to 0.27 with the

Lombard Effect. The Lombard Effect reduced the benefit of the sound-masking system by 3–9% and 7–22%. With higher ambient noise, the system is less effective; the Lombard Effect can almost cancel its benefit, resulting in increased STI (decreased privacy) with the system operating.

11:00

4aNSa7. Acoustical characteristics of Technology Educational Shops in British Columbia. Ahmed Summan and Murray Hodgson (SPPH, UBC, 5613 Montgomery Place, Vancouver, BC V6T2C8, Canada, ahmedsumman@gmail.com)

Technology Educational Shops (TES) are designed to develop high school students' technological literacy. Their acoustical conditions play a dominant role in the quality of these environments. TES are, at the same time, classrooms for learning and industrial workshops for making things. Each use has its own standards governing its acoustical characteristics: ANSI S12.60-2002 for classrooms and the Ondet & Sueur DL2 criteria for workshops. A major conflict could exist by using the same room for two different purposes. This study investigated this conflict by evaluating the acoustical characteristics of 20 unoccupied wood, metal and automotive shops. It conducted measurements of background noise level (BNL), reverberation time (RT), speech intelligibility index (SII), and DL2. Results showed that BNLs and RTs in most TES were higher than the acceptability criteria for unoccupied core learning spaces. SII values indicated bad/poor speech intelligibility for normal and raised voice levels and reasonable/good speech intelligibility for loud and shout voice levels. DL2 values were found acceptable in TES larger than 100 m^2 in floor area. In general, these results indicate the poor acoustical conditions of TES as classrooms, and the need for special sound control measures.

11:20

4aNSa8. Sound quality in small music classrooms. Iara B. Cunha, Tiago Mattos, and Stelamaris R. Bertoli (FEC - Arquitetura e Construção, UNICAMP, Av. Albert Einstein, 951 - Caixa Postal: 6021, Campinas, SP 13083-852, Brazil, iarabcunha@gmail.com)

Good conditions for teaching, learning, and practicing a musical instrument are crucial for a musician's progress. Small rooms for teaching and practicing musical instruments have specific requirements about acoustic performance and strong influence on the perception of the users. Acoustical problems in music classrooms may cause difficulties for teachers to identify mistakes from the young students performance other than those originated from the room itself. For this paper, three music classrooms were evaluated according to reverberation, background noise, and airborne sound insulation between rooms. Thus, reverberation time (RT), background noise level, and standardized level difference (DnT), as a function of frequency and according to ISO 3382-2:2008 and ISO 140-4:1998 standards, were measured and judged. As some results have disagreed from literature recommendations, the main faults of each room were highlighted, considering the type of instrument that is taught, and suggestions for acoustic adjustment were made.

11:40

4aNSa9. Nocturnal vibration from high numbers of freight trains leads to fragmented sleep. Michael G. Smith, Ilona Croy, Oscar Hammar, Mikael Ögren, and Kerstin Persson Waye (Occupational and Environ. Med., Univ. of Gothenburg, Box 414, Gothenburg 40530, Sweden, michael.smith@amm.gu.se)

There are an increasing number of freight trains on the European railway networks, and this growth has been facilitated through use of the available night time periods. Freight trains are particularly problematic with regards to generation of low frequency vibration and noise which has the potential to propagate to nearby homes and influence the sleep of residents. To investigate the potential impact we conducted a laboratory trial on 24 young healthy persons to ascertain physiological and psychological reactions to nocturnal vibration and noise from freight traffic, and to examine differences between gender and noise sensitivity. Nights with low (0.0102 m/s^2) and high (0.0204 m/s^2) peak weighted vibration amplitudes and low (20) and high (36) number of train passages were simulated with noise levels being of the same order between nights. Polysomnography was used to record sleep stage and EEG arousals and awakenings. Event related cardiac activations were analyzed using ECG recordings. Questionnaires were administered in the evenings and mornings to obtain subjective sleep parameters. Sleep was more fragmented during nights with higher vibration amplitudes and number of events. Furthermore, heart rate response was higher in the high vibration condition. Results from the subjective data showed less discrimination between nights.

THURSDAY MORNING, 6 JUNE 2013 — 511CF, 8:55 A.M. TO 10:40 A.M.

Session 4aNSb

Noise, Animal Bioacoustics, Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Future of Acoustics

Michael J. Buckingham, Cochair
SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238

Brigitte Schulte-Fortkamp, Cochair
Technische Universität Berlin, Einsteinufer 25, Berlin 10178, Germany

Chair's Introduction—8:55

Invited Papers

9:00

4aNSb1. Ocean noise: Lose it or use it. William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238, wkuperman@ucsd.edu)

Ocean noise, natural or man-made, has typically been treated as unwanted interference in the context of detecting signals. However, more recently noise has itself also become a signal of interest in which, for example, ocean or geophysical properties are embedded in the noise field. There is now significant ongoing research in trying to extract information from noise. Much of the latter has utilized either own-ship noise, surface generated noise, biological noise, and/or distant shipping noise. Here we review the evolution of research in ocean noise as it progressed from basic descriptive categorization to attempts to minimize its impact on signal detection, and finally, to its utilization as an environmental descriptor.

9:20

4aNSb2. Ambient noise measurements with deep sound in the Philippine Sea. Michael J. Buckingham and David R. Barclay (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Most measurements of ambient noise in the deep ocean have been performed using an array of hydrophones located at a fixed depth. Recently, an instrument platform known as Deep Sound has been developed, consisting of a glass sphere containing data acquisition, data storage, and system control electronics, with a pair of vertically aligned hydrophones mounted externally. Deep sound descends under gravity, jettisons a drop weight at a pre-assigned depth, and returns to the surface under buoyancy, traveling in both directions at a nominal 0.6 m/s. Throughout the descent and ascent, the hydrophones record the ambient noise over a bandwidth from 3 Hz to 30 kHz. In April–May 2009, Deep Sound was deployed to a depth of 5500 m in the Philippine Sea. The vertical coherence of the measured noise, from 1 to 10 kHz, matches accurately a simple theory of deep-water, wind-generated ambient noise, provided that the local sound speed is used in evaluating the theoretical coherence function. Moreover, the cross-correlation function of the noise, obtained by taking the Fourier transform of the coherence function, provides the basis of an inversion technique for returning the sound speed profile in the water column. [Research supported by ONR.]

9:40

4aNSb3. Soundscape-focusing on resources. Brigitte Schulte-Fortkamp (Technische Universität Berlin, Einsteinufer 25, Berlin 10178, Germany, b.schulte-fortkamp@tu-berlin.de)

In contrast to many other environmental problems, noise pollution continues to grow, and it is accompanied by an increasing number of complaints from people exposed to the noise. The growth in noise pollution involves direct, as well as cumulative, adverse health effects. It also adversely effects future generations, and has socio-cultural, aesthetic, and economic effects. Therefore, the concept of noise annoyance needed to be broaden to an integrated environmental, psychosocial, and socioeconomic assessment of the community situation to reach a more realistic basis for environmental impact and health risk assessments. Soundscape research represents a timely paradigm shift in that it combines physical, social, and psychological approaches and considers environmental sounds as a “resource” rather than a “waste” to satisfy human needs and wants. Moreover, balancing between the expertise from people living in respective areas and acoustic measurements, architectural planning will lead to a new understanding of a concept of an environment under “noise control” as soundscape suggests exploring noise in its complexity, its ambivalence, and its approach toward sound and quality of life.

10:00–10:40 Panel Discussion

THURSDAY MORNING, 6 JUNE 2013 511CF, 11:00 A.M. TO 12:00 NOON

Session 4aNSc

Noise: Children’s Perception of Noise

Irene van Kamp, Cochair
Ctr. for Environ. Health Res., RIVM, P.O. Box 1 Postbus 10, Bilthoven 3720 BA, Netherlands

Janina Fels, Cochair
Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52074, Germany

Invited Papers

11:00

4aNSc1. Current perspectives on children’s auditory perception and consequences of noise exposure effects. Sofie Fredriksson (Occupational and Environ. Medicine, Univ. of Gothenburg, Box 414, Gothenburg 40530, Sweden, sofie.fredriksson@gu.se), Janina Fels (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany), and Kerstin Persson Waye (Occupational and Environ. Medicine, Univ. of Gothenburg, Gothenburg, Sweden)

Exposure to high sound pressure levels is well known to cause auditory damage, regardless of age. There is however limited knowledge of the effects on hearing due to noise exposure early in life. In addition, no well-established model is used to describe how children perceive and experience their sound environment compared to adults. New studies of children’s hearing have revealed different directivity pattern especially at high frequencies given by the head-related transfer functions due to the anthropometric data of the children and also an ear canal resonance at considerable higher frequencies compared to adults. Recent studies also describe children feeling a great deal of discomfort when exposed to sounds with high frequency characteristics. Children today are exposed to high sound levels from an early age at preschool, school and during leisure time. Few studies have looked at general health effects or hearing in particular. It is

being discussed whether age related hearing loss, regarded as an inevitable part of life, to a large extent may be caused by a lifetime of noise exposure starting early in life. This paper will review available studies on noise induced hearing damage among children and give suggestions for future studies within this field.

11:20

4aNSc2. Laboratory study on effects of environment noise on children's short-term memory. Hui Ma and Shengnan Gong (School of Architecture, Tianjin Univ., No. 92 Weijin Rd., Nankai District, Tianjin 300072, China, mahui@tju.edu.cn)

In laboratory settings 18 children of 6-7 years old were asked to finish series of short-term memory tasks under the different sound situation. The noise stimuli used in the experiment were both road traffic noise and low frequency noise of LAeq.2min = 40, 45, and 50 dB to simulate the indoor sound condition when the children at school or at home. The result showed both road traffic noise and low frequency noise had influence on children's task achievement and significant adverse effects on subjective noise annoyance evaluation. Comparing with the linear dose-response relationship of road traffic noise, the effects brought by low frequency noise to children's short-term memory work and noise annoyance evaluation was more complicated. There was an interesting trend that higher noise annoyance was evaluated, the higher task marks was obtained. As for the gender difference, boys showed more sensitivity to low frequency noise than girls.

11:40

4aNSc3. The effects of noise disturbed sleep on children's health and cognitive development. Irene van Kamp (Ctr. for Environ. Health Res., RIVM, P.O. Box 1 Postbus 10, Bilthoven, Utrecht 3720 BA, Netherlands, irene.van.kamp@rivm.nl), Anita Gidlöf-Gunnarsson, and Kerstin Persson Waye (Occupational Medicine and the Environment, Gothenburg Univ., Gothenburg, Sweden)

Undisturbed sleep is essential for physiological and psychological health. Children have a special need for uninterrupted sleep for growth and cognitive development. Noise is an environmental factor that affects most children. In addition to noise in schools and preschools, many are exposed to potentially disturbing traffic related noise at night. The knowledge of how children's health, wellbeing and cognitive development is affected by noise disturbed sleep due to road traffic is very incomplete. Nor do we know how children are able to handle noisy situations (coping) and if a learned noise-related behavior in the long term has a negative influence on children's health and learning. The need for a restorative home environment can be particularly important when the child is simultaneously exposed to noise in the school environment. Moreover, it has been shown that although children are less sensitive for awakenings and sleep cycle shifts due to nighttime exposure, they are more sensitive for physiological effects such as blood pressure reactions and related motility during sleep. This paper aims to review existing knowledge on how children's health and cognitive development are affected by noise in the home and school environment, with special focus on the importance of noise-disturbed sleep.

THURSDAY MORNING, 6 JUNE 2013 — 519B, 9:00 A.M. TO 11:40 A.M.

Session 4aPA

Physical Acoustics: Nonlinear Acoustics I

Kent L. Gee, Chair
Brigham Young Univ., N243 ESC, Provo, UT 84602

Contributed Papers

9:00

4aPA1. Experimental verification of a wave-vector-frequency-domain nonlinear acoustic model. Yun Jing (Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695, yjing2@ncsu.edu) and Jon Cannata (HistoSonics, Ann Arbor, MI)

In this paper, a recently developed wave-vector-frequency-domain method for nonlinear wave propagation is verified by underwater experiments. A specially designed focused transducer was used to generate short high intensity pulses. 2D scans were conducted at a pre-focal distance, which were later used as the input to the numerical model to predict the acoustic field at focal and post-focal zones. Adaptive attenuation was introduced to reduce the Gibbs effect. Graphic processing units (GPU) were also employed to speed up the computation. Good agreement was observed between the simulation and experiment.

9:20

4aPA2. Ballistic shock wave localization estimation of shooter position and velocity using difference of time of arrival DTOA algorithm in orthogonally arranged discrete acoustic arrays. Murray S. Korman (Dept. of Phys., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu) and Antal A. Sarkady (Dept. of Elec. and Comput. Eng., U.S. Naval Acad., Annapolis, MD)

A mathematical algorithm was developed using the difference in time of arrival, DTOA, of the ballistic shock wave cone at each position of an N-element array. The array is thought to be made up of orthogonally arranged discrete wide-band point like microphone or piezo-electric elements. The algorithm is used to estimate the parameter space involving both the displacement of the shooter with respect to the array and the velocity of the projectile. The algorithm utilizes a nonlinear least squares parameter fit of a

difference in time of arrival equation that involves a lengthy Taylor series expansion of the exact "difference in time of arrival" theoretical equation. Results are presented showing that the model has good versatility in estimating displacement (location) and velocity in simulated trials that are presented. The method does not require any muzzle blast wave information or knowledge of the absolute time of arrival of the shock wave. Therefore, the orthogonal array located far from the shooter will be able to estimate the localization parameters as long as the projectile miss distance is not so large that ambient noise does not mask the detection of the shock wave pressure signature.

9:40

4aPA3. Compressively excited acoustic wave propagation in a three-dimensional channel bifurcated by an elastic partition. Katherine Aho and Charles Thompson (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, katherine_aho@student.uml.edu)

Fluid motion resulting from compression of the exterior walls of a three-dimensional channel that is axially partitioned by a flexible membrane is examined. The axial variation in the acoustic impedance of the partition gives rise to a pressure gradient across the partition and generates an evanescent field at its surface. The effects of these evanescent modes and their impact on spatial wavenumber selectivity are of particular interest. The Green's function for the pressure is given for the case of low fluid compressibility and channel high aspect ratio. In this limit, it is shown the evanescent modes may be modeled in terms of generalized functions. The complete solution of the pressure field inside the channel will be shown. Implication of the results on the development a MEMS transducer in which the displacement of membrane is used as the method of transduction is presented. [NSF Grant 0841392.]

10:00

4aPA4. Propagation of N-waves in a turbulent and refracting atmosphere with ground effects (laboratory-scale experiment). Sébastien Ollivier, Edouard Salze, and Philippe Blanc-Benon (LMFA - UMR CNR 5509, Univ. Lyon 1, Centre Acoustique - Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134, France, sebastien.ollivier@univ-lyon1.fr)

Sound propagation in a refracting atmosphere leads to the formation of a geometrical shadow zone close to the ground, and of an illuminated zone above a limiting ray. Without turbulence, acoustic waves can be diffracted into the shadow zone close to the ground. When propagating through a turbulent atmosphere, it is known that waves can be distorted, scattered, or focused. In order to investigate how turbulence modifies the pressure field into the shadow zone, a well controlled laboratory-scale experiment has been performed. An electrical spark source is used to generate short duration (20 μs) and high pressure (1500 Pa) N-waves. A convex surface models the effect of an upward refracting atmosphere, and a heating grid generates thermal turbulence (1% fluctuations of refraction index). To compute statistics of wave parameters variation, seven 1/8 in. microphones have been used to record 2000 waves at each position after propagation through the turbulent field. Wave parameters (peak pressure, rise time) obtained with turbulence are compared to data obtained without turbulence. Results show that turbulence scatters sound into the shadow zone, which increases significantly the noise level.

10:20

4aPA5. The influence of a focused acoustic field on mass-transfer processes at a heterogeneous boundary. Dmitry Kasyanov (Radiophysical Res. Inst., 25/12a Bolshaya Pecherskaya St., Nizhny Novgorod 603950, Russian Federation, da_kasyanov@nirfi.sci-nnov.ru)

An experiment is conducted on estimating the velocity of a Schlichting boundary flow arising when a focused field falls on a rigid boundary in a liquid. Also temperature increase into the acoustic boundary layer is experimentally estimated in this situation using optic means. The velocity of a small-scale Schlichting flow is determined by an indirect method from the characteristics of the cocurrent Rayleigh flow using the particle image velocimetry method. The velocity of the Schlichting flow attained in experiments gives the possibility of significantly accelerating mass-transfer processes at a heterogeneous boundary, which is confirmed by experimental results on acoustic intensification of rapid growth of salt monocrystals conducted under strictly controlled laboratory conditions. It is shown the increase in the temperature of the boundary layer is insufficient for significant retardation of the crystal growth.

10:40

4aPA6. Statistical properties of nonlinear N-wave propagating in thermal or kinematic turbulence. Petr V. Yuldashev (Dept. of Acoust., Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Lab. de Mécanique des Fluides et d'Acoustique, UMR CNRS 5509, Ecole Centrale de Lyon, 119991, Russian Federation, Moscow, Leninskie Gory, Moscow 119991, Russian Federation, petr@acs366.phys.msu.ru), Sébastien Ollivier (Lab. de Mécanique des Fluides et d'Acoustique, UMR CNRS 5509, Université de Lyon, Ecole Centrale de Lyon, Ecully, France), Vera Khokhlova (Dept. of Acoust., Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation), and Philippe Blanc-Benon (Lab. de Mécanique des Fluides et d'Acoustique, UMR CNRS 5509, Ecole Centrale de Lyon, Ecully, France)

Nonlinear propagation of high amplitude N-wave through turbulent layer is studied using 2D KZK-type nonlinear parabolic equation. The incident acoustic wave is assumed to have a plane wavefront and the waveform is a classical symmetrical N-wave. The turbulent layer is synthesized using a method of random Fourier modes. The modified von Karman spectra with the same values of outer and inner scales are considered for both scalar-type (temperature fluctuations) and vector-type (velocity fluctuations) turbulent fields. The rms value of the refraction index fluctuations μ is varied as a parameter. It is shown that statistical characteristics of N-wave propagating in vector-type or scalar-type turbulent fields are equivalent when μ in scalar-type turbulence is almost two times greater. The distance of most probable occurrence of caustics obtained in the KZK simulations, which account for diffraction effects is demonstrated to be inversely proportional to μ while the geometrical acoustics approach predicts the inverse proportionality to 2/3 power of μ. An effect of the initial N-wave amplitude on the peak pressure in random caustics is analyzed. The enhancement of focusing efficiency is observed for moderate initial amplitudes, whereas strong nonlinearity is shown to reduce pressure amplitudes. [Work supported by PICS RFBR 10-02-91062/CNRS 5603 grants.]

11:00

4aPA7. The numerical simulation of propagation of intensive acoustic noise. Igor Demin, Gurbatov Sergey, and Pronchatov-Rubtsov Nikolay (Acoustics, Nizhny Novgorod State Univ., 23, Gagarin Ave., Nizhny Novgorod 603950, Russian Federation, phdem56@gmail.com)

The propagation of intensive acoustic noise is of fundamental interest in nonlinear acoustics. Some of the simplest models describing such phenomena are generalized Burgers' equations for finite amplitude sound waves. An important problem in this field is to find the wave's behavior far from the emitting source for stochastic initial waveforms. The method of numerical solution of generalized Burgers equation proposed is step-by-step calculation supported on using Fast Fourier Transform of the considered signal. The general idea is to keep only Fourier image of concerned signal and update it recursively (in space). For simulating the wave evolution we used 4096 (212) point realizations and took averaging over 1000 realizations. Also the object of the present study is a numerical analysis of the spectral and bispectral functions of the intense random signals propagating in non-dispersive nonlinear media. The possibility of recovering the input spectrum from the measured spectrum and bispectrum at the output of the nonlinear medium is discusses. The analytical estimations are supported by numerical simulation. For two different types of primary spectrum evolution of jet noise were numerically simulates at a short distance and assayed bispectrum and a spectrum analysis of the signals.

11:20

4aPA8. Radiation of finite-amplitude waves from a baffled pipe. K. J. Bodon, Derek C. Thomas, Kent L. Gee, Rachael C. Bakaitis (Physics, Brigham Young Univ., Provo, UT 84602, joshuabodon@gmail.com), David T. Blackstock, and Wayne M. Wright (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The radiation of finite-amplitude waves from the open end of a baffled, circular pipe is considered as a direct continuation of work begun by Blackstock and Wright more than three decades ago [Kuhn *et al*., J. Acoust. Soc. Am. **63**, (1978), S1, S84]. Band-limited Gaussian noise, as well as 1, 1.5, and 2 kHz sinusoidal pulses, with initial sound pressure levels ranging from

552 to 1186 Pa, have been propagated down a 6.1 m pipe, whose open end (5.1 cm inner diameter) has been placed off-center in a large rectangular baffle. As the steepened or shock-like waves exit the pipe, the measured waveforms are comprised of sharp impulses that are delta function-like in nature, particularly on axis. Although linear piston theory predicts similar waveform shapes, there is also evidence that nonlinear propagation of these impulses, which can exceed 150 Pa near the pipe opening, is occurring. http://asadl.org/jasa/resource/1/jasman/v63/iS1/pS84_s5

THURSDAY MORNING, 6 JUNE 2013 — 514ABC, 8:55 A.M. TO 12:00 NOON

Session 4aPPa

Psychological and Physiological and Animal Bioacoustics: Biomechanics of Hearing

Sunil Puria, Chair
Mech. Eng., Stanford Univ., 496 Lomita Mall, Stanford, CA 94305

Chair's Introduction—8:55

Invited Papers

9:00

4aPPa1. Mechano-acoustical measurement and modeling of the outer and middle ear. W. Robert J. Funnell (BioMedical Eng. and Otolaryngol. – Head & Neck Surgery, McGill Univ., 3775, rue University, Montreal, QC H3A 2B4, Canada, robert.funnell@mcgill.ca)

Mechano-acoustical measurement and modeling have evolved together. Most early measurements of the behavior of the outer and middle ear produced either spatial averages or single-point observations, which were amenable to modeling with uniform transmission lines and lumped circuits. A major step forward was the measurement of displacement patterns on the eardrum, which called for the use of finite-element models. Other major experimental steps forward included measuring spatial sound-pressure distributions, 3-D displacement patterns, and intracochlear pressures. Use of the finite-element method made it desirable to obtain detailed 3-D shape measurements, which were made much easier by the introduction of magnetic-resonance microscopy and x-ray microCT. The finite-element method has also made it possible to exploit measurements of material properties, and several different approaches have been used recently for making such measurements. The greatest challenges may be in dealing with very small dimensions and non-linear visco-elastic behavior. There is a need for more and better 3-D multipoint vibration measurements, and for material-property measurements that are more localized and that span a broader frequency range. Important directions for modeling include better use of available shape and material-property data, more attention to experimental animals and to variability, and better integration with cochlear models.

9:20

4aPPa2. Carhart's notch: A window into mechanisms of bone-conducted hearing. Namkeun Kim, Charles R. Steele, and Sunil Puria (Dept. of Mech. Eng., Stanford Univ., 496 Lomita Mall, Stanford, CA 94305, chasst@stanford.edu)

Otosclerosis is a disease process of the ear that stiffens the stapes annular ligament and results in footplate immobilization. This produces a characteristic loss in bone-conducted (BC) hearing of about 20 dB between 1 and 2 kHz, known as "Carhart's notch," for which the specific mechanisms responsible have not yet been well understood. In this study, it is hypothesized that this observed pattern of hearing loss results from interactions between compressional and inertial mechanisms of BC hearing. Differences in the basilar-membrane velocity between a normal and otosclerotic human ear were calculated in response to compressional vibration of the cochlear capsule, translational vibration of the skull bone in various directions, and combinations of the two, using an anatomically accurate 3-D finite element model of the middle ear, cochlea, and semicircular canals. Compressional and inertial BC stimuli were found to both be necessary to capture the full behavior of clinical data, with the compressional component dominating below 0.75 kHz, the inertial component dominating above 3 kHz, and the notch between 1 and 2 kHz resulting from the suppression of an ossicular resonance due to stapes fixation. [Work supported by grant R01-DC07910 and R01-DC05960 from the NIDCD of NIH.]

9:40

4aPPa3. Fast waves, slow waves and cochlear excitation. Elizabeth S. Olson (OTO/HNS & Biomed. Eng., Columbia Univ., 630 W.168th St., P&S 11-452, New York, NY 10032, eao2004@columbia.edu)

In idealized cochlear models [especially, Peterson and Bogert, J. Acoust. Soc. Am. (1950)], intracochlear pressure is decomposed into two modes, the compression pressure (fast mode) and the traveling wave pressure (slow mode). Because the cochlear fluid is nearly incompressible, only the slow mode leads to significant motion. In the real cochlea, additional fast modes exist. These evanescent modes are similar to the traveling wave mode in driving significant fluid and tissue displacement. They are present in the region of the cochlear windows, where the anatomy is not the ideal symmetric structure of basic cochlear models. Evanescent modes also have emerged in experimental work in which cochleostomies are made in the apex. At high stimulus level fast modes are able excite hair cells, leading to auditory nerve responses. However, fast mode motions do not seem to be amplified by the cochlear amplifier. This observation supports the concept that the amplifier relies on traveling wave curvature, as has been proposed in cochlear models [for example, Yoon *et al*., Biophys. J. (2011)]. I will review experimental and theoretical results on the fast and slow waves, and propose experiments in which wavelength-based theories of amplification could be tested.

10:00–10:20 Break

10:20

4aPPa4. Comparative auditory biomechanics probed by otoacoustic emissions. Christopher Bergevin (Phys. & Astronomy, York Univ., 4700 Keele St., Petrie 240, Toronto, ON M3J 1P3, Canada, cberge@yorku.ca), Wei Dong (Columbia Univ., New York, NY), Laurel Carney (Univ. of Rochester, Rochester, NY), David S. Velenovsky, Kevin E. Bonine (Univ. of Arizona, Tucson, AZ), and James L. Jarchow (Sonora Veterinary Group, Tucson, AZ)

Since Kemp's discovery in 1978, otoacoustic emissions (OAEs) have provided valuable scientific and clinical tools for the study of the ear. For example, OAEs can provide objective measures of sensitivity and selectivity over the frequency range of "active" hearing. Given the universality of OAEs across the kingdom Animalia, comparative studies can reveal how various morphological factors affect peripheral auditory transduction and thereby what information is encoded for higher level cognition. Motivated by the complexity of cochlear mechanics and the many unknowns that currently exist, the present study describes OAEs stemming from two non-mammalian groups whose auditory periphery is relatively simpler than that of mammals: several lizard genera (Heloderma, Tiliqua, Agama, and Tupinambis) that exhibit significant relative differences in tectorial membrane structure, and a highly vocal bird species (*Melopsittacus undulatus*). By utilizing recent improvements in OAE measurement and analysis strategies combined with quantitative anatomical measures (e.g., number of hair cells), these data shed new light upon emission generation mechanisms and how such tie back to a given species' ability to encode ecologically relevant sounds. Furthermore, these data serve to inform theoretical models of auditory biophysics by clarifying what roles various morphological features do (or do not) play.

10:40

4aPPa5. Active processes and sensing in the cochlea. Karl Grosh and Julien Meaud (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu)

One key question in the biophysics of the mammalian cochlea is determining the relative contribution to cochlear amplification by the two active processes present in the outer hair cell, namely prestin-based somatic motility and hair bundle (HB) motility. In the biological cochlea, these two effects are intimately coupled as HB force generation is linked to calcium-dependent adaptation of the transduction current and somatic force generation is driven by the depolarization caused by the same transduction current. To separate these effects, we construct a global mechanical-electrical-acoustical mathematical model of the cochlea. The global cochlear model is coupled to linearizations of nonlinear somatic motility and HB motility. We find that the active HB force alone is not sufficient to power high frequency cochlear amplification while somatic motility can perform this task. We discuss the limitations to this mathematical approach along with existing seminal experiments and proposed experiments (both in the cochlea and in the auditory nerve) to map future directions for uncovering the micromechanical contributions to the system level response of the cochlea. Describing the relation between the microfluidic flow stimulating the inner hair cell HB and the active processes in the cochlea and is an important future research direction. [Support: NIH-NIDCD.]

11:00

4aPPa6. Weak lateral coupling between stereocilia of mammalian cochlear hair cells requires new stimulus methods to study the biomechanics of hearing. K. Domenica Karavitaki (Neurobiology, Harvard Med. School, 220 Longwood Ave., Goldenson 443, Boston, MA 02115, dkaravitaki@hms.harvard.edu), Paul D. Niksch, and David P. Corey (Neurobiology, Harvard Med. School and Howard Hughes Medical Inst., Boston, MA)

The forces felt by different transduction channels in a bundle depend critically on how well stereocilia remain cohesive during deflection. In the bullfrog saccule, sliding adhesion mediated by horizontal top connectors (HTC) confers coherent motion to hair cell stereocilia and parallel gating to all transduction channels. In cochlear inner and outer hair cells (IHCs and OHCs), the mature complement of HTC is established by postnatal day 12; they extend between adjacent stereocilia of both rows and columns. Contrary to our expectation that bundle cohesion should be robust in all directions, our experiments on gerbil cochleas show that lateral coupling among stereocilia of the tallest row is weak. These findings suggest that the function and molecular composition of the HTC in the cochlea are different from those in bullfrog hair cells. They also raise concern for current stimulus methods, which involve glass probes that are often small compared to the bundle width. Our data suggest that only stereocilia in contact with the probe are stimulated, and delivery of the stimulus to the remaining stereocilia is weak and inhomogeneous. To mimic the OHC stimulus delivered by the overlying tectorial membrane *in vivo*, we are developing new stimulation technologies.

11:20

4aPPa7. The elusive hair cell gating spring, a potential role for the lipid membrane. Jichul Kim, Peter M. Pinsky, Charles R. Steele, Sunil Puria (Dept. of Mech. Eng., Stanford Univ., Stanford, CA), and Anthony J. Ricci (Dept. of Otolaryngol.-HNS, Stanford Univ., 496 Lomita Mall, Stanford, CA 94305, tricci@ohns.stanford.edu)

Deflection of auditory hair cell hair bundle results in a nonlinear (i.e., non Hookean) force-displacement relationships whose molecular mechanism remains elusive. A gating spring model posits that mechanosensitive channels are in series with a spring such that channel opening puts the activation gate in series with the spring, thus reducing spring extension until further stimulation is provided. Here we present a theoretical analysis of whether the lipid membrane might be the source of nonlinearity. A hair bundle kinematic model is coupled with a lipid membrane model that includes a diffusible compartment into which the tip-link embeds and a minimally diffusive reservoir pool. Using physiological parameters, this model was capable of reproducing nonlinear force-displacement plots, including a negative stiffness component but required a standing tip-link tension. In addition, this model suggests the mechanotransducer channel is most sensitive to curvature forces that are located within 2 nm of the tip-link. [Work supported in part by Grant Nos. R01-DC07910 and R01-DC03896 from the NIDCD of NIH and by The Timoshenko fund from Mechanical Engineering Department at Stanford University].

11:40–12:00 Panel Discussion

Session 4aPPb

Psychological and Physiological Acoustics: Binaural Hearing (Poster Session)

Pavel Zahorik, Chair
Psychol. and Brain Sci., Univ. of Louisville, Life Sci. Bldg. 347, Louisville, KY KY

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

4aPPb1. Amplitude modulation detection by human listeners in reverberant sound fields: Effects of prior listening exposure. Pavel Zahorik and Paul W. Anderson (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Life Sciences Bldg. 347, Louisville, KY, pavel.zahorik@louisville.edu)

Previous work [Zahorik *et al*., POMA, **15**, 050002 (2012)] has reported that for both broadband and narrowband noise carrier signals in a simulated reverberant sound field, human sensitivity to amplitude modulation (AM) is higher than would be predicted based on the acoustical modulation transfer function (MTF) of the listening environment. These results may be suggestive of mechanisms that functionally enhance modulation in reverberant listening, although many details of this enhancement effect are unknown. Given recent findings that demonstrate improvements in speech understanding with prior exposure to reverberant listening environments, it is of interest to determine whether listening exposure to a reverberant room might also influence AM detection in the room, and perhaps contribute to the AM enhancement effect. Here, AM detection thresholds were estimated (using an adaptive 2-alternative forced-choice procedure) in each of two listening conditions: one in which consistent listening exposure to a particular room was provided, and a second that intentionally disrupted listening exposure by varying the room from trial-to-trial. Results suggest that consistent prior listening exposure contributes to enhanced AM sensitivity in rooms, but that it is not the sole determinant of the enhancement. [Work supported by the NIH/NIDCD.]

4aPPb2. Spatial consistency as a cue for segregation and localization. Brian Simpson (Battlespace Acoust., Air Force Res. Lab., 2610 Seventh St., Area B, Bldg. 441, Wright-Patterson AFB, OH 45433, brian.simpson@wpafb.af.mil), Robert Gilkey (Psychology, Wright State Univ., Dayton, OH), Eric Thompson (Ball Aerosp. and Technologies Corp., Wright-Patterson AFB, OH), Douglas Brungart (Audiol. and Speech Ctr., Walter Reed National Military Med. Ctr., Bethesda, MD), Nandini Iyer, and Griffin Romigh (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Many real-world auditory scenes are dynamic and complex, with multiple sounds that may change location over time. In this experiment, we examined the ability of listeners to localize a spatially consistent target sound in a dynamic, spatially varying auditory scene. The target and masker stimuli were composed of sequences of 60-ms bursts of uncorrelated noise (2 to 16 bursts in duration) and differed only in their degree of spatial consistency. Specifically, each target burst within a sequence came from the same spatial location (which varied from trial to trial), whereas each masker burst within a sequence came from a different, randomly chosen spatial location. The listener's task was to localize the spatially-consistent sequence. Localization errors decreased by approximately 11° with each doubling of the sequence duration, and approached quiet performance with 16-burst sequences. Adding a second masker increased localization errors by approximately 14° overall. These results suggest that spatial information can be combined across multiple observations over time to identify and localize a spatially consistent target in a dynamic auditory scene. These data will be discussed in terms of the information obtained from each burst and the manner in which the information is combined across bursts.

4aPPb3. Head movement during horizontal and median sound localization experiments in which head-rotation is allowed. Daisuke Morikawa, Yuki Toyoda, and Tatsuya Hirahara (Faculty of Eng., Toyama Prefectural Univ., 5180 Kurokawa, Imizu, Toyama 939-0398, Japan, t074001@st.pu-toyama.ac.jp)

We measured subjects' head movements during horizontal and median sound localization experiments in which head-rotation was allowed in order to know how they move their heads to localize sound in a head rotation condition. The head movements in a head-rotation condition were measured while localizing 500-Hz low-pass noise, 12-kHz high-pass noise, and white noise. With regard to horizontal plane, sound localization became easier with head-rotation than head-still condition. All subjects turned their heads toward the presented sounds, yet they did not necessarily turn their heads to face the sound. The amount of head rotation was small for WN compared to that for LPN and HPN. Sound localization also became easier with head-rotation than head-still condition for the median plane. All subjects swung their heads right and left centering on 0°, no matter what the stimulus elevation angle was.

4aPPb4. Integration of auditory input with vestibular and neck proprioceptive information in the interpretation of dynamic sound localization cues. Janet Kim (Health and Rehabilitation Sci. Program, Western Univ., 62 Essex St, London, ON N6G 1B2, Canada, jkim2223@uwo.ca), Michael Barnett-Cowan (Brain and Mind Inst., Western Univ., London, ON, Canada), and Ewan Macpherson (National Ctr. for Audiol., Western Univ., London, ON, Canada)

To determine the front/back location of a sound source via head rotation, the auditory system must integrate sensorimotor information about head motion with the dynamic acoustic cues resulting from motion of the source relative to the head. In order to determine the influence of vestibular and proprioceptive cues on processing of dynamic acoustic cues, we measured, in active, passive, and counter-rotation conditions, listeners' ability to discriminate front/rear locations of low-frequency sounds not accurately localizable without head motion. Targets were presented over headphones during head rotations using dynamic virtual auditory space methods. In the active condition, the subject performed a head-on-body rotation, which provided vestibular and neck proprioceptive information. In the passive condition, proprioceptive information was minimized by whole-body rotation with no neck movement using a motorized rotating chair. In the counter-rotation condition, the subject performed a head-on-body rotation while the body was counter-rotated, which minimized vestibular input by keeping the head still in space. Dynamic acoustic cues corresponded to the head-on-body angle. Discrimination was accurate in active and passive conditions, but near chance under counter-rotation, suggesting that vestibular inputs are

necessary and sufficient to inform the auditory system about head movement, whereas proprioceptive cues are neither necessary nor sufficient.

4aPPb5. Falling stars: Acoustic influences on meteor detection. Darlene Edewaard and Michael S. Gordon (Dept. of Psych., William Paterson Univ., 300 Pompton Rd., William Paterson U., Wayne, NJ 07470, edewaardd@student.wpunj.edu)

As particles enter the earth's atmosphere they produce a burst of electromagnetic energy, including visible and radio-wave emissions. Consequently, just as meteors can be detected visually in the night sky they can be "heard" using radio telescopes. The current project investigated the potential influence of these audio signals on meteor detection. Anecdotally, and in related research, it has been found that auditory signals can enhance or even alter visual perception of objects. The current project examined the specific effects of accompanying auditory signals on the detection of meteors. Meteors present an interesting case of audiovisual integration in that detection paradigms often entail extended vigilance and extremely brief, yet brilliant astronomical events. Experiments specifically investigated how auditory signals that varied in spectra influenced changes in visual magnitude and duration judgments of meteors. In addition, research targeted how extraneous auditory cues during a vigilant meteor search might contribute to false judgments. Results are described in terms of audiovisual integration and the relation of perceptual mechanisms to meteor detection.

4aPPb6. Time-to-arrival discrimination of multiple sound sources. Michael S. Gordon, Darlene Edewaard, and Matthew Pacailler (Psychology, William Paterson Univ., 300 Pompton Rd., Wayne, NJ 07470, gordonm10@wpunj.edu)

Previous research in auditory detection of time-to-arrival (TTA) has tended to focus on single sound sources approaching a listener. However, visual studies of TTA have suggested that there may be different perceptual strategies employed with respect to single versus multiple objects on approach. The current research is designed to directly address the capacity and informational support of listeners to make determinations on the TTA of multiple sound sources. Initial experimentation evaluated the capacity for discrimination between a stationary and moving sound source. Additional studies evaluated multiple moving sound sources, with direct manipulation of their spectra to determine how various acoustic qualities contribute to TTA determinations. Results suggest the relative influence of intensity, frequency, and sound source dynamics as they facilitate competitive discriminations between approaching sound sources. Additional conclusions demonstrate changes in the use of spectral cues for lower and higher levels of auditory clutter as they might impact TTA.

4aPPb7. Performance of a highly directional microphone array in a multi-talker reverberant environment. Sylvain Favrot, Christine R. Mason, Timothy M. Streeter (Dept. of Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, sfav@bu.edu), Joseph G. Desloge (Sensimetrics Corp., Malden, MA), and Gerald Kidd (Dept. of Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., Boston, MA)

A visual guided hearing aid (VGHA) has recently been developed, which uses an eye tracker to steer the "acoustic look direction" (ALD) of a beamforming microphone array. The current study evaluates the performance of this highly directional microphone in providing spatial release from masking (SRM) under acoustically dry and reverberant conditions. Four normal-hearing subjects participated in a speech intelligibility test with collocated and spatially separated speech maskers when listening either through the microphone array or through KEMAR to simulate "natural" binaural conditions. The results indicated that near-normal SRM was achieved by listening through the VGHA in both environments. In the acoustically dry condition, SRM was similar to the measured signal-to-noise ratio (SNR) gain from the microphone array. However, in the reverberant condition, subjects showed significantly greater SRM than predicted from the measured SNR gain from the array. This is consistent with the measured improvement in SNR for the early part of the room impulse response for the target but not for the spatially separated maskers. This indicates that in some reverberant conditions the microphone array provides substantial source selection benefits.

4aPPb8. Head tracking and source localization in reverberant cocktail party scenarios. Anthony Parks and Jonas Braasch (Program in Architectural Acoust., Rensselaer Polytechnic Inst., 126 2nd St. Apt. 12, Troy, NY 12180, abstractpoetry@gmail.com)

From experience and observation, it is known that listeners actively explore their auditory environment through a variety of head movement strategies. Beyond the resolution of front-back confusions, little is known about the mechanisms by which head movement enables listeners perform a broad range of auditory scene analysis tasks. In this experiment, an attempt was made to look at one of these tasks: how well listeners can track a source by head movement. A three-talker paradigm was utilized in a headphone-based head-tracked virtual environment spatialized with HRTFs. The task involves a target that moves from trial to trial with two stationary interferers (male target, two female interferers, all speaking phonetically balanced sentences). The listener is prompted to move her head such that she believes she is facing the target. The two independent conditions are three levels of reverberation (anechoic, strong early reflections, strong late reflections) and seven target azimuth angles (from +45 to -45 degrees in steps of 15 degrees), while the measured responses include facing accuracy as a percentage and number of times the playback button is hit before each trial is completed (to indicate task difficulty). Results are discussed within the context of both room acoustics and perception.

4aPPb9. Differences in masked localization of speech and noise. Inseok Heo (Elec. and Comput. Eng., Univ. of Wisconsin - Madison, Madison, WI), Lynn Gilbertson, An-Chieh Chang, Jacob Stamas, and Robert Lutfi (Dept. of Commun. Sci. and Disord., Univ. of Wisconsin - Madison, Madison, WI 53705, lrjohnson1@wisc.edu)

Dichotic masking studies using noise are commonly referenced in regard to their implications for "cocktail party listening" wherein target and maskers are speech. In the present study masker decision weights (MDWs) are reported suggesting that speech and noise are processed differently in dichotic masking. The stimuli were words or Gaussian-noise bursts played in sequence as masker-target-masker triads. The apparent location of words (noise bursts) from left to right was varied independently and at random on each presentation using KEMAR HTRFs. In the two-interval, forced-choice procedure listeners were instructed to identify whether the second-interval target was to the left or right of the first. For wide spatial separations between target and masker noise-MDWs were typically negative, indicating that target location was judged relative to the masker. For small spatial separations between target and masker noise-MDWs were typically positive, suggesting that target location was more often confused with the masker. For both spatial separations, however, word-MDWs were close to zero, implying that the masker served to distract attention from the target without itself being given significant weight. The results are consistent with an interpretation in which spectral dissimilarities among words generally serve to reduce confusions and relative comparisons among words.

4aPPb10. Sound-localization performance with the hearing protectors. Veronique Zimpfer (Acoust. and Protection of Soldier, ISL, Inst. of Saint Louis, BP 70034, Saint Louis 68301, France, veronique.zimpfer@isl.eu) and David Sarafian (Unite Percept., IRBA, Institut de Recherche Biomédicale des Armées, Brétigny sur Orge, France)

Hearing-protection system that provide level-dependent sound attenuation can protect the ear against potentially damaging sounds (such as loud impulsive noises), while at the same time allowing the perception of moderate-level signals (such as speech). Such systems come in two forms: passive (nonlinear-attenuation earplugs) and active (talk-through system). This study sought to quantify the effect of these systems on spatial hearing. To this aim, sound-localization performance was measured in twenty subjects, with and without ear protectors on. Five protectors (two passives and three actives) were tested. The results showed significant increases in the proportions of errors during the use of one of the systems tested. To clarify the origin of this effect, "protected head-related transfer functions" (PHRTFs), i.e., HRTFs obtained with the ear-protectors on, were measured in the horizontal plane for each of the systems tested. The comparisons of these measures between PHRTFs with HRTFs were found to be in agreement with the subjective tests.

4aPPb11. Effects of targeted pinna occlusion on pinna/spectral cues to localization in the median plane. Alan Musicant (Psychology, Middle Tennessee St. Univ., Box X063, Murfreesboro, TN 37132, Alan.Musicant@mtsu.edu) and Robert Baudo (Dept. of Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Auditory localization accuracy (in humans) in the median sagittal plane has been attributed, by some authors, to an effect of "spectral notches" that occurs in the frequency region of 4–8 kHz. Another possibility for decrements in vertical plane localization accuracy has been overlooked. Roffler and Butler (1967) and Hebrank and Wright (1974) both demonstrated that removal or absence of sound frequencies above about 8 to 10 kHz led to decrements in vertical plane localization accuracy. They did this by using carefully selected types of band pass or band limited noise. A reduction in accuracy of auditory vertical localization by occluding all or part of the pinna has been known for many years [Gardner and Gardner (1973)]. We have previously reported results that demonstrate disruption in accuracy with various partial pinna occlusions [ARO (2012)] that differs from results reported by Gardner and Gardner. We now have data that seems to indicate that the reduction in localization accuracy occurs, in part, because of disturbances in high frequency regions (above about 8 to 10 kHz) and that disruptions in "spectral notches" (4–8 kHz) has little to no effect upon vertical plane localization.

4aPPb12. Perceived elevation cued by images rotating in horizontal planes. Tianshu Qu, Haoze Sun, Ning Wang, Xihong Wu (Key Lab. of Machine Percept. (Ministry of Education), Center for Information Science, Peking Univ., Beijing 100871, China, qutianshu@gmail.com), and William M. Hartmann (Dept. of Phys. and Astronomy, Michigan State Univ., East Lansing, MI)

The sense of elevation perceived by human listeners is normally attributed to high-frequency spectral structure above 4 kHz, caused by anatomical filtering. Our research began with the conjecture that elevation information might be available below 4 kHz when it is linked to a quasi-continuous set of azimuthal cues through measured (KEMAR) head related transfer functions. In an elevation discrimination test, listeners heard two successive azimuthal rotations of 90 degrees, each in a different horizontal plane. The elevations of the horizontal planes differed by as little as 10 degrees and as much as 110 degrees. Listeners reported which rotation had the higher elevation. Results of the rotation experiments were compared with the results from experiments with fixed azimuths, similar to those of Algazi *et al*. [J. Acoust. Soc. Am. **109**, 1110–1122 (2001)]. A rotation from 0 to 90 degrees led to negligible improvement compared to a fixed azimuth of 45 degrees. By contrast, a rotation from 45 to 135 degrees appeared to be particularly advantageous. [Work supported by the National Natural Science Foundation of China Grant Nos. 61175043 and the AFOSR.]

4aPPb13. On the ecological interpretation of limits of interaural time differences sensitivity. William M. Hartmann, Brad Rakerd, and Eric J. Macaulay (Phys.-Astronomy, Michigan State Univ., East Lansing, MI 48824, hartman2@msu.edu)

Human listeners, and other animals too, use interaural time differences (ITD) to localize pure tones, but this ability abruptly diminishes as the frequency of a pure tone increases. The diminished sensitivity appears to serve a useful function. It prevents the confusion that would otherwise arise from the large interaural phase differences that occur at high frequency as sound waves diffract around the head. Possibly this benefit offers an ecological explanation for the diminished sensitivity of the nervous system. However, comparison of the frequency dependence of ITD sensitivity, as measured in headphone experiments, and the frequency dependence of the physical phase shifts, as measured in an anechoic room, reveals a bad match between these two functions. The decrease in neural sensitivity to ITD is seen to be far too rapid, casting doubt on this form of ecological reasoning. If one wants to maintain an ecological context, it is more plausible to argue that our binaural architecture, with its neurophysiological limitations, evolved when our head diameters were smaller by as much as 50%. [Work supported by the AFOSR, 11NL002.]

4aPPb14. Threshold interaural time differences and the centroid model of sound localization. William M. Hartmann (Phys. and Astronomy, Michigan State Univ., East Lansing, MI 48824, hartman2@msu.edu) and Andrew Brughera (Hearing Res. Ctr., Boston Univ., Boston, MA)

The centroid display model of sound lateralization hypothesizes an array of brain-stem cells with wide ranges of best frequencies (f) and best interaural time delays (ITD, τ). The cells are distributed according to function $p(f,\tau)$, and images are lateralized according to the centroid of an excitation pattern on this array, the rate-ITD function. The centroid display was tested by calculations using model cells for the medial superior olive, as the origin of the rate-ITD function. The cells had excitatory inputs, membrane potential increments, and time constants established by physiological measurements. Cells were driven with realistic frequency-dependent synchrony. The predictions were compared to the measured frequency dependence of ITD thresholds for human listeners. It was found that for high frequencies, 750 Hz and greater, the model could successfully account for the thresholds by making appropriate adjustments to $p(f,\tau)$. For lower frequencies, the model greatly underestimated thresholds for any reasonable $p(f,\tau)$ because integration over the wide range of τ reduced the variability in the rate-ITD function to unreasonable values. It is concluded that the centroid model fails to account for human thresholds at low frequency. [Work supported by the AFOSR and NIDCD.]

4aPPb15. Sound source localization: Bandwidth and envelope. William Yost and Xuan Zhong (Speech and Hearing Sci., ASU, P.O. Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Human listeners were asked to locate six sound sources separated by 15o in the right quarter field. Sound sources were located in a sound-deadened room (reverberation time <90 ms) at the height of the listener's pinna 1.67 meters from the listener. In experiment 1, eight, 200-ms pure tones covering the frequency range from 250 to 7011 Hz were presented. In experiment 2, 200-ms noise bursts with different bandwidths (1/6, 1/3/, 1, and 2 octaves) at three center frequencies (250, 2000, and 4000 Hz) were presented. In experiment 3, 200-ms, 4000-Hz tones were presented with transposed envelopes with rates of 50, 100, 150, and 250 Hz. Several indicators of sound source localization performance were measured including root-mean-square (rms) error in degrees. RMS error decreased with increasing bandwidth from approximately 20 degrees for pure tones to approximately 6 degrees for 2-octave wide noises. RMS error depended on center frequency much more for narrow bandwidths than for broader bandwidths. RMS error decreased slightly from 50-Hz rate of modulation to 250-Hz rate of modulation. The data suggest that stimulus bandwidth is the primary variable effecting sound source localization performance in the free-field. [Research supported by an AFOSR Grant.]

4aPPb16. The Haas Effect—Lateral extent and perceptual weighting of localization cues. M. Torben Pastore and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., 4 Irving Place, Troy, NY 12180, m.torben.pastore@gmail.com)

The Haas effect is a well-known manifestation of the precedence effect. Originally, Haas measured the echo threshold as a function of the primary auditory event and its single reflection being equally loud. What is not well known is the lateral position of the lead/lag pair as a function of inter-stimulus interval and level difference between lead and lag. How robust the Haas effect is in the localization dominance region was investigated, adjusting the level difference between lead and lag, and using 200 ms band-passed noise presented dichotically over headphones. In addition, the onset and offset cues were removed for half the trials and left intact for the other half to investigate the roles of onset and offset cues versus ongoing cues. Lateral displacement of the auditory event was recorded with an acoustic pointer. Analysis of these results help to reveal the perceptual weighting of localization cues in the Haas effect.

4aPPb17. An original paradigm to investigate pure informational masking using complex tones. Axelle Calcus (Fonds de la Recherche Scientifique, F.R.S.-FNRS, 50, av. F. Roosevelt, Brussels 1050, Belgium, acalcus@ulb.ac.be), Trevor Agus (Institut d'Etude de la Cognition, D.C. Ecole Normale Supérieure, Paris, France), Cécile Colin (UNESCOG - CRCN, Université Libre de Bruxelles, Brussels, Belgium), Régine Kolinsky (Fonds de la Recherche Scientifique, F.R.S.-FNRS, Brussels, Belgium), and Paul Deltenre (Laboratoire de Neurophysiologie Sensorielle et Cognitive, Hôpital Brugmann, Brussels, Belgium)

Most usual speech masking situations induce both energetic and informational masking. Energetic masking (E.M.) arises because both signal and maskers contain energy in the same critical bands. Informational masking (I.M.) prevents the listeners from disentangling acoustical streams even when they are well separated in frequency, and is thought to reflect central mechanisms. In order to quantify I.M. without E.M. contamination in complex auditory situations, target and maskers can by presented dichotically. However, this manipulation provides the listeners with important lateralization cues, which dramatically reduces I.M. Therefore, the current study aimed at restoring a fair amount of I.M. using complex tones in a new dichotic paradigm. Regularly repeating signals and random-frequency multitone maskers were presented dichotically, but switched from one ear to the other within a 10s sequence. Switches could either appear at a slow or rapid rate. We compared listeners' detection performance in these switching situations to that elicited in traditional diotic and dichotic situations. Results showed that the amount of I.M. induced when signal and maskers were rapidly switching throughout a sequence was significantly higher than in classical dichotic situations, and appeared to be comparable to the diotic listening situation. Therefore, this paradigm provides an original tool to evaluate auditory perception in situations of pure I.M. using complex tones.

4aPPb18. Release from masking through spatial separation in distance in hearing impaired listeners. Adam Westermann and Jörg M. Buchholz (National Acoust. Labs., Australian Hearing, 126 Greville St., Chatswood, NSW 2067, Australia, adam.westermann@nal.gov.au)

It is widely accepted that speech intelligibility improves as a speech signal and interfering masker are separated spatially in azimuth. In a previous study [Westermann *et al*. IHCON (2012)] a similarly strong improvement was found for normal hearing (NH) listeners when target and masker are separated in distance. In this study speech reception thresholds (SRTs) were measured for 16 hearing impaired (HI) listeners using the Listening in Spatialized Noise-Sentences Test (LiSN-S) and the Coordinate Response Measure (CRM). Acoustic scenarios were auralized via headphones using binaural room impulse responses recorded in an auditorium. In the first scenario, the target was presented at a distance of 0.5 m from the center of the listener's head and the interferer at a distance of 0.5 m or 10 m. In a second setup, the interferer's location was fixed and the target's location was varied. HI listeners showed a substantial release from masking as target and interferer were separated in distance. This effect was consistent for both LiSN-S and CRM, but less pronounced than for NH listeners. This study suggests that distance related cues play a significant role when listening in complex environments and are also to some extent available to HI listeners.

4aPPb19. Auditory streaming in cocktail parties and the extent of binaural benefit. Esther Schoenmaker and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Carl von Ossietzkystrasse 9-11, Oldenburg D-26129, Germany, esther.schoenmaker@uni-oldenburg.de)

Studies that investigate the advantage of spatial separation of speakers in a cocktail party generally focus on one of two processing strategies. The first assumes a top-down mechanism in which the listener focuses attention on the known location of a target speaker. Glimpses of target speech are collected and combined to form an auditory stream. The second strategy makes use of interaural differences in perceptual input and exploits these in order to suppress interfering sounds. Equalization-Cancelation (EC) models typically follow this approach. In order to investigate the contributions of both mechanisms, a headphone experiment was conducted that explores auditory streaming based on binaural cues. Sequences of logatomes spoken by one target and two interfering speakers were presented. In this experiment a new type of stimuli was introduced in which the possibility to use binaural masking release cues was eliminated for each time-frequency interval (glimpse) while the localization cues of the dominating source were preserved. Thus, listeners could attend to spatially separated glimpses, but no EC processing was possible. The effect of the availability of masking release cues on successful streaming will be analyzed and discussed.

4aPPb20. Exploring auditory gist: Comprehension of two dichotic, simultaneously presented stories. Nandini Iyer, Eric R. Thompson, Brian D. Simpson (Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Area B, Wright Patterson Air Force Base, OH 45433, Nandini.Iyer@wpafb.af.mil), Douglas S. Brungart, and Van Summers (Walter Reed National Military Med. Ctr., Bethesda, MD)

Cherry (1953) showed that when listeners were asked to selectively attend to one ear in a dichotic listening task, they were able to identify gross attributes of the signal in the unattended ear, suggesting that listeners may be able to capture the "gist" of an auditory stream even when they are asked to ignore it. This experiment explored the extraction of auditory "gist" by investigating the amount and nature of the semantic information stored in memory for later recall. In the experiment, listeners heard two dichotically presented stories; they were directed to: (1) listen to one of the two stories and answer yes-no questions about that story (Directed condition), (2) not directed (Undirected condition) and answer questions about one or both stories, and (3) listen to one of the stories and answer questions about the unattended story (Misdirected condition). Results suggest that listeners can recall the main ideas of both stories in the undirected attention condition significantly better than chance, but that their performance falls substantially below the level achieved in the directed attention condition. These findings are consistent with studies of visual gist processing, suggesting that global features, rather than details, are perceived even before attention is focused on the auditory streams.

4aPPb21. Factors influencing target detectability in realistic listening scenarios. Tobias Weller, Virginia Best, and Jörg M. Buchholz (National Acoust. Labs., 126 Greville St., Chatswood, NSW 2067, Australia, tobias.weller@nal.gov.au)

In psychoacoustics, there is an increasing demand for more realistic testing environments that better capture the real-world abilities of listeners and their hearing devices. However, there are significant challenges involved in controlling the detectability of relevant target signals in realistic environments. We conducted an extensive detection study in a simulated real-world environment to understand some of the important dimensions influencing detection. A multi-talker cafeteria scene was generated using Room simulation software and played back by means of a 3-D loudspeaker array. Detection thresholds for the target word "two" were measured adaptively for eight different target directions in the horizontal plane. Performance was then measured for fixed signal-to-noise ratios around these thresholds to obtain a psychometric function for each direction. To examine the effect of target-location uncertainty, psychometric functions were also measured with randomized target directions. Detection thresholds depended on the target direction, consistent with changes in signal-to-noise ratio caused by the head shadow. Target-location uncertainty increased thresholds globally by a small amount. These findings provide a framework for controlling the detectability of target sounds in future experiments aimed at measuring localization, identification, awareness, etc. in realistic listening environments.

4aPPb22. Spectral integration of interaural time differences in auditory localization. Nicolas Le Goff (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Lyngby 2800, Denmark, nlg@elektro.dtu.dk), Jörg M. Buchholz (National Acoust. Labs., Chatswood, NSW, Australia), and Torsten Dau (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Lyngby, Denmark)

This study investigates how the auditory system integrates spatial information across frequency. In experiment 1, discrimination thresholds for interaural time differences (ITDs) were measured as a function of both reference ITD and center frequency (CF) of noises with bandwidth of one ERB. In addition, discrimination thresholds were also measured as a function of CF for different values of interaural coherence (IC) typical of sounds in realistic acoustic environments. For both high ICs and small reference ITDs, discrimination thresholds were lowest for CFs between 700 and 1000 Hz. For smaller ICs and larger reference ITDs, this dominance region shifted towards lower CFs. A conceptual localization model was developed that used the variance of the ITD thresholds to optimally weight the contribution of the individual frequency bands before spectral integration. In experiment 2, the model was tested by asking listeners to align a broadband noise signal with an ITD that was fixed across frequency onto a broadband noise target with different ITDs in individual 1 ERB-wide subbands. The results were consistent with both the model predictions and the shift of dominance range observed in experiment one.

4aPPb23. Lateralization of noise targets with interaural level differences presented within a noise interferer. Darrin Reed and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Achternstrasse 23, Oldenburg 26122, Germany, darrinreed@hotmail.com)

The interaural level difference (ILD) of a lateralized target source is reduced when the target is presented together with background noise containing no ILD. It is unknown whether listeners simply use this reduced aggregate ILD or are still able to utilize the target ILD in a lateralization task. Behavioral experiments revealed that the temporal asynchrony between the onsets/offsets of the target and the background noise resulted in the population of listeners actually perceiving a larger ILD than the target ILD. For synchronous onsets/offsets, however, the perceived ILD depended on the coherence of the background noise. With coherent background noise, the population of subjects perceived a reduced ILD near the aggregate ILD. In contrast, the population of subjects made a reasonable estimate of the target ILD when the background noise was diffuse. Implementation of an Equalization Cancellation model and taking the compensatory level equalization that yields the lowest output as an estimate for the ILD results in the reduced ILD of the aggregate stimulus being reported regardless of the background noise coherence. However, application of an appropriate normalization factor to the model's output results in a dependence on background noise coherence for ILD lateralization as seen in the behavioral experiments.

4aPPb24. Shifts in the judgment of distance to a sound source in the presence of a sonic crystal. Ignacio Spiousas, Pablo E. Etchemendy, Esteban Calcagno, and Manuel C. Eguia (Laboratorio de Acústica y Percepción Sonora (LAPSo), Universidad Nacional de Quilmes (UNQ), Roque Sáenz Peña 352, Bernal, Buenos Aires B1876BXD, Argentina, ispiousas@unq.edu.ar)

The ability of subjects to estimate the distance to a sound source in a room relies on the integration of a number of different cues: sound intensity level, direct-to-reverberant energy ratio, spectral content and binaural cues, among others. This work examines how the perception of auditory distance is modified for a particular sound field: the transmitted field of an acoustic source through a sonic crystal slab in a semi-reverberant room. A series of experiments were performed comparing the egocentric distance to a sound source passing and not passing through the sonic crystal, using an acoustical virtual environment whereby some of the auditory distance cues could be manipulated. The results obtained show that the presence of the sonic crystal introduces significant shifts on the auditory distance perception. These shifts are correlated with the spatial and spectral variation of the acoustical properties of the sonic crystal. Also, it was possible to determine the relative influence of the manipulated cues (intensity, binaural, and reverberation).

4aPPb25. Validating a binaural head for use in jury testing. Jeremy E. Charbonneau, Colin Novak, and Helen Ule (Mech. Eng., Univ. of Windsor, 1560 Dougall Ave., Windsor, ON N8X1S1, Canada, charbo6@uwindsor.ca)

A test procedure for use in loudness perception tests must be created to completely describes a phenomenon while at the same time minimizing jury listening fatigue. One contributor to this fatigue is the amount of time necessary for the test subject to experience all the required signals. Head and torso simulators have been used for years as a means to reliably quantify the acoustic performance of a product while avoiding the influence of listener bias and fatigue. This procedure not only controls the test parameters but also removes any human error that may occur. The purpose of this investigation is to qualify a head and torso simulator for use in loudness investigations. The objective of this experiment is to correlate the results from using this equipment to human subject results for high resolution experiments on directionality of loudness. Comparisons are also made from the directionality results at various listening angles including a listener facing a sound source.

4aPPb26. Development of an underwater binaural head model. Christopher A. Bailey and Neil L. Aaronson (Natural Sci. and Mathematics, The Richard Stockton College of NJ, 54 Mark Dr., High Bridge, NJ 08829, challenbailey@gmail.com)

The human brain has difficulty localizing sound in aquatic environments, where the acoustical properties of water greatly impede the mechanisms by which the brain interprets binaural signals. In this experiment, a hollow steel sphere with antipodal hydrophones is exposed to noise bursts in an underwater environment. The sphere can be filled with various materials to alter the apparatus' rigid qualities. Interaural time and level differences (ITDs and ILDs, respectively) are calculated from recordings of these noises and compared to a theoretical model for sound propagation around a non-rigid head in an effort to better characterize binaural hearing in underwater surroundings. While the theoretical behavior of sound diffracting around a rigid head has been well documented, the similar problem involving a flexible head has largely been left to experimental methods due to the computational complexity of the task. In the current study, a new computational model, capable of predicting ITDs and ILDs for sounds encountering a non-rigid sphere in diverse environments, is used. Both the model and the experiment will be introduced in this presentation. The findings have significant implications for the future development of reliable methods for improving sound localization in underwater environments, for instance for recreational divers.

4aPPb27. Can monaural temporal masking asymmetry explain the transient and/or ongoing precedence effect? Richard L. Freyman, Charlotte Morse-Fortier, Amanda M. Griffin (Dept. of Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003, rlf@comdis.umass.edu), and Patrick M. Zurek (Sensimetrics Corp., Malden, MA)

Investigations of the precedence effect show that interaural differences within the first of two pairs of brief binaural sounds contribute more to lateralization than those within the second pair. The present study asked whether this phenomenon could be explained by asymmetries in monaural masking, and compared the results to a second experiment investigating the same question with the "ongoing" precedence effect. Leading and lagging stimuli were binaural pairs of 1-ms frozen noise bursts, with a 2-ms delay between pairs, and had ITDs of +500 and −500 ms, respectively. Detection thresholds for lead or lag in the presence of the other were assessed monaurally using a 4AFC task. Results showed that threshold for the lead was 12 dB better than that for the lag, on average. Ongoing stimuli were created by repeating the transient stimuli 63 times, with a new noise token on each repeat. Although the leading burst in each binaural pair contributes more to lateralization than the lagging burst, monaural thresholds for the leading bursts were not better than those for the lagging bursts. The results suggest that the transient precedence effect, but not the ongoing precedence effect, might be explained by temporal masking asymmetry. [Work supported by NIH DC01625].

4aPPb28. Effects of the stimulus spectrum on temporal weighting of binaural differences. G. Christopher Stecker (Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, cstecker@uw.edu)

The influence, or "perceptual weight" of binaural information typically varies over the duration of a brief sound, as characterized by the temporal weighting function (TWF). Here, TWFs were measured for binaural lateralization of Gabor click trains (GCT) varying in carrier frequency from 1 to 8 kHz, and of broadband noise-burst trains (NBT) with repeated ("frozen") or newly sampled ("fresh") noise across bursts. Interclick intervals (ICI) ranged from 2 to 10 ms. On each of many trials, human listeners judged the lateral position of a singly presented GCT or NBT, and indicated the position on a touch-sensitive display. Lateral positions varied with the overall interaural time (ITD, ranging +/− 500 μs) and level (ILD, ranging +/−5 dB) differences applied to each stimulus. Additional random variation in ITD (+/−100 μs) and ILD (+/−2dB) was applied independently to each click within a train. TWFs were calculated by multiple linear regression of normalized position judgments onto the individual click ITD and ILD values, and indicated large ICI-dependent weights on the initial click ("onset dominance"), elevated weights near offset, and lower weights for interior clicks. Flatter TWFs with reduced onset/offset weights were observed for "fresh" NBT stimuli than for GCT or "frozen" NBT stimuli. The results corroborate previous reports of temporal asymmetries in the binaural processing of periodic stimuli across frequency. [Work supported by R01 DC011548.]

4aPPb29. Frequency domain binaural model with front-back discrimination capability using artificial neural network. Tsuyoshi Usagawa, Takuro Tomita, and Yoshifumi Chisaki (Dept. of Comput. Sci. and Elec. Eng., Kumamoto Univ., 2-39-1 Kurokami, Chuo-Dist., Kumamoto 860-8555, Japan, tuie@cs.kumamoto-u.ac.jp)

Conventional binaural model or binaural hearing assistance systems have a well known ambiguity in front-back discrimination, which is called as "front-back confusion" or "cone of confusion" in psychoacoustics. It is known that spectral cue of sound provides keys to solve this confusion in binaural listening condition and the peaks and notches of spectral components play main role to estimate the vertical angle in sagittal coordinate. In this paper, a frequency domain binaural model with front-back discrimination method using artificial neural model is proposed and examined for various

HRTF catalogs. The performance of the model is examined for simulated conditions using various HRTF catalogs and the results show similar discrimination capability even if the learning process should be done for each catalogs. The experimental results using a dummy head is also provided.

4aPPb30. Simulation of the head-related transfer functions using cloud computing. Tomi Huttunen, Kimmo Tuppurainen, Antti Vanne (Kuava Ltd., Microkatu 1, Kuopio FI-70211, Finland, tomi.huttunen@kuava.fi), Pasi Ylä-Oijala, Seppo Järvenpää (Dept. of Radio Sci. and Eng., Aalto Univ., Helsinki, Finland), Asta Kärkkäinen, and Leo Kärkkäinen (Nokia Res. Ctr., Helsinki, Finland)

Due to the complexity of measurements for obtaining individual head-related transfer functions (HRTFs), numerical simulations offer an attractive alternative for generating large HRTF data bases. In this study, HRTFs are simulated using a fast multipole boundary element method (BEM). The BEM is well suited for the HRTF simulations. Namely, only the surface of the model geometry is discretized which simplifies the pre-processing compared to other full-wave simulation methods (such as finite element and finite difference methods). The BEM is formulated in frequency domain and the model is solved separately for each frequency. Since a large number of frequencies is needed in wide-band HRTF simulations, the BEM simulation greatly benefits from distributed (or parallel) computing. That is, a single computing unit takes care of a single frequency. In this study, a distributed BEM using cloud computing is introduced. Simulations are computed in a public cloud (Amazon EC2) using a realistic head and torso geometry (3D laser scanned geometry of Bruel & Kjaer HATS 4128 mannequin). The frequency range of the simulations is from 20 to 20000 Hz. The feasibility of cloud computing for simulating HRTFs is examined and first analysis results for the simulated HRTFs are shown.

4aPPb31. Effect of distant-variant/invariant head-related transfer functions on perception of a proximal sound source in virtual auditory space. Makoto Otani, Fuminari Hirata, Kazunori Itoh, Masami Hashimoto, and Mizue Kayama (Shinshu Univ., 4-17-1 Wakasato, Nagano 380-8553, Japan, otani@cs.shinshu-u.ac.jp)

A virtual auditory space can be presented to a listener based on binaural synthesis using head-related transfer functions (HRTFs) that are obtainable by measurements or numerical simulations. Due to hardware complexity, HRTF measurements are typically made for a fixed source distance though they are used in binaural synthesis for variable source distances. However, it is known that HRTFs depend on source distance especially for proximal sources for distance less than 1 m. So it is possible that binaural synthesis with HRTFs for a fixed source distance may result in degradations for proximal sound image perception. In this paper, experiments were performed to examine how the use of distant-variant or -invariant HRTFs affect the perception of a proximal sound source in a virtual auditory space in which the listener's motion is compensated by head tracking. HRTFs for source distances up to 1 m, in 5 cm steps, are numerically simulated using the boundary element method. Results show the difference between presented and perceived source distances being significantly smaller when distance-variant HRTFs were used. This indicates that the use of HRTFs corresponding to actual sound source position leads to accurate perception of a proximal source.

4aPPb32. The role of spatial detail in sound-source localization: Impact on head-related transfer function modeling and personalization. Griffin D. Romigh (711th Human Performance Wing, Air Force Res. Labs., 4064 chalfonte, beavercreek, OH 45440, griffin.romigh@wpafb.af.mil), Douglas S. Brungart (Audiol. and Speech Ctr., Walter Reed National Military Med. Ctr., Bethesda, MD), Richard M. Stern (Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA), and Brian D. Simpson (711th Human Performance Wing, Air Force Res. Labs., Dayton, OH)

Current techniques designed to personalize generic head-related transfer functions (HRTFs) have some capacity to quickly customize spatial auditory displays, but these techniques generally fall short of the level of realism and performance provided by fully individualized HRTF measurements. This residual performance deficit reflects inaccuracies due to vast amounts of spatial and spectral variation that occurs across the measured HRTFs of individual listeners. Some of this variation encodes perceptually-important directional information, but a substantial proportion does not. Kulkarni and Colburn (1998) showed that perceptually irrelevant spectral variation could be eliminated by smoothing the HRTF magnitude with a truncated Fourier-series expansion. The present study investigates a related method for smoothing the spatial variation contained in the HRTF by utilizing a truncated spherical harmonic expansion. The impact of spatial smoothing was evaluated by comparing localization performance with individualized HRTFs which were fully represented or had various degrees of spatial smoothing. Results indicate that a highly-smoothed fourth-order spherical harmonic representation can produce localization accuracy comparable to that of a full individualized HRTF. Analysis of the resulting simplified HRTF representations also uncovered a number of interesting relationships across different individuals which may provide new insights for the development of future HRTF personalization and estimation techniques.

4aPPb33. The relation between the information delivered by head-related transfer function and human spatial hearing. Vladimir Tourbabin and Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva, 84105, Israel, tourbabv@ee.bgu.ac.il)

The human auditory system is capable of performing various tasks related to spatial hearing including sound localization and source segregation. The performance of these tasks depends on many factors, including the complexity of the sound field; the way in which the information from the sound field is transferred to the ears; and the ability of the binaural hearing system to extract the required information. This study focuses on the role of the transfer system represented by the HRTFs that relate the sound field to the signals at the ears. Previously, a measure for the information delivered by the HRTFs was proposed, and the role of HRTFs in human sound localization in the horizontal and median planes was investigated. In the current study, the role of HRTFs in human sound localization is further investigated by analyzing localization in the entire three-dimensional space. Then, the proposed measure is used to investigate the role of HRTFs in source segregation as part of a spatial-release from masking task. The results show that the information delivered by the HRTFs can account for the increase in intelligibility as a function of the spatial separation between the desired source and the masker reported in the literature. However, the decrease in intelligibility with increasing number of spatially separated maskers seems to be related to the binaural hearing system, and cannot be explained by the information in the HRTFs.

4aPPb34. Simplification of head-related impulse response in early reflection simulation. Liang Zhang (Acoust. Lab., Phys. Dept., School of Sci., South China Univ. of Technol., Guangzhou, China) and Xiao-li Zhong (Acoust. Lab., Phys. Dept., & State Key Lab. of Subtropical Bldg. Sci., South China Univ. of Technol., Bldg. No.18, Wu Shan Rd. No. 381, Guangzhou, Guangdong 510641, China, xlzhong@scut.edu.cn)

In virtual auditory environment, early reflections are usually simulated by the image-source method, and binaural signals are synthesized by convolving the input stimulus with corresponding head-related impulse responses (HRIRs). Considering the limited resolution of the human hearing, this work investigates the minimal length of HRIRs needed in early reflection simulation via a simple model consisting of a single direct sound and a reflection. The direct sound is synthesized using 512-point HRIR and fixed at the position directly in front of the subject, while the reflection is synthesized by HRIRs at various directions with four time-domain lengths (512, 256, 128, and 64 points, at a sampling frequency of 44.1 kHz) as well as five time delays relative to the direct sound (from 10 to 50 ms at intervals of 10 ms). A three-interval, two-alternative forced-choice paradigm is employed in this work. Results indicate that for most spatial directions the HRIR with a length of 64-point is perceptually adequate in early reflection simulation. [Work supported by State Key Laboratory of Subtropical Building Science, South China University of Technology, Grant No. 2013KB23.]

4aPPb35. An efficient finite-impulse-response filter model of head-related impulse response. Junfeng Li, Jian Zhang (Inst. of Acoust., Chinese Acad. of Sci., No. 21, Beisihuan Xilu, Beijing 100190, China, junfeng.li.1979@gmail.com), Shuichi Sakamoto, Yoiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Yonghong Yan (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Head-related impulse responses (HRIRs) play a crucial role in binaural 3-D audio rendering. The HRIRs with a couple of hundred-sample lengths result in the high computation cost for the real-time 3-D audio applications

especially when multiple sound sources are rendered simultaneously. To overcome this problem, various modeling approaches have been reported to reduce the number of parameters of HRIRs without sacrificing the quality of rendered sounds. In this research, an efficient finite-impulse-response (FIR) model is first studied, which is essentially based on the concept of the minimum-phase modeling technique. In this method, the measured HRIRs are represented by the interaural time delay (ITD) and the magnitude spectra that are approximated by two FIR filters. To investigate the accuracy dependence of this modeling approach on the order of FIR filter, two psychoacoustic experiments on sound localization and sound quality were conducted by comparing the synthesized stimuli with the measured HRIRs and those with the FIR models of different orders. Experimental results indicated that the measured hundred-sample-length HRIRs can be sufficiently modeled by the low-order (a dozen of coefficients) FIR model from the perceptual point of view. The derived low-order FIR model can be further applied to real-time 3-D audio applications.

4aPPb36. Estimation of spectral notch frequencies of the individual head-related transfer function from anthropometry of listener's pinna. Yohji Ishii and Kazuhiro Iida (Faculty of Eng., Chiba Inst. of Technol., 2-17-1 Tsudanuma, Narashino 275-0016, Japan, s0972004QT@it-chiba.ac.jp)

Listener's own head-related transfer functions (HRTFs) are necessary for accurate sound image reproduction. The HRTFs of other listeners often cause the front-back confusion and the errors in elevation perception. It is, however, impractical to measure the HRTFs of any listener for any sound source direction because the measurement requires special apparatus and much time. On the other hand, the estimation of the entire spectrum information of listener's own HRTF still remains as an unsolved difficult issue. One of the authors has shown that the simplified HRTFs, which is recomposed only of the first spectral peak around 4 kHz (P1) and the lowest two spectral notches (N1 and N2) above P1, extracted from the listener's measured HRTFs in the median plane, provide almost the same localization accuracy as the measured HRTFs. While the frequency of P1 is almost constant independent of the sound source elevation and the listener, those of N1 and N2 are highly dependent on both the elevation and the listener. The present study proposes a method, which estimates the frequencies of N1 and N2 in the median plane for the individual listener from the anthropometry of the listener's pinna, and examines the validity of the method.

4aPPb37. Further evidences of the contribution of the ear canal to directional hearing: Design of a compensation filter. Andrea Martelloni (Inst. of Sound and Vib. Res., Univ. of Southampton, Milan, Italy), Davide A. Mauro (Institut TELECOM, TELECOM Paris Tech, CNRS-LTCI, Via Comelico 39/41, Milan 20135, Italy, mauro@di.unimi.it), and Antonio Mancuso (Lab. di Informatica Musicale (LIM), Università degli Studi di Milano, Milan, Italy)

It has been proven, and it is well documented in literature, that the directional response in HRTFs comes largely from the effect of the pinnae. However, few studies have analyzed the contribution given by the remaining part of the external ear, particularly the ear canal. This work investigates the directionally dependent response of the modeled ear canal of a dummy head, assuming that the behavior of the external ear is sufficiently linear to be approximated by an LTI system. In order to extract the ear canal's transfer function, two critical microphone placements (at the eardrum and at the beginning of the cavum conchae) have been used. The system has been evaluated in several positions, along the azimuth plane and at different degrees of elevation. The results point out a non-negligible directional dependence that is well within the normal hearing range; based on these findings, physical models of the ear canal have been analyzed and evaluated. We have also considered the practical application to binaural listening, and the coloration originated by the superimposition of the contribution of two ear canals (the listener's and the dummy head's). A compensating FIR filter with arbitrary frequency response is discussed as a possible fix.

THURSDAY MORNING, 6 JUNE 2013 — 512CG, 9:00 A.M. TO 12:00 NOON

Session 4aSA

Structural Acoustics and Vibration: Applications in Structural Acoustics and Vibration II

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Contributed Papers

9:00

4aSA1. Analysis of the acoustic scattering from a submerged bilaminar plate. Sabih I. Hayek (Eng. Sci. and Mech., Penn State Univ., 953 McCormick Ave., State College, PA 16801-6530, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NAVSEA, Div. Newport, Newport, RI)

The acoustic scattering from a submerged finite bilaminar rectangular elastic plate is modeled using the exact theory of three-dimensional elasticity. The two layers of the composite bilaminar plate have the same lateral dimensions, but have different thicknesses and material properties. The plate is set in an infinite rigid baffle and is coupled to a different acoustic medium on its two surfaces. The plate is insonified by an acoustic plane wave. The farfield backscattered and forward scattered waves are computed for various layer thicknesses and elastic material properties in contact with air or water. First, the scattering from a uniform finite, baffled, steel plate elastic plate was computed. A bilaminar plate was also analyzed with one layer made of steel and the other made of a damped elastomer. The backscattered and forward scattered acoustic farfield spectra versus frequency were computed at the on-axis point receiver due to a normally incident plane wave. The directivity functions for a normally incident plane wave insonifying the steel- or elastomer-side of the plate were computed for a range of frequencies, with either water or air backing. [Work supported by NAVSEA Division Newport under the ASEE Summer Faculty Program.]

9:20

4aSA2. Numerical modeling of the radiation by a submerged fluid-filled cylindrical shell: Observation of the S0, A0, and A waves. Serguei Iakovlev (Dept. of Eng. Mathematics and Internetworking, Dalhousie Univ., 1340 Barrington St., Halifax, NS B3J 1Y9, Canada, serguei.iakovlev@dal.ca), Hugo A. F. A. Santos (Dept. of Civil Eng. and Architecture, Tech. Univ. of Lisbon, Lisbon, Portugal), Benjamin Schulman, and Kyle Williston (Dept. of Eng. Mathematics and Internetworking, Dalhousie Univ., Halifax, NS, Canada)

We consider a submerged fluid-filled cylindrical shell subjected to an external acoustic pulse and analyze the structure of the field radiated by the shell into the fluids, both external and internal. We first propose a computationally efficient semi-analytical model of the interaction based on the Reissner-Mindlin shell theory combining some of the classical methods of mathematical physics with the finite-difference methodology, and then use the model to simulate the interaction. We demonstrate that the model accurately reproduces the wave structure of the radiated fields seen in the experiments for submerged evacuated shells, namely, both the symmetric Lamb waves S0 and the pseudo-Rayleigh waves A0. It is further observed that the internal and external wave patterns associated with the A0 waves exhibit the same alternation of the equiphase lines as the one seen in the experiments for a plate loaded by the fluid on both sides, a result that seems to be particularly relevant in the context of very limited number of experimental images of the radiated field for shells loaded by fluid from both inside and outside. Not less interestingly, we also demonstrate that the Scholte-Stoneley, or A, wave is also reproduced by the model, and we offer some insights into the non-observability of this wave for certain types of cylindrical shells reported in earlier experimental studies.

9:40

4aSA3. Modeling of wave propagation in drill strings using acoustic transfer matrix method. Je-Heon Han, Yong-Joe Kim (Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77840, jeep2000@tamu.edu), and Mansour Karkoub (Mech. Eng., Texas A&M Univ. at Qatar, Doha, Qatar)

In order to understand critical vibrations of a drill bit such as stick-slip and bit-bounce and their wave propagation characteristics through a drillstring system, it is critical to model the torsional, longitudinal, and flexural waves. The objective is to model these waves propagating through the drillstring in a computationally efficient way. Here, a modeling method based on an acoustic transfer matrix between two sets of wave variables at the ends of a cylindrical pipe is proposed. For a drillstring system with multiple pipe sections, the total acoustic transfer matrix is calculated by multiplying all individual matrices of which each is obtained for an individual pipe section. Since drillstring systems are typically extremely long, conventional numerical analysis methods such as FEM require a large number of meshes, which makes it difficult to analyze these drillstring systems. On the contrary, the "analytical" acoustic transfer matrix method requires significantly low computational costs. For the validation, experimental and numerical data are obtained from a laboratory measurement and by using a commercial FEM package, ANSYS, respectively. They are compared to the modeling results obtained by using the proposed method. It is shown that the modeling results are well matched with the experimental and numerical results.

10:00

4aSA4. Some research of mapped radiation modes and its application in analyzing the radiation surface of vibrating structures. Haijun Wu, Weikang Jiang, and Siwei Pan (State Key Lab. of Mech. System and Vib., Shanghai Jiao Tong Univ., 800 Dong Chuan Rd., Shanghai 200240, China, haijun.wu.cn@gmail.com)

Acoustic radiation modes divide the velocity patterns into groups with effective and ineffective radiation efficiencies. It suggests a promising way in the computation of sound power and near field acoustic holography. However, obtaining the radiation modes based on eigenvalue analysis suffers from issues of CPU time and memory, especially for large-scale problems. Based on the idea of equivalent source method and boundary integral equation, it is mathematically and numerically proved that the spherical basis functions is a set of linearly independent patterns on an arbitrary surface. They are termed as mapped radiation modes. The analytical sound power of a structure vibrating in its mapped radiation modes is obtained. An efficient and accurate method to compute the sound power based on the mapped radiation modes is proposed. Based on the relationship between the mapped radiation modes of a vibrating structure and its field sound pressure distribution patterns on a sphere, an improved near field acoustic holography is also developed. It does not need the inverse process but requires microphones to locate at the integral points on a sphere. Numerical examples are presented to validate advantages of applying mapped radiation modes in the sound power evaluation and near field acoustic holography.

10:20

4aSA5. Experimental investigation into sound and vibration of a torpedo-shaped structure under axial force excitation. James Leader, Jie Pan (School of Mech. and Chemical Eng., Univ. of Western Australia, 3 Brahea Place, Mt Claremont, Perth, WA 6010, Australia, 20351548@student.uwa.edu.au), Paul Dylejko (Defence Sci. and Technol. Organisation, Perth, Western Australia, Australia), and David Mathews (HMAS Stirling, Defence Sci. and Technol. Organisation, Perth, WA, Australia)

In this study, the sound radiation patterns and vibration characteristics of a torpedo-shaped structure are determined experimentally using a proof mass actuator to allow pure axial excitation of the model. Using this method, the second energy path found in previous designed structures is eliminated. Input power and driving forces are measured using four force transducers and four accelerometers, while the vibration response and mode shapes are measured using an array of accelerometers. The sound pressure and its directivity are captured by a spatially distributed microphone array inside an anechoic chamber. Motivations for this work are to investigate the effect of the complex boundary constraints: a semispherical head and conical tail on the two meter long model when compared to existing analytical solutions for simple geometries, and later the measurement will be performed in an underwater experiment to contrast the effect of fluid loading.

10:40

4aSA6. Results of an implementation of the dual surface method to treat the non-uniqueness in solving acoustic exterior problems using the boundary element method. Ralf Burgschweiger (PG Computational Acoust., Beuth Hochschule für Technik Berlin, Luxemburger Str. 10, Berlin 13353, Germany, burgi@beuth-hochschule.de), Ingo Schäfer (Underwater Acoust. and Marine Geophys. Res. Inst. (FWG), Wehrtechnische Dienststelle für Schiffe und Marinewaffen (WTD71), Kiel, Germany), Adel Mohsen (Eng. Mathematics & Phys. Dept., Eng. Faculty, Cairo Univ., Cairo, Egypt), Rafael Piscoya, Martin Ochmann (PG Computational Acoust., Beuth Hochschule für Technik Berlin, Berlin, Germany), and Bodo Nolte (Underwater Acoust. and Marine Geophys. Res. Inst. (FWG), Wehrtechnische Dienststelle für Schiffe und Marinewaffen (WTD71), Kiel, Germany)

The problem of non-uniqueness (NU) of the solution of exterior acoustic problems when using the boundary element method (BEM) is well known. Methods like the Burton-Miller technique or the CHIEF method are used to solve this challenge at the expense of more complex procedures for handling hypersingular integrals and/or higher computing times due to higher complexity of the algorithm or additional equations. The dual surface method, commonly used for electromagnetic problems, was adapted for acoustic radiation and scattering problems. The basic principles of methods to solve the NU problem are outlined and results for different models and solution procedures are presented, taking into account quality, solution time, and the numerical advantages when using iterative solvers.

11:00

4aSA7. The peculiarities of the non-axisymmetric frequency spectra of finite elastic cylinders. Dmytriy Libov (Theor. and Appl. Mech., Kiev National Taras Shevchenko Univ., 64 Volodymyrska St., Kiev 04214, Ukraine, dmytro.libov@univ.net.ua)

A rigorous solution of three-dimensional boundary-value problem concerning the forced vibrations of a finite, elastic, isotropic cylinder is constructed analytically by means of the superposition method. With this solution, the resonances in non-propagating waves were investigated.

Particularly, a survey of the frequency spectrum for an aluminum cylinder, vibrating with the circumferential order two, reveals the existence of a localized resonance, usually referred to as an end resonance, well below the cutoff frequency of the lowest real dispersion branch of an infinite cylinder. This phenomenon demonstrates the remarkable differences between the axisymmetric and non-axisymmetric end resonances of elastic cylinders. Comparison of the theoretical results with the experiments published elsewhere reveals an excellent agreement.

11:20

4aSA8. Sound generated by a wing with a flap interacting with an eddy. Avshalom Manela (Aerosp. Eng., Technion, Technion City, Haifa 32000, Israel, avshalom@aerodyne.technion.ac.il) and Lixi Huang (Mech. Eng., Univ. of Hong Kong, Hong Kong, Hong Kong)

Acoustic signature of a rigid wing, equipped with a movable downstream flap and interacting with a line vortex, is studied in a two-dimensional low-Mach number flow. The flap is attached to the airfoil via a torsion spring, and the coupled fluid-structure interaction problem is analyzed using thin-airfoil methodology and application of the Brown and Michael equation. It is found that incident vortex passage above the airfoil excites flap motion at the system natural frequency, amplified above all other frequencies contained in the forcing vortex. Far-field radiation is analyzed using Powell-Howe analogy, yielding the leading order dipole-type signature of the system. It is shown that direct flap motion has a negligible effect on total sound radiation. The characteristic acoustic signature of the system is dominated by vortex sound, consisting of relatively strong leading and trailing edge interactions of the airfoil with the incident vortex, together with late-time wake sound resulting from induced flap motion. In comparison with the counterpart rigid (non-flapped) configuration, it is found that the flap may act as sound amplifier or absorber, depending on the value of flap-fluid natural frequency. The study complements existing analyses examining sound radiation in static- and detached-flap configurations.

11:40

4aSA9. Localization and identification of three-dimensional sound source with beamforming based acoustic tomography. Hao Ding, Huancai Lu, Chunxiao Li, Jiangming Jing, Dongting Mei, and Guozhong Chai (Zhe Jiang Univ. of Technol., Chaowang Rd. 18th, HangZhou 310014, China, haodinggo@gmail.com)

Beamforming based commercial planar microphone array could only localize and identify the sound source when the distance between source and array is known. This paper presents a beamforming based acoustic tomography (BBAT) method to locate and identify the source in 3D space, say, the BBAT method can not only locate the source in X-Y plane that is parallel to the array, but also the depth Z of the source. In this method, the sound field is reconstructed on the virtual planes at different distances along depth direction (Z direction). The maximum response of sound field on every virtual reconstruction plane is tracked, where the largest value among those maximum responses appears at Z direction is the depth of the source. The location of source at X and Y directions can then be easily identified based on beamforming principle, which is utilized by commercial planar array. The BBAT method is evaluated theoretically by simulation of monopole source, the experimental evaluation is done as well in anechoic chamber. The results from both simulation and experiment indicate that this method is capable to locate and identify sound source in 3D space. However, it cannot recognize the sound source located in front of the planar array or behind because of the genetic limitation of 2D planar array in identification of source depth.

THURSDAY MORNING, 6 JUNE 2013 — 515ABC, 9:00 A.M. TO 11:40 A.M.

Session 4aSCa

Speech Communication: Auditory Feedback in Speech Production I

Anders Lofqvist, Cochair
Haskins Labs., 300 George St., New Haven, CT 06511

Charles R. Larson, Cochair
Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208

Invited Papers

9:00

4aSCa1. Individual differences in auditory-motor integration revealed by speech fluency manipulations. Torrey M. Loucks (Speech and Hearing Sci., Univ. of Illinois, 901 S. Sixth St., Champaign, IL 61820, tloucks@illinois.edu), Heecheong Chon (Div. of Speech-Language Pathol., Chosun Univ., Gwangju, South Korea), Shelly Kraft (Commun. Sci. and Disord., Wayne State, Detroit, MI), and Nicoline Ambrose (Speech and Hearing Sci., Univ. of Illinois, Champaign, IL)

A role for auditory feedback in maintaining fluency appears less specific than for pitch control, as one example, but delayed auditory feedback (DAF) clearly provides a potent manipulation of fluency. As most speakers are susceptible to DAF, we predicted DAF is particularly suited to identifying individual differences in auditory-motor integration. We conducted a series of studies to probe susceptibility to DAF-induced disfluency in 60 normally fluent speakers during conversation and oral reading. We further contrasted DAF effects on fluency with dual-task effects on fluency. During conversation and reading under DAF (250 ms delay), multivariate cluster classification indicated speakers show high, low, or intermediate susceptibility to disfluency. In contrast, dual-task effects on fluency appeared bimodal with individuals showing high or low susceptibility. DAF susceptibility was not related to dual-task disfluency in 41/60 speakers, but the remaining speakers were disfluent under DAF and dual-task conditions. When the DAF paradigm was extended to adults who stutter, most were classified as highly susceptible. The findings provide compelling evidence that individual differences need to be considered in auditory-motor integration research. Fluency is influenced by both auditory feedback and cognitive factors related to attention, which can inform theories of normal and disordered speech.

9:20

4aSCa2. Experience-dependent learning effects in speech production with spectrally degraded feedback. Elizabeth D. Casserly (Linguistics, Indiana Univ., Memorial Hall Rm. 322, Bloomington, IN 47406, casserly@indiana.edu) and David B. Pisoni (Psychol. & Brain Sci., Indiana Univ., Bloomington, IN)

This study examined the speech of normal-hearing adult participants before and during their use of a portable, real-time vocoder (PRTV). The PRTV continuously transforms environmental acoustics, including speakers' own speech feedback, via a real-time simulation of cochlear implant processing. The impacts of this substantial spectral degradation on speech production were measured in three groups of subjects: group 1 received altered acoustic feedback for one continuous 55 min session; group 2 experienced the feedback transformation for one session of 6 h total; and group 3 wore the PRTV for four consecutive sessions of 4 h each, for a total of 16 h of experience. Speakers in each group were recorded producing 114 isolated English words and 24 sentences both before their feedback manipulation began and at periodic intervals during their experimental session(s). Acoustic-phonetic analyses of the speech produced by subjects in all three groups revealed substantial effects of the spectral feedback degradation in several domains, including fluency/speaking rate, vocal affect, and vowel quality. Speakers were able to adjust and recover quickly in some of these areas, such as affect, while other changes, such as those in vowel quality and speaking rate, remained despite 16 h of experience with the acoustic transformation.

9:40

4aSCa3. Speech production in amplitude-modulated noise. Ewen N. MacDonald (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ctr. for Hearing and Speech Sci., Bldg. 352, Ørsteds Plads, Kgs Lyngby DK-2800, Denmark, emcd@elektro.dtu.dk) and Stefan Raufer (Institut für Hörtechnik und Audiologie, Jade Hochschule Oldenburg, Oldenburg, Germany)

The Lombard effect refers to the phenomenon where talkers automatically increase their level of speech in a noisy environment. While many studies have characterized how the Lombard effect influences different measures of speech production (e.g., F0, spectral tilt, etc.), few have investigated the consequences of temporally fluctuating noise. In the present study, 20 talkers produced speech in a variety of noise conditions, including both steady-state and amplitude-modulated white noise. While listening to noise over headphones, talkers produced randomly generated five word sentences. Similar to previous studies, talkers raised the level of their voice in steady-state noise. While talkers also increased the level of their voice in amplitude-modulated noise, the increase was not as large as that observed in steady-state noise. Importantly, for the 2 and 4 Hz amplitude-modulated noise conditions, talkers altered the timing of their utterances, reducing the energetic overlap with the masker by approximately 2%. However, for the 1 Hz amplitude-modulated condition, talkers increased the overlap by approximately 4%. Overall, the results demonstrate that talkers are sensitive to the temporal aspects of noisy environments and will alter their speech accordingly.

10:00

4aSCa4. Auditory plasticity and sensorimotor learning in speech production. Douglas M. Shiller (École d'orthophonie et d'audiologie, Université de Montréal, P.O. Box 6128, succursale Centre-ville, Montreal, QC H3C 3J7, Canada, douglas.shiller@umontreal.ca), Daniel R. Lametti, and David J. Ostry (Dept. of Psych., McGill Univ., Montreal, QC, Canada)

Numerous studies have shown the speech motor system to be highly flexible and responsive to changes in sensory input, revealing a central role for both auditory and somatosensory feedback in the acquisition and maintenance of speech motor control. Consistent with these studies, models of speech production have highlighted the role of accurate, stable sensory representations that serve, in part, as the goals of speech movements. A separate (and considerable) body of work has demonstrated that auditory-sensory representations of speech sounds are not perfectly stable, but rather exhibit rapid adaptation to changing input conditions in both children and adults. The plasticity of auditory representations has important implications for the control of speech production, both in early speech motor development and in the sensory-based maintenance of speech accuracy that characterizes adult speech motor control. In this talk, I will describe a series of studies that explore the link between sensory and motor plasticity in the speech motor system. The studies combine the paradigm of sensorimotor adaptation (altering auditory feedback during speech production) with measures and manipulations of auditory-perceptual representations of speech sounds. The results reveal not only that auditory speech targets are flexible under conditions of altered auditory feedback, but that changes in sensory representations can have a direct impact on speech motor learning and performance.

10:20–10:40 Break

10:40

4aSCa5. The relationship between vocal pitch feedback error and event-related brain potentials. Jeffery A. Jones, Nichole Scheerer, and Anupreet Tumber (Psych. & Laurier Ctr. for Cognit. Neurosci., Wilfrid Laurier Univ., 75 University Ave. W., Waterloo, ON N2L 3C5, Canada, jjones@wlu.ca)

Understanding the neural processing of auditory feedback during speech is essential to the development of a comprehensive model of speech motor control. Currently, the relationship between the magnitude of errors detected in feedback and the evoked neural responses is unclear. We exposed speakers to sudden changes in vocal pitch that ranged from 0 to 400 cents in magnitude. Vocal responses and auditory event-related potentials (ERPs: P1-N1-P2-N2 components) were measured. Results showed that vocal response magnitudes were relatively consistent when speakers were exposed to small feedback perturbations (<250 cents). Larger perturbations (>300 cents) caused decreased vocal response magnitudes. P1 amplitudes showed a non-specific increase when feedback was perturbed. N1 amplitudes demonstrated more specificity: smaller feedback perturbations evoked one size of response, while larger feedback perturbations elicited a larger response. P2 amplitudes increased with increases in the feedback perturbation magnitude. Moreover, a reliable relationship existed between vocal response magnitude and P2 amplitude: vocal response magnitude and P2 amplitude increased in response to perturbations between 50 and 250 cents, and then decreased in response to larger perturbations. ERPs allow us to hypothesize the stages of processing. Results will be discussed with respect to perceptual and production thresholds and implications for speech motor control.

11:00

4aSCa6. Sensorimotor integration during human self-vocalization: Insights from invasive electrophysiology. Jeremy Greenlee, Roozbeh Behroozmand, Nandakumar Narayanan (Neurosurgery, Univ. of Iowa, 1827 JCP, 200 West Hawkins Dr., Iowa City, IA 52241, jeremy-greenlee@uiowa.edu), Jonathan R. Kingyon (Dept. of Bioengineering, Univ. of Iowa, Ames, IA), Charles Larson (Speech and Commun. Disord., Northwestern Univ., Evanston, IL), Hiroyuki Oya, Hiroto Kawasaki, and Matthew A. Howard (Neurosurgery, Univ. of Iowa, Ames, IA)

Effective human speech requires the neural integration of ongoing vocal production with the auditory and somatosensory feedback signals that are produced. We are using invasive electrophysiology techniques in patient volunteers undergoing neurosurgical treatment in order to gain insights into these mechanisms and underlying neural circuits. By using multi-contact electrode arrays chronically implanted over the perisylvian temporal lobe auditory cortex (e.g., area PLST) and the inferior frontal gyrus (IFG), we can examine local field potentials and frequency-specific responses from cortical areas important for both vocal production and speech sound processing. Our initial studies have found that during self-vocalization, focal areas within higher order auditory cortex on the superior temporal gyrus (STG) show response modulation compared to the responses of the same areas during passive listening. Manipulation of the auditory feedback that a speaker receives during vocalization (e.g., pitch-shifted or delayed auditory feedback) leads to further modulation of these PLST sites. Measures of functional connectivity including electrical stimulation tract tracing or phase-synchrony analysis demonstrate that portions of PLST are functionally connected to regions of the IFG. These findings support forward models for vocal control in which efference copies of premotor cortex activity modulate sub-regions of auditory cortex.

11:20

4aSCa7. Speech motor learning alters auditory and somatosensory event-related potentials. Takayuki Ito, Joshua H. Coppola (Haskins Labs., 300 George St., New Haven, CT 06511, taka@haskins.yale.edu), and David J. Ostry (McGill Univ., Montréal, QC, Canada)

Speech motor learning is dependent upon changes to motor function, but it also results in changes to sensory systems. However, the neural mechanisms of sensory plasticity associated with speech motor learning are little understood. We here examined whether auditory and somatosensory cortical processes are changed in conjunction with speech motor learning. We tested native speakers of American English. Altered auditory feedback (AAF) training was used as a motor learning task. As subjects repeated aloud the speech utterance "head," the produced sound was feedbacked through headphones while the first formant of /ea/ was gradually decreased over 50 repetitions and held at a maximum change for 110 repetitions. In order to evaluate the effects of the resulting adaptation on cortical sensory processes, we recorded auditory and somatosensory event-related potentials (ERPs) using 64-channel electroencephalography before and after AAF training. Auditory ERPs were elicited by using the synthesized vowel sound "e." Somatosensory ERPs were elicited by facial skin deformation. We found changes to auditory and somatosensory ERPs following AAF training in individuals who showed adaptation to altered auditory feedback. The changes in ERPs were correlated with the amount of adaptation. This suggests that speech motor learning alters somatosensory and auditory cortical processing.

THURSDAY MORNING, 6 JUNE 2013 — 516, 9:00 A.M. TO 12:00 NOON

Session 4aSCb

Speech Communication: Voice and F0 Across Tasks (Poster Session)

Marc Garellek, Chair
Dept. of Linguist, UCLA, Los Angeles, CA 90095

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

4aSCb1. The relative contribution of rhythm, intonation, and lexical information to the perception of prosodic disorder. Paul Olejarczuk and Melissa A. Redford (Linguistics, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, paulo@uoregon.edu)

Acoustic studies suggest that children with autism spectrum disorder (ASD) produce atypical rhythm and intonation [Diehl and Paul (2011)]. Behavioral studies indicate that children with ASD combine prosody with lexical content in atypical ways [Peppé *et al.* (2007)]. The current study assessed the relative contribution of rhythm, intonation, and language context to perception of prosodic disorder. Short excerpts were taken from narratives produced by 18 children with ASD and 18 typically developing controls. Prior study indicated that listeners easily distinguished groups on the basis of these excerpts. Here, the excerpts were resynthesized to control for voice quality and to allow for selective inclusion of F0, duration, intensity, and lexical information. Experiment 1 investigated listeners' ability to distinguish the groups based on delexicalized samples that preserved only rhythm (duration + intensity), only intonation, or a combination of both. Experiment 2 investigated the contribution of lexical information to the judgments, and the interaction of lexical information with intonation. Results indicated that (1) listeners were less able to distinguish between groups in the Intonation Only condition, and (2) intonation had a negligible effect on performance when lexical content was present. We conclude that rhythm cues and lexical information contribute more to perceived disorder than intonation.

4aSCb2. Intonation perception in English: Effects of stimulus amplitude and listeners' language background. Katherine Morrow and Chang Liu (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, katherinemorrow@utexas.edu)

The contour of fundamental frequency (F0) of the final word is the primary acoustic cue for intonation production and perception in English utterances. On the other hand, speakers of Mandarin Chinese may have to use

other acoustic cues such as amplitude and duration rather than F0 contours to differentiate intonation contrasts since F0 contours carry lexical meaning in Mandarin Chinese. The goal of this study was to examine the role of the final word amplitude in intonation perception of English sentences. The final word amplitude was manipulated at three levels relative to the carrier sentence: −6, 0, and +6 dB. F0 contours of the final word were controlled continuously from falling to rising patterns. Listeners' task was to identify the sentence intonation: question or statement. Preliminary results showed the intonation boundary shifted from slightly falling F0 contours to slightly rising F0 contours as the final word amplitude decreased for Chinese listeners, but the boundary did not change for the three amplitudes for English listeners. These results imply that Chinese listeners may use the final word amplitude as a secondary cue to perceive intonation contrasts in English, while English listeners may primarily rely on F0 contours for intonation perception.

4aSCb3. Towards a vocal typology for American English. Tyler McPeek and James Harnsberger (U. Florida, 4131 Turlington Hall, Gainesville, FL, tylermcpeek@ufl.edu)

Prior work in the field of speaker identification has shown that individual voices are not uniformly dissimilar from one another: when misidentified, the errors are not randomly distributed but, in fact, indicate the existence of vocal stereotypes, or groups of voices that share identifiable features. Popular labels for such groups can include "rich," "droning," "gravelly," and many others. In this study, 100 American English male and female voices were separately rated for interspeaker similarity and the ratings used to posit nine vocal types for each gender. Acoustic properties corresponding to speaking rate, pitch variability, and mean pitch were the most predictive in classifying voices into types in discriminant analyses using a total of 23 measures. For female voices, chronological age strongly influenced the resulting taxonomy. Male voice types were not blocked by age, and unlike the female voices, a single type constituted a plurality (26%) of the voices in the database. The implications of this work for the modeling of other indexical properties of speech will be discussed, along with its implications in the applied area of forensic voice identification.

4aSCb4. Perceptual sensitivity to a model of the source spectrum. Marc Garellek (Dept. of Linguist, UCLA, Los Angeles, CA), Robin A. Samlan, Jody E. Kreiman, and Bruce R. Gerratt (Dept. of Head and Neck Surgery, UCLA, 31-24 Rehab. Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

A psychoacoustic model of the source spectrum has been proposed in which four spectral slope parameters describe perception of overall voice quality: H1-H2 (the difference in amplitude between the first and second harmonics), H2-H4, H4-2000 Hz (i.e., the harmonic nearest 2000 Hz), and 2000–5000 Hz. The goals of this study are to evaluate perceptual sensitivity in the mid-to-high frequency range of the model and determine how sensitivity to one parameter varies as a function of another. To determine listener sensitivity to slope changes for each parameter, just-noticeable differences were obtained for series of stimuli based on synthetic copies of one male and one female voice. Twenty listeners completed an adaptive up-down paradigm. To provide a baseline of listener sensitivity to each spectral slope parameter, the synthetic voices were manipulated so that spectral slope varied by 0.5 dB increments for each parameter while other parameters remained constant. We then assessed how listener sensitivity to a given harmonic slope parameter changes when the others covary. These results will help assess the validity of the model and determine what sources of cross-voice variability in spectral configuration are perceptible.

4aSCb5. Biomechanical models of damage and healing processes for voice health. Alba Granados, Jonas Brunskog, and Finn Jacobsen (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Blg. 352, room 111, Kongens Lyngby 2800, Denmark, algra@elektro.dtu.dk)

In voice-loading occupations, employees are required to use their voice for continuous and large periods of time, which might lead to voice problems. In this work, anomalous vocal-fold vibrations due to long-time high voice-load are investigated. Laryngeal endoscopic high-speed images within the vocal-fold plane are available. These data are used to improve existing continuum biomechanical models of the vocal-folds by analyzing the injury processes. The project is expected to result in methods that objectively demonstrate the impact of high voice-load on voice. A detailed description of the currently developing work will be presented, including a rigorous analysis of the hypothesized injury processes of the vocal folds.

4aSCb6. Perceptual consequences of changes in epilaryngeal area and shape. Robin A. Samlan and Jody E. Kreiman (Dept. of Head and Neck Surgery, UCLA School of Med., 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

Decreasing epilaryngeal area has been shown to increase glottal flow pulse skewing and harmonic amplitudes [Titze, J. Acoust. Soc. Am. **123**, 2733 (2008)]. It is not known, however, whether listeners perceive voice quality changes when epilaryngeal area is altered, or if perceived quality is different if the area change occurs at the ventricular folds or aryepiglottic (AE) folds. In this study, a kinematic vocal tract model was used to create five epilaryngeal cavity shapes resulting from constriction and retraction of the ventricular and AE folds. Four voice sources simulating varying degrees of vocal deviation were filtered through the five shapes for a total of 20 stimuli. Fourteen listeners completed a sort and rate task. Results were analyzed using multidimensional scaling (MDS). Altering the epilaryngeal cavity shape resulted in voice quality differences, and perceptual distances differed by voice source. AE fold constriction was perceived most differently from other shapes for all talkers. Ventricular fold constriction was perceived most similar to AE constriction for 3 of the 4 voice sources. Glottal flow and acoustic differences for each epilaryngeal shape will be described and related to the perceived differences in voice quality.

4aSCb7. A physiologically and perceptually motivated voice source model. Gang Chen (Elec. Eng., Univ. of California, Los Angeles, Department of Electrical Engineering, UCLA, 63-134 Engr IV, Los Angeles, CA 90095-1594, gangchen@ee.ucla.edu), Marc Garellek (Linguist, Univ. of California, Los Angeles, Los Angeles, CA), Jody Kreiman, Bruce R. Gerratt (Head and Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA), and Abeer Alwan (Elec. Eng., Univ. of California, Los Angeles, Los Angeles, CA)

Many glottal source models have been proposed, but none has been systematically validated perceptually. Our previous work showed that model fitting of the negative peak of the flow derivative is the most important predictor of perceptual similarity to the target voice. In this study, a new voice source model motivated by high-speed laryngeal videoendoscopy is proposed to capture perceptually-important source shape aspects. Six voice source models (the proposed model, two previous models developed at UCLA, as well as the Fujisaki-Ljungkvist, Liljencrants-Fant, and Rosenberg models) were fitted to 40 natural voices obtained by inverse filtering and analysis-by-synthesis (AbS). We generated synthetic copies of the voices using each modeled source pulse, with all other parameters held constant, and then conducted a visual sort-and-rate task in which listeners assessed the extent of perceived match between the original natural voice samples and each copy. Model fitting results showed that the proposed model provides a more accurate fitting to the AbS-derived source than the other models. Perceptual experiments showed that the proposed model provides a close match to the original natural voice. Perceptual studies examining the extent to which each model matches the target tokens will also be reported. [Work supported by NSF grant IIS-1018863 and NIH/NIDCD grant DC01797.]

4aSCb8. Variation of maximal Lyapunov exponent with voice disorders of pilots. Robert Ruiz (LA.R.A, Univ. of Toulouse, Toulouse, France), Philippe Plantin de Hugues (B.E.A, Bureau d'Enquêtes et d'Analyses pour la sécurité de l'aviation civile, Le Bourget, France), and Claude Legros (LA.R.A, Univ. of Toulouse, 5 allées A.Machado, Toulouse 31058 cedex 1, France, legros@univ-tlse2.fr)

The maximal Lyapunov exponent λ is a signature of chaos in the field of nonlinear dynamics. Analysis of vowels uttered by a troubled speaker can show irregularities and instabilities due to nonlinearities of the phonatory system. Therefore λ can be studied for the research of voice acoustic features able to present variations due to psychophysiological disturbances. These ones belong to the aeronautical context. Two pilots' voices have been recorded at stopovers during short-haul rotations on a day. A day of driving was used as an experimental material to study a similar workload. After being woken up in a laboratory sleep inertia experiment, another pilot is recorded. Finally, the Cockpit Voice Recorder of a crashed airplane provided a real-case corpus for the study. Vowels are extracted in all recordings and their maximal Lyapunov exponent is estimated. Regardless of whether or not the chaotic behavior of the voice, results show a large dispersion, little variations with different directions from the normal state of the speaker to the end of the recordings. On the basis of these experiments, the number of speakers involved, the choice of the calculation parameters, the phonetic material used, λ has a low sensitivity to the aeronautical psycho-physiological disturbances.

4aSCb9. Effects of supraglottic compressions on the aerodynamics and acoustics of excised canine larynges. Fariborz Alipour and Eileen Finnegan (Commun. Sci. & Disord., Univ. of Iowa, 250 Hawkins Drive, 334 WJSHC, Iowa City, IA 52242, alipour@iowa.uiowa.edu)

The purpose of this study was to examine the aerodynamic and acoustic effects due to supraglottic compressions, which may be seen in some dysphonic patients. Canine larynges were prepared and mounted and vocal fold oscillations were generated and controlled by the flow of air through the glottis. Glottal adduction was accomplished by rotating the arytenoids with a suture passed behind the vocal folds to simulate lateral cricoarytenoid muscle action. Supraglottic medial and anterior-posterior compressions were accomplished by manual squeezing at the arytenoid level and alternating between the rest and compressed conditions. The raw data, including EGG, subglottal pressure, flowrate, and microphone signals, were recorded on a DAT tape and later digitized and processed with Matlab. A video image of the superior aspect of the larynx was recorded using a stroboscopic light during the whole experiment. Results indicated that the excised larynges oscillated better and easier without the false vocal folds, but generated louder sound with false vocal folds. Medial compression always resulted in increased subglottal pressure, decreased flow rate and most often increased the sound intensity, but decreased EGG closed quotient. Both of these compressions had negative effects on the amplitude of EGG signal, suggesting disruption of vocal fold contact.

4aSCb10. The quantal larynx revisited. Scott Moisik (Linguist, Univ. of Victoria, P.O. Box 3045, Victoria, BC V8W 3P4, Canada, srmoisik@uvic.ca) and Bryan Gick (Linguist, Univ. of British Columbia, Vancouver, BC, Canada)

Quantal effects signify nonlinear relations in the properties of speech sounds, traditionally emphasizing articulatory-acoustic relations [Stevens, J. Phonet. *17*, 3–45 (1989)]; these relations hold important clues to how continuous phonetic parameters map onto discrete phonological categories. Stevens described quantal states in laryngeal speech function, showing how the vocal fold abduction-adduction continuum can be partitioned into breathy, modal, and pressed phonatory quanta. This account, however, relies on a one-dimensional conceptualization of the larynx, which recent developments in laryngeal phonetic theory reveal to be inadequate [Edmondson and Esling, Phonology *23*, 157–191 (2006)]: a more realistic model must include the epilarynx. We reopen the issue of quantal laryngeal speech behavior in the context of recent research demonstrating quantal biomechanical properties in labial articulation [Gick *et al*., Can. Acoust. *39*, 178–179 (2011)]. Our "whole larynx" approach [Moisik and Esling, ICPhS, 1406–1409 (2011)] countenances epilaryngeal influence on laryngeal articulatory and phonatory possibilities through quantal biomechanical and aero-mechanical effects. We demonstrate, through computer simulation, three novel cases of laryngeal quantality: (1) vocal-ventricular interaction in glottal stop, glottalization, and laryngealization; (2) aryepiglotto-epiglottal stricture in pharyngeal consonants; (3) epilaryngeal predisposition for growl-like or harsh phonation.

4aSCb11. Individual control of singing voice based on cepstrum manipulation. Kenji Ikeda, Kota Nakano, Masanori Morise, Masato Nakayama, and Takanobu Nishiura (Grad. School of Information Sci. and Eng., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, is002081@ed.ritsumei.ac.jp)

Desktop music (DTM) software is used to synthesize various instrumental sounds, whereas it was difficult to synthesize the natural singing voice because the singing voice is more complicated than other instrumental sounds. In the past, it had been object that we output more natural singing voice by analyzing the singing voice and extracting the spectrum envelope with high accuracy. Recently, since the algorithm is improved, the singing voice synthesis software is used. In the conventional method, it had been focused changing the original voice personality by controlling some parameters of spectrum envelope. However, the method can synthesize the singing voice by only particular voice personality. The quality of the synthesized singing voice also depends on accurate control of voice personality. In this study, we attempt to control user's impression directly and investigate the control method that is focused on the singer's individuality in the singing voice. In this paper, we propose a voice personality control method based on mapping the timbre of target singer in the cepstrum domain and demonstrate the effectiveness of the proposed method. As a result of subjective experiments, we confirmed that the proposed method can control the voice personality of singers with high quality.

4aSCb12. Modeling vocal fold asymmetries with coupled Van der Pol oscillators. Jorge C. Lucero (Dept. Comput. Sci., Univ. of Brasilia, Campus Universitario Darcy Ribeiro, Brasilia, Distrito Federal 70910-900, Brazil, lucero@unb.br) and Jean Schoentgen (Lab. of Signals, Images and Telecommunication Devices, Université Libre de Bruxelles, Brussels, Belgium)

Models of the glottal sound source are being developed to extend a recent synthesizer of disordered voices [Fraj *et al*., J. Acoust. Soc. Am. **132**, 2603–2615 (2012)]. The synthesizer was based on a nonlinear wave-shaping algorithm, which generates a glottal excitation to a concatenated-tube representation of the trachea and vocal tract. The purpose of the present work is to incorporate a physics-based model of the vibrating vocal folds in order to increase the anatomical fidelity of the synthesizer. Further, the model will permit to characterize left-right fold asymmetries and explore the effect of those asymmetries on the resultant vocal timbre. In this report, the vocal folds are represented as a system of two coupled Van der Pol oscillators with noise terms and a detuning factor between their natural frequencies. Regions of phase locked and unlocked oscillations are determined and illustrated with bifurcation diagrams. Also, the effect of frequency detuning on the resultant frequency jitter is analyzed. The results are discussed in terms of their implications for modeling abnormal vocal fold behavior. [Work supported by CNPq (Brazil) and FNRS (Belgium).]

4aSCb13. The correlation between perceptual saliency and acoustic parameters of dysarthrias. Emily Wang (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1611 West Harrison, Ste. 530, Chicago, IL 60612, emily_wang@rush.edu) and Leo Verhagen (Neurological Sci., Rush Univ. Medical Ctr., Chicago, IL)

Dysarthria is a group of speech disorders resulting from neurological disturbances in central or peripheral systems. There are six single types of dysarthria and all present with deviations at both segmental and suprasegmental level. However, it is unclear what matters more to the listener: the deficits at the segmental or suprasegmental level. In this study, reading samples were collected from subjects with any of the three types of dysarthria: scanning speech of ataxic dysarthria, spastic dysarthria, and hypokinetic dysarthria. All had slow speaking rate, monopitch, and monoloudness. Acoustic analyses were used to examine changes at both segmental and suprasegmental level. At the segmental level, parameters obtained include word and syllable per minute, vowel F1 and F2, syllable, word, sentence, and pause duration, mean F0 and vF0 at sentence and paragraph level. Peak F0 and vowel duration of stressed and unstressed vowels were also obtained. Perception experiment was conducted. Pitch contours were extracted and tested separately from those unmanipulated stimuli. Listeners made forced choice for rate and speech naturalness for the former and for overall speech intelligibility, speech rate, and speech naturalness for the latter. Effective size was used to determine the contributions of parameters at the segmental and suprasegmental level.

4aSCb14. Vowel synthesis using a vocal tract mapping interface and simulation study of inverse mapping. Kohichi Ogata and Tomohiro Hayakawa (Grad. School of Sci. and Technol., Kumamoto Univ., 2-39-1 Kurokami, Chuo-ku, Kumamoto 860-8555, Japan, ogata@cs.kumamoto-u.ac.jp)

We have developed a vocal tract mapping interface to produce vowel sounds. The interface provides an easy and effective setting of the vocal tract shape with a simple mouse click on its interface window. A vocal tract shape at the mouse position on the interface window is calculated by interpolation based on the prepared vocal tract shapes. In this paper, the features and advantages of the interface are shown through examples of the generated vowel sounds and vocal tract shapes. In addition, the inverse estimation of the vocal tract shape from formant frequencies is studied as one of its applications. In this method, the vocal tract shape is estimated by searching the point inside the region after

determining a possible region that includes the solution. The possible region on the interface window is determined based on the changes in the formant frequencies. The usefulness of the proposed method is shown through simulation results.

4aSCb15. Disruptive effect of unattended noise-vocoded speech on recall of visually presented digits: Interaction between the number of frequency bands and languages. Kazuo Ueda, Yoshitaka Nakajima (Dept. Human Sci., Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, ueda@design.kyushu-u.ac.jp), Kana Doumoto (Unit of Perceptual Psych., Kyushu Univ., Fukuoka, Japan), Wolfgang Ellermeier, and Florian Kattner (Institut für Psychologie, Technische Universität Darmstadt, Darmstadt, Germany)

To assess the effects of degraded irrelevant speech on the serial recall of visually presented digits, noise-vocoded speech was generated in Japanese and German. Effects of the participants' native language were also examined by studying 40 Japanese and 40 German listeners. The number of frequency bands used in vocoding and the language (native or not) the irrelevant sound was derived from affected performance significantly. The participants' native language had a greater disruptive effect than the non-native language, particularly in conditions in which intelligibility was moderate. Speech sounds appear to have been processed automatically although the participant was instructed to neglect them. This must have required some amount of cognitive resources, which could have been used for the recall task otherwise. This automatic interference was stronger when the native language was used, probably because it contained perceptual cues that were more difficult to degrade.

4aSCb16. Evaluation of bone-conducted ultrasonic hearing-aid regarding transmission of phonetic features. Takayuki Kagomiya and Seiji Nakagawa (Health Res. Inst., National Inst. of Adv. Sci. and Technol., 1-8-31, Midorigaoka, Ikeda, Osaka 563-8577, Japan, t-kagomiya@aist.go.jp)

Human listeners can perceive speech signals in a voice modulated ultrasonic carrier from a bone-conduction stimulator, even if the listeners are patients with sensorineural hearing loss. Considering this fact, we have been developing a bone-conducted ultrasonic hearing aid (BCUHA). The purpose of this study is to evaluate the ability of BCUHA to transmit Japanese distinctive features to the recipient. For this purpose, a series of mono-syllable intelligibility experiments was conducted. A series of sequential information transfer analyses (SINFA) were carried out to analyze what kind of articulatory features were well transmitted. Results of the SINFA showed that: in vowel perception, "openness" and "frontness" were well transmitted, while in consonant perception, Japanese "you-on" (palatalized sound) feature was well transmitted; however, transmission of other features like articulatory position or manner was limited. These results indicated that the BCUHA has sufficient frequency resolution to transmit vowel information, while some signals are masked by carrier sound. To improve this problem, further investigation and development is required.

4aSCb17. Perceptual evaluation of the functional and aesthetic degradation of speech by wind noise during recording. Iain R. Jackson, Paul Kendrick, Trevor J. Cox, Bruno M. Fazenda, and Francis F. Li (Acoust. Res. Ctr., The Univ. of Salford, Newton Bldg., Salford M5 4WT, United Kingdom, t.j.cox@salford.ac.uk)

This paper will present results from a systematic investigation into functional and aesthetic audio quality of speech recordings degraded by wind noise. The major source of wind noise tested comes from velocity fluctuations interacting with the transducer, generating pressure fluctuations at the microphone diaphragm. To better understand the effect of this type of noise, a perceptual experiment was designed to assess task performance and perceptions of quality when speech and simulated wind noise are presented together. A wind noise simulator was developed, which produces realistic audio from anemometer data, to allow the noise to be isolated from other ambient sounds, and also enable salient parameters to be controlled. Two key components of wind noise in recordings were evaluated, the average level and its temporal variance or "gustiness." Eight levels of wind noise were factorially combined with three levels of gustiness. Each of these permutations was then presented with one of 24 randomly assigned, grammatically correct, nonsense sentences. Participants were asked to type the sentence they heard, rate the difficulty of the task, and indicate overall quality of the clip. Each sentence contained four keywords—correct identification of which was used for scoring performance.

4aSCb18. Effect of source viewing on the unmasking of speech. Alan Chorubczyk, Shin-Ichi Sato, and Maira Cardozo (Sound Eng., Tres de Febrero Univ., Amenabar 1819, Ciudad Autónoma de Buenos Aires 1428, Argentina, alanchoru@gmail.com)

In this work, it is analyzed if the subject visual awareness of the speaker presence would result in an enhancement of the speech understanding capability. For this purpose, a pair comparison test with different scales of speech intelligibility conditions with and without source viewing was conducted.

4aSCb19. Detection of obstructive sleep apnea by estimation of oral and nasal cavity cross-section areas from acoustic recordings of snore. Hsu-Kang Huang, Yi-Wen Liu (Elec. Eng., National Tsing Hua Univ., 101 Kuang-Fu Rd. Sec 2, Delta Bldg. Rm. 828, Hsinchu 30013, Taiwan, ywliu@ee.nthu.edu.tw), and Rayleigh Ping-Ying Chiang (ENT, Shin Kong Hospital, Taipei, Taiwan)

Obstructive sleep apnea (OSA) refers to the condition in which a person's breathing is paused while asleep, or the airflow is decreased, due to obstruction in the upper respiratory airway. In severe cases, OSA can cause complete arousals and deprive the patient from normal sleep. Surgical intervention is sometimes recommended, but accurate identification of the site of obstruction can be difficult. In the present study, we devised signal processing methods to estimate the site and the severity of airflow obstruction from recordings of sounds of snore. The vocal tract, the oral and the nasal cavity are modeled as three branches joining at the pharynx. Each branch consists of cylindrical segments whose cross-section areas can vary during snoring. Estimation of these cross-section areas consists of two steps: First, an auto-regressive moving-average method is applied to find the linear coefficients of a pole-zero model that optimally accounts for the recorded sound. Then, the Levinson-Durbin algorithm is applied to convert the coefficients to ratios of cross-section areas between adjacent segments. The present method is applied to a set of recorded snore samples during clinically confirmed apnea episodes, and results are compared with those of simple snore. Effectiveness of the method is analyzed statistically.

4aSCb20. Study of unvoiced fricative speech production: Influence of initial conditions on flow development. Yo Fujiso, Annemie Van Hirtum (GIPSA-Lab, Grenoble Univ., 11 rue des Mathématiques, Grenoble Campus, Saint Martin d'Hères 38402, France, yo.fujiso@gipsa-lab.grenoble-inp.fr), Kazunori Nozaki, and Shigeo Wada (Grad. School of Eng. Sci., Osaka Univ., Toyonaka-city, Japan)

Human unvoiced fricative speech sounds such as [s] and [f] are produced by a complex fluid-structure interaction. Indeed, a moderate Reynolds number ($100 \leq Re \leq 10000$) turbulent jet is issued from a constriction somewhere in the vocal tract formed between the hard palate and an articulator such as tongue, teeth, or lips. By using simplified in-vitro replicas representing parts of the human vocal tract, some physical phenomena relevant to the unvoiced fricative speech production can be reproduced and more easily understood. The current study focuses on the influence of initial conditions on flow development by performing flow measurements and Large Eddy Simulation on a rectangular channel containing a tooth-shaped obstacle.

4a THU. AM

4aSCb21. Measurements of the aero-acoustic properties of the vocal folds and vocal tract by broad and narrow band probes during phonation into controlled acoustic loads. Noel N. Hanna, John Smith, and Joe Wolfe (School of Phys., The Univ. of New South Wales, Sydney, NSW 2052, Australia, n.hanna@unsw.edu.au)

The aeroacoustic properties of the vocal folds and tract are difficult to measure directly. Here, they were measured using broad- and narrow-band excitation at the mouth during phonation into various acoustic loads, including a non-resonant load provided by an acoustically infinite waveguide with cross section comparable with that of the tract. The tract is treated as a duct terminated by the larynx. Mechanical properties of the walls and terminations were determined using a microphone array [Dickens *et al.* (2007)]. The vocal fold response was monitored with an electroglottograph and wall motion was measured electromechanically. The impedance spectra show negative resistance bands at frequencies near those of phonation, consistent with regeneration at the folds. The walls give inertances consistent with thicknesses of order 1 cm and compliances consistent with distributed stiffnesses of about 100 kN/m^3 [Hanna *et al.* (2012)]. The duct resonant properties are consistent with losses several times higher than the viscothermal losses at smooth rigid walls. Dickens *et al*., J. Acoust. Soc. Am. **121**, 1471–1481 (2007). Hanna *et al*., in *Proceedings of the Australian Acoustical Society Conference* (2012).

4aSCb22. Flow development in the uniform glottis and viscosity effects. Lewis Fulcher (Phys. & Astronomy, Bowling Green State Univ., Bowling Green, OH 43403, fulcher@bgsu.edu) and Ronald Scherer (Commun. Sci. & Disord., Bowling Green State Univ., Bowling Green, OH)

Thirty-two pressure distributions at minimal diameters of $d = 0.005$, 0.0075, 0.01, 0.02, 0.04, 0.08, and 0.16 cm have been measured at a number of transglottal pressures of interest for phonation. Care is taken to identify those portions of the pressure distributions within the glottis that include substantial regions of uniform decrease with axial distance. These portions are further examined to identify their components that have a linear dependence on the volume velocity and those that have a quadratic dependence on the volume velocity. An analysis based on the Navier Stokes equation creates a natural framework for investigating corrections to the parabolic profile of fully developed flow, which leads to the Poiseuille formula. For glottal diameters between 0.0075 and 0.02 cm the Poiseuille formula is a good approximation. Overall, an inverse 2.59 power law to describe the diameter dependence of the linear coefficients is found to be superior to the inverse cube dependence of the Poiseuille formula. Glottal flow resistance is used as a means of comparing the accuracy of the two power laws.

THURSDAY MORNING, 6 JUNE 2013 — 510A, 9:00 A.M. TO 12:00 NOON

Session 4aSP

Signal Processing in Acoustics: Sensor Array Beamforming and Its Applications

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Contributed Papers

9:00

4aSP1. Circular harmonics beamforming with spheroidal baffles. Stewart Holmes (Lloyd's Register ODS, Strandvejen 104A, Hellerup 2900, Denmark, stewart.holmes@lr-ods.com)

Circular microphone arrays have been well studied when mounted on cylindrical and spherical baffles. In this paper, the oblate and prolate spheroids are explored as a baffle for a microphone array system. It is shown that the prevailing methods for analysis of circular arrays may be easily adapted to this class of baffle, and that the use of these baffles represent a continuously variable geometry with edge cases that are the cylindrical and spherical baffles, and the array in empty space with no baffle. The performance—in the form of directivity index, maximum side-lobe level and main beam resolution—of spheroidally baffled arrays is analyzed with respect to errors created by transducer noise, positioning error, and modal aliasing, for both delay-and-sum and eigenbeamforming arrays. It is shown that the noise and position errors depend strongly on the spheroidal eccentricity while the aliasing error is fairly independent of baffle shape. Simulations of example arrays are used to show that while the prolate spheroidal baffle is of little advantage compared to current systems, the oblate spheroidal baffle can be used to create a significantly smaller array with only a relatively minor performance degradation.

9:20

4aSP2. Spatial sound pick-up with a low number of microphones. Julian D. Palacino and Rozenn Nicol (FT/OLNC/OLPS/COMSERV/SVQ//TPS, Orange Labs, 2 Av Pierre Marzin, Lannion 22307, France, julian.palacino@orange.com)

Portable audio devices have become more and more popular during the last decade. Recent advances in audio compression and electronic miniaturization allow people to keep all their music and movies at hand, at all times. Nowadays people are using their smartphones everywhere as photo or video cameras. In order to provide a consumer possibility of 3D audio recording adapted to these kinds of devices, we developed a compact microphone array able to pick-up a full 3D sound scene, using less than four microphones. To compensate for the low number of microphones, which results in poor spatial selectivity, spatial post processing is applied to the microphone signals and improves the sound source localization. Another advantage is that the post processing makes the sound reproduction flexible. The 3D audio scene can be converted into any format to be rendered to any equipment and any device. The paper will describe various recording set-ups in combination with the associated post processing. As a first assessment, their performances in terms of sound localization accuracy will be compared using a new set of objective criteria descriptors.

9:40

4aSP3. Lattice theory models for space-time sampling of acoustic signals. Kaushallya Adhikari and John R. Buck (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, kadhikari@umassd.edu)

The space-time sampling of underwater acoustic signals by both fixed and towed arrays of sensors can be modeled by lattice theory. The sampling schedule of a fixed array produces a rectangular space-time lattice while the sampling schedule of an array towed at a uniform velocity produces a trapezoidal lattice. Changing the velocity of a towed array corresponds to changing the skewness of a trapezoidal lattice. Willis and Bresler [IEEE Info. Theory (1997)] established an upper bound on the space-time-bandwidth product of a signal that can be sampled time sequentially by a lattice without aliasing. This upper bound provides a valuable perspective on the tradeoff among the temporal intersample interval, the interelement spacing of sensors, the velocity of the towed array, and the spectral support of signal that can be sampled without aliasing. A towed array with the same interelement spacing and temporal intersample interval can sample signals with broader spectral support than a fixed array. Alternately, a towed array can sample a signal with the same spectral support at a slower rate than an equivalent fixed array decreasing load on the data processing system. [Work supported by ONR.]

10:00

4aSP4. Frequency-sum beamforming in an inhomogeneous environment. Shima H. Abadi, Matthew J. VanOverloop, and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 W.E.Lay Automotive Lab. 1231 Beal Ave., Ann Arbor, MI 48109, shimah@umich.edu)

The arrival directions of ray paths between a sound source and a receiving array can be determined by beamforming the array-recorded signals. And, when the array and the signal are well matched, directional resolution increases with increasing signal frequency. However, when the environment between the source and the receivers is inhomogeneous, the recorded signal may be distorted and beamforming results may be increasingly degraded with increasing signal frequency. However, this sensitivity to inhomogeneities may be altered through use of an unconventional beamforming technique that manufactures higher frequency information by summing frequencies from lower-frequency signal components via a quadratic (or higher) product of complex signal amplitudes. This presentation will describe frequency-sum beamforming, and then illustrate it with simulation results and near-field acoustic experiments made with and without a thin plastic barrier between the source and the receiving array. The experiments were conducted in a 1.0-meter-deep and 1.07-m-diameter cylindrical water tank using a single sound projector, a receiving array of 16 hydrophones, and 100 micro-second signal pulses having nominal center frequencies from 30 to 120 kHz. The results from frequency-sum beamforming will be compared to the output of conventional delay-and-sum beamforming for different center frequencies. [Work sponsored by ONR and NAVSEA.]

10:20

4aSP5. Heading and hydrophone data fusion for towed array shape estimation. Jonathan Odom and Jeffrey Krolik (Elec. and Comput. Eng., Duke Univ., P.O. Box 90291, Durham, NC 27708, jonathan.odom@duke.edu)

This paper addresses the problem of towed array shape estimation for passive, horizontal sonar arrays. Beamforming and localization techniques significantly degrade when an assumed linear array bends due to tow platform maneuvers or ocean currents. In this paper, heading sensors along the array and acoustic hydrophone data are jointly used to estimate the shape of the array. Previously, heading data have been filtered using a dynamical motion model to reduce noise during turns. In recent work, a time-varying noise field directionality estimate that incorporates a dynamical model for the acoustic field provides a second, albeit biased, estimate of the array shape. In this paper, these two estimates are combined via adaptive weights to obtain improved shape estimates during maneuvers. A multi-source simulation is used to demonstrate the robustness of the combined array shape estimate when compared to the separate heading or acoustic sensor based techniques.

10:40

4aSP6. Least squares versus non-linear cost functions for a virtual artificial head. Eugen Rasumow, Matthias Blau (Institut für Hörtechnik und Audiologie, Jade Hochschule WOE, Ofener Str. 16/19, Oldenburg D-26121, Germany, eugen.rasumow@jade-hs.de), Simon Doclo (Institut für Physik, Carl von Ossietzky Universität, Oldenburg, Germany), Martin Hansen (Institut für Hörtechnik und Audiologie, Jade Hochschule WOE, Oldenburg, Germany), Steven van de Par (Institut für Physik, Carl von Ossietzky Universität, Oldenburg, Germany), Dirk Püschel (Akustik Technologie Göttingen, Soundtec, Oldenburg, Germany), and Volker Mellert (Institut für Physik, Carl von Ossietzky Universität, Oldenburg, Germany)

In order to take into account spatial information into binaural recordings, it is common practice to use so-called artificial heads. Disadvantageously artificial heads are inherently non-individual and bulky devices. Alternatively, the individual frequency-dependent directivity pattern of human head related transfer functions (HRTFs) can also be approximated by a microphone array with appropriate filters [Rasumow *et al*. (2011)]. Such a setup may be referred to as a virtual artificial head (vah). The filters for the application of the vah can be derived by minimizing a narrow band cost function including regularization constraints. As a first approach, it is appropriate to apply a least-squares cost function. The major advantage is its closed form solution [cf. Rasumow *et al*. (2011)], whereas from a psychoacoustically point of view, it seems more reasonable to minimize the dB-error instead. The latter cost function must, however, be minimized iteratively. We propose a minimization procedure for and present first results regarding the subjective appraisal of binaural filters derived using both cost functions. Future work includes the extension of this work to binaural cost functions.

11:00

4aSP7. Signal processing for hemispherical measurement data. Markus Müller-Trapet and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, mmt@akustik.rwth-aachen.de)

To realistically model the sound propagation in rooms, a detailed knowledge of the reflection properties of the surrounding surfaces is required. In this context, the reflection properties include both the sound absorption as well as scattering. In order to be able to measure the angle-dependent reflection properties of surfaces in-situ, a hemispherical microphone array was recently designed and built. For a reduction of the required hardware an efficient, rotationally symmetric sampling was chosen, so that 28 microphones on two concentric semicircles are employed to measure a total of over 3000 positions on a hemisphere in just over 15 minutes. This contribution will give an overview over the required signal processing steps to process the measurement data from such a microphone array. Special emphasis will be placed on the determination of the microphone positions and the special case of data available on a hemispherical surface. Also, the sound field model used to determine the impedance on the surface will be explained. Preliminary results will be presented.

11:20

4aSP8. Fly-over aircraft noise measurement campaign at Montreal-Trudeau airport using a microphone array. Jean-Francois Blais (Acoust. and Vib., Bombardier Aerosp., P.O. Box 6087, Station Ctr.-Ville, Montreal, QC H3C 3G9, Canada, jean-francois.blais@aero.bombardier.com), Cédric Camier (GAUS, Univ. of Sherbrooke, Sherbrooke, QC, Canada), Mathieu Patenaude-Dufour, Robby Lapointe (Acoust. and Vib., Bombardier Aerosp., Montreal, QC, Canada), Jonathan Provencher, Thomas Padois, Philippe-Aubert Gauthier, and Alain Berry (GAUS, Univ. of Sherbrooke, Sherbrooke, QC, Canada)

Engines being quieter due to high by-pass ratios, the airframe noise, produced for instance by landing gears or high-lift devices, has become a significant contributor to the total noise radiated by aircraft during approach and landing. As part of the investigations carried out to understand noise generation mechanisms, the beamforming techniques developed over the last decade and applied to microphone array measurements have shown to be effective tools for localization and quantification of these aerodynamic noise sources. In order to validate their in-house beamforming softwares, Bombardier Aerospace and the Groupe d'Acoustique de l'Université de Sherbrooke have conducted a 5-day measurement campaign in June 2012. The 95-microphone array was located on the roof of a building next to the Montreal-Trudeau airport. Aircraft position was determined by two high-definition cameras, both synchronized with the microphone array by inter-range instrumentation group time codes generators. This paper summarizes the measurement campaign. The aircraft tracking tool and the beamforming algorithms used to characterize the noise sources are presented. Several Bombardier CRJ fly-overs were recorded during this test. Beamforming results obtained for different airlines are compared in order to evaluate the repeatability of the method.

11:40

4aSP9. Accuracy of head-related transfer functions synthesized with spherical microphone arrays. Cesar Salvador, Shuichi Sakamoto, Jorge Trevino (Grad. School of Information Sci./Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai 980-8579, Japan, salvador@ais.riec.tohoku.ac.jp), Junfeng Li, Yonghong Yan (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Yôiti Suzuki (Grad. School of Information Sci./Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan)

The spherical harmonic decomposition can be applied to present realistically localized sound sources over headphones. The acoustic field, measured by a spherical microphone array, is first decomposed into a weighted sum of spherical harmonics evaluated at the microphone positions. The resulting decomposition is used to generate a set of virtual sources at various angles. The virtual sources are thus binaurally presented by applying the corresponding head-related transfer functions (HRTF). Reproduction accuracy is heavily dependent on the spatial distribution of microphones and virtual sources. Nearly uniform sphere samplings are used in positioning the microphones so as to improve spatial accuracy. However, no previous studies have looked into the optimal arrangement for the virtual sources. We evaluate the effects of the virtual source distribution on the accuracy of the synthesized HRTF. Furthermore, our study considers the impact of spatial aliasing for a 252-channel spherical microphone array. The microphone's body is modeled as a human-head-sized rigid sphere. We evaluate the synthesis error by comparison with the target HRTF using the logarithmic spectral distance. Our study finds that 362 virtual sources, distributed on an icosahedral grid, can synthesize the HRTF in the horizontal plane up to 9 kHz with a log-spectral distance below 5 dB.

Session 4aUWa

Underwater Acoustics: Detection and Localization

Yong Min Jiang, Cochair
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Julien Bonnel, Cochair
ENSTA Bretagne, 2 rue François Verny, Brest cedex 9 29806, France

Contributed Papers

9:00

4aUWa1. Using warping processing to range bowhead whale sounds from a single receiver. Julien Bonnel (LabSTICC (UMR CNRS 6285), ENSTA Bretagne, 2 rue François Verny, Brest cedex 9 29806, France, julien.bonnel@ensta-bretagne.fr) and Aaron Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA)

In certain shallow water environments, the acoustic propagation of low-frequency marine mammal calls can be well-modeled as a discrete set of normal modes. Each mode propagates with a different group velocity, and thus in principle the range of the call can be inferred by comparing relative arrival times of the modal arrivals. Traditionally, several time-synchronized hydrophones are required to spatially filter out individual modes in order to measure relative arrival times. In this presentation a nonlinear signal processing method classed "warping" is used to identify individual mode arrival times on a single receiver, even when the mode arrivals are overlapping in time. Warping processing is limited to frequency-modulated sources with monotonic increases or decreases of frequency with time. It is thus applicable to whale calls that consist of simple frequency-modulated upsweeps or downsweeps. Once the modes are separated, the source range can be estimated using conventional modal dispersion techniques. This method is applied on several bowhead whale vocalizations recorded near Kaktovik (Alaska) in 2012. Bowhead whale calls are ranged up to 35 km under median ambient noise conditions. These single-receiver range estimates are consistent with estimated ranges previously obtained via other methods [Work supported by North Pacific Research Board and Shell Exploration and Production Company.]

9:20

4aUWa2. Source signature characterization and feature based detection of open-circuit SCUBA regulators. Kay L. Gemba and Eva-Marie Nosal (Dept. of Ocean and Resources Eng., Univ. of Hawai'i at Manoa, 2500 Campus Rd., Honolulu, HI 96822, gemba@hawaii.edu)

A goal of the Center for Island, Maritime, and Extreme Environment Security is to develop passive acoustic methods to monitor harbor and near-shore environments. To study detection of open circuit SCUBA divers, several regulator configurations were recorded and characterized under ideal conditions in a pool environment. Sound Pressure Levels (SPL) and Sound Spectral Levels were calculated over the 4 kHz to 80 kHz band. Results include SPL range over all recordings along with variation of SPL due to change in SCUBA tank pressure and regulator flow rate. Regulator broad-band signatures were used to test three diver detection algorithms under varying synthetic and real ambient noise conditions. The first detector is a broad band energy detector exploiting *a priori* knowledge of the underlying signal length. The other two detectors are an envelope and a cepstral detector which estimate the divers' fundamental breathing frequency and require at least two breaths for a positive result. In order to maximize SNR, regulator signatures were band pass filtered to exploit their respective dominant features. Receiver operating characteristic curves were calculated to compare detector performances. [Work funded by the U.S. Department of Homeland Security through the Center for Island, Maritime, and Extreme Environment Security.]

9:40

4aUWa3. Data-based sensitivity kernel in a highly reverberating cavity. Selda Yildiz, Christian Marandet, Sandrine T. Rakotonarivo (Marine Physical Lab., Scripps Inst. of Oceanogr./UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, syildiz@ucsd.edu), Philippe Roux (Institut des Sci. de la Terre, Université Joseph Fourier, Grenoble, France), and W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr./UCSD, La Jolla, CA)

Our goal is to acoustically localize medium inhomogeneities, (i.e., scatterers) in a very complex medium without having to resort to constructing an accurate propagation model. Instead, we use a data-based sensitivity kernel approach to characterize medium changes in this complex medium which, in this study, is a highly reverberating cavity. The efficacy of the method is confirmed in an experiment with a moving aggregate of ping-pong balls inside a fish tank of 5.6 m diameter and water depth of 1.05 m in the ~10 kHz frequency regime; acoustic sources and receivers are on the periphery of the tank. Using a sensitivity kernel constructed from field data for scatterers at a sparse set of known positions, we demonstrate that we can localize other scatterers at unknown positions.

10:00

4aUWa4. Active detection of a moving target in a waveguide with strong masking echoes. Yoann Benoit and Claire Prada (Institut Langevin, ESPCI ParisTech, CNRS UMR 7587, 1 rue Jussieu, Paris 75006, France, claire.prada-julia@espci.fr)

In shallow water, active detection of a small moving target can be difficult because of strong echoes from large fixed obstacles. To cancel strong unwanted echoes, differences between successive acquisitions can be achieved, however they are very sensitive to fluctuations. A projection method combined with a fast acquisition technique is proposed as a robust alternative. An ultrasonic experiment is presented: a 64 transducers linear vertical array is used to detect a small target moving above a large obstacle in a waveguide. To reduce acquisition time, eight groups of adjacent elements transmit linear frequency modulations with increasing delays in a single emission. The 8x64 array response matrix is then obtained by correlations and time windowing. The projection is achieved between two acquisitions obtained while the target is moving, in order to remove the obstacle's contribution. Namely, the second acquired matrix is projected on the space orthogonal to the eight singular vectors of the first acquired matrix. Then, it is shown that the first singular vector of the projected matrix focuses on the second target's position. Comparisons are made with the decomposition of the time reversal operator in differential mode and conventional beamforming.

10:20

4aUWa5. Underwater source localization using a hydrophone-equipped glider. Yong Min Jiang and John Osler (Ctr. for Maritime Res. and Experimentation, Viale San Bartolomeo 400, la Spezia, (SP) 19126, Italy, jiang@cmre.nato.int)

Buoyancy-driven underwater gliders are autonomous underwater vehicles that were originally developed to collect oceanographic data. CMRE is studying the use of this technology for the characterization of denied areas, including alternate sensor payloads and applications. During the Rapid

Environmental Assessment phase of the Noble Mariner 2012 NATO exercise, Conducted in Gulf of Lions in September 2012, an omnidirectional hydrophone was mounted on a shallow water glider to sample the spatial distribution of the acoustic and oceanographic fields at different ranges and depths. This paper presents a study of the potential to localize acoustic sources by using the acoustic and environmental data collected by the glider. During the experiment, a bottom moored acoustic source was deployed in an area with benign bathymetry. Continuous wave and frequency modulated pulses were broadcast for approximately 6 h. The glider was flying along predefined tracks and the distances from the source were typically from 5 to 9 km. A Ray tracing model is used to evaluate the arrival structures of the acoustic signal, and to estimate the source location. The impact of the range dependent water column sound speed profile on the uncertainty of the source localization is also discussed.

10:40

4aUWa6. Three-dimensional localization of multiple sources in an uncertain ocean environment. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no) and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper develops a Bayesian focalization approach to simultaneous three-dimensional localization of multiple sources in shallow water with uncertain environmental properties, for application to horizontal line array (HLA) data. The algorithm maximizes the posterior probability density over unknown environmental (seabed and water column) and source parameters (includes source-bearing) using an adaptive hybrid optimization algorithm. Maximum-likelihood expressions for source strengths and noise variance are used that allows these parameters to be sampled implicitly rather than explicitly. An extension to the algorithm that optimizes for *a priori* unknown number of sources, based on minimizing the Bayes information criterion, is developed and presented. The algorithm is applied to simulated multi-frequency data in a continental shelf environment, and to data recorded on a HLA deployed on the seafloor in an experiment conducted in the Barents Sea.

11:00

4aUWa7. Small boat localization using adaptive three-dimensional beamforming on a tetrahedral and vertical line array. John Gebbie, Martin Siderius (Elec. and Comput. Eng. Dept., Portland State Univ., 1900 SW 4th Ave., Ste.160, Portland, OR 97201, jgebbie@ece.pdx.edu), Peter Nielsen, James H. Miller (Res. Dept., STO-CMRE, La Spezia, Italy), Steven Crocker (Sensors & SONAR Systems Dept., NUWC, Newport, RI), and Jennifer Giard (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Passive acoustic detection and localization of small surface craft has a number of practical applications, such as monitoring and protecting sensitive marine habitats. Moored passive equipment can be cumbersome to deploy and communicate with, so AUV-mounted devices are being investigated as an alternative. The GLASS'12 experiment was designed to assess the feasibility of using a hybrid autonomous underwater vehicle outfitted with a compact volumetric nose array as a data collection platform. The array consisted of five vertical elements and 4 in a tetrahedral arrangement, and the hybrid underwater vehicle had the capability operating in either glider or propeller-driven modes. The rigid design of the array minimized element location mismatch and enabled the use of aggressive adaptive beamforming in 3-D. This facilitated isolation of broadband multipath arrivals originating from the motor of a small rubber boat. Cross-correlation of beams enabled the time-lag between the arrivals to be measured, which, in turn yielded information about the target range. The underlying formulation bears similarity to the passive fathometer [J. Acoust. Soc. Am. **120**(3) (2006)], which exploits surface wave noise rather than ship noise. This presentation will focus on the array beamforming and potential applications for localization and environmental sensing.

11:20

4aUWa8. Separation of moving ship striation patterns using physics-based filtering. Yann Le Gall and Julien Bonnel (ENSTA Bretagne, Lab-STICC (UMR CNRS 6285), 2, rue François Verny, Brest 29806, France, yann.le_gall@ensta-bretagne.fr)

When a ship is moving toward an acoustic receiver in an oceanic waveguide, the time-frequency representation of the recorded signal exhibits a striation pattern that can be useful in numerous applications such as ship localization or geoacoustic inversion. If there are many ships, the striation patterns add up and they must be separated if one wants to study them separately. In this paper, a physics-based filtering scheme for passive underwater acoustics has been developed. The algorithm allows separating the time-frequency striations of two different moving ships. The proposed method considers filtering the 2D Fourier transform of the received spectrogram. The filter design is based on the waveguide invariant principle and on some *a priori* knowledge on the oceanic waveguide. The noise nature on the spectrogram is taken into account by introducing a nonlinearity to the filtering scheme. The algorithm thus corresponds to a nonlinear homomorphic filter. The method is validated on both simulated data and experimental marine data. This filtering scheme offers good prospects for all applications using ship noise and a single receiver.

Session 4aUWb

Underwater Acoustics and Acoustical Oceanography: Propagation and Scattering

Kyle M. Becker, Chair
Ocean Acoust. Program, Office of Naval Res., 875 N. Randolph St., Arlington, VA 22203-1995

Contributed Papers

9:00

4aUWb1. Experimental verification of enhanced sound transmission from water to air at low frequencies. David C. Calvo, Michael Nicholas, and Gregory J. Orris (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, david.calvo@nrl.navy.mil)

Laboratory-tank measurement of enhanced sound transmission from water to air at low frequencies is presented. Findings are consistent with the theory of anomalous transparency of the water-air interface in which almost all of the acoustic energy emitted by a shallow submerged source is emitted into the air [Godin, Phys. Rev. Lett. **97** (2006)]. The classical picture in the water remains very much the same: the monopole source suffers a radiation efficiency decrease due to interference with the strong surface reflection [McDonald and Calvo, J. Acoust. Soc. Am. **122** (2007)]. For source depths progressively less than a fraction of an acoustic wavelength in water, the measured radiation pattern in the air becomes progressively omnidirectional. The wider radiation pattern owes itself to the conversion of inhomogeneous (evanescent waves) into propagating waves that fill the angular space outside the usual 13-degree cone. On-axis point measurements using a microphone in air and a hydrophone in water, along with the measured directivities, are consistent with previously published power transmission ratios from water to air. Asymptotic expressions for the radiated field in the air are also presented. [Work sponsored by the Office of Naval Research.]

9:20

4aUWb2. Measured scattering of a first-order vortex beam by a sphere: Cross-helicity and helicity-neutral near-forward scattering and helicity modulation. Viktor Bollen, David J. Zartman, Timothy M. Marston, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu)

The wavefield of a traveling wave acoustic vortex beam has an axial null and an angular phase ramp. An appropriately phased four-element transducer array can be used to generate a first order vortex beam [Hefner and Marston, J. Acoust. Soc. Am. **106**, 3313–3316 (1999)]. The direction of the phase ramp determines the helicity of the beam. Superposition of signals from an appropriately positioned four-element receiver array gives a helicity selective detector and commutation of diagonal source elements can be used to reverse the source helicity [Marston and Marston, J. Acoust. Soc. Am. **127**, 1856 (2010)]. These techniques were used to investigate the near forward scattering by a small sphere placed on or near a beam's axis. The forward scattering vanishes in the on-axis case [Marston, J. Acoust. Soc. Am. **124**, 2905–2910 (2008)]. As the sphere is moved off axis the scattering to a helicity neutral receiver is found to increase linearly in the displacement with a first order phase swirl as a function of the sphere coordinates. For cross-helicity detection (detection opposite the beam's helicity) as required by symmetry, the signal is approximately quadratic in the displacement with a second-order phase swirl. [Work supported by ONR.]

9:40

4aUWb3. A numerically stable rational approximant for the split-step Padé propagator. David E. Roberts (34 Craiglockhart Loan, Edinburgh, Scotland, EH14 1JS) and David J. Thomson (733 Lomax Rd., Victoria, BC V9C 4A4, Canada, drdjt@shaw.ca)

The split-step Padé algorithm due to Collins [J. Acoust. Soc. Am. **93**, 1736–1742 (1993)] provides a fast and accurate method for solving the parabolic equation (PE). The formal solution to the PE propagator, which involves the pseudo-differential operator $(1+X)^{1/2}$, is replaced by an $[n/n]$-Padé rational approximant. This approximant can be expanded either as a product or a sum of rational-linear terms, each term leading to a tridiagonal system of equations in X which is readily solved numerically. To ensure adequate suppression of undesirable contributions from the evanescent part of the spectrum ($X < -1$), stability constraints must be imposed. In contrast to this approach, we follow the suggestion of Lu and Ho [Optics Lett. **27**, 683–685 (2002)] and examine the use of an $[n-1/n]$-Padé approximant that inherently dampens these evanescent components of the spectrum. An algorithm for generating the necessary coefficients is described. Transmission losses computed using both rational approximants are compared for a typical shallow-water configuration.

10:00

4aUWb4. Three-dimensional acoustic propagation under a rough sea surface. Megan S. Ballard (Appl. Res. Labs. at the Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, suedewitt@gmail.com)

A three-dimensional propagation model using stepwise coupled modes is applied to calculate the acoustic field under a rough sea surface. The model is formulated in a cylindrical coordinate system and the solution for the three-dimensional acoustic field is approximated by accounting for mode coupling in the radial direction and including horizontal refraction in the azimuthal direction. The atmosphere above the sea surface is modeled as an acoustic half space having the properties of air and sea surface height is allowed to vary arbitrarily as a function of range and azimuth. For the sea surfaces presented in this work, the amplitude spectrum of the surface waves is modeled according to the JONSWAP spectrum and the directionality is included by assuming cosine-squared spreading. The acoustic field is calculated for sea surfaces determined for varying levels of wind intensity and fetch. A modal decomposition of the acoustic field is used to provide insight into the effects of the rough sea surface on the predicted transmission loss. The importance of using three-dimensional versus two-dimensional models for acoustic propagation under rough sea surfaces is investigated. [Work supported by ONR.]

10:20

4aUWb5. Acoustic interface treatment with an adjoint operator for linear range-dependent ocean index of refraction inversions. Edward Richards and Gopu Potty (Univ. of Rhode Island, Narragansett Bay Campus, South Ferry Rd., Narragansett, RI 02882, edwardlrichards@gmail.com)

Hurskey *et al.* [J. Acoust. Soc. Am. **115**(2), 607–619 (2004)] introduced the adjoint method and incorporated local sound-speed measurements into range-dependent ocean sound-speed inversion. They left two practical issues for the implementation of this method unresolved in this paper: the collection of appropriate environmental measurements and the implementation of bottom boundary

conditions. The first of these issues was considered in a simulation study that used an oceanographic glider to collect range-dependent sound-speed measurements [Richards, CMRE memorandum (2012)]. A covariance matrix was constructed from the changes observed in the range-dependent sound-speed field. The adjoint inversion was performed in a reduced element subset of the empirical orthogonal functions (EOF) base of the covariance matrix. The second issue is the focus of this paper, which describes a bottom boundary condition with a defined adjoint operator. A horizontal fluid-bottom interface is implemented using the implicit finite difference form of the parabolic equation introduced by McDaniel and Lee [J. Acoust. Soc. Am. **71**(4), 855–858 (1982)]. Combined with local range-dependent sound-speed statistics gathered with gliders, this development may provide a method of near real-time acoustic measurement of ocean sound-speed variations between an acoustic source and vertical hydrophone array.

10:40

4aUWb6. Applicability of two-dimensional boundary scattering models as a proxy for three-dimensional models. Bryant M. Tran (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bmtran@gmail.com), Sumedh M. Joshi (Ctr. for Appl. Mathematics, Cornell Univ., Ithaca, NY), and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Three-dimensional numerical models offer unique insight into the nature of scattering from rough surfaces. However, use of these models is computationally prohibitive for any application more time-sensitive than basic research. This work seeks to determine a proxy for full three-dimensional rough boundary scatting models using appropriate two-dimensional models. Specifically, a Monte Carlo Kirchhoff approximation model in 2D with a derived proxy relationship applied is compared to a similar model in 3D. The region of validity of the proxy will be explored. The usage of the proxy function when applied to an finite element method model will also be discussed. [Work supported by ONR Ocean Acoustics.]

11:00

4aUWb7. Comparison of two-dimensional axial-symmetric and two-in-one-half dimensional Green's function methods for three-dimensional shallow water acoustic propagation using finite element methods. Benjamin M. Goldsberry and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, 7201 Wood Hollow Dr., Apt.# 436, Austin, TX 78731, bmg08c@my.fsu.edu)

In shallow water acoustic propagation, performing a fully three-dimensional finite element model is currently unfeasible due to difficulty in implementation and limits in computational power. Therefore, alternative representations of the 3D acoustic field are sought. Two promising methods to represent the 3D field are a locally invariant, 2.5D Green's Function kernel, and a 2D axial-symmetric reduction of the 3D Helmholtz equation. When a spherical source is used, azimuthal symmetry of the acoustic propagation is assumed, and these two methods can be compared in 2D planes of the 3D field. Because the 2.5D method takes more computational time than the axial-symmetric method, the accuracy of the pressure field are compared to see if the axial-symmetric method can be used in place of the 2.5D method. First, the two methods are compared for a flat ocean surface with stratified media. Then, a wedge-shaped ocean surface is considered, and the two methods are compared with 2D PE solutions. These comparisons will show if the axial-symmetric method produces similar results to the 2.5D method, and if so, under which geometrical and physical situations the axial-symmetric method can be used in place of the 2.5D method. [Work sponsored by the Office of Naval Research, Ocean Acoustics.]

11:20

4aUWb8. The rate of convergence and error distribution of Galerkin approximations to eigenvalues in underwater acoustics. Richard B. Evans (Mathematical Sci., RPI, 99F Hugo Rd., N. Stonington, CT 06359, rbevans@99main.com)

The eigenfunctions of the depth separated wave equation can be expanded in terms of a known finite basis set. The expansion coefficients are found by requiring that the error is orthogonal to each of the M members of the basis. This is the Galerkin approximation, wherein one obtains an MxM matrix eigenvalue problem that can be solved by existing software packages. The convergence of the matrix eigenvalues depends on the suitability of the chosen basis set. Typically, the errors in the matrix eigenvalues are bounded by $1/(M^r)$, for large M, where the exponent $r > 0$ is the rate of convergence: Consider basis sets consisting of trigonometric (Fourier) or orthogonal (Legrendre) polynomials. The density discontinuity at the bottom of the ocean creates a corner in the eigenfunctions that should be built into the basis sets. A corner in the sound speed profile (e.g., at the bottom of the mixed-layer) yields $r = 3/2$, which assures convergence, but is still a practical consideration. The distribution of errors is a determining factor in the choice between Fourier-Galerkin and Legendre-Galerkin. The errors in the first (2/pi)M of the eigenvalues are orders of magnitude smaller with Legendre-Galerkin, in the problem presented.

THURSDAY AFTERNOON, 6 JUNE 2013 513ABC, 1:00 P.M. TO 5:20 P.M.

Session 4pAAa

Architectural Acoustics: Room Acoustics Computer Simulation II

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Invited Papers

1:00

4pAAa1. The differences and though the equivalence in the detection methods of particle, ray, and beam tracing. Uwe M. Stephenson (HafenCity Univ. Hamburg, Hebebrandstr. 1, Hamburg 22297, Germany, post@umstephenson.de)

Within the numerical methods used in room acoustics, the geometrical and energetic methods of sound particle, ray and beam tracing are often confused. This rather tutorial paper does not treat the tracing algorithms but rather aims to explain the differences in the physical models and the corresponding detection and evaluation methods. While ray tracing needs spherical detectors as receivers to count rays, the particle model is based on a weighting of the energies of the particles with their inner crossing distances to compute the local sound energy densities. For beam tracing, receiver points are sufficient. In its core, this paper shows the convergence of the evaluated intensities computed from immitted sound particle energies to those predicted by the well-known 1/R<+>2<+>-distance law for the free field—as applied with the mirror image source method and beam tracing as its efficient implementation. Finally, the geometrical methods are classified depending on their efficiency with higher orders of reflection and their extensibility by scattering and diffraction.

1:20

4pAAa2. Modeling (non-)uniform scattering distributions in geometrical acoustics. Dirk Schröder (LCAV, EPFL, Station 14, Lausanne 1015, Switzerland, dirk.schroeder@epfl.ch) and Alexander Pohl (HCU Hamburg, Hamburg, Germany)

In most cases, a surface is not ideally smooth. It rather contains regular and irregular dents, bumps, and other textures that influence the reflection of the incident wave. A reflection on such a corrugated surface causes a frequency-dependent redirection of the incident sound energy outside the specular direction, called scattering. While the computation of the specular part is well elaborated today, a model that thoroughly captures the wave phenomenon of scattering is still under discussion. Here, the most common assumption is that scattered energy follows a uniform Lambert distribution, which has proven to be a good approximation, especially in room acoustical applications. In this contribution, we will discuss Lambert-based scattering models (specular/diffuse sound field decomposition and vector mixing) and their implementations in methods of Geometrical Acoustics. We will analyze benefits and flaws of the respective models and investigate possibilities to introduce angle-dependent scattering for use cases where the uniform Lambertian distribution becomes invalid.

1:40

4pAAa3. A hybrid acoustic model for room impulse response synthesis. Alexander Southern and Samuel Siltanen (Dept. of Media Technol., Aalto Univ. School of Sci., Otaniemi, Finland, samuel.siltanen@aalto.fi)

The prediction and synthesis of room impulse responses (RIR) has wide application from computer gaming to architectural acoustics. When a level of physical accuracy is important, a single acoustic modeling technique is usually limited by its computational load. Hybrid acoustic models target different time/frequency regions of the RIR with different modeling techniques. This paper introduces a hybrid acoustic model consisting of a physical FDTD model for low-mid frequencies, beam-tracing, and the acoustic radiance transfer method in the early part and late parts at high frequencies respectively. In this work, attention is given to establishing the equivalence of the boundary characteristics in each modeling domain. Good agreement is demonstrated indicating that mixing the separate model responses leads to an energetically consistent RIR.

2:00

4pAAa4. Comparison of sound field measurements and predictions in coupled volumes between numerical methods and scale model measurements. Paul Luizard (LIMSI-CNRS, BP 133, Université Paris Sud, Orsay 91403, France, paul.luizard@limsi.fr), Makoto Otani (Faculty of Eng., Shinshu Univ., Nagano, Japan), Jonathan Botts, Lauri Savioja (Dept. of Media Technol., Aalto Univ. School of Sci., Aalto, Finland), and Brian F. Katz (LIMSI-CNRS, Orsay, France)

Prediction of sound fields in closed spaces can be achieved by various methods, either physical or numerical, based on different theoretical features. While the benefits and limitations of many methods have been examined for single volume spaces, there has been little effort in examining these effects for coupled volume situations. The present study presents a case study comparing theoretical,

experimentally physical measurements on a scale model, and various numerical methods, namely boundary element method (BEM), finite-difference time-domain (FDTD), and ray-tracing through the commercial software CATT-ACOUSTIC and ODEON. Although these numerical methods all use 3D numerical models of the architecture, each is different. Ray-tracing is more suitable to geometries with larger planes; BEM requires a more regular finer surface mesh; and FDTD requires a volumetric mesh of the propagation medium. A simple common geometry based on the scale model is used as a basis to compare these different approaches. Application to coupled spaces raises issues linked to later parts in the decay due to multi-slope decay rates, as well as diffraction phenomenon due to acoustic energy traveling between coupling surfaces from one volume to another. The ability of these numerical methods to adequately model these effects is the question under study.

2:20

4pAAa5. Inversion of a room acoustics model for the determination of acoustical surface properties in enclosed spaces. Soenke Pelzer and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany, spe@akustik.rwth-aachen.de)

Acoustic consultants are often in charge of treating spaces to fix problems or improve their room acoustics. To assess the situation and to find a solution, it is common practice to perform computer simulations. This technique is well established, cheap and effective. But it requires a CAD model of the room as well as properties of its boundaries, such as absorption and scattering coefficients. The CAD model is usually easy to obtain by asking the architect or measuring yourself, but quantifying the absorption and scattering coefficients of every single wall is a challenging task. This contribution presents a method that automatically matches absorption coefficients for every single wall by applying an inverse room acoustics model which bases on geometrical acoustics. The inversion is done numerically using a non-linear least-squares optimization process in MATLAB. The independent variables are all absorption coefficients and the goal is to minimize the error between measured and simulated impulse responses at all measured positions in the room. In addition to the acquisition of absorption and scattering coefficients, the goal after the optimization process is to perform interactive binaural auralizations that have a high perceptual congruence with the existing space.

2:40

4pAAa6. Construction and optimization techniques for high order schemes for the two-dimensional wave equation. Stefan Bilbao and Brian Hamilton (Music, Univ. of Edinburgh, Rm. 7306B, JCMB, Kings Bldgs., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, sbilbao@staffmail.ed.ac.uk)

With the advent of high performance parallel computing, audio rate room auralization using finite difference time domain (FDTD) methods is becoming possible in a reasonable computation time. Yet, there are still deficiencies in the methods, which are used for this purpose, particularly with regard to minimizing numerical dispersion over the full range of audible frequencies. This paper is concerned with construction techniques for families of methods for the test case of the 2D wave equation. Such methods are explicit, can be of very high accuracy, and operate over a small local stencil. Such schemes can be attractive in a parallel computation environment. As such methods will depend, invariably, on a set of free parameters, including the Courant number, a major concern is optimization. The remainder of this paper approaches the problem of setting up such an optimization problem in terms of various constraints and a suitable cost function. Some of the constraints follow from consistency, stability, isotropy, and accuracy of the resulting scheme, and others from perceptual considerations peculiar to audio. Simulation results will be presented.

3:00–3:20 Break

Contributed Papers

3:20

4pAAa7. Speech intelligibility prediction in very large sacral venues. Wolfgang Ahnert and Tobias Behrens (Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

In very large sacral venues like cathedrals or mosques the intelligibility of spoken words is very important especially during praying. For such venues with volumes of up to more than one million m3 special routines are needed for simulation to obtain predicted STI values by using of up to more than 1000 sound sources. Special cloud computing has been developed which allow to do the calculation by providing the needed memory size and by cutting the calculation time from days or weeks to hours. Here also modern binaural or ambisonic B-format impulse responses are derived. Additionally the absorption behavior of typical floor materials in such venues has to be known like worshipers in church pews or sitting or kneeing on carpets in mosques. This absorption is often the only one in sacral venues to reduce the reverberation time. For mosque projects the measurement of absorption coefficients of persons in typical postures and arrangements has been done according to the reverberation room method. Persons have been tested on a carpet as a 10 m<+>2<+> sample within a surrounding reflective barrier while standing, kneeing on carpet or being in Muslim specific praying posture.

3:40

4pAAa8. The effect of edge caused diffusion on the reverberation time - A semi analytical approach. Stefan Drechsler and Uwe Stephenson (HafenCity Univ., Hebebrandstr. 1, Hamburg 22297, Germany, drechsler@anklick-bar.de)

The basic numerical model here is the Anisotropic Reverberation Model (ARM). This geometrical/energetic model assumes a homogeneous but anisotropic sound field in room acoustics. Its system of linear differential equations describes the redistribution of sound energy to different directional ranges by wall reflections, which may be specular or diffuse, where the diffuse reflections are caused also by the edges (edge effect). The reverberation times result from eigenvalues and eigenvectors of the differential equation system. Recently, an analytical formula has been found, that calculates the diffracted, angle dependent sound field, averaged over the octave band even for arbitrarily shaped polygons. The reverberation times calculated with the ARM extended by that edge diffraction are presented. So, first time, not only for the

typical shoe-box room with an absorbing floor and reflecting walls realistic reverberation times can be calculated, taking the edge effect into account.

4:00

4pAAa9. Numerical models for predicting absorption/insulation performance of acoustic elements. Naohisa Inoue and Tetsuya Sakuma (The Univ. of Tokyo, 5-1-5 Kashiwanoha, Kashiwa 2778563, Japan, naohisa.inoue7@gmail.com)

With a great improvement of computer resource availability, numerical analysis is widely used to investigate acoustic characteristics of various materials. A further expectation will be to predict absorption/insulation performance of acoustic elements used for buildings, automobile and so on, which can be the alternative to the actual measurements. This paper presents general numerical models for predicting the absorption coefficient and the transmission loss of acoustic elements with arbitrary shape and material composition. The features of the models are: (1) A test sample is mounted in the cavity or aperture on a thick rigid baffle: (2) FEM is employed for the materials and the air in the cavity, and coupled with sound fields out of the baffle by BEM: (3) Acoustical indices are calculated from the incidence power and the absorption/transmission power on the interfaces. Numerical simulation demonstrates the oblique incidence absorption coefficients and transmission losses of single- and multi-layered materials, and the influence of the sample's area is discussed in comparison with theoretical values for the infinite area. Additionally, the influence of the thickness of the cavity/aperture is examined, which is so-called "niche-effect."

4:20

4pAAa10. Hexagonal vs. rectilinear grids for explicit finite difference schemes for the two-dimensional wave equation. Brian Hamilton and Stefan Bilbao (Acoust. & Fluid Dynam. Group, Univ. of Edinburgh, Rm. 4350, JCMB, Kings Bldgs., Mayfield Rd., Edinburhg EH9 3JZ, United Kingdom, b.hamilton-2@sms.ed.ac.uk)

Finite difference schemes for the 2-D wave equation operating on hexagonal grids and the accompanying numerical dispersion properties have received little attention in comparison to schemes operating on rectilinear grids. This paper considers the hexagonal tiling of the wavenumber plane in order to show that the hexagonal grid is a more natural choice to emulate the isotropy of the Laplacian operator and the wave equation. Performance of the 7-pt scheme on a hexagonal grid is better than previously reported as long as the correct stability limit and tiling of the wavenumber plane are taken into account. Numerical dispersion is analyzed as a function of temporal frequency to demonstrate directional cutoff frequencies. A comparison to 9-pt compact explicit schemes on rectilinear grids is presented using metrics relevant to acoustical simulation. It is shown that the 7-pt hexagonal scheme has better computational efficiency than parameterized 9-pt compact explicit rectilinear schemes and that the error remains isotropic to fourth-order. Simulation results are presented.

4:40

4pAAa11. Acoustic propagation modeled by the curvilinear Fourier pseudospectral time-domain method. Maarten Hornikx and Daan Steeghs (Eindhoven Univ. of Technol., P.O. Box 513, Eindhoven 5600 MB, Netherlands, m.c.j.hornikx@tue.nl)

The Fourier pseudospectral time-domain method is an efficient domain-discretization wave-based method to model sound propagation in inhomogeneous bounded media. The method was successfully applied to model atmospheric sound propagation and acoustics in urban environments. One of the limitations of the method is its restriction to a Cartesian grid, confining it to staircase-like geometries. When applying a transform from the Cartesian coordinate system to the curvilinear coordinate system, more arbitrary geometries may be solved by the method. In free field, the frequency dependent accuracy of the curvilinear Fourier pseudospectral time-domain method is investigated as a function of the deformation angle of the grid. Further, the performance of the pseudospectral method with a curvilinear grid as well as a Cartesian grid for scattering of elementary objects as an inclined plate and a cylinder is studied. Finally, sound propagation in a room with non-parallel boundaries and over a building with gabled roof is computed with the pseudospectral method with a curvilinear grid and compared with results obtained from the boundary element method. All computed results are in 2D.

5:00

4pAAa12. Three-dimensional point-cloud room model in room acoustics simulations. Milos Markovic, Søren K. Olesen, and Dorte Hammershøi (Section of Acoust., Dept. of Electron. Systems, Aalborg Univ., Fredrik Bajers Vej 7 B4-209, Aalborg Ø 9220, Denmark, mio@es.aau.dk)

Telepresence applications require communication with the feeling of being together and sharing the same environment. One important task in these applications is to render the acoustics of the distant room for the telepresence system user. This paper presents a fast method for the room geometry acquisition and its representation with a 3D point-cloud model, as well as utilization of such a model for the room acoustics simulations. A room is scanned with a commercially available input device (Kinect for Xbox360) in two different ways; the first one involves the device placed in the middle of the room and rotated around the vertical axis while for the second one the device is moved within the room. Benefits of both approaches were analyzed. The device's depth sensor provides a set of points in a three-dimensional coordinate system which represents scanned surfaces of the room interior. These data are used to build a 3D point-cloud model of the room. Several models are created to meet requirements of different room acoustics simulation algorithms: plane fitting and uniform voxel grid for geometric methods and triangulation mesh for the numerical methods. Advantages of the proposed method over the traditional approaches are discussed.

Session 4pAAb

Architectural Acoustics and Noise: Control of Impact and Airborne Noise in Buildings

Jean-Philippe Migneron, Chair
Ecole d'architecture, Universite Laval, 1, cote de la Fabrique, Quebec City, QC G1K 7P4, Canada

Invited Papers

1:00

4pAAb1. Global understanding of important parameters for improvement of impact insulation. Jean-Philippe Migneron and Jean-Gabriel Migneron (Ecole d'architecture, Universite Laval, 1, cote de la Fabrique, Quebec City, QC G1K 7P4, Canada, jean-philippe.migneron.1@ulaval.ca)

Floors impact insulation performances can be very different from one assembly to another. Many years of research and development have been done in this area. Now, it seems that a new solution or another product is commercialized every week. From buyers' point of view, there is a need to decide which topping and underlay will suit some noise requirement to the lowest cost. However, acousticians and specialists in noise control might consider a more complex problem, especially in multi-family dwellings. In lightweight construction, the relation between the floor and the ceiling underneath also affect the overall performance in terms of IIC, or even in risk of complaints. Knowing that it is often difficult to compare a real situation to a datasheet from a manufacturer or to building codes, few key ideas should be remembered. This paper aims to briefly review some conclusions of previous works done in impact sound insulation and to analyze how fundamental parameters can be applied to real installations. An example of variable modification on a topping sample also tries to demonstrate the influence of basic aspects without according most attention to single number ratings.

1:20

4pAAb2. Effects of flooring, topping, and underlayment on impact sound insulation of wood-joisted floor-ceiling assemblies. Lin Hu, Anes Omeranovic (Bldg. Systems, FPInnovations, 319 rue Franquet, Quebec, QC G1P 4R4, Canada, lin.hu@fpinnovations.ca), and Richard Dufour (Res. and Development, Feutre National Felt Inc., St-Narcisse, QC, Canada)

Footstep impact noise transmission through floor-ceiling assemblies is a major source of complaints in wood-framed multi-family buildings. Experience shows that adding a finishing-topping-underlayment sandwich on a base floor-ceiling assembly significantly affects the noise transmission. So far, no reliable tool has been developed for designing the proper sandwich and most proposed solutions rely basically on trial and error. FPInnovations has launched a major research project to develop such tools. First phase of the project is focused on understanding the effects of flooring, topping and underlayment on the impact sound insulation. A mock-up assembly simulating a pair of typical stacked rooms was built. Standard field impact sound transmission tests were conducted on floors topped with a 1.2 m by 1.2 m four-layer sandwich patch. By varying the combination of flooring, topping and underlayment, over 50 patches were tested to better understand the effects of the type of materials in each layer on the assembly's overall impact sound insulation. Based on a large number of tests conducted so far, it is evident that proper combination of flooring, topping and underlayment produces satisfactory impact sound insulation on wood-joisted floor-ceiling assembly. Verification testing on the assembly fully covered by the 4-layer samples is under way.

1:40

4pAAb3. On the relevance of impact source impedance at low frequencies. Berndt Zeitler, Stefan Schoenwald, and Brad Gover (Construction, NRC Canada, 1200 Montreal Rd., Bldg. M-27, Ottawa, ON K1A 0R6, Canada, berndt.zeitler@nrc.ca)

Many researchers have posed the question of whether the standard tapping machine simulates the impedance of real sources well enough to properly judge the impact sound insulation performance of a floor. Proposed solutions such as the modified tapping machine, the bang machine, and ball were results of these investigations. Recent data collected on bare (wood and concrete) floors, suggest that in the low frequency range, the impedance of the source has no influence on the power injected into the floor. This is presumably due to the fact that the bare floors have much higher impedances than most common sources, meaning only the blocked force of the source influences the injected power. This furthermore suggests that modifying or redeveloping the source is not necessary, and that through use of an appropriate weighting curve a single number rating that correlates well with subjective measurements can be defined. Supporting objective and subjective results will be presented.

Contributed Papers

2:00

4pAAb4. Impact noise isolation provided by a bare concrete slab evaluated according to European and American regulations and two prediction software compared to field measurements. Nicolas Lévêque (MJM Acoust.al Consultant, 6555 Côte des Neiges, Bureau 440, Montréal, QC H3S 2A6, Canada, nicolas.leveque00@gmail.com)

European regulation EN12354-2:2000 outlines a procedure to predict impact noise isolation provided by a floor covering installed on a concrete slab. In Annex B of this regulation, one can found a formula to calculate the Normalized Impact Sound Pressure Levels (NISPL) for a bare concrete slab. In North America, ASTM 2179-03 standard describes a procedure to measure impact noise insertion loss provided by a floor covering in laboratory conditions, using NISPL of a reference bare concrete slab. INSUL software uses Cremer's point force excitation theory to evaluate NISPL of concrete slab whereas BASTIAN software uses the procedure described in EN12354-2:2000 European regulation. This paper presents a comparison between NISPL calculated for 8 to 10 inches thick bare concrete slab according to procedures and software listed above and field NISPL measurements of thirty-five bare concrete slabs varying in thickness from 8 to 10 inches

presented in a article published in the Journal of Canadian Acoustical Association [Morin (2009)]. This comparison suggests that a statistical approach is required to evaluate accurately NISPL provided by a floor covering installed on a concrete slab according to the procedures and software listed above.

2:20

4pAAb5. Partition intersections and their effect on transmission loss and apparent sound transmission class. Jean-François Latour (Acoust. and Vib., SNC-Lavalin Inc., 2271 Fernand-Lafontaine, Longueuil, QC J5G 2R7, Canada, jean-francois.latour@snclavalin.com)

It is widely recognized and accepted that poorly designed intersections between partitions can significantly reduce the sound isolation that is achieved. However, the effect of specific intersection details are not widely reported and available. This presentation will focus on a case study in which different intersection details have been tested in the same conditions using the same source and receiving rooms. In terms of ASTC and transmission loss, *in situ* results (i.e., ASTM E 336) will be presented and compared with expected performance based upon laboratory test results (i.e., ASTM E 90) for the same partition type. Observed differences between laboratory and field performance will also be compared with results from similar previous studies.

2:40

4pAAb6. A study of a real world transmission loss chamber and the Kinetics UniBrace-L technology. Scott Hulteen, Eric McGowan, and Dominique J. Chéenne (Columbia College Chicago, 4118 N Ashland Ave., Chicago, IL 60613, scott.hulteen@loop.colum.edu)

This study performed at Columbia College Chicago (CCC) had two purposes: The first was to test the functionality of the recently developed real-world transmission loss (RWTL) chamber, while the second was to evaluate the performance of the Kinetics UniBrace-L product on a double wall construction assembly. CCC's RWTL chamber is designed to illustrate issues that have influence when testing partitions in the field. It is smaller in volume than a certified STC chamber, which results in modal effects on both sides of the chamber. Numerous microphone positions are available and are used to display the modal effects of the rooms and to determine an average sound pressure level in both spaces during testing. Absorption values are also substantially different between the sending and the receiving spaces (each side can be switched as either sending or receiving room) and diffuse-field conditions are not achieved in either side of the chamber. As such, the RWTL chamber will typically yield a lower STC value than a certified STC Chamber and will also yield lower values than what may be experienced when performing a test that would follow the standard ASTM E-336 or associated procedures.

THURSDAY AFTERNOON, 6 JUNE 2013 — 510B, 12:55 P.M. TO 4:40 P.M.

Session 4pAB

Animal Bioacoustics: Animal Vocal Modification in Noise

Susan Parks, Chair
Syracuse Univ., 114 Life Sci. Complex, Syracuse, NY 13244

Chair's Introduction—12:55

Invited Papers

1:00

4pAB1. Calling in gray treefrog choruses: Modifications and mysteries. Mark A. Bee (Ecology, Evolution and Behavior, Univ. of Minnesota, St. Paul, MN) and Joshua J. Schwartz (Biology, Pace Univ., Pleasantville, NY, NY 10570, JSCHWARTZ2@PACE.EDU)

Frogs are well known model systems in the study of communication for investigating the influences of noise on both signaling behavior and auditory processing. The best-studied frogs in this regard are two sister-species in the *Hyla versicolor* species complex (*H. versicolor* and *H. chrysoscelis*). Males of both species produce loud, pulsatile advertisement calls that function to attract females. In the competitive social environment of a breeding chorus, males commonly shift to producing longer calls (with more pulses) at slower rates when the level of competition increases. These behavioral modifications can be evoked in controlled laboratory experiments using playbacks of calls and chorus-shaped noise. In contrast to birds and mammals, however, there is no evidence that males increase the amplitude of their vocalizations (the Lombard Effect) in response to increasing noise levels. In addition, current evidence suggests that males do not necessarily profit significantly from producing longer calls at slower rates in terms of increasing their overall attractiveness to females, overcoming interference by overlapping calls, or increasing the detectability of their calls in noise. Despite the robust and directional nature of call modifications in noise, the evolutionary function of these modifications remains obscure.

1:20

4pAB2. Anthropogenic noise constrains acoustic communication in urban-dwelling frogs. Kirsten M. Parris (School of Botany, The Univ. of Melbourne, Parkville, VIC 3010, Australia, k.parris@unimelb.edu.au)

Urban noise may hinder acoustic communication in a diversity of animal groups by reducing the distance over which vocal signals can be detected. Given the importance of such signals for mate attraction and territory defence, this acoustic interference may have wide-ranging consequences for individual fitness. I will present a mathematical model of the active space of frog calls in urban noise as a function of body size. Despite having lower auditory thresholds, larger species with lower-frequency calls are predicted to suffer the greatest reduction in communication distance in noisy urban environments. During a field study in Melbourne, Australia, my colleagues and I found that the southern brown tree frog *Litoria ewingii* called at a higher frequency in traffic noise. However, modeling indicates that the observed frequency shift would confer only a modest increase in active space. Furthermore, as females of certain frog species

appear to prefer lower-frequency advertisement calls, this strategy may improve the audibility of calls but reduce attractiveness to potential mates. Calling more loudly would result in a larger increase in active space, but the high metabolic cost of this strategy could limit chorus tenure and ultimately reduce breeding success.

1:40

4pAB3. Acoustic invasion: How invasive species can impact native species acoustic niche? Camila Both (Programa de Pós-graduação em Ecologia e Evolução, Universidade Federal de Goiás, Campus Samambaia, Cx. 131, Goiânia, Goiás 74001970, Brazil, camilaboth@gmail.com) and Taran Grant (Instituto de Biociências, Universidade de São Paulo, São Paulo, Brazil)

The effects of invasive species on native taxa due to direct predation, food, and space competition, and disease transmission are well documented. However, the effects of acoustic invaders on animal communication have not been explored. We simulated an invasion of the acoustic niche by exposing calling native male white-banded tree frogs (*Hypsiboas albomarginatus*, harmonics at 60–1430 Hz and 2720–2780 Hz or 2280–2850 Hz) to recorded calls of the invasive American bullfrog (*Lithobates catesbeianus*, frequencies from 90 to >4000 Hz) at a non-invaded site in the Brazilian Atlantic Forest. In response, tree frogs immediately shifted calls to significantly higher frequencies. In the post-stimulus period, they continued to use higher frequencies and also decreased signal duration. Tree frogs did not change calling rate or inter-call interval. Acoustic signals are the primary basis of mate selection in many anurans, and such changes could negatively affect the reproductive success of native species. The effects of bullfrog vocalizations on acoustic communities are expected to be especially severe due to their broad frequency band, which masks the calls of multiple species simultaneously. These results show that invasive species could affect native species by interfering in their acoustic niche.

Contributed Paper

2:00

4pAB4. Impacts of acoustic competition between invasive Cuban treefrogs and native treefrogs in southern Florida. Jennifer B. Tennessen (Dept. of Biology, Penn State Univ., 208 Mueller Lab., University Park, PA 16802, jbt148@psu.edu), Susan E. Parks (Dept. of Biology, Syracuse Univ., Syracuse, NY), Ray W. Snow (U.S. Dept. of the Interior, National Park Service, Everglades National Park, Homestead, FL), and Tracy L. Langkilde (Dept. of Biology, Penn State Univ., University Park, PA)

The natural acoustic environment has undergone substantial changes over the past century due to human activities, creating novel soundscapes. Much research has focused on the impacts of anthropogenic noise on acoustic communication, including noise from transportation, construction, energy development, and defense. The impact of acoustic invasive species has been largely overlooked in bioacoustic studies on the behavioral and ecological consequences of noise. We conducted a passive monitoring experiment and a playback experiment to quantify the impact of invasive Cuban treefrog (*Osteopilus septentrionalis*) acoustic signals on the acoustic environment and on native treefrog acoustic behavior. Our results show that Cuban treefrog chorus altered the soundscape in Everglades National Park and affected the acoustic behavior of native treefrogs. Collectively, these results suggest that acoustic invasive species are important yet rarely considered sources of noise that can have ecological consequences at scales ranging from the individual to the ecosystem.

Invited Papers

2:20

4pAB5. Modification of humpback whale social sound repertoire and vocal source levels with increased noise. Rebecca Dunlop, Michael Noad (School of Veterinary Sci., Univ. of Queensland, Cetacean Ecology and Acoustics Lab., Gatton, QLD QLD 4343, Australia, r.dunlop@uq.edu.au), and Douglas Cato (Inst. of Marine Sci., Univ. of Sydney, Sydney, NSW, Australia)

In acoustic communication, high background noise is an important obstacle in successful receiver signal detection and perception of an intended acoustic signal. To overcome this problem, many animals modify acoustic signals by increasing the repetition rate, duration, amplitude, or frequency range of the signal. Humpback whales are the most vocal of the baleen species in that they use a wide and varied catalog of social sounds. More than 36 different sound types (vocal sounds and surface-generated sounds from energetic surface behaviors) were found during a three year study on migrating humpback whales. During periods of high wind noise (where there were no audible boats or singing whales in the area), humpback whales modify both their acoustic repertoire as well as vocal signal properties. We found that humpback whale groups gradually switched from primarily vocal to primarily surface-generated communication in increasing wind speeds and background noise levels, but kept both signal types in their repertoire. We also found evidence of the Lombard effect, in that in increased wind-dominated background noise levels, humpback whale groups tended to increase the amplitude of their vocalizations. Determining how whales modify their vocal behavior in increasing levels of background noise will give us an important insight into how they might cope with increasing levels of anthropogenic noise.

2:40–3:00 Break

3:00

4pAB6. Variation in the vocal behavior of southern right whales (*Eubalaena australis*) in coastal Brazilian waters. Susan Parks (Biology Dept., Syracuse Univ., 114 Life Science Complex, Syracuse, NY 13244, sparks@syr.edu), Karina Groch (Projeto Baleia Franca, Instituto Australis, Florianópolis, SC, Brazil), Paulo A. C. Flores (Centro Mamíferos Aquáticos – CMA, Centro Nacional de Pesquisa e Conservação de Mamíferos Aquáticos, ICMBio, MMA – CMA SC, Florianópolis, SC, Brazil), Renata S. Sousa-Lima (LaB - Laboratório de Bioacústica, Departamento de Fisiologia, Centro de Biociências, Universidade Federal do Rio Grande do Norte, Natal, RN, Brazil), and Ildar R. Urazghildiiev (Bioacoustics Res. Program, Cornell Univ., Ithaca, NY)

Currently, there are three recognized species of right whales. The largest population is the southern right whale (*Eubalaena australis*), with circumpolar distribution in the southern hemisphere. One calving area for this population is in Brazilian waters, where increasing numbers of right whales have been sighted over the past decade along with an increase in anthropogenic activities including

4p THU. PM

shipping traffic and fishing. The goals of this study were to describe the vocal behavior of southern right whales in Brazilian waters, assess the difference in vocalizations between areas with low and high human activity, and compare these results to studies conducted with North Atlantic right whales (*Eubalaena glacialis*) in the Western North Atlantic. Bottom-mounted archival acoustic recorders were deployed in October and November 2011 in two coastal locations in central Santa Catarina State, southern Brazil. One recorder was placed off Gamboa (27056'S and 48039'W, low traffic) and a second off Ribanceira (28011'S and 48037'W, high traffic). Automated detectors and noise statistic analysis tools developed for North Atlantic right whale upcalls were utilized to analyze the dataset. Calls produced by Brazilian right whales were significantly lower in fundamental frequency than North Atlantic right whale calls and the implications for these results will be discussed.

3:20

4pAB7. Are there metabolic costs of vocal responses to noise in marine mammals? Marla M. Holt (Conservation Biology Div., NOAA NMFS Northwest Fisheries Sci. Ctr., Seattle, WA), Robin C. Dunkin (Long Marine Lab., Dept. of Ecology and Evolutionary Biology, Univ. of California, 100 Shaffer Rd., Santa Cruz, CA 95060, dunkin@biology.ucsc.edu), Dawn P. Noren (Conservation Biology Div., NOAA NMFS Northwest Fisheries Sci. Ctr., Seattle, WA, United Kingdom), and Terrie M. Williams (Dept. of Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA)

Many species respond to increases in environmental noise by increasing the amplitude, duration, and/or repetition rate of their vocalizations. Potential costs of noise-induced vocal modifications include increased energetic costs but no empirical data in marine mammals exist. This study's objective was to compare the metabolic costs of communicative sounds produced by captive bottlenose dolphins ($N = 2$) under two conditions (low- and high-amplitude vocalization trials) to assess energetic costs of vocal responses to noise. An experimental trial consisted of a 10-min rest period to determine resting metabolic rate, followed by a two-minute vocalization period, and concluded with another 10-min rest period to measure recovery. Open-flow respiratory was used to measure oxygen consumption during each trial component. Vocalizations were recorded using a calibrated hydrophone for analysis. Both dolphins tended to produce longer vocalizations during high-amplitude trials. Thus, metabolic rates were related to total sound energy of all vocalizations produced during the vocal period for each trial. Metabolic costs tended to be higher during high sound energy trials, but only verged on statistical significance when vocal-performance differences were at least 10 dB (in cumulative sound exposure level). This study provides key data to assess biological consequences of anthropogenic noise exposure in marine mammals.

3:40

4pAB8. Vocal modifications in primates: Effects of noise and behavioral context on vocalization structure. Cara F. Hotchkin (NAVFAC Atlantic, 6506 Hampton Blvd, Norfolk, VA 23508, hotchkin.cara@gmail.com), Susan E. Parks (Dept. of Biology, Syracuse Univ., Syracuse, NY), and Daniel J. Weiss (Dept. of Psych., The Penn State Univ., University Park, PA)

During increased noise, modifications of the acoustic structure of vocalizations (amplitude, temporal, and spectral parameters) may allow release from masking, potentially conferring fitness benefits to vocally flexible signalers. Among primates, humans have demonstrated extreme vocal flexibility during noise, with modifications to all three speech parameters affected by both noise type and motivational state of the signaler. While non-human primates have also demonstrated changes to call amplitude and temporal characteristics, to the best of our knowledge spectral modifications have not been observed and the influence of behavioral context remains unknown. This experiment used playbacks of broad (10 kHz) and narrowband (5 kHz) white noise to investigate the effects of noise level and bandwidth on chirps and combination long calls (CLCs) produced by cotton-top tamarins. Noise amplitude and frequency content both influenced the structure of vocalizations; modifications included increased call amplitude (the Lombard effect), changes to call durations, and previously undocumented spectral shifts. Behavioral context was also relevant; modifications to CLCs were different from those observed in chirps. These results provide the first evidence of noise-induced spectral shifts in non-human primates, and emphasize the importance of behavioral context in vocal noise compensation.

Contributed Papers

4:00

4pAB9. Active ultrasonic vocal communication channel found in Mongolian gerbils through the cochlear microphonics with noise exposure. Hiroshi Riquimaroux, Keizo Fukushima, and Kohta I. Kobayasi (Life and Med. Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

Mongolian gerbils (*Meriones unguiculatus*) use ultrasonic vocal communication in frequency range of 22–45 kHz. However, hearing threshold of this frequency range reported has been very high, which is not suited for vocal communication. We examined possible active amplification created by the outer hair cells for frequency range of 22–45 kHz. In this study, we evaluated the amount of active amplification by the cochlear microphonics (CM) combined with temporary damage created by noise exposure. Adult gerbils received surgical implantation of a silver wire electrode on the round window of their cochlea through the middle ear to record CM. They were exposed to broadband noise (0.5 to 45 kHz) at 90 dB SPL for 5 min. CMs were recorded for tone bursts of 1 to 45 kHz. The following results were obtained. First, we observed the largest CM reduction just after the noise exposure. Second, decrements in CM amplitude depended on frequency. Low sensitivity frequency range above 22 kHz produced large reduction in CM amplitude. Third, decrease in CM amplitude was greater for lower stimulus intensities. Fourth, for testing frequencies, which produced large CM decrements, it took a longer period to recover back to pre-noise exposure amplitude levels. These findings indicate that reduction in CM amplitude appeared to be related to the cochlear nonlinearity generated by the outer hair cells.

4:20

4pAB10. Benign exclusion of birds using acoustic parametric arrays. Eric A. Dieckman, Elizabeth Skinner (Dept of Appl. Sci., College of William and Mary, P.O. Box 8795, Williamsburg, VA 23187, eric.dieckman@gmail.com), Ghazi Mahjoub, John Swaddle (Dept. of Biology, College of William and Mary, Williamsburg, VA), and Mark Hinders (Dept. of Appl. Sci., College of William and Mary, Williamsburg, VA)

Excluding birds from areas can be important in aviation safety, agriculture, and facilities maintenance. Presenting audible stimuli or predator vocalizations in the affected area often has initial success, but has a limited effect over the long-term, even if the signals are varied to reduce the chances of the birds habituating to the sounds and objects. Many birds are highly vocal and rely on auditory communication in almost every aspect of their life history. By creating noise specifically targeted to be within the vocal range of the

nuisance species, we hypothesize that the birds will be less able to communicate and will move to more acoustically suitable environments. To avoid introducing noise pollution to the surrounding environment we create spatially well-controlled "sonic nets" using a mix of speakers and acoustic parametric arrays. To better understand the interaction of the sound field and the environment we combine finite difference solutions of the KZK equation with 3D acoustic finite integration simulation. These simulations allow us to propagate a nonlinear acoustic beam to a real-world target and then study the scattering from the target. We discuss initial experiments with a parametric array in an aviary on the exclusion of starlings from a food source.

THURSDAY AFTERNOON, 6 JUNE 2013 — 510D, 1:20 P.M. TO 4:20 P.M.

Session 4pAO

Acoustical Oceanography and Underwater Acoustics: Biologic and Non-Biologic Scatterers

James Lynch, Cochair
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Ikuo Matsuo, Cochair
Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan

Contributed Papers

1:20

4pAO1. Acoustic scattering from a water-filled cylindrical shell: Mode identification and interpretation via finite element and analytical models. Aubrey L. Espana, Kevin L. Williams (Acoust. Dept., Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105, aespana@apl.washington.edu), Daniel S. Plotnick, and Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA)

Understanding the physics governing the interaction of sound with targets in an underwater environment is essential to improving upon existing target detection and classification algorithms. Simple models are viable tools for meaningful interpretation of scattering results. To illustrate this, two modeling techniques are employed to study the acoustic scattering from a water-filled cylindrical shell. The first model is a hybrid 2-D/3-D finite element (FE) model, whereby the scattering in close proximity to the target is handled via a 2-D axisymmetric FE model, and the subsequent 3-D propagation to the far-field is determined via a Helmholtz integral. This model is characterized by the decomposition of the fluid pressure and its derivative in a series of azimuthal Fourier modes, a technique that has previously facilitated mode identification [Espana *et al*., J. Acoust. Soc. Am. **130**, 2332 (2011)]. The second is an analytical solution for an infinitely long cylindrical shell, coupled with a simple approximation that converts the results to an analogous finite length form function. These two model results, when examined together on a mode-by-mode basis, offer visualization of the mode dynamics and the ability to distinguish the different physics driving the target response (i.e., structural modes versus water-waveguide modes). [Work supported by ONR.]

1:40

4pAO2. Testing of an extended target for use in high frequency sonar calibration. John L. Heaton (Mech. Eng., Univ. of New Hampshire, Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., 24 Colovos Rd., Durham, NH 03824, jheaton@ccom.unh.edu), Thomas C. Weber (Ocean/Mechanical Eng., Univ. of New Hampshire, Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., Durham, NH), Glen Rice (NOAA Office of Coast Survey, Univ. of New Hampshire, Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., Durham, NH), and Xavier Lurton (IMN/NSE/AS, IFREMER, Plouzane, Bretagne, France)

Acoustic backscatter tests were performed in a tank with a 200-kHz, 7°, SIMRAD EK60 Split-Beam Echo-Sounder, and a 256-beam RESON SeaBat 7125 Multi-Beam Echo-sounder. Tests were done in order to investigate the angular and range dependency of the scattering strength of a new test target in order to validate its use in sonar testing. This target was constructed of small chain links arranged in a "curtain" simulating an extended scattering surface, such as the seafloor. Target strength for individual links was collected as the links were rotated 360°. The links are combined into an extended surface target, spacing between scatterers being approximately 1 cm. The scattering network irregularity is enough to ensure random phase at the wavelength considered. The target scattering strength was measured as a function of grazing angle and range, hence varying the number of scatterers within the beam footprint. These tests suggest that the amplitude envelope of the scattered signals is Rayleigh distributed and that the backscatter strength depends linearly on the number of active scatterers, all desirable features for calibrating sonars used to make measurements of similarly random surfaces such as the seafloor. Results show a promising in-tank calibration technique when extended surface targets are desirable.

2:00

4pAO3. Observing natural methane seep variability in the northern Gulf of Mexico with an 18-kilohertz split-beam scientific echosounder. Kevin Jerram, Thomas C. Weber, and Jonathan Beaudoin (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, kjerram@ccom.unh.edu)

Underwater methane seeps support diverse biological communities on the seafloor and, in cases of bubble survival to the surface, contribute to the quantity of atmospheric methane. The National Oceanic and Atmospheric Administration (NOAA) ship *Okeanos Explorer* completed two research cruises for seep mapping and characterization in the northern Gulf of Mexico during August and September of 2011 and April of 2012. Seeps originating at depths of approximately 1500 m were observed during multiple transects with a 30-kHz Kongsberg EM 302 multibeam echosounder (MBES) and an 18-kHz Simrad EK60 split-beam scientific echosounder calibrated for backscatter. A methodology for determining vessel offsets for the EK60 using MBES seep observations as benchmarks is discussed as part of a larger framework for transformation of seep targets from the split-beam echosounder reference frame to the geographical reference frame. Utilizing sound speed and attitude data collected for the MBES, several EK60 observations of strong individual seeps are scrutinized for variability of seep position and target strength between 2011 and 2012.

2:20

4pAO4. *In situ* measurement of the individual target strength of crustacean zooplankton with concurrent optical identification. Christian Briseño-Avena, Jules S. Jaffe, Paul L. Roberts, and Peter J. Franks (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0208, cbriseno@ucsd.edu)

Acoustic methods are common tools in the study of zooplankton distributions and behaviors. However, relating acoustic scattering to zooplankton abundance is difficult because zooplankton acoustic properties have been

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challenging to obtain *in situ*. Most of the acoustic measurements of zooplankton come from either preserved or recently dead organisms, or are derived from computer models. Other attempts to measure *in situ* acoustic target strength (TS) of zooplankton have targeted larger taxa such as euphausiids and amphipods. *In situ* measurements of copepods and other small crustaceans are extremely desirable as these taxa can be dominant in marine ecosystems. Here we present *in situ* measurement of TS using frequencies of 1.5–2.5 MHz with concurrent stereoscopic imaging. The concurrent calibration of the optics and acoustics permits the quantification of individual acoustic TS associated with the individual organism that gave rise to the echo. Furthermore, new technological advances have allowed us to measure organisms with TS as small as -125 dB. The results of this work will permit improvements in extant acoustic models, and enhance our interpretation of acoustic data collected in the field.

2:40

4pAO5. Complimentary ultrasound methods for the estimation of sound speed in macroalgae. Jo Randall (Environ. Hydroacoustics Lab., Universite Libre de Bruxelles, Bruxelles, Belgium), Jean-Pierre Hermand (Environ. Hydroacoustics Lab., Universite Libre de Bruxelles, Ave. Franklin Roosevelt, Bruxelles B-1050, Belgium, jhermand@ulb.ac.be), Marie-Elise Arnould (Laborelec, Linkebeek, Belgium), Jeff Ross, and Craig Johnson (Inst. for Marine and Antarctic Studies, Univ. of Tasmania, Sandy Bay, TAS, Australia)

Temperate kelp forests are among the most productive ecosystems in the world. However, there is mounting evidence that these habitats are in decline, both in range and productivity. Acoustic propagation modeling has been used to identify primary productivity in seagrass beds, and work is ongoing in development as a method of providing large scale measurements of productivity in macroalgae forests. Acoustic predictive models require knowledge of the material properties of interest, yet little is known about the acoustic properties of seaweed species. As a preliminary step towards acoustic modeling of seaweed systems, this study investigates the acoustic properties of *Ecklonia radiata*, a key species in temperate Australian marine systems. Measuring sound speed in macroalgae, as with other biological material, provides unique challenges due to their intrinsic morphological and anatomical characteristics. Using a range of frequencies between 2 and 10 MHz different methods are proposed to measure sound speed both directly and indirectly. The measurements show a consistent result, with variation according to tissue type. This research provides an important first step toward the development of acoustic propagation models in kelp forest ecosystems.

3:00–3:20 Break

3:20

4pAO6. Modeling the acoustic scattering from large fish schools using the Bloch-Floquet theorem. Jason A. Kulpe, Michael J. Leamy, and Karim G. Sabra (Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, jkulpe@gatech.edu)

Scattering from large fish schools can significantly contribute to volume reverberation in the open ocean measured by mid-frequency long-range SONAR systems (1–10 kHz). This can potentially cause large false-alarms, especially if the resonance frequencies of the fish' air-filled swim bladder is excited. Hence, to ultimately improve the detection performance of long-range SONAR systems, we seek an efficient modeling technique for the acoustic scattering created by large school of fish, which readily accounts for the fish bio-acoustic properties, school's spatial configuration and multiple scattering effects. We exploit here a key observation to simplify our problem: fish in larger schools tend to swim in a periodic arrangement whereby we approximate the large school as an infinite system with a periodic collection of fish' air filled swimbladders. Thus, the Bloch-Floquet theorem, governing waves in periodic media, allows predictions of the acoustic field in an infinite media by simply modeling the dynamic response of a single unit cell only (containing one fish). This approach allows one to rapidly predict the frequency dependent reflection and transmission coefficient on a semi-infinite fish school for various incident waves. Good agreement was found with the results obtained from finite element modeling of realistic, finite sized fish schools.

3:40

4pAO7. Clustering of acoustic fish features tracked by broadband split-beam echo sounder. Masanori Ito, Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, ito@cs.tohoku-gakuin.ac.jp), Tomohito Imaizumi, Tomonari Akamatsu (National Res. Inst. Fisheries Eng., Fisheries Res. Agency, Kamisu, Japan), Yong Wang, and Yasushi Nishimori (Furuno Electric Co., Nishinomiya, Japan)

Monitoring fish species and amount with acoustical instruments is a challenging problem to survey fish resources in the ocean. Broadband split-beam echo sounder was useful to observe individual fish behavior in schools. Echoes from fish schools were measured from an anchored vessel for several hours. Echo signals were gathered into one block within a certain depth and a period of time and analyzed to track individual fish in the schools. Target strength was calculated from the individual fish echo and associated with tilt angle which could be estimated by using the tracking result. Feature in each block was statistically calculated by averaging target strengths according to the tilt angles. The blocks of the signals were clustered by K-means method with the features and divided into some clusters. The distribution of the clusters in time and depth was investigated. It appeared that the distributions of the clusters were dependent on both time and depth. Clustering of the features would be effective to monitor diversity of fish in the ocean. [Research supported by JST, CREST.]

4:00

4pAO8. Scattering and reverberation from fish schools in the 500–1500 Hertz band. Arthur Newhall, James Lynch, Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr., Bigelow 213, WHOI, Woods Hole, MA 02543, anewhall@whoi.edu), Thomas Grothues (Inst. of Marine and Coastal Sci., Rutgers Univ., Nw Brunswick, NJ), and Glen Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanogr., Woods Hole, MA)

We report here on the preliminary results from an experiment off Cape Hatteras, North Carolina, to look at acoustic scattering and reverberation from fish schools in the 500–1500 Hz band. The experiment, which was performed during the period May 12–29, 2012, was a joint acoustics, biology and physical oceanography effort, with distinct, but coordinated, goals in each area. Acoustically, our goal was to examine the scattering of sound from fish schools over a full range of azimuthal angles. To do this, we employed a source mounted on an autonomous vehicle and a moored, four element hydrophone array receiver. The source traveled around the fish school and the receiver, giving the desired angular diversity. Biologically, we were interested in mapping and imaging/classifying the fish (both individually and as schools) with the sidescan sonars on the vehicles, and contrasting/verifying this information with video images from high definition cameras attached to the vehicles. Oceanographically, the correlation between the ocean temperature field and the fish species encountered was of first order interest. Results from all three areas will be presented, including some interesting video images, and directions for analysis and further research will be discussed. [Work sponsored by the Office of Naval Research.]

Session 4pBA

Biomedical Acoustics: High-Frequency Ultrasound (20–80 MHz)

Michael Oelze, Chair
UIUC, 405 N Mathews, Urbana, IL 61801

Chair's Introduction—12:55

Invited Papers

1:00

4pBA1. Acoustic and photoacoustic imaging of spheroids. Michael C. Kolios, Elizabeth S. Berndl, Lauren C. Wirtzeld, Eric M. Strohm (Phys., Ryerson Univ., 350 Victoria St., Ontario, Toronto, ON M5B2K3, Canada, mkolios@ryerson.ca), and Gregory J. Czarnota (Med. Biophys., Univ. of Toronto, Toronto, ON, Canada)

Acoustic and photoacoustic high frequency imaging (50–100 MHz) can be used to generate images of cell constructs and spheroids with good spatial resolution and contrast. Here we demonstrate how co-registered acoustic and photoacoustic imaging can be used for imaging spheroids. Spheroids are widely used in cancer research and biology since they emulate a three-dimensional environment such as that experienced in tumors. Spheroids were made by the hanging-drop method using the MCF-7 cancer cell line. To generate photoacoustic contrast, MCF-7 cells were incubated with optical absorbing nanoparticles (e.g., gold nanorods, 780 nm absorption) for 24 h and mixed with native MCF-7 cells prior to spheroid formation. The spheroids were between 0.5 mm and 1mm in diameter. Imaging was performed with the VisualSonics VEVO 770 (25–55 MHz) and a high-resolution SASAM acoustic/photoacoustic microscope for frequencies over 80 MHz (Kibero GmbH, Germany). The spheroid was imaged first using pulse echo ultrasound, then with photoacoustics immediately after. The necrotic core of the spheroid had a 20 dB increase in ultrasound backscatter compared the viable cells surrounding the core, and the ultrasound/photoacoustic images of the spheroid were co-registered showing the distribution of the optical absorbing agents.

1:20

4pBA2. High-frame-rate retrospective imaging of mouse-embryo cardiac function using annular array and Doppler-derived gating. Jeffrey A. Ketterling, Erwan Filoux (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10038, jketterling@riversideresearch.org), Orlando Aristizabal (Skirball Inst. of Biomolecular Medicine, NYU School of Medicine, New York, NY), and Jonathan Mamou (Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY)

A high-frequency (HF) imaging system based on a custom 5-element, 40 MHz annular array has been used to study the cardiovascular development of mouse embryos. High-frame-rate imaging of the heart dynamics was achieved using a retrospective reconstruction method based on Doppler-derived electrocardiogram (ECG) waveforms and respiratory gating was used to suppress motion artifacts. The ECG signals were obtained by measuring blood-flow velocities in major arteries of *in-vivo* mouse embryos using a custom HF Doppler apparatus made from two 20 MHz, single-element, PZT transducers with a Doppler sample volume of 15 mm. Co-registered M-mode data were acquired from the annular array excited with a 5-channel pulser/receiver. A synthetic-focusing (SF) algorithm was used to improve spatial resolution ($<$100 μm), depth-of-field ($>$10 mm) and signal-to-noise ratio ($>$45 dB). This technique was used on embryos aged from 11.5 to 14.5 days and provided high-resolution, morphologically correct B-mode cine-loops of the heart chamber dynamics at frame rates of 1 kHz. The ultra-fine temporal resolution (1 ms) allowed for precise quantification of the mean cardiac cycle length and detailed visualization of fast events such as opening and closing of the mitral valve. The speckle characteristics of the high-resolution images could be used to assess blood flow and to quantify myocardial strain at each developmental stage of the embryonic heart.

1:40

4pBA3. Quantitative ultrasound evaluation of tumor cell death response in locally advanced breast cancer patients to chemotherapy treatment administration. Gregory Czarnota, Ali Sadeghi-Naini, Naum Papanicolau, Omar Falou (Imaging Res. and Radiation Oncology, Sunnybrook Health Sci. Ctr., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, gregory.czarnota@sunnybrook.ca), Rebecca Dent, Sunil Verma, Maureen Trudeau (Medical Oncology, Sunnybrook Health Sci. Ctr. and the Univ. of Toronto, Toronto, ON, Canada), Jean-Francois Boileau (Surgical Oncology, Sunnybrook Health Sci. Ctr. and the Univ. of Toronto, Toronto, ON, Canada), Jacqueline Spayne (Radiation Oncology and Medical Biophys., Univ. of Toronto, Toronto, ON, Canada), Sara Iradji, Ervis Sofroni, Justin Lee (Imaging Res. and Radiation Oncology, Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), Sharon Lemon-Wong (Nursing, Odette Cancer Ctr. and Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), Martin Yaffe (Imaging Res. and Radiation Oncology, Sunnybrook Health Sci. Ctr., Toronto, ON, Canada), and Michael Kolios (Phys., Ryerson Univ., Toronto, ON, Canada)

A clinical study was undertaken investigating the efficacy of ultrasound to quantify cell death in tumor responses with cancer treatment. Patients ($n = 25$) with locally advanced breast cancer received anthracyline and taxane-based chemotherapy treatments over four to six months. The majority of patients went on to have a modified radical mastectomy and correlative whole mount histopathology. Data collection was carried out using an Ultrasonix-RP and an L15-5 6 cm transducer pulsed at 10 MHz with RF data collected five times during neoadjuvant chemotherapy. Data indicated increases of approximately 9 dBr (+/$-$1.67) maximally in ultrasound backscatter in patients who clinically responded to treatment. Patients assessed as responding poorly demonstrated significantly lower increases

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(2.3 +/− 1.7 dBr). Increases in 0-MHz intercept followed similar trends while increases in spectral slope were observed locally from tumor regions demonstrating increases in tissue echogenicity. This study demonstrates the potential of ultrasound to quantify changes in tumors in response to cancer treatment administration in a clinical setting. The results indicate that such responses can be detected early during a course of chemotherapy and should permit ineffective treatments to be changed to more efficacious ones potentially leading to improved treatment outcomes.

2:00

4pBA4. High-frequency quantitative ultrasound approaches for cancer detection in freshly-excised lymph nodes. Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org), Alain Coron (Laboratoire d'Imagerie Paramétrique, CNRS and UMPC, Univ Paris 06, Paris, France), Emi Saegusa-Beecroft (Dept. of Surgery, Kuakini Med. Ctr. and Univ. of Hawaii, Honolulu, HI), Masaki Hata (Dept. of Surgery, Juntendo Med. Ctr., Tokyo, Japan), Michael L. Oelze (Bioacoustics Res. Lab., Univ. of Illinois, Urbana-Champaign, IL), Eugene Yanagihara (Dept. of Surgery, Kuakini Med. Ctr. and Univ. of Hawaii, Honolulu, HI), Tadashi Yamaguchi (Res. Ctr. for Frontier Med. Eng., Chiba Univ., Chiba, Japan), Pascal Laugier (Laboratoire d'Imagerie Paramétrique, CNRS and UMPC, Univ Paris 06, Paris, France), Junji Machi (Dept. of Surgery, Kuakini Med. Ctr. and Univ. of Hawaii, Honolulu, HI), and Ernest J. Feleppa (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY)

Histology performed to assess lymph nodes excised during node-dissection surgeries from cancer patients suffers an unsatisfactory rate of false-negative determinations due to labor and time constraints. In this study, more than 300 lymph nodes were scanned in 3D using a 26-MHz high-frequency ultrasound transducer. Following scanning, individual nodes underwent a special histology procedure that involved step-sectioning each node at 50-μm intervals to guarantee that no significant cancer foci were missed. The 3D radio-frequency ultrasound dataset was analyzed using overlapping 3D regions-of-interests that were individually processed to yield 13 quantitative ultrasound (QUS) estimates associated with tissue microstructure and were hypothesized to show contrast between normal and cancerous regions in lymph nodes. Step-wise linear discriminant analyses were performed to yield an optimal QUS-based classifier. ROC curves and areas under the ROC curves (AUCs) were obtained to assess cancer-detection performance. The AUC for the linear combination of four QUS estimates was 0.83 for a dataset of 110 axillary nodes of breast-cancer patients. Similarly, using five QUS estimates, an AUC of 0.97 was obtained for a dataset of 180 nodes of gastrointestinal-cancer patients. These studies demonstrate that QUS methods may provide an effective tool to guide pathologist towards suspicious regions in lymph nodes.

2:20

4pBA5. Radial shear strain elastography imaging of carotid atherosclerotic plaques in a porcine model. Guy Cloutier, Younes Majdouline, Damien Garcia, Louise Allard, Sophie Lerouge (Lab. of Biorheology and Med. Ultrason., Univ. of Montreal Hospital, 2099 Alexandre de Sève, Montreal, QC H2L 2W5, Canada, guy.cloutier@umontreal.ca), Jacques Ohayon (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Grenoble, France), and Gilles Soulez (Lab. of Biorheology and Medical Ultrason., Univ. of Montreal Hospital, Montreal, QC, Canada)

The objective is to show the feasibility of shear strain elastography (SSE) *in vivo* with intravascular ultrasound (IVUS) radio-frequency data acquired at 20 MHz in carotid arteries of pigs. We previously proposed the Lagrangian Speckle Model Estimator (LSME) to estimate the strain tensor including the shear strain that could be involved in atherosclerotic plaque hemorrhage, inflammation, and rupture mechanisms. However, the LSME performance to compute SSE had never been validated. Atherosclerotic pigs with significant plaques on carotids were studied. To induce atherosclerosis, pigs were put on an atherogenic diet and partial ligations of common carotid arteries were performed. Diabetes was induced by selective intra-arterial injection of streptozotocin. IVUS acquisitions were performed before sacrifice and histologic analysis were realized on fixed dissected carotids. SSE maps at end diastole were estimated with a new implementation of the LSME and matching with histology sections was realized. SSE clearly identified all plaques; reported figures show SSE mapping characterized by cohabitation of high positive and high negative shear values in the specific region of the plaque. This study demonstrates the performance of the LSME implementation to estimate accurately the shear strain distribution, and the feasibility of SSE to highlight atherosclerotic plaque vulnerability characteristics.

2:40

4pBA6. A method to validate quantitative high-frequency power Doppler ultrasound with fluorescence *in vivo* video microscopy. James C. Lacefield (Dept. of Med. Biophys., Western Univ., Thompson Engineering Bldg., Rm. 279, London, ON N6A 5B9, Canada, jlacefie@uwo.ca), Stephen Z. Pinter (Biomedical Eng. Graduate Program, Western Univ., London, ON, Canada), Dae-Ro Kim (Dept. of Med. Biophys., Western Univ., London, ON, Canada), M. Nicole Hague (London Regional Cancer Program, London, ON, Canada), Ian C. MacDonald (Dept. of Med. Biophys., Western Univ., London, ON, Canada), and Ann F. Chambers (London Regional Cancer Program, London, ON, Canada)

Flow quantification with high-frequency power Doppler ultrasound can be performed using the wall-filter selection curve (WFSC) method [Elfarnawany *et al.*, Ultrasound Med. Biol. **38**, 1429–1439 (2012)]. The WFSC method plots color pixel density (CPD) as a function of wall filter cut-off velocity as a means of objectively selecting an operating point cut-off velocity. In this study, an *in vivo* video microscopy (IVVM) system was used to measure the size of small (140–400 μm diameter) mouse testicular vessels immediately after the vessels were imaged with 30 MHz power Doppler. The mouse remained on the same platform throughout ultrasound and IVVM imaging. Measurements in four image planes from three mice demonstrated that, similar to previously reported flow-phantom data, *in vivo* WFSCs exhibit distinct, sloped "characteristic intervals" at cut-off velocities where the CPD approaches the gold-standard IVVM estimate of vascular volume fraction. A wide range of operating point cut-off velocities (4.5 to 12 mm/s) was obtained, which indicates that use of a predetermined cut-off can produce substantial errors in cross-sectional studies that employ power Doppler to quantify vascularity. The WFSC method is a promising strategy for adapting the cut-off velocity to intersubject and longitudinal variations in blood flow during microvascular imaging experiments.

3:00–3:20 Break

3:20

4pBA7. Determining breast pathology in surgical margins with high-frequency ultrasound: phantom and numerical simulations. Timothy E. Doyle, Monica Cervantes (Physics, Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Laurel A. Thompson (Chemistry, Utah Valley Univ., Orem, UT), Joseph E. Roring, Matthew A. Grover (Physics, Utah Valley Univ., Orem, UT), J. Andrew Chappell, Bradley J. Curtis, Janeese E. Stiles, and Brett D. Borget (Biology, Utah Valley Univ., Orem, UT)

Two parameters in high-frequency ultrasound (20–80 MHz) have been found to be sensitive to a range of pathologies in resected margins from breast conservation surgery: The number of peaks (the peak density) in the waveform spectrum and the slope of the Fourier transform of the waveform spectrum. Previous studies have indicated that peak density and slope may correlate to microscopic heterogeneity in tissue structure, which is modified by atypical and malignant processes. To test this hypothesis, through-transmission and pulse-echo measurements were acquired from gelatin-based phantoms containing polyethylene microspheres and nylon fibers (2.5–10% volume concentration). Multipole methods were also used to model through-transmission measurements of tumor progression in lobular carcinoma *in situ*. The simulated breast tissue contained 1000–2000 nucleated cells with random lobular cavities. The peak densities of the heterogeneous phantoms were significantly greater than those of the homogeneous control samples, whereas the slopes were less. Similarly, the models produced spectra with peak densities that increased with malignant cell proliferation. The results are consistent with breast tissue data, and provide a physical mechanism for the use of peak density and slope in the imaging of breast tissues with atypical and malignant pathologies. [Work supported by Utah Valley University.]

3:40

4pBA8. Molecular profiling of breast cancer using high-frequency ultrasound. Timothy E. Doyle (Physics, Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu), Janice E. Sugiyama, Bradley J. Curtis, Mandy H. Marvel, Marcus J. Payne, and Janeese E. Stiles (Biology, Utah Valley Univ., Orem, UT)

In addition to the traditional classifications based on histopathology, breast cancer can also be classified by gene expression profiling into five molecular subtypes that have been found to be more predictive of patient prognosis and treatment response. The purpose of this study was to determine if high-frequency (HF) ultrasound (20–80 MHz) is sensitive to the molecular subtypes of breast cancer, and can differentiate between the more aggressive subtypes such as triple-negative and Her2+ from the less aggressive, more treatable subtypes. Recently, mutations associated with triple-negative and Her2+ have been discovered that are associated with the actin cytoskeleton, extracellular matrix (ECM), and integrin signaling. These mutations may alter the biomechanical and thus ultrasonic properties of tumor cells. This hypothesis was tested using both numerical and cell culture studies. Multipole expansions were used to simulate micro-level ultrasonic scattering from malignant cells with a range of cytoskeletal and ECM properties. Modest property changes produced large variations in simulated spectra. Cell lines of different molecular subtypes were also cultured as monolayers and tested with HF ultrasound. Results from the cell culture tests and their correlation to the models will be discussed. [Work supported by a Utah Valley University Presidential Fellowship Award.]

4:00

4pBA9. Hepatitis fibrosis characterization by a multiparametric study. Mahmoud Meziri, Razika Bouzitoune (Laboratoire LM2S, Université, BP, 790, RP, Annaba 23000, Algeria, mahmoud.meziri@gmail.com), Naamane Remita (Physique, Université, Skikda, Algeria), Christiano Machado, Wagner.C. A Pereira (COPPE/UFRJ, Université, Rio de Janeiro, Brazil), and Frédéric Padilla (Inserm, U1032, LabTau, Université, Lyon, France)

The ultrasonic tissue characterization (UTC) is primarily based on radio-frequency (RF) signals' analysis. The processing of these signals allowed the estimation of different quantitative ultrasonic parameters (backscattered coefficients—ICB-, velocity—SoS, etc). It is known that the RF contains information that can be used to noninvasively characterize the structural and mechanical properties of tissue. Scatterer diameter (or size) from ICB measurements have been already used to discriminate fibrosis from normal tissue. From these findings, our goal was to evaluate the scatterer diameter to test its potential in the discrimination of fibrosis groups (F0, F1, F2, F3, and F4, METAVIR scale) from 20 *in-vitro* human liver samples, explored at 20 MHz. The mean scatterer diameters (μm) measured were: 42.14 □ 4.90 (F0), 40.18 □ 10.51 (F1), 38.82 □ 6.05 (F3), and 40.30 □ 2.22 (F4). The Kolmogorov-Smirnov test has shown a non-significant level ($p > 0.05$) indicating that the scatterer size estimation alone cannot differentiate between all fibrosis groups; an obvious overlap between groups appears. However, for the two different combinations (ICB, Size) and (ICB, Size, SoS), the discriminant analysis has correctly classified 75% and 85%, respectively, of liver samples at a significant level ($p < 0.00005$). The multiparametric study could play an important role to aid in the diagnostic of liver fibrosis.

4:20

4pBA10. Probability distribution variation in high-frequency ultrasound blood echogenicity under *in-vitro* and *in-vivo* blood flow. Tae-Hoon Bok, Kweon-Ho Nam, Dong-Guk Paeng, and Juho Kim (Ocean System Eng., Jeju National Univ., 102 Jejudaehakno, Jeju 690-756, South Korea, bth012@jejunu.ac.kr)

The dynamic phenomena of erythrocyte aggregation (EA) need to be analyzed statistically since EA varies spatially and temporally. In the present study, the cross-sectional B-mode images were acquired from a mock circulatory system with varying blood flow velocity under steady flow, and the human radial artery using an ultrasound biomicroscopy system at 20 MHz. The kurtosis (K) and skewness (S) coefficients, and the Nakagami parameter (m) were computed for each image. For the *in-vitro* experiment, both K and S increased about 0.87 ± 0.18 and 0.63 ± 0.09, respectively; while m decreased about 0.90 ± 0.20 with increasing blood velocity from 12 to 44 cm/s. *In-vivo* experimental results also showed that K, S, and m varied during a cycle. When the blood velocity varied from 5 to 15 cm/s during a cardiac cycle, K and S increased about 1.42 ± 0.64 and 0.44 ± 0.11, respectively; while m decreased about 0.97 ± 0.26. The *in-vivo* results seemed to be consistent with the *in-vitro* results in the sense that K and S increased with blood velocity while m decreased with velocity. This study suggests that the statistical analysis of blood echogenicity can be useful for *in-vivo* hemorheology and blood characterization. [Work supported by NRF-2012-0005005 and NIPA-2012-H0401-12-2006.]

4:40

4pBA11. Improvement of an intravascular ultrasound elasticity modulus imaging approach for detecting vulnerable atherosclerotic plaques. Zahra Keshavarz-Motamed (Lab. of Biorheology and Med. Ultrason., Univ. of Montreal HospitalRes. Ctr., Montreal, QC, Canada, zahra.keshavarz@crchum.qc.ca), Simon Le Floc'h, Jacques Ohayon (Lab. TIMC-IMAG-UJF-CNRS UMR 5525, Univ. Joseph-Fourier, Grenoble, France), and Guy Cloutier (Lab. of Biorheology and Med. Ultrason., Univ. of Montreal Hospital, Montreal, QC, Canada)

Atherosclerotic plaque rupture is the major cause of acute coronary syndrome, myocardial infarction, and stroke in the western world. Stress concentration is recognized to be a good indicator of vulnerable plaques (VP). The Lagrangian speckle model estimator (LSME) for vascular ultrasound elastography, developed by our group, provides the strain field within the plaque. However, evaluation of the stress field relies on a precise identification of the mechanical properties of plaque components. As a response to this need, our group recently developed an approach called imaging modulography (iMOD). iMOD uses a continuum-mechanics-based segmentation method and the inverse finite-element method to reconstruct elasticity maps (or modulograms) of atheroma plaques based on the radial strain field calculated by the LSME. The present theoretical study was designed to further develop segmentation and optimization procedures of iMOD to incorporate both radial and shear components of the strain tensor. Simulated IVUS images of coronary lesions with

known material properties and known stress fields were used to validate the new iMOD algorithm and assess its robustness and performance in detection and quantification of VPs. The results demonstrate promising benefits of the new optimized iMOD-LSME clinical imaging method for VP detection.

5:00

4pBA12. Acoustical imaging of internal spheroid structures at a variety of frequencies. Elizabeth S. Berndl and Michael C. Kolios (Physics, Ryerson Univ., 350 Victoria St., Toronto, ON M5N 2K3, Canada, eberndl@ryerson.ca)

We have previously shown that ultrasound is capable of identifying apoptosis in individual cells and cell pellets due to changes in the ultrasound attenuation, speed of sound, and backscatter. Spheroids can be used to more accurately model non-vascularized tumors due to their three-dimensional growth pattern, cell-cell interaction, disorganized growth, and development of a necrotic core when grown to sufficiently large sizes. To examine cell death in spheroids due to necrosis or chemotherapy, quantitative ultrasound methods were used on the ultrasound backscatter power spectrum throughout the spheroid. MCF7 spheroids ranging in size from 100 to 1000 μm were probed at 25 and 55 MHz using a VEVO770 high frequency ultrasound machine, and at 80 and 200 MHz using an acoustic microscope. Changes in spheroid structure as the necrotic core develops, and after it is exposed to chemotherapeutic agents were recorded and analyzed. An increase in the ultrasound backscatter amplitude from necrotic cells within the core of the spheroid versus the viable cells around the core was observed. Changes in the ultrasound backscatter were also observed for spheroids treated with chemotherapeutics to induce apoptosis. This work furthers our understanding of non-invasively identifying the viability of cancerous tumors, and the efficacy of chemotherapeutic treatments.

THURSDAY AFTERNOON, 6 JUNE 2013 512AE, 1:00 P.M. TO 5:20 P.M.

Session 4pEAa

Engineering Acoustics: Sound Field Control in the Ear Canal

Pablo Hoffmann, Cochair
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Janina Fels, Cochair
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Invited Papers

1:00

4pEAa1. Individual *in-situ* calibration of insert headphones. Marko Hiipakka and Ville Pulkki (Aalto Univ., Otakaari 5 A, Espoo 02150, Finland, Marko.Hiipakka@aalto.fi)

An important procedure in binaural reproduction is the calibration of headphones, which is commonly achieved by first measuring the headphone transfer functions (HpTFs). The commonly used methods of measuring the HpTF are not applicable for insert headphones, since the inserts block the ear canal entrance and since the transducer ports of the inserts are inside the ear canals. Recently, an alternative technique of obtaining HpTFs of inserts using measurements with in-ear microphones, computational modeling, and electro-acoustic Norton-type source models of the inserts has been proposed. In this study, the technique is evaluated using measurements at the eardrums of eight human subjects and computational modeling with normal human ear canal parameters. In addition, different methods of obtaining the electro-acoustic source model parameters are compared. It is shown that the most reliable method of obtaining the Norton source parameters of insert headphones is through measurements with a miniature-sized particle velocity sensor and several tubes with different cross-sectional diameters as acoustic loads. The evaluations show that the proposed technique of obtaining the HpTFs of insert headphones is accurate and reliable at least up to 8 kHz, which bolsters the applicability of the technique for individual *in-situ* calibration in binaural reproduction.

1:20

4pEAa2. Sound transmission in a simple model of the ear canal and tympanic membrane. Antonio Gonzalez-Herrera (Civil Eng., Univ. of Malaga, Calle Diego de Siloe, 29013 Malaga, Malaga, Spain, agh.uma.es@gmail.com), Kapil Wattamwar (Biomedical Eng., Columbia Univ., New York, NY), Christopher Bergevin (Phys. and Astronomy, York Univ., Toronto, ON, Canada), and Elizabeth S. Olson (OTO/HNS & Biomed. Eng., Columbia Univ., New York, NY)

The classic picture of middle ear (ME) transmission has the tympanic membrane (TM) as a piston and the ME space as a vacuum. However, the TM moves in a complex wavy pattern and modern theories link this behavior to the ME's broadband transmission [e.g., Fay *et al*, Proc. Natl. Acad. Sci. U.S.A. (2006), Parent and Allen, J. Acoust. Soc. Am. (2007)]. Furthermore, Rabbitt [J. Acoust. Soc. Am. (1990)] predicted that the TM radiated considerable sound pressure into the ME space that could in return affect TM motion. This study explores these ideas with a simple model, using a tube terminated with a plastic membrane to model the EC and TM. We measured membrane motion via laser interferometry, and pressure (on both sides) via micro-sensors that allowed positioning very close to the membrane (10 micrometers) without disturbance. We made a finite element model of the system to complement the experimental results. The theoretical results show the interaction of acoustic and mechanical resonances, and both theoretical and experimental results show strong transmission of sound through the membrane at some of its resonant frequencies.

1:40

4pEAa3. Pole-zero modeling of the middle ear based on acoustic reflectance measurements. Sarah Robinson and Jont Allen (Elec. Eng., Univ. of Illinois at Urbana-Champaign, 209 N Coler Ave, APT 2, Urbana, IL 61801, srrobin2@illinois.edu)

Fitting poles and zeros to complex acoustic reflectance (CAR) data using a "rational approximation method" [Gustavsen and Semlyen (1999)] allows for a precise parameterization of complex real-ear measurements. CAR is measured using a foam-tipped probe sealed in the ear canal, containing a microphone and receiver (i.e., MEPA3 system, Mimosa Acoustics). From the complex pressure response to a broadband stimulus, the acoustic impedance and reflectance of the middle ear can be calculated as functions of frequency. The goal of this work is to establish a quantitative connection between the fitted pole-zero locations and underlying physical properties of the CAR and impedance of the middle ear. It was found that (1) the contribution of the ear canal may be approximated as the lossless all-pass component of the factored reflectance fit, (2) individual CAR magnitude variations for normal middle ears in the 1 to 4 kHz range give rise to closely placed pole-zero pairs, and (3) the locations of the poles and zeros in the s-plane may differ between normal and pathological middle ears. Pole-zero fitting allows for concise characterization of individual CAR measurements, providing a foundation for modeling individual and pathological variations of middle ears.

2:00

4pEAa4. Mitigation of excessive acoustic compliance and trapped volume insertion gain in ear-sealing listening devices: Toward a safe, full-frequency-response hearing aid. Stephen D. Ambrose and Samuel P. Gido (Asius Technologies LLC, 1257 Whitehall Dr., Longmont, CO 80504, stephen.ambrose@asiustechnologies.com)

When a sound producing device is sealed in the ear canal, acoustical compliances resulting from pressurization of the trapped volume lead to dramatic boosts in SPL, up to 60 dB, especially at low frequencies. This has been found to result in listener fatigue, and to trigger the acoustic (stapedius) reflex, as well as producing temporary threshold shift. Repeated exposure can cause temporary threshold shift to become permanent. Hearing aids avoid this problem by suppressing frequencies below about 300 Hz, where the effect is most pronounced. Other devices, such as ear buds and professional in-ear monitors, offer wider frequency response and thus expose listeners to potentially dangerous sound pressures. The acoustical compliance and trapped volume insertion gain is measured for ear buds and hearing aids by comparing SPL, measured in the ear canal, for sealed and unsealed conditions. New ear sealing technology is demonstrated that allows release of the excess acoustical compliance and thus mitigates the trapped volume insertion gain: (1) a vent covered with a flexible membrane, and (2) an inflatable bubble seal. This novel technology has allowed the creation of a hybrid device with hearing aid functionality that also has the broad frequency response of professional in-ear monitors.

2:20

4pEAa5. A comparison of methods for measuring the acoustic input impedance of ear canals for hearing aid applications. Tobias Sankowsky-Rothe, Simon Köhler, Matthias Blau (Institut für Hörtechnik und Audiologie, Jade Hochschule WOE, Ofener Straße 16-19, Oldenburg, Niedersachsen 26121, Germany, Tobias.Sankowsky@jade-hs.de), and Alfred Stirnemann (Adv. Products, Phonak AG, Stäfa, Switzerland)

In hearing aid fitting the sound pressure at the ear drum is a reference quantity, since all *real ear* characteristic values refer to it. Typically, the sound pressure at the ear drum is estimated by a model of an average ear canal (e.g., a coupler). Such a model cannot account for inter-individual differences. Alternatively, there are methods to predict the acoustics of the individual ear canal. Some of these methods make use of the acoustic input impedance of the ear canal. In general, the accuracy of the measured impedance depends on the effort that will be made. Therefore, different methods of impedance measurements were investigated concerning accuracy and effort. The methods differ in the number of calibration measurements (and calibration parameters). They were compared on the basis of impedance measurements on different model ear canals. Measurements were done with an impedance probe consisting of a typical hearing aid receiver and a hearing aid microphone. The measurements were compared to measurements with a reference impedance probe and method. As a result, it was observed that with a single calibration measurement the maximum absolute error of the transfer impedance was smaller than 3 dB up to 8 kHz.

Contributed Paper

2:40

4pEAa6. A comparison of methods for estimating individual real-ear-to-coupler-differences in hearing aid fitting. Simon Köhler, Tobias Sankowsky-Rothe, Matthias Blau (Institut für Hörtechnik und Audiologie, Jade Hochschule WOE, Ofener Str. 16/19, Oldenburg 26121, Germany, simon.koehler@jade-hs.de), and Alfred Stirnemann (Adv. Products, Phonak AG, Stäfa, Switzerland)

The sound pressure at the ear drum is *the* reference quantity for almost all applications of sound delivery to the ear, especially in hearing aid fitting. Since hearing aids are typically calibrated using the so called 2cc-coupler, the link to the individual sound pressure at the ear drum is given by the *real-ear-to-coupler-difference* (RECD). Nowadays, averaged RECDs are used for hearing aid fitting, which do not account for inter-individual differences in ear canal acoustics. As a consequence, resulting coupling errors may reach 15 dB for frequencies up to 10 kHz. Alternatively, there are methods for estimating individual RECDs, based on acoustic impedance measurements at the inner face of the ear mold. These methods differ in effort (e.g., the complexity of the ear canal model and fitting algorithm) and accuracy. By using an integrated ear canal microphone, individual RECD estimation could be feasible in future hearing aid fitting. In this research, six different methods to predict individual RECDs were compared using simulations as well as real ear measurements with open and closed ear molds. As a result, it appeared that relatively simple cylindrical and conical ear canal models give the best compromise between effort and accuracy.

Invited Paper

3:00

4pEAa7. Using inter-individual standard deviation of hearing thresholds as a criterion to compare methods aimed at quantifying the acoustic input to the human auditory system in occluded ear scenarios. Matthias Blau, Tobias Sankowsky-Rothe, Simon Köhler (Institut für Hörtechnik und Audiologie, Jade Hochschule Wilhelmshaven/Oldenburg/Elsfleth, Ofener Str. 16/19, Oldenburg D-26121, Germany, matthias.blau@jade-hs.de), and Jan-Henning Schmidt (Physikalisch-Technische Bundesanstalt, Braunschweig, Germany)

Occluded ear scenarios are found in many applications, e.g., hearing aids or insert ear phones. Unfortunately, the correct quantification of the acoustic input delivered to the auditory system in such a scenario is complicated by the individual character of our outer ear anatomy. For instance, one can easily observe inter-individual differences in ear drum pressure level of up to 30 dB at 10 kHz with one and the same sound source. We may thus ask: (1) Is the sensitivity of our auditory system at threshold adapted to our outer ear anatomy? and (2) what is the best method to quantify the acoustic input? We propose to use the inter-individual standard deviation of hearing thresholds as a means to answer these questions: The quantity that is best suited to describe the input to the auditory system should result in the lowest inter-individual standard deviation of thresholds. Preliminary results based on tests with custom ear shells and with foam ear plugs show that up to 6 kHz, there are no significant differences between the methods tested, whereas in the 6–9 kHz frequency range, individual estimates of the sound pressure at the ear drum yield a significantly lower inter-individual standard deviation than, e.g., the ear simulator pressure.

Contributed Papers

3:20

4pEAa8. Acoustics of enclosed spaces: The differences when the dimensions are millimeters vs. tens of meters. Martin Kuster (Sci. & Technol., Phonak AG, Laubisrütistr. 28, Stäfa 8712, Switzerland, kuster_martin@hotmail.com)

The dimensions of the ear canal are at least 3 orders of magnitude smaller than those typically encountered in room acoustics but at the same time the range of wavelengths for audio applications is identical. This results in a disparity not only in length scale but also a disparity in time scale. The influence of these disparities on well-known room acoustics parameters or features such as the reflection density, the direct-to-reverberant ratio, the critical distance, or transfer function nulls is reviewed and highlighted. The nature of the two substantially different sound fields is also important for active sound control. Consequently, the respective relevance of total absorption as well as values of source and sink impedance are also compared.

3:40

4pEAa9. Effect of the middle ear cavity on the response of the human auditory system. Antonio Garcia-Gonzalez and Antonio Gonzalez-Herrera (Civil Eng., Univ. of Malaga, AVDA. SALVADOR ALLENDE, 322, MALAGA, MALAGA 29017, Spain, AGH@UMA.ES)

The effect of the acoustic cavities on the response of the auditory system has been usually focused on the influence of the external ear canal (EEC). The presence of the middle ear cavity (MEC) has been ignored. Experimental difficulties to obtain information inside this cavity without altering the whole system make difficult its study. In order to explore the influence of this cavity, a numerical study is made. This is made by means of a complete finite element (FE) model including the tympanic membrane, ossicular chain, and acoustic cavities. Different FE models are used to analyze the influence of each component. By means of different calculations removing these components from the model, their relative effects can be distinguished. At low frequencies (below 2 kHz) the influence of the MEC is negligible. Piston-like motion is dominant. Nevertheless, at higher frequencies a new resonant peak appears at a frequency of 4 kHz. This is due to the presence of the MEC. It combine with the pressure gain due to the ear canal (at a frequency of 3 kHz) increasing the response of the system in terms of Umbo velocity. This effect is observed in different published experimental results.

4:00

4pEAa10. Estimation of ideal open-cavity middle-ear responses from responses with partial cavity opening. Nima Maftoon (BioMedical Eng., McGill Univ., 3775, rue University, Montréal, QC H3A 2B4, Canada, nima.maftoon@mail.mcgill.ca), W. Robert J. Funnell (BioMedical Eng. and Otolaryngol. – Head & Neck Surgery, McGill Univ., Montréal, QC, Canada), and Sam J. Daniel (Paediatric Surgery and Otolaryngol. – Head & Neck Surgery, McGill Univ., Montréal, QC, Canada)

An important step in developing mathematical models of the middle ear is the validation of simplified models without the middle-ear air cavity. However, open-cavity experimental results are often collected with only a partial opening of the middle-ear cavity, due to experimental limitations. The partial opening introduces an anti-resonance that obscures features of the frequency response in its neighborhood. In this study, we suggest a numerical method for estimating ideal open-cavity responses from experimental results with partial openings. We fit rational-fraction polynomials to portions of the response in order to parametrically identify the transfer function associated with the anti-resonance. The ideal open-cavity response is then estimated by dividing the experimentally measured frequency response by the identified anti-resonance transfer function. The method has been validated against synthesized transfer functions with features similar to those caused by partial opening of the cavity and against responses calculated using models of the middle ear with a partially open cavity.

4:20

4pEAa11. Finite-element modeling of the newborn ear canal and middle ear. Hamid Motallebzadeh, Brian Gariepy, Nima Maftoon (BioMedical Eng. Dept., McGill Univ., 3775, rue University, Rm. 303, Montreal, QC H3A 2B4, Canada, hamid.motallebzadeh@mail.mcgill.ca), W. Robert J. Funnell (BioMedical Eng. and Otolaryngol. – Head & Neck Surgery, McGill Univ., Montreal, QC, Canada), and Sam J. Daniel (Paediatric Surgery and Otolaryngol. – Head & Neck Surgery, McGill Univ., Montreal, QC, Canada)

Available hearing-screening procedures cannot distinguish clearly between conductive and sensorineural hearing loss in newborns, and the results of available diagnostic tests in very young infants are difficult to interpret. Admittance measurements can help to detect conductive losses but do not provide reliable results for newborns, where the ear is anatomically different from the adult ear. Finite-element models of the newborn ear canal and middle ear were developed and their responses were studied for frequencies up to 2000 Hz. Material properties were taken from previous measurements and estimates, and the sensitivities of the models to these different parameters were examined. The simulation results were validated through comparison with previous experimental measurements. Simulations indicate that at frequencies up to 250 Hz the admittance of the canal wall is comparable to that of the middle ear in the newborn. Above 250 Hz, the canal-wall admittance remains almost constant but for the middle ear there is a clearly defined resonance peak, which produces an admittance much larger than that of the canal wall. These results suggest that admittance measurements in the vicinity of the middle-ear resonance frequency can provide clinically useful information about the newborn middle ear.

4:40

4pEAa12. Harmonic hydromechanical movement. Santos Tieso, Francisco Messina, Lucas V. Fantini, Nicolás Casco Richiedei, Nahuel Cacavelos, Ignacio Talento, Nicolás Vallese, Rodrigo Fernández, and Adrián Saavedra (Ciencia y Tecnología, Universidad Nacional de Tres de Febrero, zapiola, Mendoza, Capital federal 1428, Argentina, francisco.messina@gmail.com)

The current theoretical physics tools used to describe the acoustic phenomena that occur outside the ear, such as the specific impedance, the acoustic impedance and the mechanical impedance are not applicable to describe the cochlear mechanics. For this reason, this study uses the hydro-mechanical impedance concept. The latter is only applicable to a harmonichydro-mechanical systems, which consist of a rigid recipient, filled with liquid and two elastic windows that relate

the system with a sound environment, considering that the distance between them should be much smaller than the sound wavelength. This system could be considered as the most primitive model inner ear to build. The movement of the contained fluid in this system has particular characteristics that differentiate it from the wave motion and from a simple mass-spring-dumping vibration system. In order to demonstrate the existence of the harmonic hydro-mechanical movement, was modeled and built an equivalent harmonic electrical system, which results corresponded with the ones from the theoretical mathematical model.

5:00

4pEAa13. Secondary path variation in human listeners and its effect on an active noise cancellation system. Jinjun Xiao, Buye Xu, and Tao Zhang (Starkey Hearing Technologies, 6600 Washington Ave. S., Eden Prairie, MN 55344, jinjun_xiao@starkey.com)

Active noise cancellation (ANC) has been applied to cancel the penetrated ambient noise in the ear canal for hearing impaired listeners [Zhang *et al.* (2012)]. The performance of the proposed ANC system depends on the characteristics of the secondary path (SP). In this study, we developed an in-ear ANC system where the error microphone and the miniature loudspeaker were both placed in the ear canal. In such a case, the SP response depends on the error microphone response, the loudspeaker response, and how the microphone, the loudspeaker, and the ear canal are acoustically coupled. A robust method was proposed to measure the SP response of the proposed ANC system under various coupling conditions using human listeners. Variations of the measured SP responses were analyzed both within each individual and across different individuals. The effect of the SP variations on the performance of the proposed ANC system was evaluated. The implications for improving the proposed ANC system will be discussed.

THURSDAY AFTERNOON, 6 JUNE 2013 512BF, 1:00 P.M. TO 5:20 P.M.

Session 4pEAb

Engineering Acoustics: Fields and Devices

Daniel Armstrong, Chair
Mech. Eng., McGill Univ., MacDonald Eng. Bldg., Rm. 364, 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada

Contributed Papers

1:00

4pEAb1. Collocation analysis of junction conditions for waveguides at high-frequencies. Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338-2854, jerry.ginsberg@me.gatech.edu)

Analyses of high-frequency transmission and scattering at junctions of waveguide segments having different cross-sectional area must account for the generation, reflection, and transmission of higher order modes. The usual analysis relies on modal orthogonality properties, which are awkward to implement if a junction does not have a regular geometrical configuration. An alternative formulation uses a collocation procedure, in which continuity and boundary conditions at the interface are enforced exactly at a set of discrete points. The viability of that approach was examined recently [J. Acoust. Soc. Am. **132**, 1955 (2012)], but time restrictions only permitted assessment of the technique for a waveguide whose end is excited by a discontinuous velocity distribution. It was found that the convergence and computational requirements of the collocation method are essentially the same as those of an analysis that uses modal orthogonality. The present work extends the formulation to address situations where waveguide segments having different cross-sections are joined. A least squares solution of the junction equations for the modal coefficients is described and illustrated by an example.

1:20

4pEAb2. Inspectability of interfaces between composite and metallic layers using ultrasonic interface waves. Michael D. Gardner (Grad. Program in Acoust., The Penn State Univ., 225 Strouse Ave., State College, PA 16803, gardnerfrance@gmail.com), Joseph L. Rose (Dept. of Eng. Sci. and Mech., The Penn State Univ., University Park, PA), Kevin L. Koudela, and Clark A. Moose (Appl. Res. Lab., The Penn State Univ., University Park, PA)

The interface between an anisotropic composite material and a metallic material is inspected non-destructively for disbonded regions using ultrasonic guided waves. The material properties of the composite and metal have been tailored to demonstrate their effect on inspectability. The material properties have been designed to be either favorable or unfavorable to the existence of propagating Stoneley waves. Stoneley waves can exist because the layer thicknesses are large enough compared to the wavelength to be considered half-spaces. The existence of Stoneley waves between generally anisotropic materials depends on the elastic constants and densities in a complicated way. The range of material properties that allow Stoneley waves is small; however, when the vertically polarized shear wave speeds are similar in the two materials, the existence of Stoneley waves is generally possible. If the conditions do not strictly allow Stoneley waves, other interface waves can still exist such as leaky waves. Disbonds are inserted into the materials before bonding and are inspected using interface waves. Sensitivity to disbonds is determined and thus inspectability is demonstrated for cases that are favorable and unfavorable to Stoneley waves. Both numerical and physical experimental results are shown.

1:40

4pEAb3. Case studies of casing inspection with multi-functional ultrasonic imaging logging tool. Zhifeng Sun, Honghai Chen, and Xien Liu (WellTech, China Oilfield Services Ltd., G.P.O. Box 232, Beijing 101149, China, sunzf@cosl.com.cn)

The multi-functional ultrasonic imaging logging tool can provide casing inspection and cement bonding evaluation by using ultrasonic pulse echo technique. The casing's internal conditions can be inspected from echo amplitude and transit time curves. The casing thickness and the cement impedance can be calculated from resonance frequency and resonance decay. The paper describes three well cases histories of casing inspection in detail with this instrument. The first case reports the evaluation of perforation interval in the cased hole; hole enlargement of perforation interval and perforations can be seen from echo amplitude and casing internal radius imaging curves. In the second case, we cannot distinguish corrosion or deposits in the casing surface from the echo amplitude and transit time curves, but casing thickness imaging curve characterize some slight corrosion. Finally, the last case history describes evaluation of casing mechanical wear in a deviated well; casing thickness imaging curve shows mechanical wear is caused by logging

instrument in low side well. All cases shows that the multi-functional ultrasonic imaging logging tool can provide both quantitative and qualitative evaluation and diagnosis of casing problems.

2:00

4pEAb4. Enhancing noise control in a cavity using Helmholtz resonators. Ch. Surya Narayana V. Reddi and Chandramouli Padmanabhan (Mech. Eng., Indian Inst. of Technol. Madras, 406 Machine Design Section, Chennai, Tamilnadu 600036, India, mouli@iitm.ac.in)

In this paper, it is shown that there is a decrease in sound levels, not only when a resonator is tuned exactly to a cavity mode, but also when it is tuned to a frequency slightly different from the natural frequency of the chosen cavity mode. The finite element (FE) method is used for numerical modeling of the coupled Helmholtz resonator-cavity system. To validate the FE model prediction, a boundary element (BE) analysis is performed by specifying an impedance boundary condition on the element where the resonator is mounted. This impedance has been calculated, from a BE model of the resonator alone, using a plane wave excitation, over a range of frequencies around the cavity mode of interest. Numerical experiments have been performed in a cavity with two close modes (155 and 173 Hz) and it is observed that when a resonator is tuned to a frequency slightly lower than the first cavity mode, the performance of the resonator is much better than when it is tuned exactly to the first cavity mode or tuned to a slightly higher frequency. A detailed parametric study has been carried out and guidelines for tuning resonators for superior noise control is proposed.

2:20

4pEAb5. Acoustical impedance characterization of liners using a Bayesian approach. Yorick Buot de l'Epine, Jean-Daniel Chazot, and Jean-Michel Ville (CNRS UMR6253 Roberval, Université Technologique de Compiègne, Centre de Recherche Royallieu, BP20529, Compiègne 60205, France, ybuotdel@utc.fr)

Acoustic liners composed of perforated plates and honeycomb layers are used in several applications. These liners are used for example in aerospace as acoustic treatments for aircraft nacelles. To describe their behavior, empirical models or standing wave tube experiments can be employed. However, the resulting impedance is not always accurate to describe the real behavior of the liner when submitted to several planes waves at various angles (higher order modes in the duct), or when submitted to a grazing flow. In this paper, an inverse method based on a Bayesian approach is presented in order to characterize acoustic liners in real conditions. An analytical solution and experimental data are used to calculate the likelihood function of the estimated impedance. The posterior probability density function can then be obtained by adding the prior information. Finally, an evolutionary Markov Chain Monte Carlo method (eMCMC) is implemented to explore the probability density space. This inverse method is first validated on simulated data. Then experimental data are used.

2:40

4pEAb6. Cylindrical cyclic acoustic imaging with a Bayesian approach for cyclostationary sources reconstruction. Sebastien Personne (Laboratoire Roberval CNRS, UTC, BP 20529 cedex, Compiegne 60205, France, sebastien.personne@hotmail.fr), Jerome Antoni (LVA, INSA, Lyon, France), and Jean Daniel Chazot (Laboratoire Roberval CNRS, UTC, Compiegne, France)

Standard acoustic imaging techniques, such as beamforming or near acoustical holography, are now widely used in engineering contexts. However, large arrays of microphones are sometimes required to have a good resolution. Besides new challenges arise, particularly in the field of non stationary sources, which need to be identified and solved. Cyclostationary sound sources, a specific kind of non stationary signals, are characterized by statistical properties evolving periodically in time. In practice, the first-order statistical properties contain some periodic components while the second orders may be random with a periodic flow of energy. The present work tackles the acoustic imaging of cyclostationary sources with a scanning microphone, i.e., without any array. Cylindrical surfaces, adapted to standard rotating machines, are considered. The reconstruction difficulty of acoustic sources from discrete measurements is addressed here thanks to the cyclostationary properties. A cyclic sound field is hence extracted from the discrete measurements. Finally, a Bayesian formulation, gathering both physical and probabilistic information on this inverse problem, is used to back propagate the sound over the radiating surface.

3:00

4pEAb7. Pressure mapping system based on guided waves reflection. Nicolas Quaegebeur, Patrice Masson (GAUS - Dept. Mech. Eng., Université de Sherbrooke, 2500 Blvd Universite, Sherbrooke, QC J1K2R1, Canada, nicolas.quaegebeur@usherbrooke.ca), Nicolas Beaudet, and Philippe Sarret (Dept of Physiol. and Biophys., Université de Sherbrooke, Sherbrooke, QC, Canada)

In this paper, guided wave interaction is used to develop a pressure mapping system for medical and touch-screen applications. The principle is based on interaction of guided waves in the presence of an added local mass and in the presence of a local pressure. For this purpose, piezoceramics are used for injecting guided waves into a thin structure and to measure the reflected waves due to the presence of an added mass or pressure. SHM imaging algorithms, based on time-of-flight (EUSR) or correlation (Excitelet), are implemented in order to obtain cartography of the reflections and deduce the presence, localization, and intensity of local contact spots. Analytical and numerical models are first derived to assess the critical parameters in order to maximize the reflection of guided waves (first order modes A0 and S0). It is shown that the sensitivity of the guided waves with respect to an added mass and pressure is highly related to the Young's modulus of the host structure. Validation on a 0.5 mm thick plane aluminum plate prototype is addressed using 4 sensor/actuator pairs. It is observed that imaging of single pressure spot and multiple or extended pressure spots can be achieved using S0 mode around 300 kHz with a resolution of 0.5 mm.

3:20

4pEAb8. Microbubble histogram reconstruction by nonlinear frequency mixing. Matthieu Cavaro (CEA, DEN, Lab. of Instrumentation and Technolog. Test, Cadarache, Bâtiment 202, Saint Paul lez Durance 13108, France, matthieu.cavaro@cea.fr) and Cédric Payan (Aix-Marseille Université, Aix en Provence, France)

In the 4th generation sodium fast nuclear reactors (SFR), different phenomena can lead to gaseous microbubbles presence in the primary liquid sodium pool. This paper investigates the ability of nonlinear acoustics techniques to characterize these microbubbles presence. The goal is here to determine the void fraction (volume fraction of free gas) and the histogram of bubbles radius. Different acoustic techniques are currently developed at CEA. Among others, the nonlinear mixing of two frequencies [V. L. Newhouse and P. M. Shankar, J. Acoust. Soc. Am. **75**(5), 1473–1477 (1984)] is under study. Based on the nonlinear behavior of bubble resonance, this technique allows determining the radius histogram of a bubble cloud. Two different mixing techniques are here presented: the mixing of two high frequencies and the mixing of a high and a low frequency. The first step is an air-water experimental set-up. Microbubbles clouds are generated with a like dissolved air flotation process and an optical device gives us reference measures. Generated bubbles have radii in the range of several microns to several tens of microns. The developed experimental procedure allows us to determine the bubble size's histograms with accuracy never reported yet.

3:40

4pEAb9. Effects of a boundary layer trip in the prediction of noise from flow past tandem cylinders. Daniel Armstrong and Luc Mongeau (Mech. Eng., McGill Univ., MacDonald Eng. Bldg., Rm. 364, 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada, daniel.armstrong2@mail.mcgill.ca)

The use of boundary layer trips in wind tunnel experiments forces transition to fully turbulent flow, which affects the resulting acoustic signature. Boundary layer trips are often too small to be modeled geometrically in computational simulations, making it difficult to consider their influence. The goal of this study was to determine the best method for representing the effects of a boundary layer trip in the benchmark aeroacoustic problem of flow over tandem cylinders in a wind tunnel. This case is pertinent to the study of aircraft landing gear noise because periodic vortex shedding results in pronounced acoustic tones in addition to broadband noise from turbulent

wake interactions. A hybrid lattice-Boltzmann Method/Ffowcs-Williams Hawkings technique was used to predict the transient flow field and the resulting noise. This study compared four methods for simulating the trip's effects: (1) applying a surface roughness on the upstream cylinder; (2) reducing the viscosity of the fluid; (3) forcing velocity perturbations in the incoming free stream; and (4) applying a ridge to the upstream cylinder that is one volume element thick. Preliminary results have shown that reducing the fluid's viscosity is an effective way to reproduce the highly turbulent flow patterns and the acoustic signature.

4:00

4pEAb10. Characteristics of ultrasonic complex vibration for hole machining in brittle materials: Comparison of longitudinal and complex vibration sources. Takuya Asami and Hikaru Miura (Nihon Univ., 1-8-14 Kanda-Surugadai, Chiyoda-ku, 325 Rm., Tokyo 101-8308, Japan, asami.takuya@gmail.com)

Ceramic materials have the advantage of abrasion resistance, heat resistance, and corrosion resistance compared with metal materials. The combination of ultrasonic vibration and polishing slurry has been shown to be an effective method for machining holes in brittle materials. However, conventional ultrasonic methods use only longitudinal vibration. Complex vibration sources with diagonal slits have been applied to ultrasonic motors and ultrasonic welding; in contrast, few studies have been conducted on ultrasonic machining using complex vibration and polishing slurry. Removal rates and machining accuracy are expected to be improved by using ultrasonic complex (longitudinal-torsional) vibration. Therefore, we have developed a new method using polishing slurry together with ultrasonic longitudinal and torsional vibration sources with diagonal slits for hole machining of brittle materials. Torsional vibration is considered to improve the processing of the hole side of ceramic materials such that the polishing slurry can circulate more easily. We assume improvement of removal rate and machining accuracy for that reason. In experiments, soda-lime glass is used as the processing material in ultrasonic complex vibration or ultrasonic longitudinal vibration, and machining time is measured to assess the hole machining characteristics.

4:20

4pEAb11. Sound generation using photoacoustic effect. Kaoru Yamabe, Yasuhiro Oikawa, and Yoshio Yamasaki (Intermedia Art and Sci., Waseda Univ., 59-407-2, 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, shikisokuzeku@fuji.waseda.jp)

It is highly important to generate sound sources in mid air for several applications such as virtual realty and rigorous acoustic measurement. One possible solution for generating sound sources in mid air is photoacoustic effect that generates sounds from the alternate-current component of air expansion due to the heat generated by light absorption of materials when the materials are radiated light modulated with acoustic signal. Our previous research confirmed that audible sound could be generated by radiating light modulated with acoustic signal to charcoal, which has high absorptive power. Therefore, it is possible to generate point sound source in mid air by applying this method to molecule of gases in mid air. However, gases are difficult to absorb light since gases have low density of molecule. Thus in this paper, as the preliminary step applying photoacoustic effect to gas molecule, it is discussed to generate audible sound by radiating light modulated with acoustic signal to liquid phase of H_2O and solid phase of CO_2. These molecules of greenhouse gases can absorb infrared light that is safer than ultraviolet light that is absorbed by monatomic molecules such as N_2 and O_2.

4:40

4pEAb12. Acoustic compressor coupled with fluidic diodes. Sonu K. Thomas and T. M. Muruganadam (Dept. of Aerosp. Eng., Indian Inst. of Technol., Madras, Rarefied Gas-dynamics Lab, IIT, Madras, Chennai, Tamil Nadu 600036, India, thomas.sonu91@gmail.com)

Performance of an acoustic compressor coupled with a fluidic diode is studied. The acoustic compressor works on the idea of Resonant Macrosonic Synthesis (RMS) technology demonstrated by Lawrenson *et al*. The main idea is to replace non-return valves by no moving part fluidic device. Fluidic diode rectifies an oscillatory flow analogous to rectifying an electric AC to obtain DC output. Synthetic jets analogous to AC current falls in the category of jet-driven acoustic streaming, which has a zero mean mass flux. The synthetic jet can be combined with no-moving part fluidic device to generate Hybrid Synthetic Jets with non zero mean mass flux. Non-zero mass flow rate was achieved by coupling fluidic diode and the resonator. In the present study, better rectification is achieved by having number of fluidic diodes in series. Experiments were done for the case of 3 and 4 diodes in series. In the case of three diodes in series the mass flow was 4.6 l/min and in case of four diodes it was 4.9 l/min. Full paper will present the pressure and mass flow measurements for 1 to 5 diodes in series. Thus, RMS cavities with fluid diodes can work as a pump.

5:00

4pEAb13. Plane wave echo particle image velocimetry. Samuel Rodriguez, Xavier Jacob, and Vincent Gibiat (Université Paul Sabatier Toulouse III, PHASE Lab., Bât. 3R1-B2, 118, route de Narbonne, Toulouse 31062, France, vincent.gibiat@univ-tlse3.fr)

This paper deals with the application of topological imaging to ultrasonic echo-particle image velocimetry (Echo-PIV). Echo-PIV is a recent alternative to optical PIV for measuring the instantaneous velocity field of a fluid flow previously seeded with small particles. It consists in imaging the flow with a ultrasonic array at a high frame rate. Topological imaging is a method that benefits from the refocusing properties of the time-reversal principle in a systematic way, so that a single plane wave illumination of the medium leads to a fine resolution. Multiple insonifications are then possible at very high speed allowing not only static images of the medium but successive images of a moving medium. Experimental results are presented for a fluid seeded with stone powder. Two cases are studied: a vortex flow and the propagation of water surface waves.

Session 4pMU

Musical Acoustics: Measurements, Modeling, and Simulations of Brass Instruments

James W. Beauchamp, Cochair
School of Music and Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824

Wilfried Kausel, Cochair
Inst. of Music Acoust., Univ. of Music and Performing Arts, Anton-von-Webern-Platz 1, Vienna 1030, Austria

Thomas Moore, Cochair
Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789

Invited Papers

1:20

4pMU1. Do trumpeters tune resonances of their vocal tract? Jer-Ming Chen, John Smith, and Joe Wolfe (The Univ. of New South Wales, School of Phys. UNSW, Sydney, NSW 2052, Australia, jerming@unsw.edu.au)

In most wind instruments, the acoustic output is generated by airflow through a non-linear valve, whose sounding frequency is largely determined by resonances in the bore of the instrument (an acoustic duct downstream of the valve) and mechanical properties of the non-linear valve that converts DC to AC power. The player's vocal tract (a second duct, upstream) also has acoustic resonances, which—in particular cases—play a significant role in performance technique. For example, when executing advanced techniques (e.g., pitch-bending, altissimo playing) on the clarinet and saxophone, we showed that expert control of vocal tract resonances is essential for performance [Chen *et al*., *Science*, **319**, 726 (2008)]. To understand how such a tract-valve-bore system might interact during trumpet performance, we measured the acoustic impedance spectrum in seven trumpeters' mouths as they played normal notes, high-register notes and while pitch-bending below and above the normal note. Unlike the behavior seen in saxophonists and clarinetists, none of the trumpeters studied showed any systematic adjustment of their vocal tract resonances to the notes played. The much greater control that trumpeters have over the natural frequency of the vibrating valve may explain the difference with clarinetists and saxophonists.

1:40

4pMU2. How can we deduce playing frequencies from measured resonance frequencies for trumpets? René E. Caussé, Pauline Eveno (Instrumental Acoust., IRCAM, 1 place Igor Stravinsky, Paris 75004, France, Rene.Causse@ircam.fr), Joël Gilbert (LAUM, Universite du Maine, Le Mans, France), and Jean-François Petiot (IRCCYN, Ecole Centrale de Nantes, Nantes, France)

Lip-type valve ("striking outwards" type) is responsible for sound production for brass instruments. The operation of the valve is controlled by feedback from a passive resonator. The purpose of this study is to compare experimentally how far the resonance frequencies of instrument, taken from their input impedance (which does not involve the intervention of the player's lips) are able to give informations about the playing frequencies. A family of three trumpets made from a basic instrument for which the lead pipe will be slightly modified for each model, were considered for the experiment. Four expert musicians were asked to play the first five playable notes, for four different fingerings and for three nuances. This exercise was repeated three times. All these notes allow to make a quantitative assessment of the relation between the resonance frequencies and the playing frequencies, using in particular statistical methods. Several results will be presented: the influence of the player on the overall intonation, the effect of nuances on the pitch and the relation between small changes of geometry and playing frequencies. Functions made from the input impedance, such as the « sum function » proposed by Wogram, do not bring more relevant informations than the input impedance itself.

2:00

4pMU3. A trombone model emphasizing acoustic accuracy and playability. Tamara Smyth (Music, Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, trsmyth@ucsd.edu) and Frederick Scott (Computing Sci., Simon Fraser Univ., Vancouver, BC, Canada)

This work contributes a physical synthesis model of the trombone, a virtual musical instrument emphasizing quality sound production, and interactivity. The focus is on modeling and coupling four parts of the trombone: the instrument bore, the bell, the vibrating lips, and the mouthpiece. The model of the instrument is made parametrically flexible by using a combination of filter elements modeled either using known theory or, for elements not well described theoretically, from acoustic measurement. In particular, acoustic accuracy of the bell reflection and transmission is explored by comparing results obtained from measurement, to those obtained from a piecewise conical model. In addition, the playability and performance characteristics of the complete sounding model—when coupled to a mouthpiece and a configurable generalized reed model—is discussed with reference to expected acoustic behavior.

2:20

4pMU4. Influence of the bell profile of the trombone on sound reflection and radiation. D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk), Arnold Myers (Edinburgh College of Art, Univ. of Edinburgh, Edinburgh, United Kingdom), and John Chick (School of Eng., Univ. of Edinburgh, Edinburgh, United Kingdom)

One of the most striking external features of a modern trombone is its wide and rapidly flaring bell. The bore profile of this final section of the instrument influences its musical behavior in a number of different ways, since it determines both the strength of the acoustical feedback from the instrument to the lips of the player and the nature of the radiated sound field. These effects have been explored in an experimental study in which a number of trombones have been progressively modified by the removal of annular sections of the bells. Measurements of input impedance, transfer function, and directivity of radiated sound are presented, and the implications for the timbre and playability of the instruments are discussed.

2:40

4pMU5. Brass instrument power efficiency and the relationship between input impedance and transfer function. Wilfred Kausel (Inst. of Musical Acoust., Univ. of Music ad Performing Arts, Institut f. Wiener Klangstil, Vienna, Austria, Kausel@mdw.ac.at), James W. Beauchamp (School of Music and Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Sandra Carral (Inst. of Musical Acoust., Univ. of Music ad Performing Arts, Vienna, Austria)

From observation of graphs of brass input impedance magnitude and transfer function vs. frequency, it is obvious that there is a strong relationship between the two. Both exhibit a series of strong resonances extending from a low frequency limit to a cutoff frequency, which is inversely proportional to the instrument's bell radius ($f_c = c/(\pi a)$. However, the maxima of the impedance function correspond to the minima of the transfer function. As previously shown [Elliott *et al.*, J. Acoust. Soc. Am. (1982)], the relationship can be seen through a formula for *efficiency* given by $power_{out}/power_{in}$. This formula leads to the squared transfer function being proportional to the efficiency times the real part of the reciprocal of the input impedance, divided by the real part of the radiation admittance. Curves for input impedance, transfer function, and efficiency have been measured, simulated, and compared for several brass instruments. For frequencies below cutoff, the efficiency has an approximate monotonically increasing relationship with frequency, where the log-log slope is dependent on internal losses.

3:00–3:20 Break

3:20

4pMU6. Comparing steady-state and transient phenomena in brass instruments. Peter L. Hoekje (Phys. and Astronomy, Baldwin Wallace Univ., 275 Eastland Rd., Berea, OH 44017, phoekje@bw.edu)

Input impedance and radiation patterns are well-known examples of important brass instrument characteristics measured under steady state conditions. Transient phenomena are less studied, but potentially as important to the player and listener. For example, the heights and harmonicity of the peaks of the instrument's input impedance affect its steady-state playing response, which the player might describe on a range from "stuffy" to "open" or "free-blowing." However, the player also wants an instrument that will facilitate clean and reliable attacks. The development of the instrument's pressure spectrum during the onset transient can serve as an additional diagnostic tool to reveal information about the instrument response under playing conditions. The contribution of the instrument body vibrations to the radiated sound field is small and generally imperceptible under steady-state conditions. However, the bandwidths of the body vibration resonances are generally much narrower than those of the air column resonances and accordingly their transient responses are much longer. This leads to a time signature that enhances their detection by the listener. A complete picture requires consideration of both the steady-state and transient phenomena.

3:40

4pMU7. Nonlinear wall vibration and wave steepening contributing to tonal metallicness and brassiness in a horn. Takayasu Ebihara (YAMAHA Corp., 203 Matsunokijima, Iwata, Shizuoka 438-0192, Japan, prawn_taka@yahoo.co.jp) and Shigeru Yoshikawa (Grad. School of Design, Kyushu Univ., Fukuoka, Japan)

It is well known that wave steepening and shock-wave formation due to nonlinear propagation through the bore are responsible for tonal *brassiness* of brass instruments. On the other hand, penetrating metallic tones are produced by hand-stopping the French horn. The present study demonstrates that the mechanism account for tonal *metallicness* of the French horn is nonlinear wall vibration of the bell. The measured waveforms of radiated pressure of the stopped tones indicate rapidly corrugating changes, which are not observed in brassy tones. Also, their spectra show much larger amplitudes of higher harmonics than those in normal mezzo-forte playing. The measurement of the wall vibration at the bell in hand stopping demonstrates similar characteristics. These results suggest that the bell wall vibration is responsible for the radiated tone color. Excitation experiments on the bell are carried out to elucidate the mechanism how the higher harmonic vibration is generated in hand stopping. They indicate that wall vibrations over 3 kHz are excited by the superharmonic generation derived from the geometrical nonlinearity of the bell. Moreover, for a direct support to our inference above, sound pressure of the stopped tone radiated when the horn bell is heavily damped will be examined.

4:00

4pMU8. Analysis of vibroacoustics of trombone bells thanks to an adaptation of the Miller experiment. Francois Gautier, Mathieu Secail, and Joel Gilbert (Laboratoire d'Acoustique de l'Université du Maine, Université du Maine, Avenue O. Messiaen, Le Mans 72000, France, francois.gautier@univ-lemans.fr)

The influence of wall vibrations on the sound produced by a wind instrument is an open question. If it is clear that the vibrations of bells vibrations can be felt and measured, the influence of these vibrations on the radiated sound is more difficult to bring to light, because the fluid-structure couplings involved are particularly weak except when coincidence effects occur. We propose to study the case of a trombone bell, which is large and thin, favoring the vibrations of large amplitudes and thus the vibroacoustic coupling between

the wall and the air column. For studying the light fluid-structure interaction in organ pipes, Miller [Science **29**(735), 161–171 (1909)] developed one century ago an experiment consisting in blowing an organ pipe surrounded by water. A water tank can be filled progressively in order to modify the mechanical modes of the system in a continuous manner. We propose an adaptation of this experiment to the case of a trombone bell. The acoustic impedance, the acoustical and mechanical responses of a trombone bell excited by a loudspeaker or a shaker are measured for different levels of water allowing an analysis of the vibroacoustic couplings.

Contributed Papers

4:20

4pMU9. Axial vibrations of brass wind instruments. Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu), Wilfried Kausel, Vasileios Chatziioannou (Inst. of Music Acoust., Univ. of Music and Performing Arts, Vienna, Austria), Nikki Etchenique, and Britta Gorman (Dept. of Phys., Rollins College, Winter Park, FL)

It has been proposed that axial vibrations of the bells of brass wind instruments can lead to audible effects in the sound [Kausel *et al*. (2010)]. Using both laser Doppler vibrometry and a novel implementation of electronic speckle pattern interferometry, we have demonstrated that these vibrations exist, and that the magnitude is of the order predicted. [Work supported in part by a grant from the National Science Foundation.]

4:40

4pMU10. The soft-source impedance of the lip-reed: Experimental measurements and computational simulation. Michael Newton, Reginald Harrison Harrison (Reid School of Music, Univ. of Edinburgh, Alison House, Nicolson Square, Edinburgh EH8 9DF, United Kingdom, michael.newton@ed.ac.uk), and Jonathan Kemp (Dept. Music, Univ. of St Andrews, St Andrews, United Kingdom)

Most theoretical descriptions of the brass instrument lip-reed consider the acoustical condition at the lips to be a closed, rigid termination, corresponding to a unitary reflectance. This assumption is carried through to many computational models as well. In reality, the protrusion of the player's lips into the mouthpiece causes a periodic shortening/extension of the acoustical tube downstream, an effect sometimes but not always incorporated into such models. Of interest here is the absorption properties of the lip termination, the so-called "soft source impedance." This provides a further modification to the boundary condition at the lips, since the soft, deformable nature of the lips are likely to cause some extra damping of the acoustic standing wave. Measurements are presented to demonstrate this damping effect using an artificial mouth. This is achieved through measurements of the lip reflectance from downstream of the lips, from where it is shown that the reflectance shows a dip at the peak absorbance frequency of the lips. The frequency of the absorbance is shown to vary as the lip parameters are changed. A simple computational model is described to account for the effect.

Session 4pNSa

Noise, Architectural Acoustics, and Psychological and Physiological Acoustics: Effects of Noise on Human Performance and Comfort II

Lily M. Wang, Cochair
Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

Arianna Astolfi, Cochair
Politecnico di Torino, Corso D.C. degli Abruzzi, 24, Turin 10124, Italy

Chair's Introduction—12:55

Invited Papers

1:00

4pNSa1. Effects of background noise alternating between two levels at varying time intervals on human perception and performance. Andrew Hathaway and Lily M. Wang (Durham School of Architectural Eng. and Constr., Univ. of Nebraska – Lincoln, Peter Kiewit Inst., 4014 Burt St., Omaha, NE 68131, ahathaway@unomaha.edu)

Heating, ventilation, and air-conditioning (HVAC) systems commonly produce noise in the built environment, and the increased noise levels that these systems can produce have been shown to impact occupant comfort. However, relatively little is understood about how the fluctuations in HVAC noise over time can impact human perception and performance. This research aims to measure human responses under HVAC-like background noise that is alternating between two different levels (one low and one high) at varying time intervals. Twenty-seven participants were tested over four 30 min sessions during which they were subjected to broadband noise at room criteria ratings of RC-29(H) and RC-47(RV) that alternated at certain time intervals. The time intervals of variation tested were 2, 5, 8, and 10 min, and would remain the same during one 30 min test session. The results of an arithmetic test dealing with short-term memory and a subjective questionnaire are presented to show whether or not shorter time intervals of variation have different effects than longer ones. [Work supported by a NASA Nebraska Space Grant.]

1:20

4pNSa2. Background noise in Chinese schools – Student and teacher perceptions. Kenneth P. Roy (R&D, Armstrong World Industries, Innovation Ctr., 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com) and Jerry Li (Regional Res. Ctr., Armstrong W.I., China Ltd., Shanghai, China)

A significant research study was conducted in 2011/2012 to evaluate the Acoustic Comfort in 10 primary/middle schools throughout China as a joint effort by the Green Campus workgroup (Tongji University, Shanghai), and Armstrong World Industries (Research and Development). In the fall of 2012, an entire grade school in Nanjing, China, was involved in a combination of renovated and new construction with a focus on the requirements of Chinese Standard GB50118 for acoustic design/performance of classrooms. These research programs all included both objective measurements of acoustic performance, and subjective perception by students and teachers of both speech clarity (architecture) and distractions and comprehension (noise). The noise aspects of these measurements and surveys will be addressed in this paper. The primary noise sources in these schools are mainly from outside the classroom itself and involve student activities in corridors and other classrooms, and adjacent transportation noise. Effectiveness of in-room acoustic treatments on these noise perceptions is also reviewed.

1:40

4pNSa3. Influence of classroom acoustics on the vocal behavior of teachers. Arianna Astolfi (Dept. of Energy, Politecnico di Torino, Corso Duca degli Abruzzi, 24, Turin 10124, Italy, arianna.astolfi@polito.it), Alessio Carullo, Alberto Vallan (Dept. of Electron. and Telecommunications, Politecnico di Torino, Turin, Italy), and Lorenzo Pavese (Dept. of Energy, Politecnico di Torino, Turin, Italy)

Erroneous vocal behavior of teachers and their changes in the voice production due to poor acoustics in classrooms, can be investigated through recently developed voice-monitoring devices. These devices are portable analyzers that use a miniature contact microphone glued to the jugular notch in order to sense the skin acceleration level due to the vibration of the vocal folds. They estimate the Sound Pressure Level (SPL) at a certain distance from the speaker's mouth, provided that a suitable calibration procedure is performed, the fundamental frequency and the time dose. Two different devices are compared in this work: the former is a commercial device, whose phonation sensor is a small accelerometer; the latter, recently developed by the authors, uses an electret condenser microphone to sense the skin acceleration level. SPL and fundamental frequency are estimated over 30 ms- and 50 ms-length frame and the results that refer to a sample of 40 primary school teachers and some university professors are analyzed. The length of the voice and pause frames is analyzed in order to detect the maximum of occurrence and accumulation in different conditions of noise and reverberation. A method for the detection and analysis of the emphatic speaking is also proposed.

2:00

4pNSa4. Effects of reverberation and noise on speech comprehension by native and non-native English-speaking listeners. Zhao Peng, Lily M. Wang, Siu-Kit Lau, and Adam M. Steinbach (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182, zpeng@unomaha.edu)

Previous studies have demonstrated the negative impact of adverse signal-to-noise-ratios on non-native English-speaking listeners' performance on speech recognition using recall tasks, as well as implied that comprehension skills were more impaired than recognition skills under reverberation and noise. The authors have themselves previously conducted a pilot study on three native and three non-native English-speaking listeners to examine the effects of reverberation and noise using speech comprehension tasks. Those results suggested that speech comprehension performance is worse under longer reverberation times (RT), and that a longer RT is more detrimental to speech comprehension by non-native listeners than native listeners. This paper reports on the refined full study, in which a larger number (up to 30) of each group was tested. Each participant was exposed to 15 acoustic conditions, created from combinations of five RTs (0.4 to 1.2 s) and three background noise levels (RC-30, 40, and 50). Speech comprehension performance under each condition was recorded. Confounders related to general speech comprehension abilities were screened for, including listening span, oral comprehension abilities, and English verbal skills. Results are presented and compared between native and non-native listeners. [Work supported by a UNL Durham School Seed Grant and the Paul S. Veneklasen Research Foundation.]

2:20

4pNSa5. Open plan office: The appropriate privacy and material metrics. Kenneth W. Good (Armstrong, 2500 Columbia Ave., Lancaster, PA 17601, kwgoodjr@armstrong.com)

Recent metrics for speech privacy may be misapplied to the open plan environment resulting in false and misleading conclusions. This paper will review the state of privacy and material metrics, discuss the application and perspective of "privacy" in the open plan. And present the real effects of these metrics in the open plan environment.

Contributed Papers

2:40

4pNSa6. The teachers perspective on noise in the classroom. Ana M. Jaramillo, Michael Ermann, and Patrick Miller (School of Architecture + Design, Virginia Tech, 424 C Harding Ave., Blacksburg, VA 24060, anaja@vt.edu)

A survey was sent to third grade teachers in Orange County, FL, to find out about their noise awareness and coping strategies. Results of the survey were also correlated to mechanical system type and achievement data. Preliminary analyses show very little awareness on mechanical noise by teachers but a good range of coping strategies when noise sources are present (mostly activity noise). The survey also helped to better understand the classroom environment. For example, most classrooms have a frequent use of computers or projectors and a few schools are still open-plan. These facts create new questions about noise in the classroom that need to be addressed in further studies.

3:00–3:20 Break

3:20

4pNSa7. Perception and evaluation of noise sources in open plan office. Marjorie Pierrette, Etienne Parizet (Laboratoire Vibrations Acoustique, INSA, 25 bis, av. J. Capelle, Villeurbanne F-69621, France, marjorie.pierrette@insa-lyon.fr), and Patrick Chevret (Laboratoire "Réduction du bruit au travail", INRS Centre de Lorraine, Vandoeuvre Les Nancy, France)

Open plan offices are now the most common form of workspaces organization. They can improve communication between workers while saving space. Their main drawbacks are the lack of intimacy for occupants and the increase of noise level. Noise is one of the most important annoyance factor as described by workers (see SBISB study, 2010). This paper describes a study aiming at a better knowledge of most annoying noise sources in an open plan offices. It consisted in interviews and questionnaires conducted in offices together with physical measurement. This provided some information about sources and tasks for which workers are mainly disturbed. The analysis of recorded answers allowed to evaluate the influence of this annoyance on job stress and health and emphasize the influence of environmental and individual factors in the assessment of noise annoyance.

3:40

4pNSa8. Work performance and mental workload in multiple talker environments. Ange Ebissou, Patrick Chevret (Institut National de Recherche et de Sécurité (INRS), Rue du Morvan, CS 60027, Vandoeuvre-les-Nancy 54519, France, ange.ebissou@inrs.fr), and Etienne Parizet (Laboratoire Vibration et Acoustique, INSA-Lyon, Villeurbanne Cédex, France)

The impairment of cognitive performance resulting from the presence of speech sounds is known to increase as the intelligibility of the speech signals is improved. For that reason, speech intelligibility measures are used to quantify the nuisance potential of an unattended voice. However, most of these indexes struggle with situations in which the level of the masking sound is fluctuating. This is the case in open-plan offices, where competing voices are involved. This paper relates a set of experiments in which subjects had to carry out a basic memory task in various noise settings. In addition to a target speech, the masking sounds were made up of speech and differed in temporal variability. The signal-to-noise ratios and the overall long-term spectra were kept constant. Disturbance was assessed both through objective measurements of performance and subjective reports of workload. The results highlight the importance of taking into account the temporal fluctuations of the overall ambient sound when trying to ascertain the influence of speech intelligibility on observed and perceived disturbance during the performing of a mental activity. Insights are provided, which could lead to the use of a speech intelligibility measure better equipped to deal with multi-sources environments.

4:00

4pNSa9. Planned versus achieved acoustical performance and user satisfaction for an office fit-out. Ryan Bessey (Acoust., Noise and Vib. Group, Golder Assoc. Ltd., 141 Adelaide St. West, Ste. 1220, Toronto, ON M5H 3L5, Canada, ryan_bessey@golder.com)

The purpose of the project was to move 500 office workers separated between two aging buildings into one new larger building. To this end, a team was formed to create an office fit-out design including several open plan offices, private offices, meeting rooms, boardrooms, a conference center, and a cafeteria to connect with the base-building infrastructure. Our role, as part of the design team, was to provide guidance such that the acoustical performance for new spaces was at least equivalent but preferably better than

their existing counterparts. Factors that were considered included acoustic comfort, background sound level, speech privacy, and speech intelligibility. This paper summarizes our approach to the project, some difficulties that were encountered during construction, measured acoustical performance in both the newly constructed and existing buildings, and the statistical change in user satisfaction before and after the move determined via surveys.

4:20

4pNSa10. Noise stress for patients in hospitals – A literature survey. Gert Notbohm and Silvester Siegmann (Inst. of Occupational Medicine, Univ. of Duesseldorf, Universitaetsstr. 1, Duesseldorf 40225, Germany, notbohm@uni-duesseldorf.de)

The growing number of publications on noise in hospitals reflects not only a rising interest in this theme during the last decades, but also an increasing noise exposure of the patients: the average SPL reported in literature between 1960 and 2005 has risen from 57 to 72 dBA in daytime and from 42 to 60 dBA at night. The hospitals in question differ substantially with regard to type of construction, technical equipment, and organizational issues. But especially for intensive care units (ICUs), the main sources of noise described in international research are similar: sounds from technical appliance such as alarms, noise caused by the staff talking or handling material, and communication systems such as overhead paging. With regard to patients in ICUs, sleep disturbances in terms of falling asleep and sleeping through are the greatest problem as assessed by questionnaires or by physiological measurements. They might have harmful effects on the outcome of the medical treatment influencing the duration of recovery and the need for sedative medication. Several intervention programmes for noise reduction are reported in literature combining a variety of methods such as acoustical insulation, sound level reduction with regard to equipment, and especially behavior modification of the staff.

4:40

4pNSa11. Vocal strain in UK teachers: An investigation into the acoustic causes and cures. Nick Durup, Bridget Shield, Stephen Dance (Dept. of Urban Eng., London South Bank Univ., 14 Lynton Dr., Ely cb6 1dq, United Kingdom, nicksenate@hotmail.com), and Rory Sullivan (Sharps Redmore Partnership Ltd., Ipswich, United Kingdom)

Recent surveys indicate that approximately 60% of UK teachers experience voice problems during their career. This costs £15M annually in teacher absence and can have a significant human cost for those involved. This study investigated the impact of classroom acoustics on teachers' voice levels to determine if acoustic modifications of classrooms could reduce the vocal load placed on teachers. Measurements of teachers' voice levels were made using an ambulatory phonation monitor (APM), which measures voice parameters directly from skin vibrations on the neck. Simultaneous sound level meter measurements of various parameters were also carried out in the classrooms. The room acoustic parameters of the classrooms were measured separately to the APM measurements. Measurements have been taken in a range of classrooms as part of a pilot study. Results will be reported as to the effects of different acoustic environments in the classroom on the teachers' voice levels.

5:00

4pNSa12. Acoustic quality on board ships. Robin D. Seiler and Gerd Holbach (Naval Architecture Ocean Eng., Berlin Inst. of Technol., Salzufer 17-19, Bldg. SG1, Secr. SG 6, Berlin 10587, Germany, r.seiler@tu-berlin.de)

Approaches to determine acoustic quality on board ships are usually based on a three- or five-stage classification system using critical A-weighted sound pressure levels. At times other criteria such as the sound insulation between cabins, impact noise from upper decks, speech interference levels or general noise-rating curves are also taken into account. With regard to increasing requirements of passenger comfort and crew accommodation, a more detailed evaluation of auditory perception on board would be worthwhile. Furthermore, other industrial sectors have stated that psychoacoustic or room acoustic models are useful tools to analyze and guarantee product-sound quality. In order to find better indicators for the quantification of acoustic performance of (luxury) vessels, audio material was acquired at a sea trial and evaluated with the help of a paired-comparison listening test in the laboratory by 30 test-persons. Also, the possibility to comment the judgments was given to the subjects. The results were then analyzed by using the statistic model of the "Law of Comparative Judgment" and compared by correlation analysis with physical, psychoacoustic, and room acoustic parameters. Highly correlating parameters could be identified.

THURSDAY AFTERNOON, 6 JUNE 2013 — 511CF, 1:00 P.M. TO 3:20 P.M.

Session 4pNSb

Noise and Architectural Acoustics: Noise Control

Fabian Probst, Chair
Res. & Development, DataKustik GmbH, Gewerbering 5, Greifenberg 86926, Germany

Contributed Papers

1:00

4pNSb1. Airborne sound insulation as a measure for noise annoyance. Reinhard O. Neubauer and Jian Kang (School of Architecture, Univ. of Sheffield, Theresienstr. 28, Ingolstadt, Bavaria 85049, Germany, r.neubauer@sheffield.ac.uk)

There is currently a lack of measure to describe airborne sound insulation in terms of subjective evaluation of noise annoyance. With a given sound insulation value, different kinds of sound signals could produce rather different hearing sensation levels. Physical noise measurements to describe airborne sound insulation often cannot solve problems in terms of noise annoyance, and psychoacoustic metrics are increasingly used. Recently, new results of evaluating sound insulation spectra by single-numbers have been adapted for practical applications such as in ISO 16717-1. In this paper, comparisons are carried out to demonstrate how single-number ratings are affected by non-steady-state sounds. The effect of a sound insulation having a frequency dip of 6 dB has also been examined. It is well known that noises with tonal components could be rather annoying, so that it would be of significance to examine if a frequency depending sound insulation can act as a filter for tonal components. In this paper, it will be shown that psychoacoustic magnitudes like loudness, sharpness, and fluctuation strength can largely account for different aspects, especially if airborne sound insulation is supposed to describe hearing sensation.

1:20

4pNSb2. Noise reduction from large machineries by using sound enclosures. Hyun-Sil Kim, Jae-Seung Kim, Seong-Hyun Lee, and Yun-Ho Seo (Acoust. and Noise Res. Team, Korea Inst. of Machinery and Mater., Yusung-Gu Jangdong 171, Daejeon 305-343, South Korea, hskim@kimm.re.kr)

A sound enclosure is an effective measure to reduce the noise emitting from the large noise sources such as diesel engines and gas turbines. In this study, insertion loss prediction of the large enclosure is presented. Inside the enclosure, diffuse sound field is assumed, and there exist no air leakages. Insertion loss is predicted by using statistical energy analysis (SEA). From the energy equilibrium equations, sound pressure inside the enclosure is derived in terms of the acoustic power from the machinery. Insertion loss is defined as the ratio between acoustic power inside and transmitted power outside the enclosure. It is shown that sound radiation from the panel vibration can be neglected compared to that transmitted through panel. Insertion loss predictions are compared to the measurements. The enclosure size is 6.4 m x 2.65 m x 4.8 m (L x W x H) and 4.5 m x 2.5 m x 2.0 m, where panel consists of 1.5 mm steel plate and 70 mm mineral wool. The comparisons show good agreements. It is concluded that to increase the insertion loss, panel must have a large sound transmission loss and sound absorption coefficient inside the enclosure must be high.

1:40

4pNSb3. Noise reduction in working areas by the application of absorbing baffle-systems. Fabian Probst (Res. & Development, DataKustik GmbH, Gewerbering 5, Greifenberg 86926, Germany, info@datakustik.com)

Baffle systems are arrangements of absorbing panels that allow free flow of air and therefore do not disturb the acoustic climate. This is one of the reasons why they are often used in industrial environments because there is no need to take into account aspects of thermal isolation that may be a problem with closed suspended ceilings. A method to determine the absorption coefficient of baffle systems was derived and published in 2008 [Probst W.: "Sound absorption of baffle systems", Lärmbekämpfung Nr.2 (2008)]. A method is now presented how such systems can be taken into account if the acoustic behavior of even complex rooms is determined by computer modeling. For simple cases, the mentioned analytical method can be applied, and it is shown that the results are in good agreement with the detailed simulation. But this detailed simulation allows to determine the acoustic influence of partially covered areas with different heights and otherwise complex layouts with absorbing appliances.

2:00

4pNSb4. Experimental study on sound absorbing performance of rubber crumb. Davide Borelli, Corrado Schenone, and Pittaluga Ilaria (DIME - Sez. TEC, Università Degli Studi di Genova, Via all'Opera Pia 15/A, Genova, GE 16145, Italy, davide.borelli@unige.it)

The present paper describes an experimental campaign aimed at the determination of acoustical properties of vulcanized rubber crumbs obtained by the shredding of used tires. In particular, their performance as sound absorbing material in lined ducts has been investigated. The most innovative aspect that is addressed in the study is the use of a waste material such as rubber tires reduced into small grains as a sound absorbing material: tires are in fact usually used at the end of their life cycle as fuel and burned in cement kilns in order to take advantage of their high heating value, with all the problems of pollution that this solution produces. Two kinds of rubber crumbs have been investigated in terms of characteristic dimension of the grains, porosity, and sound absorbing coefficient, while their "in situ" performance when used inside lined and parallel-baffle rectangular ducts has been evaluated measuring their insertion loss. The results of this research show that the acoustical behavior of the tested rubber crumbs is the typical behavior of the granular materials, showing a noteworthy performance of the tested material in the low frequency range, opening a scenery of possible applications where noise has relevant tonal components below 315 Hz.

2:20

4pNSb5. Shape optimization of reactive mufflers using threshold acceptance and finite element method methods. Abdelkader Khamchane, Youcef Khelfaoui, and B. Hamtache (Material Technol. and Eng. Process Lab., Univ. of Abderahman Mira of Bejaia, Route de Targa Ouzemour, Béjaia 06000, Algeria, abdelkader.khamchane@yahoo.fr)

Recently, research on the acoustic performance of reactive mufflers under space constraint becomes important. In this paper, the attenuation performance of single and double expansion-chambers under space constraint is presented. To assess the reactive mufflers, a shape optimization analysis is performed using a novel scheme called threshold acceptance (TA), the best design obtained by the shape optimization method are analyzed by Finite Element Method (FEM). The numerical assessment is based on the maximization of the sound transmission loss (STL) using the Transfer Matrix Method (TMM), a modeling method based on the plane wave propagation model. The FEM solution used to analyze the STL of the shape optimized mufflers is based on the Acoustic Power method, a standard computational code COMSOL Multiphysics is used to analyse in 3D the sound attenuation of the mufflers by the FE method. The acoustical ability of the mufflers obtained is than assessed by comparing the FEM solution with the analytical method. Results show that the maximal STL is precisely located at the desired targeted tone. In addition, the acoustical performance of mufflers with double expansion-chamber is found to be superiors to the other one. Consequently, this approach provides a quick and novel scheme for the shape optimization of reactive mufflers.

2:40

4pNSb6. Numerical mode-matching approach for acoustic attenuation prediction of expansion chambers with single inlet and double outlets. Zhenlin Ji and Zhi Fang (School of Power and Energy Eng., Harbin Eng. Univ., No. 145 Nantong St., Nangang District, Harbin City, Heilongjiang Province, Harbin, Heilongjiang 150001, China, zhenlinji@yahoo.com)

Numerical mode matching (NMM) method is developed to predict the acoustic attenuation performance of expansion chambers with single-inlet and double-outlets. The two-dimensional finite element method is employed to calculate the transversal eigenvalues and eigenvectors, and the mode matching technique is used to determine the modal amplitudes and transmission loss of expansion chamber silencers by combining the boundary conditions at inlet and outlets. For the purpose of validation, the transmission loss predictions of elliptical expansion chambers with single-inlet and double-outlets from the present NMM method and the three-dimensional finite element method (FEM) are compared, and good agreements between them are observed. Then the NMM method is used to investigate the effects of extended lengths and locations of inlet and outlets on the acoustic attenuation performance of elliptical expansion chambers.

3:00

4pNSb7. The impact of neck material on the sound absorption performance of Helmholz resonator. Dong Yang, Min Zhu (Dept. of Thermal Eng., Tsinghua Univ., Rm. 110, Gas Turbine Inst., Bei Jing 100084, China, yd.tsinghua@gmail.com), and Xiaolin Wang (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

Helmholtz resonator is an effective acoustic attenuation device at low frequencies and is generally used as passive damper. In this work, parallel perforated ceramics with different perforation diameters were used to improve acoustic impedance at the entry of the resonator and thus achieve better acoustic absorption coefficient and better absorption bandwidth simultaneously. With experimental measurement, ceramics with different perforation diameters are found to improve sound absorption performance of Helmholtz resonator in different extent. At the same time, a model is developed to calculate the resonator's neck mouth impedance and further to predict sound absorption coefficient. Particularly, resonance resistance is considered based on the nonlinear correction to Darcy's law. The results show that large resonance resistance with large perforation diameter materials are due to non-fully developed factor. The largest velocity oscillation amplitude in the resonator neck will lead the Reynolds number up to more than 3000 near the resonance frequency and thus make the nonlinear Forchheimer revision coefficient decrease as Reynolds number increase. Helmholtz resonator with neck filled with sound absorption materials has improved sound absorption capacity. This prediction agrees well with the experiment results and this model can be used to optimize the sound absorption system with Helmholtz resonators.

Session 4pPAa

Physical Acoustics: Nonlinear Acoustics II

Murray S. Korman, Chair
Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402

Contributed Papers

1:00

4pPAa1. Acoustic instability of vortices. Konstantin A. Naugolnykh (Physics, Univ. of Colorado, 325 Broadway, Boulder, CO 80305, konstantin.naugolnykh@noaa.gov)

Large plane is a powerful source of vortex. Disturbances of an axial vortex in a compressible fluid are unstable with respect to Kelvin wave development and produce acoustic wave generation. On the other hand system of counter-rotating vortices (Lamb dipole), with different intensity of vortices, as a result of instability collapse into the center of rotation of vortices. The characteristic time of collapse is in the reciprocal proportion the vortices intensity difference. The evolution of Lamb dipole, determined by the two competing processes of instability, is considered in the presented paper. The sound radiation of Lamb dipole can be used to estimate the vortex structure produced by large plane. The advent of large aircraft, with their attendant large and strong trailing vortex structures, made this a problem of considerable practical concern.

1:20

4pPAa2. General nonlinear acoustical equation of relaxing media and its stationary solutions. Nonna Molevich, Rinat N. Galimov (Theor. Phys., P. N. Lebedev Physical Inst. of RAS, Samara Branch, NovoSadovaya 221, Samara 443011, Russian Federation, molevich@fian.smr.ru), Vladimir G. Makaryan, Dmitriy I. Zavershinskii, and Igor P. Zavershinskii (Physics, Samara State Aerosp. Univ., Samara, Russian Federation)

During previous years, the conditions for the negative second (bulk) viscosity existence were found in a large number of nonequilibrium media. The media with negative viscosity possess a number of new properties including acoustical activity. In the present paper, we investigate the nonlinear stage of acoustical perturbation evolution in acoustically active nonequilibrium media using three models: the vibrationally excited gas with the exponential model of relaxation, the chemical active two component mixture with a nonequilibrium reaction and media with the general heat-loss function. The general nonlinear acoustical equation describing stationary density profiles behind the shock wave front in these media is obtained and solved. Its low- and high- frequency limits correspond to the Kuramoto-Sivashinsky equation and the Burgers equation with a source, respectively. Stationary structures of general equation, the conditions of their establishment and all their parameters are found analytically and numerically. In acoustically active media, it is predicted the existence of the stationary solitary pulse. Unstable weak shock waves disintegrate into the sequence of solitary pulses. Their amplitude, form, and speed are rigidly defined by the nonequilibrium degree and do not depend on the initial weak perturbation amplitude. For weak nonequilibrium degree, this solitary pulse is described analytically.

1:40

4pPAa3. Perturbation methods for the spectral analysis of a weakly nonlinear acoustic field generated by a transient insonation. Hassina Khelladi (Faculté d'Electronique et Informatique, Département Instrumentation, Université des Sci. et de la Technologie Houari Boumediene, BP32, El Alia, Alger, Bab Ezzouar 16111, Algeria, hassinakhelladi@yahoo.fr) and Fahim Rahmi (Faculté des Sci. de l'Ingénieur, Université M'Hamed Bougara, Boumerdes, Algeria)

In this study an infinite plane piston is considered which oscillates with finite amplitude in unbounded homogeneous fluid. To illustrate the shape of the weakly nonlinear acoustic field generated by a transient insonation, the function defined by Funch/Muller representing a damped sinusoid is used to simulate the temporal waveform of the piston vibration. The acoustic transient wave generates harmonic components as result of nonlinearities in the material properties of the fluid and in the convective terms of the propagation equation. The mathematical approach is based upon the generalized Burgers' equation, which is a good approximation of the exact equation for the nonlinear propagation when diffraction effects are assumed to be negligible. The pressure amplitude of the fundamental is considered large enough to produce the second harmonic wave. Under the quasi-linear approximation, an analytical description of the fundamental and the second harmonic waves is elaborated. To simulate the spectrum of the weakly nonlinear acoustic field, the pressure field is written in a perturbation series where the first term is the linear acoustic field that results from an infinitesimal oscillation of the piston and the second term contains the first nonlinear contribution to the acoustic field due to the finite amplitude effects.

2:00

4pPAa4. Strongly nonlinear waves – A new trend of nonlinear acoustics. Oleg V. Rudenko (Radiophysics, Nizhni Novgorod State Univ., Campus Grasvik, Karlskrona, Blekinge 37179, Sweden, oru@bth.se) and Claes M. Hedberg (School of Eng., Blekinge Inst. of Technol., Karlskrona, Blekinge, Sweden)

Strongly nonlinear waves (SNWs) and extreme states of matter are key physical concepts. A SNW is a wave whose amplitude is on the order of the material's internal strength. High-intensity light is a weak nonlinear wave (WNW) if its electric field is weaker than the intra-atomic: $E \ll 10^{11}$ V/m. A SNW irreversibly modifies a medium, up to its destruction. In vacuum a wave 10^{18} V/m is strong, when it creates electron-positron pairs. In acoustics SNWs must be distinguished from WNWs which also can display strongly nonlinear phenomena. When a shock front appears at a distance of 10^2-10^3 wavelengths in water, nonlinearity is weak but strongly expressed. The acoustic pressure is 10^5-10^6 Pa, much less than the internal pressure 2.2×10^9 Pa. However, impurities decrease the breaking strength, and waves create bubbles at smaller pressures. An explosive wave is also a SNW, breaking solids. Nuclear explosions may even create new chemical elements. For WNWs the equation of state can be expanded in power or functional series. However, these cannot be used in three cases. First, if the equation contains singularities, like for "clapping" and Hertz nonlinearities of heterogeneous solids. Second, if the series is divergent. Third, when the linear term is absent and the higher nonlinearities dominate. Such SNWs appear in mechanics and in quantum field theory. Mathematical models of SNW, solutions, and new phenomena observed experimentally will be presented.

2:20

4pPAa5. Chaos and beyond in a water filled ultrasonic resonance system. Laszlo Adler (Ohio State Univ./Adler Consultiants Inc., 1560 Gulf Blvd #1002, Clearwater, FL 33767, ladler1@aol.com), William T. Yost, and John H. Cantrell (NASA-Langley Res. Ctr., Hampton, VA)

Finite amplitude ultrasonic wave resonances in a one dimensional liquid-filled cavity are reported. The resonances are observed to include not only the expected harmonic and subharmonic signals but chaotic signals as well. The nonlinear features of this system were recently investigated and are the focus of this presentation. An ultrasonic interferometer having

optical precision was constructed. The transducers having the frequency range of 1–10 MHz, driven by a high power amplifier. Both an optical diffraction system and a receiving transducer were used to assess the generated resonance response in the cavity. At least five regions of excitation are identified: (1) Linear region: at low intensity of the ultrasonic wave the diffraction pattern of a light beam is symmetric. (2) Nonlinear region: with increased sound amplitude the diffraction pattern becomes asymmetrical. (3) Subharmonic region: further increase of the amplitude above a threshold value leads to the generation of subharmonics. (4) Chaos: increasing the drive amplitude to a second threshold level the diffraction pattern is smeared out indicating a time-chaotic region. (5) Beyond chaos: further increase of the amplitude results again a stable diffraction pattern. A first-principle-based explanation of the experimental findings is presented. [Work supported by the Aircraft Aging Program, at NASA Langley Research Center. Pending approving by NASA.]

2:40

4pPAa6. Compressed parametric difference frequency sound with chirp signal. Hideyuki Nomura, Hideo Adachi, Tomoo Kamakura (The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu-shi 182-8585, Japan, h.nomura@uec.ac.jp), and Gregory T. Clement (Harvard Med. School, Boston, MA)

The directivity of parametric difference frequency sound is narrower than that of linear sound with same frequency radiated from a sound source with same aperture size. In addition, the parametric difference frequency sound can propagate a long distance in a dissipative medium, because sound absorption at low frequency is less than those at usual ultrasound frequency for measurements and medical imaging. However, for applications of parametric difference frequency sound on measurements and imaging, that has the disadvantage of low spatial resolution because. In this study, we proposed the application of pulse compression technique with chirp signal to parametric difference frequency sound for the improvement of spatial resolution. Nonlinear propagation of ultrasound in water was numerically simulated to confirm the realization of compressed parametric difference frequency sound. A sound source at center frequency of 1 MHz was driven by up and down linear chirp signals to generate chirp difference frequency with band width of 100 to 400 kHz, and the autocorrelation function of generated difference frequency sound was calculate to archive pulse compression. The results indicated the realization of pulse compressed parametric difference frequency sound with desired pulse width which is inversely proportional to the band width.

3:00

4pPAa7. Exploration of third-order nonlinear acoustics for projection of narrow-beam lower-frequency underwater beams. Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil), Richard A. Katz (Sensors & Sonar Systems, Naval Undersea Warfare Ctr., Newport, RI), Allan D. Pierce (P.O. Box 339, East Sandwich, MA), and Derke R. Hughes (Sensors & Sonar Systems, Naval Undersea Warfare Ctr., Newport, RI)

Projection systems are considered where two or three frequencies (e.g., f1, f2, and/or f3) are simultaneously projected into water in a parallel fashion. High near-field amplitudes produce beams of frequencies equal to any linear combination of f1, f2, and f3, with integer coefficients n1, n2, and n3 (possibly zero or negative). Interest here is in the case where the magnitudes of the coefficients sum to three, associated with a third-order nonlinearity. The question addressed is that of how large the amplitude of the far-field signal will be. The considered causes of the nonlinearities are (1) the convective derivative term in the total time derivative of the fluid velocity, and (2) the higher coefficients in the expansion of the fluid density in terms of the deviation of the pressure from its ambient value. These coefficients are derived from data reported by Holton *et al*. [J. Acoust. Soc. Am. (1968)] on the sound speed in water. A perturbation technique is explored starting with the basic nonlinear equations of compressible time-dependent fluid dynamics, where at each step one has a simultaneous set of coupled linear and homogeneous equations with the source terms dependent on the solutions of the analogous equations corresponding to the previously considered orders.

THURSDAY AFTERNOON, 6 JUNE 2013 — 519B, 3:20 P.M. TO 5:00 P.M.

Session 4pPAb

Physical Acoustics: Thermoacoustics II

Albert Migliori, Chair
Los Alamos National Laboratory, Los Alamos, NM 87545

Contributed Papers

3:20

4pPAb1. Hysteresis of mode transitions with varying cavity length of bottle-shaped thermoacoustic prime movers. Bonnie Andersen, David Pease, and Jacob Wright (Physics, Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84057, bonniem@uvu.edu)

Transition regions to higher resonant modes of a bottle-shaped thermoacoustic prime mover (neck: 5.39 cm long, 1.91 cm ID; variable cavity with a sliding piston: up to 38 cm long, 4.76 ID) were studied. The neck and cavity regions behave as coupled resonators. A variable cavity with a sliding piston was constructed to study the nature of the device as the cavity length is varied. The dominant mode of operation depends on the length of the cavity, favoring successively higher modes as the cavity length increases, occurring roughly where the higher mode overlaps with the fundamental frequency of the neck region. As the cavity length is increased, the transition of the dominant frequency from the fundamental to the first overtone occurs. However, when the length is then shortened, transition back to the fundamental does occur at the same piston position, revealing hysteresis. Three transitions to higher modes were observed. The hysteresis was studied as a function of input power (12.0–16.5 W) and stack volume filling factor (3.0–4.9%). Preliminary results indicate that the transition region occurs shallower in the cavity and the hysteresis widens as the input power is increased. Decreasing the stack mass causes an increase of the hysteresis width, but has no strong effect on the hysteresis depth.

3:40

4pPAb2. Numerical simulations of a transient behavior in the onset of thermoacoustic marginal oscillations in a looped tube. Dai Shimizu and Nobumasa Sugimoto (Mech. Sci., Grad. School of Eng. Sci., Osaka Univ., 1-3, Machikaneyama, Toyonaka, Osaka 560-8531, Japan, dai_shimizu@me.es.osaka-u.ac.jp)

This paper simulates a transient behavior in the onset of spontaneous, thermoacoustic oscillations of a gas in a looped tube with a so-called stack sandwiched by hot and cold heat exchangers. Numerical

simulations are performed to solve an initial-value problem by employing the linearized boundary-layer theory. Initial conditions are chosen in such a way that the gas under a uniform pressure is given impulses at two locations in the outside of the stack to cancel with each other. Because of installation of the stack and heat exchangers, account is taken of discontinuity in the temperature gradient, the cross-sectional area and the wetted perimeter of the gas passages by imposing continuity of mass and energy fluxes. Except for a special value of the temperature ratio of the two heat exchangers, the pressure fluctuates significantly around the initial value transiently, but it eventually tends to grow indefinitely or decay out. At a marginal case, it is observed that a traveling wave tends to emerge spontaneously. The traveling wave always propagates in the sense from the hot to cold heat exchangers in the tube outside of the stack. It is shown qualitatively that the traveling wave is enhanced as the porosity lowers.

4:00

4pPAb3. Marginal conditions of thermoacoustic oscillations in a looped tube based on thick and thin diffusion-layer theories. Hiroaki Hyodo (Dept. of Mech. Sci., Grad. School of Eng. Sci., Univ. of Osaka, Motoyama-kitamati3-4-25, Kobe 6580003, Japan, h_hyodo@mars.me.es.osaka-u.ac.jp) and Nobumasa Sugimoto (Dept. of Mech. Sci., Grad. School of Eng. Sci., Univ. of Osaka, Toyonaka, Japan)

Marginal conditions for the onset of thermoacoustic oscillations of a gas in a looped tube with a "stack" inserted are examined by using two approximate equations for thick and thin thermoviscous diffusion layers in comparison with a span length of a gas passage. The equations are derived from the general thermoacoustic-wave equation valid for any thickness of the layer in the linear framework. Applying those approximate equations, respectively, to the gas in the stack and that in the outside of the stack, a frequency equation is derived by imposing matching conditions at both ends of the stack. Seeking a real solution for the frequency, the marginal conditions are obtained numerically for the temperature ratio at both ends of the stack. The ratio depends not only on the span length of one passage in the stack but also on its porosity. It is revealed that the temperature ratio decreases with increasing the span length and the porosity as well. This is the case when the thick diffusion layer is assumed in the stack. It is also revealed that a traveling wave tends to emerge in the tube outside of the stack in the sense from the hot end to cold end.

4:20

4pPAb4. Introduction of conical phase adjuster for thermoacoustic system. Shin-ichi Sakamoto (Univ. of Shiga Prefecture, 2500 Hassaka, Hikone 522-8533, Japan, sakamoto.s@usp.ac.jp), Manabu Inoue, Yosuke Nakano (Doshisha Univ., Kyotanabe, Japan), Yuichiro Orino, Yoshitaka Inui, Takumi Ikenoue (Univ. of Shiga Prefecture, Hikone, Japan), and Yoshiaki Wanatane (Doshisha Univ., Kyotanabe, Japan)

We have proposed a phase adjuster for the thermoacoustic system and succeeded in improving the energy conversion efficiency from heat to the sound. A phase adjuster in a cylindrical shape was used in the past experiments. In this report, a conical phase adjuster is introduced. The inside diameter in one side of the phase adjuster is 20.5 mm, and the other is 39.5 mm. The length of phase adjuster is 45 mm.The phase adjuster is placed in two ways; the larger inner diameter of the phase adjuster is placed in the left side and the smaller is in the right (with PA L39.5 R20.5); the smaller inner diameter is in the left and the larger is in the right (with PA L20.5 R39.5).The total length of the loop is 3.3 m and the phase adjuster is placed 1.125 m away from the upper end of the prime mover stack in the clockwise direction. The measurements are conducted in three conditions: without phase adjuster and with each phase adjusters. Both the sound pressure and sound intensity generated in the thermoacoustic system with phase adjuster are greater than those without phase adjuster. The biggest sound intensity is observed with PA L39.5 R20.5.

4:40

4pPAb5. Early onset of sound in Rijke tube with abrupt contraction. Konstantin Matveev and Rafael Hernandez (School of Mech. and Mater. Eng., Washington State Univ., MME School, WSU, Pullman, WA 99164-2920, matveev@wsu.edu)

Rijke tube is a system convenient for studying thermoacoustic instabilities both experimentally and theoretically. Common Rijke setups involve tubes with constant cross-sections. With the goal to reduce supplied heat necessary to excite acoustic modes, a segmented Rijke tube is constructed comprising two pipes of different diameters. Experiments with regulated mean flow and supplied heat demonstrate that thermoacoustic instability in the segmented tube occurs at about half of the heat addition rate required for sound onset in a tube with uniform cross-section. This finding suggests that simpler experimental means can be used for studying thermoacoustic instabilities due to significant reduction of required heat and highest temperatures in the system. Also, some practical devices with complicated resonators may appear to be more prone to instabilities of this sort. Stronger coupling between acoustic modes due to their enhanced non-orthogonality in segmented resonators can result in richer nonlinear effects. A simplified energy-based model is developed that predicts the onset of instability in both straight and segmented Rijke tubes.

Session 4pPP

Psychological and Physiological Acoustics and Speech Communication: Computational Modeling of Sensorineural Hearing Loss: Models and Applications

Michael G. Heinz, Cochair
Speech, Lang., and Hearing Sci. & Biomedical Eng., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907

Torsten Dau, Cochair
Ctr. for Appl. Hearing Res., Technical Univ. of Denmark, Kongens, Lyngby 2800, Denmark

Chair's Introduction—12:55

Invited Papers

1:00

4pPP1. Hearing impaired cochlear response simulation. Marcos F. Simon Galvez and Stephen J. Elliott (SPCG, ISVR, Rm. 3049, Bldg. 13, ISVR, Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, mfsg1e10@soton.ac.uk)

A model is introduced which allows the vibration of the basilar membrane to be estimated for different degrees of hearing loss. The model is based on a discrete lumped parameter model of the human cochlea, which uses a three dimensional description of the fluid coupling. The hearing losses are assumed to be caused by the combined malfunction of the outer hair cells (OHCs), the inner hair cells (IHCs), and the endocochlear potential driving the system. OHC loss and damage to endochoclear potential are modeled by a reduction of the cochlear amplifier gain, which is obtained by reducing the feedback gain of the OHCs. IHC loss is modeled as an overall reduction in basilar membrane response. The distribution of OHC and IHC loss along the cochlea are derived using an iterative method, which matches the output vibration amplitude of the model to that assumed to generate the hearing impaired audiogram.

1:20

4pPP2. Modeling disrupted tonotopicity of temporal coding following sensorineural hearing loss. Michael G. Heinz and Kenneth S. Henry (Speech, Lang., and Hearing Sci. & Biomedical Eng., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, mheinz@purdue.edu)

Perceptual studies suggest that sensorineural hearing loss (SNHL) affects neural coding of temporal fine structure (TFS) more than envelope (ENV). Although the "quantity" of TFS coding is degraded only in background noise, Wiener-kernel analyses suggest SNHL disrupts tonotopicity (i.e., the "quality") of TFS coding for complex sounds more than ENV coding. Specifically, auditory-nerve (AN) fibers in noise-exposed chinchillas can have their dominant TFS component located within their tuning-curve tail (i.e., the wrong place) while their ENV response remains centered at CF. Here, the ability of a AN model [Zilany and Bruce (2007)] to replicate this dissociation between TFS and ENV tonotopicity was evaluated. By varying the degree of outer- and inner-hair-cell damage, hypothesized factors such as hypersensitive tails and tip-to-tail ratio were evaluated. The model predicted the main trends in our physiological data: (1) no loss of tonotopicity for lower CFs without a clear tip/tail distinction, (2) more easily disrupted TFS tonotopicity than ENV (without requiring hypersensitive tails), and (3) disruption of both TFS and ENV tonotopicity for severely degraded tips. This computational approach allows exploration of the interaction between tip-tail ratio and phase-locking roll-off, and whether amplification strategies can restore cochlear tonotopicity. [Work supported by NIH grants R01-DC009838 and F32-DC012236.]

1:40

4pPP3. Physiological prediction of masking release for normal-hearing and hearing-impaired listeners. Ian C. Bruce (Dept. of Elec. & Comput. Eng., McMaster Univ., Rm. ITB-A213, 1280 Main St W, Hamilton, ON L8S 4K1, Canada, ibruce@ieee.org), Agnès C. Léger (Institut d'Etude de la Cognition, École normale supérieure, Paris, France), Brian C. Moore (Dept. of Experimental Psych., Univ. of Cambridge, Cambridge, United Kingdom), and Christian Lorenzi (Institut d'Etude de la Cognition, École normale supérieure, Paris, France)

Léger *et al.* [J. Acoust. Soc. Am. (2012)] measured the intelligibility of speech in steady and spectrally or temporally modulated maskers for stimuli filtered into low- (<1.5 kHz) and mid-frequency (1–3 kHz) regions. Listeners with high-frequency hearing loss but near to clinically normal audiograms in the low- and mid-frequency regions showed poorer performance than a control group with normal hearing, but showed preserved spectral and temporal masking release. Here, we investigated whether a physiologically accurate model of the auditory periphery [Zilany *et al.*, J. Acoust. Soc. Am. (2009)] can explain these masking release data. Intelligibility was predicted using the Neurogram SIMilarity (NSIM) metric of Hines and Harte [Speech Commun. (2010) and (2012)]. This metric can make use of either an "all-information" neurogram with small time bins or a "mean-rate" neurogram with large time bins. The average audiograms of the different groups of listeners from the study of Léger *et al.* were simulated in the model by applying different mixes of outer and/or inner hair cell impairment. Very accurate predictions of the human data for both normal-hearing and hearing-impaired groups were obtained from the all-information NSIM metric (i.e., taking into account phase-locking information) with threshold shifts produced predominantly by OHC impairment (and minimal IHC impairment).

2:00

4pPP4. Neural-scaled entropy as a model of information for speech perception. Joshua M. Alexander and Varsha Hariram (Speech, Lang., and Hearing Sci., Purdue Univ., Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, alexan14@purdue.edu)

Neural-Scaled Entropy (NSE) is an objective metric used to quantify "information" available in speech consequent hearing loss, hearing aid signal processing, and distortion from various environmental factors. One pursuit is to use NSE to find optimum hearing aid settings that maximize speech perception. Inspired by the Cochlear-Scaled Entropy model [Stilp *et al*., J. Acoust. Soc. Am. 2112–2126 (2010)], NSE uses the neural spike output at the inner hair cell synapse of an auditory nerve model [Zilany *et al*., J. Acoust. Soc. Am. 126, 2390–2412 (2009)]. Probability of spike output from fibers sampled at equidistant places along the model cochlea is computed for short duration time frames. Potential information is estimated by using the Kullback-Liebler Divergence to describe how the pattern of neural firing at each frame differs from preceding frames in an auto-regressive manner. NSE was tested using nonsense syllables from various perceptual studies that included different signal processing schemes and was compared to performance for different vowel-defining parameters, consonant features, and talker gender. NSE has potential to serve as a model predictor of speech perception, and to capture the effects of sensorineural hearing loss beyond simple filter broadening. [Work supported by NIDCD RC1DC010601.]

2:20

4pPP5. Modeling detection of 500-hertz tones in reproducible noise for listeners with sensorineural hearing loss. Laurel H. Carney (Biomedical Eng. and Neurobiology & Anatomy, Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu), Junwen Mao (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY), Kelly-Jo Koch (Biomedical Eng. and Neurobiology & Anatomy, Univ. of Rochester, Rochester, NY), and Karen A. Doherty (Commun. Sci. and Disord., Syracuse Univ., Syracuse, NY)

Detection of tones in reproducible noises provides detailed patterns of hit and false-alarm rates across sets of masker waveforms. Analysis of these detection patterns can identify the cues or combination of cues listeners use for detection in narrowband and wideband noise. Recent work has shown that diotic detection patterns of listeners with normal hearing (NH) are significantly correlated to energy and envelope cues; fine-structure cues also contribute for wideband maskers. Detection patterns are best predicted by an optimal cue-combination model based on signal-detection theory. In this study, listeners with mild to moderate sensorineural hearing loss (HL) were tested using the same waveforms. Their diotic detection patterns were best predicted by energy or envelope cues, with little contribution of fine-structure timing. Also, unlike NH patterns, predictions of HL patterns were rarely improved by an optimal combination of cues. For dichotic detection, NH patterns were better predicted by the slope of the interaural envelope difference (SIED) than by ITD or ILD cues. For HL patterns, the SIED cue, a nonlinear combination of ITD and ILD cues, generally did not predict detection patterns. These results illustrate differences between NH and HL listeners in the use and combination of cues for detection in noise.

2:40

4pPP6. A perceptual model of auditory deafferentation. Enrique A. Lopez-Poveda and Pablo Barrios (Inst. of Neurosci. of Castilla y Leon, Univ. of Salamanca, Calle Pintor Fernando Gallego 1, Salamanca, Salamanca 37007, Spain, ealopezpoveda@usal.es)

Overexposure to intense sound produces temporary threshold shifts but permanent loss of afferent nerve terminals. Here, we present a vocoder designed to explore the perceptual consequences of this type of damage. The basic idea is that the spike train produced by an individual auditory afferent resembles a stochastically digitized binary version of the stimulus waveform and that the quality of the waveform representation in the whole nerve depends on the number of aggregated spike trains. Sounds were processed by filtering them into ten adjacent frequency bands. For the signal in each band, multiple spike trains were then obtained in an attempt to mimic the different representations of that signal conveyed by different auditory afferents innervating a given cochlear region. The aggregated spike train was multiplied by the original signal to obtain an acoustic version of the simulated nerve waveform. Tone-in-noise and speech-in-noise perception tests were performed by young, normal-hearing listeners using different numbers of afferents per frequency band. Results support that deafferentation impairs perception in noise more than in quiet. The proposed vocoder may be extended to model other types of hearing damage and to guide the design of hearing aids and cochlear implants.

3:00–3:20 Break

3:20

4pPP7. Understanding hearing impairment through model predictions of brainstem responses. Sarah Verhulst, Hari Bharadwaj (Auditory Neurosci. Lab, Boston Univ., 677 Beacon St., Boston, MA 02215, save@bu.edu), Golbarg Mehraei (Boston, Massachusetts), and Barbara Shinn-Cunningham (Auditory Neurosci. Lab, Boston Univ., Boston, MA)

Latencies of auditory brainstem response (ABR) wave-V decrease with increasing stimulus level, an effect often ascribed to broadened auditory filters. Following this hypothesis, hearing-impaired subjects with broad auditory filters should exhibit shorter wave-V latencies than normal-hearing listeners. Hearing anomalies resulting from the preferential degradation of low spontaneous rate (LS) auditory nerve (AN) fibers with intact thresholds have recently received attention. However, their effect on the ABR wave-V latency are yet to be elucidated. Here, a model of ABR investigates the relationships between wave-V latency and various forms of hearing damage. ABR wave-Vs are predicted from a model consisting of a nonlinear cochlear model (Verhulst *et al*., J. Acoust. Soc. Am. (in press)], an AN synapse model [Zilany *et al*., J. Acoust. Soc. Am. **126** (2009)], and a model of the cochlear nucleus (CN) and IC [Nelson and Carney, J. Acoust. Soc. Am. **116** (2004)]. Simulations predict that level changes cause smaller latency shifts in AN than in the IC, likely due to how inhibition/excitation shapes CN and IC responses. Furthermore, the increase in wave-V latency with decreasing click-to-noise ratios is predicted from LS fiber responses at low click-to-noise ratios. Preliminary simulation results suggest that wave-V latencies at different click-to-noise ratios may help diagnose LS damage.

3:40

4pPP8. Computational modeling of tinnitus development. Roland Schaette (UCL Ear Inst., 332 Gray's Inn Rd., London WC1X 8EE, United Kingdom, r.schaette@ucl.ac.uk)

Animal models and human neuroimaging studies have shown that tinnitus is generated through pathologically altered spontaneous activity of neurons in the central auditory system. Sensorineural hearing loss has been identified as an important trigger for the development of these aberrant patterns of neuronal activity, but the functional mechanisms that underlie this process have not yet been pinpointed. Using computational models, we have investigated which neuronal plasticity mechanisms could account for the development of neuronal correlates of tinnitus after hearing loss. We could show that a model based on the principle of activity stabilization through homeostatic plasticity can explain the development of neuronal hyperactivity as observed in animal studies. Moreover, the model's predictions of tinnitus frequencies from the audiograms of patients with noise-induced hearing loss and tonal tinnitus are close to the observed tinnitus pitch. The model thus proposes a specific mechanism for how plasticity in the central auditory system could lead to the development of tinnitus after cochlear damage. The model also predicts that central auditory structures may show increased response gain, which could explain why tinnitus and hyperacusis often occur together. Moreover, the homeostasis model is consistent with recent experimental findings from tinnitus patients with normal audiograms, and it explains why auditory deprivation through an earplug can lead to the occurrence of phantom sounds.

4:00

4pPP9. An auditory model for intelligibility and quality predictions. James Kates (Speech Lang. Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

The perceptual effects of audio processing in devices such as hearing aids can be predicted by comparing auditory model outputs for the processed signal to the model outputs for a clean reference signal. This paper presents an improved auditory model that can be used for both intelligibility and quality predictions. The model starts with a middle-ear filter, followed by a gammatone auditory filter bank. Two-tone suppression is provided by setting the bandwidth of the control filters wider than that of the associated analysis filters. The analysis filter bandwidths are increased in response to increasing signal intensity, and compensation is provided for the variation in group delay across the auditory filter bank. Temporal alignment is also built into the model to facilitate the comparison of the unprocessed reference with the hearing-aid processed signals. The amplitude of the analysis filter outputs is modified by outer hair-cell dynamic-range compression and inner-hair cell firing-rate adaptation. Hearing loss is incorporated into the model as a shift in auditory threshold, an increase in the analysis filter bandwidths, and a reduction in the dynamic-range compression ratio. The model outputs include both the signal envelope and scaled basilar-membrane vibration in each auditory filter band.

4:20

4pPP10. Modeling loudness for impaired ears and applications to fitting hearing aids. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Models of loudness for impaired ears are based on the assumption that cochlear hearing loss can be partitioned into a component due to outer hair cell dysfunction, resulting in reduced frequency selectivity and a more rapid growth of neural response with increasing level (reduced compression), and a component due to inner hair cell dysfunction, which reduces the neural response at all levels. In the first two models that were developed in Cambridge, the filtering and compression that take place on the basilar membrane were modeled as sequential processes, which is not physiologically realistic. Nevertheless, the models were able to account for many aspects of loudness, and were used to develop methods of fitting multi-channel compression hearing aids that have proven to be effective. More recently, a model of loudness has been developed in which the filtering and compression are modeled using a physiologically plausible nonlinear filter bank. This has also been applied to the fitting of hearing aids. Factors not included in the models include central plasticity resulting from altered auditory input, possible consequences of the operation of the efferent system, and the influence of cognitive factors such as perceived distance of the sound source and perceived vocal effort.

4:40

4pPP11. Modeling music perception in impaired listeners. Martin McKinney, Kelly Fitz (Starkey Hearing Technologies, 6600 Washington Ave S, Eden Prairie, MN 55344, martin_mckinney@starkey.com), Sridhar Kalluri, and Brent Edwards (Starkey Hearing Res. Ctr., Berkeley, CA)

We employ computational models of loudness and pitch perception to better understand the impact of sensorineural hearing loss on music perception, with the aim of guiding technology development for hearing-impaired listeners. Traditionally, hearing aid development has been geared towards improving speech intelligibility and has largely failed to provide adequate restoration of music to those with hearing loss. One difficulty with trying to improve music perception in impaired listeners is the absence of a good quantitative measure of music reception, analogous to speech reception measures like word-recognition rate. Psychoacoustic models for loudness and pitch allow us to guage quantitative parameters relevant to music perception and make predictions about the type of deficits listeners face. We examine the impact of hearing loss to predicted measures of loudness, specific loudness, pitch, and consonance and make suggestions on possible methods for restoration.

5:00

4pPP12. A model-based hearing aid: Psychoacoustics, models, and algorithms. Stephan D. Ewert, Steffen Kortlang, and Volker Hohmann (Medizinische Physik, Universität Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, Stephan.ewert@uni-oldenburg.de)

In hearing aids, amplification and dynamic range compression typically aim at compensating the deficits associated with outer-hair cell (OHC) loss. Nevertheless, success shows large inter-individual variability and hearing-impaired listeners generally still have considerable problems in complex acoustic communication situations including noise and reverberation. These problems could be related to inner-hair cell (IHC) damage and reduced frequency selectivity resulting in a loss of spectro-temporal coding fidelity. Here a model-based, fast-acting dynamic compression algorithm which aims at approximating the normal-hearing BM input-output function in hearing-impaired listeners is suggested. The algorithm is fitted by estimating low-level gain loss (OHC loss) from adaptive categorical loudness scaling data and audiometric thresholds based on Ewert and Grimm [*Proc. ISAAR* (2012)] and Jürgens *et al*. Hear. Res. **270**, 177 (2011)]. Aided speech intelligibility was measured in stationary and fluctuating noise and related to the estimated OHC loss. To improve diagnostics of OHC and IHC loss, a series of five psychoacoustic measurements was conducted aiming at a direct quantification of IHC damage in a group of six young and elderly normal-hearing and 12 hearing-impaired listeners. A model is suggested to account for the temporal fine-structure detection and discrimination data. [Work funded by BMBF 01EZ0741 and DFG FOR1732.]

THURSDAY AFTERNOON, 6 JUNE 2013 512CG, 1:00 P.M. TO 5:00 P.M.

Session 4pSA

Structural Acoustics and Vibration: Applications in Structural Acoustics and Vibration III

Linda P. Franzoni, Cochair
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James E. Phillips, Cochair
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Contributed Papers

1:00

4pSA1. Dynamics and stability of pneumatically isolated systems. Vyacheslav Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vyacheslav.ryaboy@newport.com)

Pneumatic vibration isolation is the most widespread effective method for creating vibration-free environments that are vital for precise experiments and manufacturing operations in optoelectronics, life sciences, microelectronics, nanotechnology, and other areas. The modeling and design principles of a dual-chamber pneumatic vibration isolator continue to attract attention of researchers. On the other hand, behavior of *systems* of such isolators was never explained in the literature in sufficient detail. After a brief summary of the theory and a model of a single standalone isolator, the dynamics of a system of isolators supporting a payload is considered with main attention directed to three aspects of their behavior: first, the static stability of payloads with high positions of the center of gravity; second, role of gravity terms in the vibration transmissibility; third, the dynamic stability of the feedback system formed by mechanical leveling valves. The direct method of calculating the maximum stable position of the center of gravity is presented and illustrated by three-dimensional stability domains. A numerical method for feedback stability analysis of self-leveling valve systems is provided, and the results are compared with the analytical estimates for a single isolator. The relation between the static and dynamic phenomena is discussed.

1:20

4pSA2. Dissipative effects in the response of an elastic medium to a localized force. Douglas Photiadis (NRL, 4555 Overlook Ave. SW, Washington, DC 20375, douglas.photiadis@nrl.navy.mil)

The effect of dissipation on the real part of the admittance of an elastic half-space is typically thought to be unimportant if the loss factor of the elastic medium is small. However, dissipation induces losses in the near field of the source and, provided the size of the source is small enough, this phenomenon can be more important than elastic wave radiation. Such losses give rise to a fundamental limit in the quality factor of an oscillator attached to a substrate. Near field losses associated with strains in the elastic substrate can actually be larger than intrinsic losses in the oscillator itself if the internal friction of the substrate is larger than the internal friction of the oscillator. [Research sponsored by the Office of Naval Research.]

1:40

4pSA3. Modal active control applied to simplified string musical instrument. Simon Benacchio, Adrien Mamou-Mani (Instrumental Acoust., IRCAM, 1 place Stravinsky, Paris, France, simon.benacchio@ircam.fr), Baptiste Chomette, and François Ollivier (Institut d'Alembert, Universite Pierre et Marie Curie, Paris, France)

This study aims to control the vibrational eigenmodes of soundboards in order to modify the timbre of string instruments. These structures are wooden plates of complex shape, excited by a string through a bridge. Their modal parameters are first identified using modal analysis algorithms on experimental measurements. Then a digital controller is designed using these parameters and classic active control methods. The effects of this controller are first studied thanks to time simulation. Prior to applying experimentally this controller, an optimization procedure is carried out to determine the quantity, dimensions and positions of sensors and actuators needed for the control. These best possible specifications are obtained according to the controllability, observability and other optimization criteria. Finally, a real time system using the control procedure is tested on a simplified musical instrument. The experiment is conducted on a rectangular spruce plate, clamped at its boundary and excited by means of a single string. This simple case study is presented here and its results are discussed in terms of eigenmodes modifications.

2:00

4pSA4. Simulation of structural dynamics using a non-polynomial method. Teemu Luostari, Tomi Huttunen (Dept. of Appl. Phys., Univ. of Eastern Finland, P.O. Box 1627, Kuopio 70211, Finland, teemu.luostari@uef.fi), and Peter Monk (Dept. of Mathematical Sci., Univ. of Delaware, Newark, DE)

In structural dynamics, the modeling of steady-state thin plate bending is an important but, especially at high frequencies, computationally challenging problem. When solving the displacement of an elastic thin plate, a fourth order partial differential equation (Kirchhoff's plate equation) needs to be solved. In addition, two boundary conditions are needed in order to uniquely solve the problem. Polynomial methods, such as the finite element method (FEM) and discontinuous Galerkin method (DGM), are generally used to solve the plate dynamics. At higher frequencies the computational burden of a low order FEM becomes rapidly unbearable. Consequently, non-polynomial modeling methods are investigated because of their

capability to solve the problem more efficiently than the standard FEM. The non-polynomial method used in this study is called the ultra weak variational formulation (UWVF). The UWVF uses finite element meshes and it is essentially an upwind DGM with a special choice of basis functions. To date, the UWVF has been successfully used in electromagnetism, acoustics and linear elasticity. We shall show, using a mixture of theory and numerical examples, that the UWVF is feasible for thin plate problems. For these problems the UWVF basis consists of plane wave and evanescent (corner) wave functions.

2:20

4pSA5. Impact of mass ratio and bandwidth on apparent damping of a harmonic oscillator with subordinate oscillator array. Aldo A. Glean, Joseph F. Vignola, John A. Judge, and Teresa J. Ryan (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, 10glean@cardinalmail.cua.edu)

The response of a lightly damped resonator with a set of substantially less massive attached oscillators has been studied. The collection of attached oscillators is known as a subordinate oscillator array (SOA). An SOA can function as an energy sink, extracting vibration energy from the primary mass and thus adding apparent damping to the system. We have shown that the limit of apparent damping achievable for this class of system is the inverse of non-dimensional bandwidth (ratio of the bandwidth to the fundamental frequency of the primary oscillator). In practice, the utility of this result is limited because a great deal of mass (~25% of primary) is required to approach critical damping. The mass of the subordinate set required to achieve the most rapid energy transfer from the primary is proportional to the non-dimensional bandwidth squared. Low apparent Q is achieved by increasing non-dimensional bandwidth. The presentation will describe numerical optimizations that investigate the impact of the SOA bandwidth, the mass ratio (the ratio of the total mass of the SOA to the mass of the primary structure) and the apparent damping of the system.

2:40

4pSA6. Imaging crack orientation using the time reversed elastic nonlinearity diagnostic with three component time reversal. Brian E. Anderson, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, bea@lanl.gov)

The time reversed elastic nonlinearity diagnostic (TREND) is a method to allow one to nondestructively evaluate a sample by locating nonlinear scatterers. In the TREND method one creates a localized focus of energy using time reversal at each point of interest. The localized nature of the focus, which is at a higher energy level relative to the wave field nearby thereby amplifying the potential nonlinear signature of the focal location, allows one to image localized nonlinearities. It has also been shown that a focus of energy may be individually created in each of the three independent vector components of vibration using time reversal. Here we show that the use of TREND scans in each of the three vector component directions allows imaging of a crack's orientation. This work is conducted on steel samples, each with cracks at known orientations that were created in a controlled manner. The scaling subtraction method is also used at each scan point to classify the nonlinearity. [Work supported by Institutional Support (LDRD) at the Los Alamos National Laboratory.]

3:00–3:20 Break

3:20

4pSA7. Shock dynamics of random structures. Mauro Caresta, Robin S. Langley, and Jim Woodhouse (Engineering, Univ. of Cambridge, Trumpington St., Cambridge CB21PZ, United Kingdom, maurorestaca@yahoo.it)

Predicting the response of a structure following an impact is of interest in situations where parts of a complex assembly may come into contact. Standard approaches are based on the knowledge of the impulse response function, requiring the knowledge of the modes and the natural frequencies of the structure. In real engineering structures the statistics of higher natural frequencies follows those of the Gaussian Orthogonal Ensemble, this allows the application of random point process theory to get a mean impulse response function by the knowledge of the modal density of the structure. An ensemble averaged time history for both the response and the impact force can be predicted. Once the impact characteristics are known in the time domain, a simple Fourier Transform allows the frequency range of the impact excitation to be calculated. Experimental and numerical results for beams, plates, and cylinders are presented to confirm the validity of the method.

3:40

4pSA8. Dynamic analysis of annular sector plate with general boundary supports. Dongyan Shi, Xianjie Shi (College of Mech. and Elec. Eng., Harbin Eng. Univ., No 145, Nantong St., Harbin, Heilongjiang 150001, China, dongyanshi@gmail.com), Wen L. Li (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI), and Qingshan Wang (College of Mech. and Elec. Eng., Harbin Eng. Univ., Harbin, China)

Dynamic behavior of annular sector plate is an important research topic since they have been extensively used in practical engineering applications. However, the dynamic analysis of annular sector plates with general boundary supports is rarely studied in literature. In this investigation, an analytical method is presented for the vibration analysis of annular sector plates with general elastic boundary supports. Unlike most existing framework, arbitrary elastic boundary supports can be easily realized by setting the stiffness of the two types restraining springs. The displacement field is universally expressed as a new form of trigonometric series expansions with a drastically improved convergence as compared with the conventional Fourier series. Mathematically, such a double Furies series is capable of representing any function (including the exact displacement solution) whose third-order partial derivatives are continuous over the area of the plate. Thus, the double Fourier series solution to the dynamic analysis of the structure is obtained by employing the Raleigh-Ritz method. The accuracy and reliability of the current method are validated by both FEA and reference results under various boundary conditions. The present method can be directly applied to other more complicated boundary conditions and other shape plates.

4:00

4pSA9. Structural element vibration analysis. Kamel Falek, Lila Chalah-Rezgui, Farid Chalah (Faculty of Civil Eng., Usthb, Algiers 16111, Algeria, kfalekgc@yahoo.fr), Abderrahim Bali (Polytechnic National School of Algiers, Algiers, Algeria), and Amar Nechnech (Faculty of Civil Eng., Usthb, Algiers, Algeria)

Various approaches are usually used in the dynamic analysis of beams vibrating transversally. For this, numerical methods allowing the solving of the general eigenvalue problem are utilized. The equilibrium equations, describing the movement, result from the solution of a fourth order differential equation. Our investigation is based on the finite element method. The findings of these investigations are the vibration frequencies, obtained by the Jacobi method. Two types of elementary mass matrix are considered, representing a uniform distribution of the mass along the element and concentrated ones located at fixed points whose number is increased progressively separated by equal distances at each evaluation stage. The studied beams have different boundary constraints representing several classical situations. Comparisons are made for beams where the distributed mass is replaced by n concentrated masses. As expected, the first calculus stage is to obtain the lowest number of the beam parts that gives a frequency comparable to that issued from the Rayleigh formula. The obtained values are then compared to theoretical results based on the assumptions of the Bernoulli-Euler theory. These steps are used after for the second type mass representation in the same manner.

4:20

4pSA10. Study on error of vibration intensity measurement in a flat plate with multiple energy flows entering from outside. Mototaka Hibinaga and Masato Mikami (Grad. School of Sci. and Eng., Yamaguchi Univ., 2-16-1 Tokiwadai, Ube, Yamaguchi, Japan, 1-20-308 Yamakado, Ube, Yamaguchi, Japan, Ube, Yamaguchi 755-0097, Japan, s032ve@yamaguchi-u.ac.jp)

Vibration Intensity (VI) method can show flows of vibration energy in surface of mechanical structures as a vector quantity and is one of the techniques to identify vibration source and vibration transmission path. Some previous studies have showed that if there are multiple vibration sources in the analyzed plate, VI causes error due to superposition of multiple waves. In actual structures, however, the vibration source often exists outside of the analyzed plate which is surrounded by the boundary with reflection, such as curved parts and staged parts. The VI error has not been investigated in such

cases close to actual structures. The purpose of this research is to investigate the influence of superposition of multiple waves which come across the stepped boundary into a flat plate on VI by using finite element method analysis. This model also simulates the case with the superposition of progressive wave entering from the outside of the plate and reflected wave caused at a boundary of the plate. The results of calculation showed that the superposition of waves causes a difference between VI and real vibration energy flow, depending on frequency and plate size.

4:40

4pSA11. Mid-frequency vibrations of a double-leaf plate with random inhomogeneities. Hyuck Chung (SCSM, Auckland Univ. of Technol., PB 92006 Auckland, Auckland 1142, New Zealand, hchung@aut.ac.nz)

Predicting vibrations of composite structures such as double-leaf plates is difficult because of the large number of components and random inhomogeneities in the components. In the low and high frequency ranges, the components may be homogenized, and consequently the model of a structure becomes simple enough to be mathematically and computationally tractable. However the vibrations in the mid-frequency range cannot be predicted using such methods because the wavelengths are comparable to the size of the components and junctions between components. Simply adding more details, e.g., higher resolution in finite element mesh, will not result in more accurate predictions. In this paper a double-leaf plate is modeled using the Kirchhoff plate and Euler beam theories. The elastic moduli and junctions are allowed to be inhomogeneous over the plates and beams. These inhomogeneities are simulated as continuous smooth random functions rather than series of discrete random numbers. The random functions are incorporated into the model using the variational formulation and the Fourier expansion of the vibration field. Various probability density functions are tested for the inhomogeneities. Then the distribution of resonant frequencies and the vibration field are studied and compared with other models.

THURSDAY AFTERNOON, 6 JUNE 2013 — 515ABC, 1:00 P.M. TO 3:20 P.M.

Session 4pSCa

Speech Communication: Auditory Feedback in Speech Production II

Anders Lofqvist, Cochair
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Charles R. Larson, Cochair
Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208

Invited Papers

1:00

4pSCa1. Cortical mechanisms of integrating auditory feedback with vocal pitch control. Jean Mary Zarate (Psychology, New York Univ., 6 Washington Place, Rm. 275-276, New York, NY 10003, jean.m.zarate@nyu.edu)

Precise vocal pitch regulation is crucial for both speech and song. The pitch of a speaker's voice can indicate the intent of a sentence, set the emotional context of a conversation, or distinguish meanings in tonal languages. In singing, accurate vocal pitch is the single most important element needed to properly produce notes and melodies. Vocal pitch regulation requires the integration of auditory feedback processing with the vocal motor system, also known as audio-vocal integration; however, the neural substrates governing this integration have been elusive. Recent functional magnetic resonance imaging (fMRI) studies of singing with pitch-shifted feedback are presented here to outline the neural mechanisms of audio-vocal integration for voluntary vocal pitch regulation, and to discuss the effects of long-term vocal training on vocal performance and neural activity during vocal pitch regulation.

1:20

4pSCa2. Cortical plasticity in the sensorimotor control of voice induced by auditory cognitive training. Hanjun Liu, Weifeng Li, and Zhaocong Chen (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

Multiple lines of evidence have shown that motor-related brain regions can be activated during passively musical listening or beat perception, indicating a connection between auditory and sensorimotor systems. In the present study, we sought to examine whether the neural processing of auditory-vocal integration can be shaped by short-term cognitive training related to auditory attention and working memory. Auditory cognitive training consisted of a ten-day backward digit span task, in which digits embedded in various noise at different SNR levels were presented and subjects were required to repeat the digits in the reverse order. Before and after the cognitive training, subjects also participated in a vocal motor task, in which they heard their pitch auditory feedback unexpectedly altered upwards (50 and 200 cents) during sustained vocalization and their neurophysiological responses were recorded. The results revealed a significantly improved performance on the backward digit span task after the training. Moreover, cortical responses indexed by P2 amplitude to pitch perturbations in voice auditory feedback were significantly increased after the training compared with those before the training. These findings provide evidence that plastic cortical changes in the sensorimotor control of voice can be caused by auditory cognitive training.

1:40

4pSCa3. Neural evidence for state feedback control of speaking. John F. Houde (Otolaryngol. – Head and Neck Surgery, Univ. of California San Francisco, 513 Parnassus Ave., S362, San Francisco, CA 94143, houde@phy.ucsf.edu) and Srikantan S. Nagarajan (Radiology, Univ. of California San Francisco, San Francisco, CA)

An important recent development in neuroscience has been the use of models based on state feedback control (SFC) to explain the role of the central nervous system in motor control. In SFC, control is based on internal feedback of an estimate of the dynamic state of the thing (e.g., arm) being controlled. Within the internal loop, the state is predicted from outgoing motor commands and corrected by comparing the feedback expected to result from this state with actual incoming sensory feedback. SFC has received scant attention in the speech community, but the indirect role it suggests for feedback can account for much of what is known about the role of feedback in speech motor control. Our lab has been investigating how well SFC also accounts for the neural correlates of auditory feedback processing during speaking. Our principal approach has used magnetoencephalography to record the cortical activity of speakers as they hear themselves speaking, but recently, we have also completed an auditory feedback study based on electrocorticography. Many of the results of these studies have supported the SFC model, but some have posed challenges for it, which will be discussed. [Work supported by NSF grant BCS-0926196 and NIH grant R01-DC010145.]

2:00–2:20 Break

2:20

4pSCa4. Intentionality and categories in speech motor control. Takashi Mitsuya (Psychology, Queen's Univ., 62 Arch St., Humphrey Hall, Kingston, ON K7L3N6, Canada, takashi.mitsuya@queensu.ca) and Kevin G. Munhall (Psych. & Otolaryngol., Queen's Univ., Kingston, ON, Canada)

Actions are organized around goals or intentions. In speech production, there has been no agreement on how best to discuss speech goals. However, the auditory feedback perturbation methodology provides a window into the nature of speech goals. To the extent that subjects are sensitive to variation in an acoustic attribute, this attribute must be part of the controlled intention of articulation. In this presentation, we will review a series of studies that speak to this issue. In one study, we examined how intentionality of speech production influences compensatory formant production by instructing subjects to use a cognitive strategy in order to make the feedback sound consistent with the intended vowel. In other studies, we have explored the specificity of vowel formant compensation by comparing cross-language differences. The results indicate that speech goals are (1) very specific, defined by a phonemic category and its relationship with neighboring categories, and (2) multivariate. We will discuss these results by contrasting compensatory behaviors in reaching and limb movements to those observed in speech studies. The presence of a system of categories in speech may result in differences in the way speech goals are represented.

2:40

4pSCa5. Exploring auditory-motor interactions in normal and disordered speech. Jason A. Tourville, Shanqing Cai, and Frank H. Guenther (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 677 Beacon St, Boston, MA 02215, jtour@bu.edu)

Auditory feedback plays an important role in speech motor learning and in the online correction of speech movements. Speakers can detect and correct auditory feedback errors at the segmental and suprasegmental levels during ongoing speech. The frontal brain regions that contribute to these corrective movements have also been shown to be more active during speech in persons who stutter (PWS) compared to fluent speakers. Further, various types of altered auditory feedback can temporarily improve the fluency of PWS, suggesting that atypical auditory-motor interactions during speech may contribute to stuttering disfluencies. To investigate this possibility, we have developed and improved Audapter, a software that enables configurable dynamic perturbation of the spatial and temporal content of the speech auditory signal in real time. Using Audapter, we have measured the compensatory responses of PWS to static and dynamic perturbations of the formant content of auditory feedback and compared these responses with those from matched fluent controls. Our findings indicate deficient utilization of auditory feedback by PWS for short-latency online control of the spatial and temporal parameters of articulation during vowel production and during running speech. These findings provide further evidence that stuttering is associated with aberrant auditory-motor integration during speech.

3:00

4pSCa6. The role of auditory feedback in speech development: A study of compensation strategies for a lip-tube perturbation. Lucie Menard (Phonet. Lab., Universite du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca), Pascal Perrier (Speech and Cognition, GIPSA-lab, Grenoble, France), and Jerome Aubin (Phonet. Lab., Universite du Quebec a Montreal, Montreal, QC, Canada)

The role of auditory feedback in speech development was investigated through a study of compensation strategies for a lip-tube perturbation. Acoustic, articulatory, and perceptual analyses of the vowels /i/, /y/, and /u/ produced by ten 4-year-old French speakers and ten adult French speakers were conducted under two conditions: normal and with a 15-mm-diameter tube (for /y/ and /u/) or a 5-mm-diameter tube (for /i/) inserted between the lips. Ultrasound and acoustic recordings of isolated vowels were made in normal condition before any perturbation (N1), for each of the 20 trials in the perturbed condition (P), and in normal condition after the perturbed trials (N2). Data reveal that adult participants moved their tongue in the P condition more than children subjects, to compensate for F1 and F2 alteration induced by the tube. Except for /y/, the perturbation was generally at least partly compensated during the perturbed trials in adults and children, but children did not show a typical learning effect. Results are analyzed from the perspective of (i) goal specification in speech production in the acoustic and/or somatosensory domain, and (ii) the maturity of representations of the motor apparatus in the brain.

Session 4pSCb

Speech Communication: Production and Perception I: Beyond the Speech Segment (Poster Session)

Sam Tilsen, Chair
Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

4pSCb1. Prosodic effects on speech gestures: A shape analysis based on functional data analysis. Christine Mooshammer (Linguist Dept., USC, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693, mooshamm@usc.edu), Lasse Bombien (Inst. for Phonet. and Speech Processing, LMU, Munich, Germany), and Jelena Krivokapic (Linguist Dept., Yale, New Haven, CT)

Prosodic phrasing and prominence modulate the production of speech gestures by increasing their duration and often also the amplitude. Here we investigate whether the shape of bilabial and dorsal constriction and release gestures is affected by phrase boundary strength and lexical stress. Articulatory movements of four speakers of German were recorded with 3D EMA. Eight bisyllabic test words starting with the clusters /kn, kl, ps, pl/ and either stressed on the first or on the second syllable were embedded in sentences that elicited phrase boundaries of different strengths. Vertical tongue back movements and lip aperture from the constriction onset to the release offset of the initial consonant were extracted and time and amplitude normalized. In order to investigate shape differences a functional principal component analysis [see Ramsay and Silverman (2002)] was performed. The resulting factor scores quantify gestural shape differences. The results indicate that boundary strength affects the skewness, i.e., targets are achieved later for stronger preceding boundaries. Furthermore, movement curves of initial consonants in unstressed syllables were more peaked than in stressed ones. By applying this method, global shape differences due to prosodic modulation of articulatory gestures can be extracted without recourse to specific landmarks. [Work supported by NIH DC03172.]

4pSCb2. Exploring prosodic boundaries: Gradiency and categoricity of prosodic boundaries in articulation. Jelena Krivokapic (Linguistics, Yale Univ., 370 Temple St., 204, New Haven, CT 06520-8366, jelena.krivokapic@yale.edu), Christine Mooshammer (Linguistics, Univ. of Southern California, Los Angeles, CA), and Mark Tiede (Haskins Labs., New Haven, CT)

The prosodic hierarchy is a core concept of prosodic theory. Despite this, the number of categories in the hierarchy, and the structural relationships between them, is not clear. For English, a major and a minor category above the word level is usually assumed, and some studies have suggested additional categories [cf. Shattuck-Hufnagel and Turk (1996)]. Each of these category levels is assumed to be marked by categorically distinct boundaries, but experimental evidence for this view is sparse. In this work, EMA is used to investigate this notion of the prosodic hierarchy [following a preliminary analysis in Krivokapić and Ananthakrishnan (2007)]. Seven subjects read six repetitions of 48 sentences, each containing one, two or three prosodic boundaries, for a total of 56 boundaries per speaker. The predicted boundary strength varied from a weak clitic boundary to a strong sentential boundary. The produced boundaries are evaluated using a Gaussian mixture model analysis. The results bear on the question whether prosodic boundaries behave categorically (i.e., prosodic boundary values cluster within a small number of categories), or in a gradient manner (i.e., they are more evenly spread), thus supporting an alternative view of the prosodic hierarchy. [Work supported by NIH.]

4pSCb3. Acoustic characteristics of intervocalic stop lenition in American English. Dominique Bouavichith and Lisa Davidson (Linguistics, New York Univ., 10 Washington Place, New York, NY 10003, bouavichith@nyu.edu)

Descriptions of English and other languages have claimed that intervocalic stops are often lenited to fricatives or approximants in connected speech, but few acoustic analyses of factors that affect lenition have been reported for American English [cf. Lavoie (2001)]. In this analysis, intervocalic voiced stops produced in bi- and trisyllabic words during story reading are examined (participants $N = 14$). The first result shows that American English speakers never lenite to fricatives, but rather produce approximants whenever lenition occurs. Second, stress plays an essential role: 51% of stops are lenited when stress is on the first syllable (e.g., "yoga"), but only 7% of stops lenite when stress is on the second syllable (e.g., "begin"). Overall, approximant productions are significantly higher for /d/ (which becomes a flap–63%) and /g/ (70%) as compared to /b/ (43%). For both stress placements, stop productions are longer and lower in intensity than approximant productions. Another significant factor is frequency, as higher frequency words are produced as approximants more often. These acoustic findings are generally consistent with Kingston's (2006) claim that lenition within words and phrases occurs to minimize interruptions to the prosodic unit and indicate that the current constituent is ongoing.

4pSCb4. Variability attenuates sensitivity to acoustic detail in cross-language speech production. Sean Martin, Lisa Davidson (Linguistics, New York Univ., 10 Washington Place, New York, NY 10003, sean.martin@nyu.edu), and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Baltimore, MD)

In the production of non-native consonant clusters, speakers' systematic errors have been attributed to the influence of native-language phonotactics [Dupoux *et al*. (1999)]. However, recent models of non-native speech production suggest that speakers are also sensitive to acoustic details [Wilson *et al*. (2012)]. We examine whether speakers' sensitivity to phonetic detail is modulated by variability in the speech signal, and whether they abstract away from subphonemic detail given sufficient variability. This was tested by presenting English speakers with ill-formed clusters (e.g., bdafa, tmape, zgade) containing systematically manipulated sub-phonemic acoustic properties: stop burst duration and amplitude for stop-initial clusters, and the presence/absence of pre-obstruent voicing (POV) for voiced clusters. In experiment 1, which presented stimuli produced by a single Russian talker, significant effects were found for the duration manipulations on the rates of epenthesis, the amplitude manipulation on consonant change/deletion errors, and the POV manipulation on the rate of prothesis. In experiment 2, which contained stimuli produced by three talkers, there was a substantial attenuation of the influence of the acoustic manipulations on speakers' productions. These results suggest that an account of non-native speech production that models the relative contribution of phonotactics and phonetic detail must incorporate information about variability in the environment.

4pSCb5. Deriving functional load of phonemes from a prosodically extended neighborhood analysis. Mafuyu Kitahara (School of Law, Waseda Univ., 1-6-1 Nishiwaseda, Shinjuku-Ku, Tokyo 1698050, Japan, kitahara@waseda.jp), Keiichi Tajima (Dept. of Psych., Hosei Univeristy, Tokyo, Japan), and Kiyoko Yoneyama (Dept. of English Lang., Daito Bunka Univeristy, Tokyo, Japan)

The functional load of phonemes is a long-standing, but not a mainstream notion in modern linguistics: that some pairs of phonemes distinguish more words than other pairs is intuitively plausible, but hard to quantify. Meanwhile, neighborhood effects in word recognition and production have been one of the central topics in psycholinguistics, leading to a wide variety of investigations. However, the Greenberg-Jenkins calculation, the most common definition of phonological neighborhood, deals only with deletion, addition, and substitution of phonemes, lacking any consideration of prosody. For example, homophones, which cannot be segmental neighbors and thus excluded in most neighborhood research, can be distinctive if lexical accent is specified. The role of onset/rhyme distinction in neighborhood calculation has been discussed, but morae, another basic unit of prosody, were not mentioned in the literature. We propose a novel method for calculating the functional load based on a prosodically extended neighborhood analysis. It is a frequency-weighted neighborhood density summed across neighbors for a particular phoneme. Accentual distinctions, morae or syllables, and context effects within a word are taken into account. The proposed method gives a better account for the difference in the acquisition order of segments across languages. [Work supported by JSPS.]

4pSCb6. Durational characteristics of English by Chinese learners of English: A case of the northeast dialect speakers of Chinese. Kiyoko Yoneyama (Daito Bunka Univ., 1-9-1 Takashimadaira, Itabashi, Tokyo 175-8571, Japan, yoneyama@ic.daito.ac.jp)

This study examined the durational patterns of English production by Chinese learners of English. The same production experiment of Mochizuki-Sudo and Kiritani (J. Phonet. **19**, 231–248) was conducted to investigate how Chinese speakers of English control the durational properties of inter-stressed intervals (ISIs) and target stressed vowels by compressing the stressed vowels when unstressed syllables are added. Five male and three female English non-proficient Chinese speakers of the northeast dialect participated in the study. They were all recruited from Changchun city in Jilin province of China. They graduated from the same high school, and they did not have any experience of oversea study in any English speaking country. Durations of the ISIs and the target vowel were analyzed. The tentative analysis revealed that the ISI durations produced by the non-proficient Chinese learners of English showed the similar durational patterns like the non-proficient Japanese learners of English did in Mochizuki-Sudo and Kiritani (1991). The analysis of stressed vowel durations showed that when they produced the vowel durations, they didn't show the similar durational patterns like the non-proficient Japanese, but those to the American speakers. The implications from the results will be discussed in the paper. [Work supported by JSPS.]

4pSCb7. Auditory free classification of nonnative speech by nonnative listeners. Eriko Atagi and Tessa Bent (Speech and Hearing Sci., Indiana Univ., 200 S Jordan Ave., Bloomington, IN 47405, eatagi@indiana.edu)

Nonnative listeners are less accurate than native listeners at classifying talkers by regional dialect [Clopper and Bradlow (2009)]. This decrement may be due to less robust knowledge about the underlying sound structure of the target language or less extensive experience with socio-cultural phonetic variation in the target language. To disentangle the contribution of these two factors, this study examined native and nonnative listeners' abilities to classify talkers who varied on another sociophonetic dimension: foreign accent. Unlike regional dialect variation, nonnative listeners typically have more experience with nonnative speech than native listeners, particularly for talkers with the same native language background. Using auditory free classification, native listeners of English and native Korean listeners classified talkers by perceived native language. Talkers consisted of nonnative talkers from six native language backgrounds and native talkers. Results demonstrated that native listeners were nearly perfect at grouping the native talkers together, but Korean listeners were much less accurate. Further, Korean listeners did not show an advantage for grouping Korean-accented talkers together. These results suggest that nonnative listeners' less robust linguistic representations of the target language can hinder their abilities to attend to the acoustic-phonetic features that index dialect and accent categories. [Work supported by NIH-NIDCD Grant R21DC010027.]

4pSCb8. Categorization of regional, international, and nonnative accents. Amal Akbik, Eriko Atagi, and Tessa Bent (Speech and Hearing, Indiana Univ., 200 S Jordan Ave., Bloomington, IN 47405, aakbik@indiana.edu)

Auditory free classification—a task in which listeners classify auditory samples into unconstrained groups—has provided insight into perceptual representation and categorization for several sources of speech variability including U.S. regional dialects, nonnative accents, and foreign languages. Within these studies, phonological markedness and geography have emerged as central organizing principles. However, previous studies were limited by including only one source of variability. To address this gap, the perception of U.S. regional dialects, international English dialects, and nonnative accents was investigated within one classification task. Listeners categorized talkers based on perceived location of origin. Cluster analysis demonstrated a perceptual divide between native and nonnative talkers. Native talkers were further delimited by geographic proximity into Southern Hemisphere, U.S., and United Kingdom groups. One exception was the consistent grouping of Southern U.S. talkers with talkers from England. Nonnative talkers were grouped into three major branches: French and German, Asian, and other. The "other" branch primarily consisted of less familiar accents. The results suggest that native and nonnative accents are perceived as separate categories regardless of accent markedness. Additionally, when listeners are presented with a wide range of dialects and accents, geography remains an important organizing principle. [Research supported by IU Hutton Honors College.]

4pSCb9. Is consonant harmony assimilatory? Ian Maddieson (Dept. of Linguist, UNM, Univ. of New Mexico, MSC03-2130, Albuquerque, NM 87131-0001, ianm@berkeley.edu)

Ohala (1990) claimed that vowel harmony is in origin a product of vowel-to-vowel assimilation across intervening consonants. Gafos (1999) essentially argued that consonant harmony may similarly be assimilatory. For this to be the case, intervening segments—typically vowels—must be capable of transmitting the harmonizing property. For some properties, such as nasality or lip-rounding such "spreading" is non-problematic as these can be properties of either consonant or vowels. An alternative view, e.g., in Hansson (2010), is that consonant harmony (albeit more narrowly defined) is a correspondence or copying process, not an assimilatory effect. In this paper a range of attested varieties of consonant harmony will be evaluated in terms of how plausibly an assimilatory component might be involved. The analysis indicates that consonant harmony patterns vary along a scale of their likelihood to be explicable as assimilatory in nature. Processes such as sibilant harmony may have an assimilatory part, as suggested by Whalen *et al*. (2011) and supported by a limited acoustic study reported here. However, harmony involving certain phonatory and laryngeal features, such as voicing (given that vowels are prototypically already voiced) or ejective production, does not plausibly involve assimilatory transmission of the harmonizing property.

4pSCb10. French listeners' perceptions of prominence and phrasing are differentially affected by instruction set. Caroline Smith (Linguistics, Univ. of New Mexico, MSC 03 2130, Albuquerque, NM 87131-0001, caroline@unm.edu)

Listeners' perception of prosodic structure may differ depending on whether they are instructed to attend to the meaning of a spoken passage, or to the acoustics. Real-time perceptions of prominence and phrasal boundaries were obtained from Rapid Prosody Transcription [Cole *et al*. (2010)]. Twenty naive French listeners were divided into two groups that were given either meaning- or acoustically based instructions for listening to passages of spontaneous speech. While listening, they read an orthographic unpunctuated transcript of the speech. Half of each group first labeled words they perceived as prominent in five passages, then phrasal boundaries in five different passages; the other half performed the tasks in the opposite order. Consistent with previous results, listener agreement (measured by kappa) was higher for labeling boundaries than for prominence. The mean kappa was 0.80 for both meaning- and acoustically based responses, but the

difference between prominence and boundaries was greater in meaning-based responses. Listeners labeled more words as prominent than they labeled boundaries, with a greater difference in meaning-based responses than acoustically based, although the frequency of labeling did not differ overall between acoustic- and meaning-based instructions. The divergence between prominence and boundary labeling challenges the assumption that prominence in French derives from pre-boundary position.

4pSCb11. The influence of production latencies and phonological neighborhood density on vowel dispersion. Benjamin Munson (Speech-Lang.-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, munso005@umn.edu)

A number of studies have shown that in read laboratory speech talkers produce vowels in words from dense phonological neighborhoods closer to the periphery of the F1/F2 space than vowels in words from sparse neighborhoods [e.g., Wright (2004)]. The reason for this pattern is hotly debated: some argue that it reflects listener-directed partial compensation for the negative effect of high neighborhood density on perception [Munson and Solomon (2004)]. Others argue that this effect is due to the coactivation of similar words during speech production [Baese-Berk and Goldrick (2009)]. If the patterns are due to the coactivation of phonologically similar words in production, then the degree of hyperarticulation should be related to another measure presumably related to the coactivation of words, production latencies. Specifically, hyperarticulation should be greatest when the production latencies are shortest, as coactivation would presumably speed production [Vitevitch (2002)]. Production latencies and vowel dispersion were measured in read productions of single words varying in phonological neighborhood density. A preliminary analysis of data from 11 adults partially supports Baese-Berk and Goldrick's hypothesis: vowel-space dispersion was better predicted by production latencies than by neighborhood density, albeit in the opposite-than-predicted direction. Analysis of a larger cohort of talkers is ongoing.

4pSCb12. The influence of multiple narrators on adults' listening comprehension. Brittan A. Barker and Cornetta Mosely (COMD, Louisiana State Univ., 63 Hatcher Hall, Baton Rouge, LA 70803, barkerb@lsu.edu)

Research has demonstrated that variable, talker information—such as the number of talkers—affects listeners' perception and processing of linguistic information during various laboratory tasks. In particular, the detrimental effects of multiple talkers are highlighted during online speech perception tasks with little contextual support [isolated word recognition; e.g., Mullennix *et al.* (1989), Ryalls and Pisoni (1997), Sommers and Barcroft (2011)]. Nonetheless, it is unclear how multiple talkers might affect listeners' perception of linguistic information in more complex spoken language tasks utilizing real-time, fluent speech. The present experiments were conducted determine if information contributed by multiple talkers influences adults' auditory story comprehension in the presence of both quiet and background noise. The accuracy and reaction time data did not support the hypothesis that talker information affects the perception of linguistic information during auditory story comprehension. Thus these data bring to light theoretical perspectives that emphasize the importance of looking across experimental tasks to better understand talker-specific information's pattern of influence on spoken language processing [e.g., Sommers and Barcroft (2006), Werker and Curtin (2005)].

4pSCb13. Multi-subject atlas built from structural tongue magnetic resonance images. Jonghye Woo (Dept. of Neural and Pain Sci., Univ. of Maryland School of Dentistry, Baltimore, MD, jschant@gmail.com), Junghoon Lee, John Bogovic (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Emi Z. Murano (Dept. of Otolaryngol., Johns Hopkins Univ., Baltimore, MD), Fangxu Xing (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Maureen Stone (Dept. of Neural and Pain Sci., Univ. of Maryland School of Dentistry, Baltimore, MD), and Jerry L. Prince (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

Magnetic resonance imaging (MRI) is a widely used technology for non-invasive tongue imaging. MRI can detail tongue and muscle shapes and their variability in both healthy and diseased populations. Such detail can aid significantly in the interpretation of muscle interactions in the tongue, and their relation in normal and disordered speech production. However, the size or shape of the tongue and muscles may vary from one subject to another. In addition, there exists no comprehensive and systematic framework to assess the difference and variability of tongue and muscles in a normalized space. In the present work, we built a multi-subject atlas from 20 normal subjects that are acquired using structural MRI to offer a normalized space on which all subjects from a target population can be mapped and compared. In order to find accurate one-to-one correspondences, we bound the tongue so that each volume had the same vocal tract features. For registration, we utilize symmetric diffeomorphic image registration with cross-correlation, which is widely used in brain image analysis. The atlas facilitates a template-based segmentation in assigning anatomical labels in the images. The tongue atlas is unprecedented and opens new vistas for exploring normal and diseased oral structures and function.

4pSCb14. An examination of the articulatory characteristics of prominence in function and content words using real-time magnetic resonance imaging. Zhaojun Yang, Vikram Ramanarayanan (Elec. Eng., Univ. of Southern California, 1124 W. 29th ST, Apt 2, Los Angeles, CA 90007, zhaojuny@usc.edu), Dany Byrd (Linguistics, Univ. of Southern California, Los Angeles, CA), and Shri Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

We examine the functional coupling between articulatory characteristics of prominence, such as articulator speed, and its acoustic characteristics, such as F0 and acoustic energy, for content words (nouns) and function words (e.g., articles, prepositions and conjunctions) using real-time magnetic resonance imaging data. We use Granger causality ideas to test the degree and direction of causal influence between the chosen articulatory and acoustic measures for function and content words. We further apply functional canonical correlation analysis to these measures to understand the covariant behavioral modes of the articulatory and acoustic measures. After controlling for word duration, we observe that articulatory speed generally has a significant causal influence on F0, especially for longer content words, but observe no such effect in the opposite direction. Notably, we do not observe this effect for function words in most cases. We further observe a tighter coupling of canonical weight functions of articulatory speed and acoustic-prominence characteristics (F0 and energy) for the content words as compared to the function words considered. These observations provide support for the hypothesis that prominence realized during content words may result from a close coupling between articulatory and acoustic characteristics such as articulatory speed and F0, with suggestions of a directional causal relationship. [Work supported by NIH.]

4pSCb15. Do formant frequencies correlate with Japanese accent? Yukiko Sugiyama (Sci. and Technol., Keio Univ., Hiyoshi 4-1-1, Kohoku-ku, Yokohama 223-8521, Japan, yukiko_sugiyama@mac.com) and Tsuyoshi Moriyama (Media and Image Technol., Tokyo Polytechnic Univ., Atsugi, Japan)

Formant frequencies were examined as a possible acoustic correlate to Japanese accent. A perception study conducted by the first author of the present study used synthetic speech stimuli in which the harmonic structure of each spectrum (F0) was removed from speech produced in a normal manner and replaced by white noise. The stimuli created this way ensured that the only property altered from the normal speech would be the presence or absence of the F0. The result found that Japanese listeners reliably identified minimal pairs of words that differed only by accent, indicating that F0 is not the only acoustic cue to Japanese accent. Since previous studies on whispered speech report a positive correlation between the pitch height intended by speakers and formant frequencies, formant frequencies were measured as a possible correlate to accent. However, the analysis did not find any correlation between the F0 movements observed in the original normal speech and the formant frequencies measured in the synthetic speech. This result suggests that acoustic properties other than the F0 are also affected in whispered speech and claims made about the positive correlation between the intended pitch and formant frequencies in whispered speech do not hold in normal speech.

4pSCb16. The perception of formant-frequency range is affected by veridical and judged fundamental frequency. Santiago Barreda and Terrance M. Nearey (Dept. of Linguist, Univ. of AB, Edmonton, AB T6G 2E7, Canada, sbarreda@ualberta.ca)

The vowels produced by different speakers vary in terms of their fundamental frequency (f0) and formant frequencies (FFs). Variation in the production of a given vowel category between speakers of different sizes is

primarily according to a single multiplicative parameter (related to speaker vocal-tract length). This parameter, which we refer to as FF-scaling, has an associated perceptual quality that listeners may use to determine apparent speaker characteristics and vowel quality. In a previous experiment [Barreda and Nearey, J. Acoust. Soc. Am. **129**, 2661 (2011)], listeners were trained to identify a limited set of voices based on FF-scaling and f0 differences. The current study presented listeners with large number of voices ($n = 4000$) varying in FF-scaling and f0, arranged in a two-dimensional space where one dimension corresponded to each acoustic characteristic. Listeners were played a voice, and asked to indicate its location on the board, thereby providing an f0 and FF-scaling estimate for the voice. Results indicate that listeners are able to identify voice FF-scaling, and that this decision is informed primarily by veridical voice FF-scaling. However, there is a complicated relationship between perceived f0 and FF-scaling, suggesting an interdependent relationship in the perception of these characteristics.

4pSCb17. Relationship between the durations of rhythm unit with primary and secondary stresses in English speech. Shizuka Nakamura (Grad. School of Lang. and Culture, Osaka Univ., 1-8 Machikaneyama-cho, Toyonaka-Shi, Osaka 5600043, Japan, shizuka@akane.waseda.jp)

In this author's previous study [Nakamura, J. Acoust. Soc. Am. **131**(4, Pt. 2), 3347 (2012)] on the acoustical analysis of the duration structure of rhythm in English speech observed in short sentences uttered by native speakers, the durations of the following rhythm unit showed the smallest variance among native speakers: 1/4 of the preceding unstressed syllable(s) + stressed syllable + 3/4 of the succeeding unstressed syllable(s). The durations of the rhythm unit with a secondary stress were concentrated at half of those of the unit with a primary stress. Therefore, the rhythm can be described by a series of rhythm units with primary and secondary stresses where the latter unit is half the duration of the former. In this study, the relationship between the rhythm units with primary and secondary stresses was investigated from viewpoints of the position of syllables with primary and secondary stresses in a sentence, correlation among rhythm units in a sentence, and individual differences among native speakers. The results show that rhythm units in a sentence that are not only adjacent but also remote can adjust their durations mutually to realize the two to one ratio of the duration of the rhythm unit with primary and secondary stresses.

4pSCb18. Gestural reorganization under rate pressure interacts with learned language-specific phonotactics. Ioana Chitoran (Universite Paris 7 Denis Diderot, 175 rue du Chevaleret, Paris 75013, France, ioana.chitoran@dartmouth.edu) and Mark Tiede (Haskins Labs., New Haven, CT)

Studies of articulatory reorganization occurring under rate-driven production pressure can provide a window into speech planning. Previous work shows evidence for stable coordinative structures in speech in which VC patterns reorganize to CV, VCC to CCV, and coronal-labial to labial-coronal order. Such stable modes are argued to result from general physical-biological constraints imposed by the articulatory/auditory system. Here we examine whether stable modes can also arise from linguistic patterns learned on a language-specific basis. The case study is Georgian, which licenses complex onsets disregarding sonority, following instead a phasing pattern whereby degree of overlap varies with order of constriction location (front-to-back /pt/ sequences are more overlapped than back-to-front /tp/). We analyze preliminary data from native speakers repeating the Georgian words [pata] and [tapa] as they tracked an accelerating metronome. Results show: (1) pAta > patA > pta (stress shift followed by elision, licensed by Georgian phonotactics); (2) tApa > tAp (elision only; consistent with Georgian order constraints, not with biomechanical constraints). These patterns are contrasted with similar data from French in which elision is not observed, consistent with French phonotactics. The data thus provide an example in which language-specific structure rather than biomechanical constraints alone mediate gestural reorganization. [Work supported by Fulbright-Hays.]

4pSCb19. The effect of interpretation bias on the production of disambiguating prosody. Wook Kyung Choe and Melissa A. Redford (Dept. of Linguist, 1290 Univ. of Oregon, Eugene, OR 97403-1290, wchoe1@uoregon.edu)

Syntactically ambiguous sentences are frequently strongly biased toward one meaning over another [see, e.g., Tanenhaus and Trueswell (1995)]. This interpretation bias influences listeners' use of disambiguating prosody [Wales and Toner (1979)]. The current study investigated the effect on production. In experiment 1, the default interpretation of a heterogeneous set of 18 syntactically ambiguous sentences was investigated in 40 participants, who completed a question-and-answer task designed to identify intended meaning without making participants aware of potential ambiguity. Results were that 90% of the participants interpreted 11 of the sentences in just one way. There was a weaker interpretation bias for the remaining 7 sentences. In experiment 2, ten speakers were provided with and taught the alternate meanings of the 18 sentences from experiment 1, and then asked to disambiguate the meanings using prosody. Temporal and F0 measures indicated that while all speakers differentiated between meanings in production, only sentences with weak interpretation biases were consistently prosodically disambiguated. Prosodic cues to structure were applied inconsistently to differentiate meaning in sentences with strong interpretation biases. We conclude that disambiguating prosody is grammaticalized only when required by the interpretative norms of the speech community.

4pSCb20. Mechanisms for remembering roots versus affixes in complex words. Anne Pycha (Linguistics, Univ. of Wisconsin, Milwuakee, 3243 N. Downer Ave., Curtin Hall 537, P.O. Box 413, Milwaukee, WI 53211, pycha@uwm.edu)

Previous research has demonstrated that listeners remember low-frequency words (fob) through explicit recollection, but high-frequency words (money) through implicit familiarity [Joordens and Hockley (2000)]. We hypothesize that a similar asymmetry in remembering occurs in morphologically complex words (bleakish), where root frequency (bleak) is always low relative to affix frequency (ish). In our experiment, which modifies a technique developed by Goldinger *et al*. (1999), participants hear both complex and simple words at study. At test, they hear old words in which a portion of the stimulus is masked with soft or loud background noise. For complex words, the masked portion is either the root or the affix (**bleak**ish, bleak**ish**); for simple words, it is the corresponding pseudo-morpheme (**rel**ish, rel**ish**). Participants indicate whether they heard the word previously by making an old/new judgment, followed by a remember/know judgment [Tulving (1985)]. Preliminary results indicate that listeners are more likely to make "old" judgments when morphemes occur in soft (versus loud) background noise, but that this illusion effect is stronger for roots than affixes. Thus, clarity of perceptual input influences the memory of a complex word, but in an asymmetric fashion, suggesting that listeners remember roots and affixes via different mechanisms.

4pSCb21. Lexical bias and prosodic cues: An eye-tracking study of compound/phrase disambiguation. Jessica L. Gamache (Michigan State Univ., 461 Rampart Way, Apartment 106, East Lansing, MI 48823, gamache1@msu.edu)

Despite compounds and phrases exhibiting distinct prosodic cues (e.g., pitch and duration), adults fail to use these cues in certain contexts. In two eye-tracking experiments, I examined English speakers' attention to prosody in adjective-noun string disambiguation to explore a lexical bias previously reported (e.g., "hot dog" is interpreted as the food, regardless of prosody.) In experiment 1, 24 participants were presented with pictures of the compound and phrasal representations for possible compounds (20 known, 10 novel) accompanied by an audio presentation of either a phrasal or compound production. Replicating previous findings, participants sometimes ignored prosody and exhibited biases towards compound pictures for common compounds and towards phrasal pictures for novel compounds. The fixation data reveals similar patterns in which participants appeared to not consider the alternative picture, even when it matched available prosodic cues. In experiment 2, the phrasal/compound pictures were decoupled and placed with unrelated distractors and participants were asked whether either picture matched the audio presentation of a phrasal or compound production. This experiment helps to determine whether the exhibited lexical bias is an artifact of experimental designs using minimal pairs or whether prosody is still ignored when frequency and novelty are not directly competing.

4pSCb22. How robust are lexical effects on phonetic categorization? Mary M. Flaherty and James R. Sawusch (Psychology, SUNY Buffalo, 392 Park Hall, North Campus, Buffalo, NY 14260, maryflah@buffalo.edu)

Previous work has shown that lexical knowledge (whether a stimulus token makes a word or nonword) influences phonetic categorization [e.g., Fox (1984)]. Recent work in our lab examined the effect of lexical influences on speech perception using two tasks (phoneme identification and AXB discrimination) and uncovered some unexpected findings. Listeners who performed identification first showed a robust effect of lexical status on phonetic categorization. However, listeners who performed the discrimination task first showed no effect of lexical status. Since prior research has shown that the lexical effect is fairly robust [see Pitt and Samuel (1993)], the finding that the influence of lexical status can be eliminated by first placing listeners in an AXB discrimination task is the focus of the current research. The AXB task may focus attention on the prelexical, phonetic representation and differences within each phonetic category. This, in turn, eliminates the influence of higher level processes on phonetic perception in the identification task. The present study seeks to replicate this finding and investigate whether other lexical influences on perception (the influence of lexical neighborhood) are also altered by experience with AXB discrimination. Results are discussed in terms of the flow of information during phonetic perception and word recognition.

4pSCb23. Neural processing of voices—Familiarity. Lisa Gustavsson, Petter Kallioinen, Eeva Klintfors (Phonet./Linguist., Stockholm Univ., Stockholm 106 91, Sweden, lisag@ling.su.se), and Jonas Lindh (Inst. of Neurosci. and Physiol., Gothenburg, Sweden)

Brain responses to familiar and unfamiliar voices were investigated with ERPs (Event Related Potentials). Presentation of a stream of one syllable utterances from a female voice established a standard expectation, and similar samples from four other male voices where inserted as unexpected deviants in a typical mismatch paradigm. The participants were 12 students from the basic course in linguistics. Two of the deviant voices were familiar voices of their teachers. The two other deviant voices were matched (same age, sex, and dialect) but unfamiliar to the participants. A typical MMN (Mismatch Negativity) was elicited, i.e., a more negative response to the deviants compared to the standards. In contrast to verbal reports, where only one participant identified any of the deviant voices, the MMN response differed on group level between familiar and unfamiliar voices. MMN to familiar voices was larger. Using teachers' voices ensured naturalistic long term exposure, but did not allow for random assignment to conditions of familiarity making the design quasi-experimental. Thus, acoustic analysis of voice characteristics as well as follow up studies with randomized exposure to voices are needed to rule out possible confounds and establish a causal effect of voice familiarity.

4pSCb24. Effects of prosodic strengthening and lexical boundary on /s/-stop sequences in English. Yoonjeong Lee (Linguistics, Univ. of Southern California, C-128, 3701 Overland Ave., Los Angeles, CA 90034, yoonjeol@usc.edu)

This study examined how the effects of prosodic strengthening (from prosodic boundary and accent) and lexical boundary (e.g., "ice # can" vs. "eye # scan") are acoustic-phonetically realized on English /s/-stop sequences in a sentence. First, the domain-initial strengthening effect was not strictly confined to the first segment, but could extend into the second consonant and, at least partially, into the following vowel in the #/sCV/ sequence (e.g., in "scan"). Second, the accent-induced strengthening effect was robust in all acoustic measures for the #/sCV/ sequence. Third, prosodic strengthening arising with boundary and accent gave rise to the "shortened" VOT for the voiceless stop in the #/sCV/ sequence, suggesting that prosodic strengthening can operate on the phonetic manifestation of a phonological rule to reinforce the language-specific phonetic feature, which is, in this case, {-spread glottis}. Fourth, domain-initial strengthening and accent-induced strengthening differ substantially in some aspects, suggesting that they may be encoded separately in speech production process. Finally, "ice # can" and "eye # scan" were indeed very differently realized, suggesting that the underlying lexical boundary is signaled by fine-phonetic details even when the sequences occurred phrase-internally where they appeared to be homophonous, at least impressionistically, and syllabified the same.

4pSCb25. Syntactic predictability influences duration. Claire Moore-Cantwell (Linguistics, UMass Amherst, 26 Wright Ave., Northampton, MA 01060, cmooreca@linguist.umass.edu)

Building on work by Gahl and Garnsey, 2004, this paper demonstrates that speakers "buy time" during the planning of upcoming low-probability syntactic structures by producing prosodic boundaries with longer duration before low-probability than before high-probability structures. Subject extraction cleft sentences ("It was Edward who (t) loved Lucy.") are more common in corpora than object extraction cleft sentences ("It was Edward who Lucy loved (t).") [Roland *et al*. (2007)], and are also easier to process [e.g., Gibson (1998)]. The duration of the clefted constituent ("Edward") was measured in planned productions of subject- and object-extraction clefts in English. In order to disentangle the probability of each structure from its difficulty level, the probabilities were manipulated within the experiment through training. Participants read aloud: First, two SE and two OE clefts; second, eight SE or eight OE clefts; and finally, another two SE and two OE clefts. Before training, the clefted constituent was longer in OE clefts (mean 407 ms) than in SE clefts (370ms, $t = 2.4$, $p = 0.02$). This difference was no longer present after OE training (OE: 385 ms, SE: 397 ms), but was still present after SE training (OE: 448 ms, SE: 388 ms).

4pSCb26. Coarticulation in a whole event model of speech production. Bryan Gick (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada, gick@mail.ubc.ca) and Ian Stavness (Comput. Sci., Univ. of Saskatchewan, Saskatoon, SK, Canada)

Previous models of coarticulation have used varying combinations of advance planning and on-line calculation of weighted averages to determine how temporally overlapping speech sounds interact [see Farnetani and Recasens, Hbk. Phonet. Sci. Wiley-Blackwell (2010)]. A robust model of coarticulation should be able to predict such local interactions, as well as to describe changes resulting from rate variation, which should influence degree of temporal overlap between adjacent events. Using a model of tongue-jaw-hyoid biomechanics [Stavness *et al*., J. Biomech. (2012)], the present paper demonstrates that typical cases of lingual coarticulation can be attributed to the intrinsic biomechanics of the human body in an entirely feed-forward model with no additional machinery. Biomechanical modeling outcomes are compared to speech articulations at different speech rates, and show that naturalistic coarticulatory patterns emerge simply by varying degrees of temporal overlap in a biomechanically realistic model. The built-in mechanics of the human body can handle coarticulatory interactions with no extrinsic model at all, save one that identifies a) the right body parts, and b) the time-course of events. Results are interpreted in a "Whole Event" model of speech. [Research funded by NSERC.]

4pSCb27. Perceptual integration of indexical information in bilingual speech. Charlotte Vaughn and Susanne Brouwer (Linguistics, Northwestern Univ., 2016 Sheridan Rd, Evanston, IL 60208, crvaughn@u.northwestern.edu)

The present research examines how different types of indexical information, namely talker information and the language being spoken, are perceptually integrated in bilingual speech. Using a speeded classification paradigm [Garner (1974)], variability in characteristics of the talker (gender in experiment 1 and specific talker in experiment 2) and in the language being spoken (Mandarin vs. English) was manipulated. Listeners from two different language backgrounds, English monolinguals and Mandarin-English bilinguals, were asked to classify short, meaningful sentences obtained from different Mandarin-English bilingual talkers on these indexical dimensions. Results for the gender-language classification (Exp. 1) showed a significant, symmetrical interference effect for both listener groups, indicating that gender information and language are processed in an integral manner. For talker-language classification (Exp. 2), language interfered more with talker than vice versa for the English monolinguals, but symmetrical interference was found for the Mandarin-English bilinguals. These results suggest both that talker-specificity is not fully segregated from language-specificity, and that bilinguals exhibit more balanced classification along various indexical dimensions of speech. Currently, follow-up studies investigate this talker-language dependency for bilingual listeners who do not speak Mandarin in order to disentangle the role of bilingualism versus language familiarity.

4pSCb28. Prosodic characteristics of two focus types in emphatic context in Thai. Alif Silpachai (Linguist, Univ. of Southern California, 3170 Aintree Lane, Apt/Ste., Los Angeles, CA 90023, silpacha@usc.edu)

This study presents an acoustic analysis of narrow focus (early focus) and broad focus, each in emphatic context (tune) in Thai, with the goal of providing a basic characterization of their prosody. To investigate prosodic realizations, target words from each of the 5 lexical tones in Thai were

placed in subject positions of sentences with SVO structure. Each target word was placed in a sentence in which each syllable contained the same lexical tone as that of the target word. Preliminary results show that F0 measures, especially F0 maximum, minimum, and range, differed between focus types. In particular, narrow focused words were distinguished from non-narrow focused by higher F0 maximum, minimum, and range, while post-focal words contained lower F0 measures. Syllable duration also played a role in signaling narrow focus: focal words in narrow focus sentences were significantly longer than their non-focal counterparts in broad focus sentences. Interestingly, a pitch reset seemed to occur after focused words. Findings from four additional Thai speakers will be presented and there will be a discussion of their relevance to the intonational phonology of Thai.

4pSCb29. Speech rhythm and speech rate affect segmentation of reduced function words in continuous speech. Tuuli Morrill, Laura Dilley, J. Devin McAuley (Michigan State Univ., Oyer Ctr., 1026 Red Cedar Rd., East Lansing, MI 48824, tmorrill@msu.edu), and Mark Pitt (Psychology, The Ohio State Univ., Columbus, OH)

Recent work [Dilley and Pitt, Psychol. Sci. (2010)] has demonstrated that reduced function words in speech can perceptually disappear if the rate of surrounding speech is slowed, even when the acoustic properties of the function word (FW) and its immediate phonetic environment are held constant. An experiment was conducted to determine whether this disappearing word effect could be elicited through a manipulation involving speech rhythm, realized as binary and ternary alternations of high and low tones, as well as through manipulations to context speech rate. 74 participants transcribed 32 sentences containing a FW in which the preceding speech within the utterance was resynthesized with a binary or ternary speech rhythm presented at one of three context speech rates. A binary rhythm in the preceding speech context yielded lower FW report rates than the ternary rhythm. These results suggest that listeners' expectations about speech rhythm and/or syllable grouping affected the number of syllables and words perceived, indicating that such properties may play an important role in word segmentation and lexical access. [Work supported by NSF grant BCS-0847653.]

4pSCb30. Prosody and syntactic structures in continuous speech in French. Sarah Massicotte-Laforge (Psychologie, Université du Québec à Montréal, Groupe de recherche sur le langage, Département de psychologie, section développement, Université du Québec à Montréal, C.P.8888, Succursale Centre-Ville, Montréal, QC H3C 3P8, Canada, massicotte-laforge.sarah@courrier.uqam.ca), Andréane Melançon, and Rushen Shi (Psychologie, Université du Québec à Montréal, Montréal, QC, Canada)

Infant-directed speech contains dominantly multi-word utterances. Segmenting speech into linguistic units is crucial for language acquisition. This study inquires if prosodic cues exist in speech and mark syntactic categories. Participants were Quebec-French speakers. In experiment 1 participants read determiner+noun and pronoun+verb utterances. Nouns and verbs were pseudo-words (e.g., mige, crale) counterbalanced in their occurrences in the utterances, and their prosodic properties (duration, pitch, intensity) were measured. Results showed that the two types of utterances did not differ in prosody; noun versus verb productions of these pseudo-words were equivalent prosodically. Experiment 2 tested whether larger utterances were produced with prosodic cues supporting syntactic units. The same pseudo-words were the final words (counterbalanced) in (1) [determiner+adjective+noun] and (2) [[determiner+-noun]+verb] structures. Results showed that the last word as nouns versus verbs differed significantly in duration, pitch and intensity. Moreover, the initial consonant of verb productions was longer, with a distinct preceding pause. The word preceding the verb (2) exhibited boundary cues, differing significantly from the word preceding the noun in (1) in duration, pitch, and intensity. We suggest that these acoustic cues may help children first parse larger utterances and then acquire the syntactic properties of phrases and words based on their distribution.

4pSCb31. Tonogenesis in contemporary Korean with special reference to the onset-tone interaction and the loss of a consonant opposition. Mi-Ryoung Kim (Practical English, Korea Soongsil Cyber Univ., 474 East 14th Alley #9, Eugene, Oregon 97401, kmrg@mail.kcu.ac)

Recent studies show that, besides their effect on the fundamental frequency (f0) contour of the following vowel, Korean stops are undergoing a sound change in which a partial or completer voice onset time (VOT) merger is taking place between aspirated and lax stops. The purpose of this study is to see whether the sound change holds across the three major dialects of Korea (Seoul, Pusan, and Gwangju). The three acoustic parameters, VOT, f0, and H1-H2, were examined. The results show that the effects of onsets on f0 (i.e., onset-tone interaction) were robust across dialects whereas the merging of aspirated and lax stops was not. With respect to H1-H2, lax consonants showed higher breathiness than aspirated counterparts whereas tense consonants did not. With respect to VOT, most aspirated and lax stops were produced with long-lag voicing whereas tense stops were produced with short-lag voicing. However, interspeaker variations were noticeable even within each dialect, indicating that the sound change is still ongoing among Korean speakers. The phenomena correspond to typical tonogenesis properties, characterized by onset-tone interaction and merging of a consonantal opposition. The findings suggest that the sound change in contemporary Korean can be viewed as undergoing tonogenesis.

4pSCb32. Towards a model of Singaporean English intonational phonology. Adam J. Chong (Linguistics, UCLA, 1380 Veteran Ave., Apt. 103, Los Angeles, CA 90024, ajchong@ucla.edu)

Singaporean English (SgE) is a variety of English spoken in Singapore. Recent research has sought to identify the systematic features that make SgE distinct from other varieties of English. Although the intonation of SgE has been described previously [Deterding (1994), Lim (2004), Ng (2011)], no phonological model has yet been proposed. This paper proposes a model of intonational phonology for SgE within the Autosegmental-Metrical phonology framework. Three native speakers were recorded reading declarative and question sentences of varying length and stress pattern. Preliminary results suggest that SgE has three prosodic units above the word: the Accentual Phrase (AP), Intermediate Phrase (ip) and Intonational Phrase (IP). An AP is slightly larger than a word and is characterized by a general LH (rising) contour. The L can be attributable to either an L* tone on a lexically-stressed syllable or an L initial boundary tone if the stressed syllable occurs late in the AP. The AP-final syllable always has a phonologically High boundary tone (Ha). The initial AP is realized in a large pitch range, and subsequent APs within the same ip are realized in successively reduced pitch ranges. Tones of larger prosodic units will also be discussed.

4pSCb33. Calibrating the detection of spontaneous speech: From sentences to noun phrases. Sara Parker and Jennifer Pardo (Psychology, Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, saraphyn@gmail.com)

Many studies have examined the differences between speech that is produced spontaneously as opposed to read from a prepared script. Most of these studies have focused on prosodic measures taken from clauses, sentences, or connected discourse. Furthermore, studies have shown that listeners are able to identify the context of production when presented with sentence-length utterances. The current study examined whether a listener can identify the context for utterances that are briefer than a sentence. A set of 20 talkers (10 male) produced spontaneous descriptions of maps that they then read aloud in a separate session at least one week later. Pairs of sentences that matched in fluency across both contexts were selected, and listeners judged which member of a pair was produced spontaneously. In separate blocks, listeners heard either full sentences, sentence beginnings, sentence endings, or two-word noun phrases excised from sentences. Overall, listeners could identify the spontaneously produced utterances, but only for excerpts longer than two-word noun phrases. These findings indicate that the information present in two-word noun phrases is not sufficient to support perception of spontaneous versus read speaking style.

4pSCb34. Effects of musical experience on perception of audiovisual synchrony for speech and music. Dawn M. Behne, Magnus Alm, Aleksander Berg, Thomas Engell, Camilla Foyn, Canutte Johnsen, Thulasy Srigaran, and Ane Eir Torsdottir (Psych. Dept., NTNU, Trondheim NO7491, Norway, dawn.behne@svt.ntnu.no)

Perception of audiovisual synchrony relies on matching temporal attributes across sensory modalities. To investigate the influence of experience on cross-modal temporal integration, the effect of musical experience on the perception of audiovisual synchrony was studied with speech and music stimuli. Nine musicians and nine non-musicians meeting strict group criteria provided simultaneity judgments to audiovisual /ba/ and guitar-strum stimuli, each with 23 levels of audiovisual alignment. Although results for the speech and music stimuli differed, the two groups did not differ in their responses to the two types of stimuli. Consistent with previous research, responses from both groups show less temporal sensitivity to stimuli with video-lead than audio-lead. No significant between-group difference was found for video-lead thresholds. However, both for the speech and music stimuli, musicians had an audio-lead threshold significantly closer to the point of physical synchrony than non-musicians, indicating the musicians' greater acuity for audiovisual temporal coherence. Overall this leads to a non-significant tendency for a narrower window of synchrony for musicians than non-musicians. Findings are consistent with predictions that cross-modal temporal experience increases threshold acuity for audio-lead, but not for video-lead, and also support theories suggesting greater efficiency with relevant experience.

4pSCb35. Speech rhythm in Korean: Experiments in speech cycling. Younah Chung (Linguistics, UCSD, 8520 Costa Verde Blvd., APT 3420, San Diego, CA 92122, yachung@ucsd.edu) and Amalia Arvaniti (English Lang. & Linguist, Univ. of Kent, Canterbury, United Kingdom)

Korean has not been unanimously classified for rhythm class, and it lacks stress. Thus, it does not fit into views that rhythm rests on alternations of metrical strength. The goal was to examine what, if any, elements are used in Korean for rhythm purposes. It was hypothesized that the onsets of accentual phrases act as beats. The materials were 6 sentences; each was 9 syllables and three APs long. The number of syllables in each AP varied. Syllable composition also varied between CV and CVC. Native speakers repeated each sentence, fitting each repetition into beat intervals at three different metronome rates. Each AP was expressed as a ratio of the entire cycle. Two experiments were conducted. The first experiment suggests that speakers keep AP onsets in phase although syllable count and composition also affect phase. The results support our hypothesis that AP onsets operate similarly to stresses. The second experiment that used waltz rhythm showed that it is the only level of prominence, and no differentiation between the strength of these beats, such that it would produce waltz rhythm, is possible. The results suggest that Korean rhythm is not characterized by multiple levels of alternation between strong and weak constituents.

4pSCb36. Acoustic vowel space size and perceived speech tempo. Melanie Weirich and Adrian P. Simpson (Institut für Germanistische Sprachwissenschaft, Friedrich-Schiller-Universität Jena, Fürstengraben 30, Jena 07743, Germany, melanie.weirich@uni-jena.de)

"Females speak faster than males." Although several studies have proved this stereotype to be wrong [Byrd (1994)], it is still a widespread belief in many languages and within both genders. The interesting question is why. Two findings are particularly relevant regarding this stereotype: First, females reveal a greater acoustic vowel space than males [Hillenbrand *et al*. (1995)]. Second, a stimulus with a moving f0-contour is perceived as faster than the same stimulus with a monotonous contour [Lehiste (1976)]. From that, we might propose that if a dynamic f0 contour triggers the perception of a faster speaking rate, then a larger acoustic vowel space might have the same effect. The reason for female speakers being perceived as speaking at a faster tempo, then, is that they traverse on average a larger acoustic vowel space within the same time-frame than male speakers do. Furthermore, we could also expect a relationship between vowel space size and perceived speech tempo within the same gender. A perception test was conducted with temporally aligned stimuli from 56 female speakers who vary in their vowel space sizes. Results reveal a significant positive correlation between vowel space size and perceived tempo ($r = 0.36$, $p < 0.001$).

4pSCb37. English native monolingual and simultaneous English/Spanish bilingual listeners' perception of foreign accented speech: Cross-language effects on accented speech perception. Somang Moon and Su-Hyun Jin (Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, somang.moon@gmail.com)

The current study was designed to explore whether English monolingual listeners (ENM) perceive foreign- accented speech differently from English/Spanish simultaneous bilingual listeners (ESB) who learned both languages simultaneously at early age. Previous studies suggest that listener's perception of foreign-accented speech is affected by listener's L1 phonological systems. When exposed to two languages simultaneously, bilinguals might be able to exploit the phonetic categories of the two languages in speech perception [Best (1994), Goetry and Kolinsky (2000)]. It would be possible that simultaneous exposure to multi-languages at the early age helps SB listeners to tolerate foreign accents in speech more resulting in better understanding of accented speech. It would be also possible that if two phonological systems of SB listeners interact each other resulting in poorer understanding of foreign accented speech than ENM listeners. ENM and SB listeners completed two speech perception tasks: accent ratings of an English passage and identification of English vowels spoken by Korean-native speakers. Results might suggest the effect of language exposure on accented speech perception.

4pSCb38. An investigation of the three tone system in Tsuut'ina (Dene). Joyce McDonough, Jared O'Loughlin (Dept. of Linguist, Univ. of Rochester, Rochester, NY 14625, joyce.mcdonough@rochester.edu), and Christopher Cox (Dept. of Linguist, Univ. of Alberta, Edmonton, AB, Canada)

This study is part of the documentation and conservation of Tsuut'ina (formerly Sarcee, Sarsi; ISO 639-3: srs), a northern Dene (Athabascan) language by a collaboration of academic and community members. Tsuut'ina is a tone language. Contrary to Dene tonogenesis theory and unlike reports on all other Dene tone languages, Tsuut'ina is reported to have three tones, H, L, M. The tonal system in Dene family has been argued to arise from the loss of laryngealized sonorants in monosyllabic stem codas and incorporation of laryngealization into the nucleus of the stem, resulting in H and L tonal contrasts. The Dene languages additionally exhibit "tonal reversal", a tendency for the Dene tone languages to show "reversed" tonal patterns that postdate the original tonogenesis. In this study we investigate the tonal distribution, realization patterns and tonal alignment in data collected from two fluent speakers reciting prepared wordlists and short discourses. Preliminary investigation indicates that, as reported, three tonal patterns emerge, M tone associated most often with a falling tone, with distinct distribution patterns arguably related to morphological factors. Furthermore M tone is more highly variable. We lay out distribution patterns and interactions with morphology and statistical analyses associated with the data.

4pSCb39. Prosodic correlates of smiled-speech. Caroline Émond (Linguistique, Université du Québec à Montréal, C.P. 8888, Succ. Centre-ville, Montréal, Québec, QC H3C 3P8, Canada, caroemond@hotmail.com) and Marty Laforest (Lettres et Commun. sociale, Université du Québec à Trois-Rivières, Trois-Rivières, QC, Canada)

Smiling is a visible expression and an audible one too when it is synchronous with speech. Very few studies have documented the perceptual prosodic cues associated with perceived smiling speech. The first aim of this paper is to study the perception of smiled-speech according to the listeners' gender. The reaction time and the intensity of the perceived smiled-speech were also investigated. The second aim is to identify a combination of prosodic parameters which would allow a phonetic description of smiled-speech. 140 utterances were extracted from spontaneous data (Montréal 1995 corpus) and used as stimuli for a perception test administered to 40 Québec French listeners (20 men, 20 women). Results show that men and women do not perceived smiled-speech in the same way, and women are quicker than men to make their decisions. Moreover, reaction times are faster for utterances perceived as smiling with a high degree of intensity, for both men and women, than those with lower intensity. Perceived prosodic parameters related to pitch height, pitch range, rhythm, and speech rate in relation to smiled-speech and its intensity are also discussed.

4pSCb40. Vowel production in Mandarin accented English and American English: Kinematic and acoustic data from the Marquette University Mandarin accented English corpus. An Ji (Elec. & Comput. Eng., Marquette Univ., Milwaukee, WI), Jeffrey J. Berry (Speech Pathol. & Audiol., Marquette Univ., P.O. Box 1881, Milwaukee, WI 53201-1881, jeffrey.berry@marquette.edu), and Michael T. Johnson (Elec. & Comput. Eng., Marquette Univ., Milwaukee, WI)

Few electromagnetic articulography (EMA) datasets are publicly available, and none have focused systematically on non-native accented speech. We introduce a kinematic-acoustic database of speech from 40 (gender and dialect balanced) participants producing upper-Midwestern American English (AE) L1 or Mandarin Accented English (MAE) L2 (Beijing or Shanghai dialect base). The Marquette University EMA-MAE corpus will be released publicly to help

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advance research in areas such as pronunciation modeling, acoustic-articulatory inversion, L1-L2 comparisons, pronunciation error detection, and accent modification training. EMA data were collected at a 400 Hz sampling rate with synchronous audio using the NDI Wave System. Articulatory sensors were placed on the midsagittal lips, lower incisors, and tongue blade and dorsum, as well as on the lip corner and lateral tongue body. Sensors provide five degree-of-freedom measurements including three-dimensional sensor position and two-dimensional orientation (pitch and roll). In the current work we analyze kinematic and acoustic variability between L1 and L2 vowels. We address the hypothesis that MAE is characterized by larger differences in the articulation of back vowels than front vowels and smaller vowel spaces compared to AE. The current results provide a seminal comparison of the kinematics and acoustics of vowel production between MAE and AE speakers.

4pSCb41. Phonetic alignment and phonological association in Tashlhiyt Berber. Timo B. Röttger (IfL Phonetik, Univ. of Cologne, Herbert-Levin-Str. 6, Köln D-50931, Germany, timo.roettger@uni-koeln.de), Rachid Ridouane (Laboratoire de Phonétique et Phonologie (UMR 7018), CNRS/Sorbonne Nouvelle, Paris, France), and Martine Grice (IfL Phonetik, Univ. of Cologne, Köln, Germany)

Although Tashlhiyt Berber uses intonation to mark sentence modality, the location of f0 events is severely constrained by its notorious predominance of consonantal nuclei (cf. (1) where syllable nuclei are underlined) (1) [ts.sk.ʃ f.tstt] "you dried it (fem.)" Here we report on the alignment of f0 peaks in disyllabic target words in polar questions and contrastive statements in the language. Data from four native speakers revealed that questions tend to have later f0 peaks than statements. This was reflected in discrete association patterns when more than one tone bearing unit was available: in questions the f0 peak occurred significantly more often on the final syllable than in statements. Interestingly, if no association distinction was made, there was a difference in alignment of this peak within a tone bearing unit: the peak was aligned significantly later in questions. Thus, discrete phonological association patterns were mirrored by phonetic alignment detail. These data question the traditional dichotomy between phonological association and phonetic alignment.

4pSCb42. Articulatory parameterization in Trique tone production: Distinguishing co-production from coarticulation. Christian DiCanio and Hosung Nam (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, dicanio@haskins.yale.edu)

The production of a tone in a tonal language is typically influenced by adjacent tonal targets. Within the literature, all such influences on F0 are considered part of tonal coarticulation. Yet, conflating all these effects under "coarticulation" results in an assortment of different processes within a tone language which lack a common motivating principle. In this talk, we present original tone production data from Itunyoso Trique (Oto-Manguean). The data consists of five repetitions of 24 sentences spoken at two speech rates (fast/normal) by eight native speakers. The medial target word was one of four tones (/45/,/4/,/32/,/2/), while the adjacent words were one of six tones (/45/,/43/,/32/,/3/,/2/,/1/). F0 data was extracted and time-normalized. Two patterns were observed. First, adjacent tones influenced F0 at the onset and offset of target tones. Second, global changes in F0 contour occurred for certain tones. All such effects were stronger during fast speech rate. We argue that these effects, often grouped together as coarticulation, have distinct explanations within an Articulatory Phonology framework. Transitional effects at tonal onsets and offsets are modeled by temporally modulating gestural activation intervals, resulting in articulatory undershoot between tones, whereas global changes in F0 contour are modeled by modulating gestural target parameters.

4pSCb43. Perception modeling of native and foreign-accented Japanese speech based on prosodic features of pitch accent. Ashleigh R. Gonzales (Linguistics, Simon Fraser Univ., 8888 University Dr., 9201 Robert C. Brown Bldg., Burnaby, BC V5A 1S6, Canada, agonzale@sfu.ca), Shunichi Ishihara (School of Culture, History and Lang., Australian National Univ., Canberra, ACT, Australia), and Chiharu Tsurutani (School of Lang. and Linguist, Griffith Univ., Nathan, QLD, Australia)

This study investigates the influence acoustic measures of pitch accent have on L1 and Australian English (AusE) L2 Japanese speech perception, expanding Tsurutani (2010) and Ishihara, Tsurutani, and Tsukada (2011), and motivated by Munro and Derwing (2001), which studies the role of speaking rate on judgments of L2 speech. We establish native and advanced AusE listeners of Japanese differ in their judgments of foreign accent in terms of accentedness and comprehensibility [Munro and Derwing (1995, 1999)] through a listening task. Selected acoustic measures of pitch accent from the speech stimuli, which displayed significant variance across listener groups—delta-pitch, max and mean max delta-intensity, and duration per mora—are correlated with L1 and L2 listener data. Testing for a relationship between each of the acoustic measures and listener judgments, the regression analyses show a considerable relationship between comprehensibility judgments and duration and intensity features, ranging from adjusted $R2 = 14.3\%$ to 24.6% across listeners, and indicating the degree of variance between judgments can be attributed to these acoustic measures. We can interpret that comprehensibility is linked to intensity and duration, which supports the authors' prior findings that timing is considered more important than pitch in the detection of foreign-accented speech.

4pSCb44. Can co-speech hand gestures facilitate learning of non-native tones? Katelyn Eng, Beverly Hannah, Lindsay Leung, and Yue Wang (Linguistics, Simon Fraser Univ., 8888 University Dr., RCB 9213, Burnaby, BC V5A 1S6, Canada, yuew@sfu.ca)

Speech perception research has indicated that information from multiple input modalities (e.g., auditory, visual) facilitates second language (L2) speech learning. However, co-speech gestural information has shown mixed results. While L2 learners may benefit from this additional channel of information, it may also be inhibitory as learners may experience excessive cognitive load. This study examines the role of metaphoric hand gestures in L2 lexical tone learning using previously established laboratory training procedures. Training stimuli include Mandarin tones produced by native Mandarin speakers, with concurrent hand gestures mimicking pitch contours in space. Native Canadian English speakers are trained to perceive tones presented in one of three modalities: audio-visual (AV, speaker voice and face), audio-gesture (AG, speaker voice and hand gestures) and audio-visual-gesture (AVG). The effects of training are assessed by comparing the pre-training and post-training tone identification results. Greater improvements for the AVG compared to AV group would indicate the facilitative role of gestures. However, greater improvements for the AG or AV compared to AVG group would support the cognitive overload account. Findings are discussed in terms of how sensory-motor and cognitive domains cooperate functionally in speech perception and learning. [Equal contributions by KE, BH, and YW; work supported by SSHRC.]

4pSCb45. Consonant harmony in Moroccan Arabic: Similarity and incomplete neutralization. Georgia Zellou (Linguistics, Univ. of Pennsylvania, 800 N. 48th St., #26, Philadelphia, PA 19139, gzellou@sas.upenn.edu)

Moroccan Arabic (MA) displays a synchronic consonant harmony alternation where underlying alveolar sibilants can assimilate in place of articulation to a following palatal sibilant, e.g., seʒera ~ ʃeʒera "tree". This study investigates the phonetic realization of the assimilated sibilant variant in consonant harmony forms. This consonant harmony process is typologically unusual since avoidance of similarity of root consonants has been proposed to be a pervasive tendency for the Semitic languages, including Arabic. Hence, it is predicted that even though a phonological change has resulted in adjacent stem consonants with identical features, similarity avoidance tendencies will act at the level of the phonetic representation to ensure that adjacent consonants are not articulatorily identical. An acoustic investigation using a center of gravity (COG) measure of MA sibilants was conducted on monolingual MA speakers to test this hypothesis. The results indicate that the harmonized palatal sibilants (i.e., ʃeʒera) are produced with a higher COG, suggesting a further front place of articulation, compared to regular (non-harmonized) palatal sibilants. In other words, the harmonized sibilants exemplify a case of incomplete neutralization, where the phonetic trace of a disappeared consonant remains. Furthermore, these results suggest that similarity avoidance in MA is maintained through sub-phonemic, gradient differences.

4pSCb46. The role of prosody in speech segmentation: Comparisons between monolinguals and French-English bilinguals. Meghan Spring, Linda Polka (School of Commun. Sci. and Disord., McGill Univ., 46 Tiffany Crescent, Kanata, Ontario K2K1W2, Canada, meghan.spring@mail.mcgill.ca), and Suzanne Curtin (Dept. of Psych., Univ. of Calgary, Calgary, AB, Canada)

Monolinguals harness language-specific prosodic cues for the purpose of segmenting out words from the speech stream. However, if and how bilinguals are able to do so in both their languages is less certain. In the current study, 26 English monolinguals, 28 French monolinguals, and 41 English-French adult bilinguals heard streams of both English- and French- accented nonsense syllables. While there were clear differences between the monolingual English and French groups, there was no difference between the performance of English-dominant and French- dominant bilinguals, nor between simultaneous versus sequential bilinguals. As a group, English-French bilinguals did show evidence of different segmentation strategies between language streams. It is therefore concluded that in certain conditions, bilinguals appear to be able to switch stress-based segmentation strategies between their languages. The use of the Hearing in Noise Test (HINT) as a promising new method for measuring language dominance in bilinguals is also discussed.

4pSCb47. Perceived prosodic boundaries in Taiwanese and Swedish.
Grace Kuo (Linguistics, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, gracekuo@humnet.ucla.edu)

Earlier studies have shown that listeners are not only able to detect the presence or the absence of a prosodic boundary but also able to distinguish between different boundary types. This study examined whether Taiwanese listeners ($n=18$) and English listeners ($n=7$) were able to predict the occurrence and the strength of the upcoming prosodic boundaries in Taiwanese and Swedish. For this purpose, we conducted a perceptual rating experiment, whose stimuli consisted of fragments with different boundaries (word, phonological phrase/tone sandhi domain, and intonational phrase), length (2-second and one-word) and quality (low-pass filtered and unfiltered.) Results show that both Taiwanese and English listeners can detect the occurrence and distinguish the boundaries in a foreign language when they are presented with longer fragments. Our finding strengthens the notion proposed in Carlson *et al*. (2005) that lexical information is not a necessary cue for prosodic boundary detection Another supporting evidence is that they could do the task nearly as well when the utterances were low-pass filtered. Significant correlations between ratings and the following relevant measures are found: f0 and voice quality.

4pSCb48. Decrease of pitch perception ambiguity in tone language processing.
Xiao Perdereau (Burgundy Univ., 9, A. Savary, BP 47870, Dijon 21078, France, xiao.chen-perdereau@u-bourgogne.fr)

Native tone language speakers were presented with speech materials in their language produced by non-native speakers. The speech materials were selected sound streams according to acoustic characteristics. They were made of monosyllabic words, disyllabic words and polysyllabic short sentences in spoken Mandarin. Participants were required to recognize the speeches in as short time as possible. Results revealed that the essential time to identify the speech is longer for shorter sequences, suggesting that ambiguities reside mostly in the lexical tone level. The ambiguity due to pitch perception decreases when the segmented speech events increase. Although it contributes to word meaning, pitch perception is less important in a polysyllables group of words or sentence processing than in monosyllabic word identification. We will also present some applications of these findings.

4pSCb49. Towards a model of intonational phonology of Turkish: Neutral intonation.
Canan Ipek and Sun-Ah Jun (Linguistics, USC, 123 S Figueroa Apt. 835, Los Angeles, CA 90012, canan.ipek@gmail.com)

This study proposes an Autosegmental-Metrical model of Turkish intonation based on sentences produced in neutral focus, as part of our ongoing research investigating Turkish intonational phonology. Tonal patterns of utterances were examined by varying the length of a word and a phrase, the location of stress, syntactic structures, and sentence types. Preliminary results suggest that Turkish has a H* pitch accent, realized on the stressed syllable of most content words. Each content word forms one Prosodic Word (PW) whose left edge is marked by an L tone. There are two prosodic units higher than PW: an Intermediate Phrase (ip) marked by a final rising (LH) tone and an Intonational Phrase (IP) marked by various types of a final boundary tone. These three prosodic units are also distinguished by the degree of juncture. Interestingly, the ip-final LH boundary tone marks the right edge of a heavy syntactic constituent regardless of the length of the unit. Furthermore, the left edge of a nuclear pitch accent is also marked by a rising tone (LH), which is realized on the last syllable of the immediately preceding PW. The ip-final LH tone and the pre-nuclear LH tone are phonetically different and perceptually distinct.

4pSCb50. Fundamental frequency as cue to intonation: Focus on Ika Igbo and English rising intonation patterns.
Joy O. Uguru (Linguist, Igbo and other Nigerian Lang., Univ. of Nigeria, No. 01 Louis Mbanefo st., Nsuukka +234, Nigeria, joyolug@yahoo.com)

This paper shows that fundamental frequency, Fo, can be a cue to type of intonation. The work centers on three main intonation patterns in Ika Igbo and English. Ika Igbo is a language that manifests intonation in addition to lexical tone. These intonation patterns are Low Rise (LR), High Rise (LR), and Fall Rise (FR). The Fos of these intonation patterns were analyzed acoustically in utterances with similar phonemes and tunes in both languages. Eighteen utterances were used for the study. The analyses show that the Fos of LR and FR intonation were generally lower than those of HR. Hence, it can be concluded that high intonation has high Fo while low intonation has low Fo. It can therefore be concluded that Fo is a cue to type of intonation.

4pSCb51. Downstep exceptions in Ibibio.
Afton L. Coombs (Linguistics, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, acoombs@usc.edu)

Downdrift and downstep are processes which may cause lowering of high tone syllables. Downdrift is intonational, occurring at phrasal or utterance level, while downstep is a phonological process which acts from one tone-bearing unit to the next such that H tones lower successively. The relationship between these larger tonal lowering processes and individual tone units is complicated, however, by processes which may raise or preserve original high tone pitches. Ibibio, a Niger-Congo language spoken in southeastern Nigeria, is a terraced tone language with contrastive H and L tones. H tones in Ibibio experience automatic and non-automatic downstep, lowering both in sequences of high tones and around intervening lows. This study aims to determine those factors which counteract or overrule the downstep process. Average pitch readings were taken of entire syllables and compared with readings of other syllables within the same word. The main finding of this study is that while single words show acoustically measurable downtrends, they also show non-lowering and even raising of high tones, specifically in HHL contexts. This complicates how downtrends act across tone-bearing syllables, and may indicate that a high tone is raised in order to increase contrast with a following low.

4pSCb52. Effect of speech variable rate on the coarticulation in the right vocalic context of Arabic utterances VCV.
Leila Falek, Hocine Teffahi (Electron. and Comput. Sci. Faculty, USTHB FEI Algiers, USTHB FEI Algiers Algieria, Algiers 16111, Algeria, falek.leila@gmail.com), and Amar Djeradi (faculté d'electronique, USTHB, Algiers, Algeria)

Our study consists of analysing Arabic utterances VCVα in brief vocalic context with Vα and speech rate as variables in order to observe the impact of the "right" context and speech rate on the coarticulation. Thus, we have to look for some invariance in the speech signal explaining the coarticulation phenomenon related to speech rate. So, we have analyzed the formant tracking of the arabic pharyngeal / / (or (ع) in arabic) in vocalises contexts /a/, /u/ and /i/ with variables speech rates (normal, fast and slow) in interrogative sentences. That is in order to confirm the anticipatory phenomenon and observed the influence of speech rate variation on the vocal tract in articulation. The observed results have shown in our case the existence of anticipatory articulation and that it depends on speech rate. (Non existence in slow rate, however, with more prominence in normal or fast rate.)

4pSCb53. Acoustic features of English sentence production for English and Chinese native speakers: Intonational and temporal patterns.
Ashley Woodall, Chang Liu, Brenna Thomas, and Katherine Reistroffer (The Univ. of Texas at Austin, 2504 A Whitis Ave A1100, Austin, TX 78712, ash.woodall@gmail.com)

Intonation is utilized by languages in order to differently convey intention, meaning, and emotion. Chinese, a tonal language, assigns F0 formant frequencies to lexically important components within words; whereas English, a stress language, changes F0 patterns according to the speaker's intended meaning or emotion, especially changing near the end of a phrase. As Chinese speakers produce other languages, especially a non-tonal language such as English, it is uncertain whether their intonation is the same as a native speaker. The purpose of this study is to compare intonation across Chinese and English speakers. An acoustical analysis was completed on 16

English and 32 Chinese speakers producing List 1 of the Hearing In Noise Test (HINT) sentences. Preliminary results show that Chinese speakers produce sentences with longer absolute duration than English speakers. Intonation features of sentence production such as F0 contour and temporal contours, as well as temporal features of English sentences such as temporal gap and sentence and word duration will also be compared. Findings will show the effect of L1 tonal language on producing a stress language, such as English.

4pSCb54. Rate variation as a talker-specific/language-general property in bilingual speakers. Midam Kim (Linguistics, Northwestern Univ., 425 Hurricane Ln, Lawrence, Kansas 66049, midamkim@gmail.com), Lauren Ackerman, L. Ann Burchfield, Lisa Dawdy-Hesterberg, Jenna Luque, Kelsey Mok, and Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL)

Nonnative talkers tend to exhibit slower speech rates than native talkers at the group level. Here we ask whether individual variation in rate is language-general to the extent that L1 rate is a significant predictor of L2 rate within bilinguals. 62 nonnative English talkers participated in three speech production tasks in both their L1 (14 Cantonese, 14 Mandarin, 11 Korean, 4 Portuguese-Brazilian, 6 Spanish, 13 Turkish) and L2 (English), namely, reading a paragraph, spontaneously answering questions, and spontaneously describing a picture story. Two measurements of rate were automatically extracted from the recordings: speech rate (syllables per second), and articulation rate (syllables per second excluding silent pauses). As expected, L2 speech and articulation rates were overall slower than L1 speech and articulation rates for all tasks. Importantly, L2 speech rates and articulation rates were positively related to L1 speech rates and articulation rates, respectively. There were also significant differences in L2 speech rates and L2 articulation rates depending on L1 background and tasks. However, the positive relationship between L1 and L2 rates still holds with these other effects taken into consideration, suggesting that overall rate variation is partially an individual-specific property that transcends L1 and L2 within bilinguals. Acknowledgments: Vanessa Dopker and Chun Liang Chan. [Work supported by Grant R01-DC005794 from NIH-NIDCD.]

THURSDAY AFTERNOON, 6 JUNE 2013 — 510A, 1:00 P.M. TO 5:00 P.M.

Session 4pSP

Signal Processing in Acoustics, Acoustical Oceanography, and Architectural Acoustics: Sampling Methods for Bayesian Analysis and Inversions in Acoustic Applications

Cameron Fackler, Cochair
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Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180

Invited Papers

1:00

4pSP1. Sampling methods for uncertainty quantification in source localization and geoacoustic inversion in the ocean. Tao Lin and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd, Newark, NJ 07102, michalop@njit.edu)

Iterative and sequential Bayesian filtering approaches have been successfully employed for the estimation of select features of received acoustic signals—namely, arrival times and amplitudes of paths that have interacted with the propagation medium. These are subsequently utilized in source localization and environmental property estimation. Sequential filtering has the advantage of relating arrival times across spatially separated hydrophones of a receiving array, providing "tighter" estimates of arrival times and amplitudes and, thus, probability densities with a reduced "spread" in inversion. We demonstrate that sequential methods are superior to solely iterative ones by linking estimates of times and amplitudes to propagation models and estimating source location and environmental parameters. The inversion component of the problem is approached with an efficient approach, which relies on a novel implementation of linearization of the relationship that links parameters of the propagation medium to the received sound field. [Work supported by ONR].

1:20

4pSP2. Bayesian localization of an unknown number of ocean acoustic sources. Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper considers localizing an unknown number of ocean acoustic sources when properties of the environment are poorly known. A Bayesian formulation is developed in which environmental parameters, noise statistics, and the number, locations, and complex spectra (amplitudes and phases) of multiple sources are considered unknown random variables constrained by acoustic data and prior information. The number of sources is determined during a burn-in stage by minimizing the Bayesian information criterion using hybrid optimization with an efficient source birth/death scheme. Optimal estimates and marginal posterior probability distributions for source locations are computed employing a variety of sampling approaches. Environmental properties and source locations and are treated as explicit parameters and marginalized using Markov-chain Monte Carlo sampling methods. In particular, environmental parameters are treated using Metropolis-Hastings sampling applied efficiently in a principal-component space, and source locations are treated using Gibbs sampling since the corresponding conditional probability distributions can be computed efficiently using normal-mode methods. Source and noise spectra are sampled implicitly by applying analytic maximum-likelihood solutions expressed in terms of the explicit parameters. This represents an empirical Bayesian approximation within a hierarchical formulation, and significantly reduces the dimensionality and improves sampling efficiency in the inversion.

1:40

4pSP3. Probabilistic two dimensional joint water-column and seabed inversion. Jan Dettmer and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3P6, Canada, jand@uvic.ca)

This paper develops a probabilistic two-dimensional (2D) inversion for geoacoustic seabed and water-column parameters in a strongly range-dependent environment. Range-dependent environments in shelf and shelf-break regions are of increasing importance to the acoustical-oceanography community, and recent advances in nonlinear inverse theory and sampling methods are applied here for efficient probabilistic inversion in 2D. The 2D seabed and water column are parameterized by highly efficient, self-adapting irregular grids which match the local resolving power of the data and provide parsimonious solutions requiring few parameters to capture complex environments. The self-adapting parameterization in the water-column and seabed is achieved by implementing the irregular grid as a trans-dimensional hierarchical Bayesian model which is sampled with the Metropolis-Hastings-Green algorithm. To improve sampling, population Monte Carlo is applied with a large number of interacting parallel Markov chains employing a loadbalancing algorithm on a computer cluster. The inversion is applied to simulated data for a vertical line array and several source locations to several kilometers range. Complex pressure fields are computed using a parabolic equation model and results are considered in terms of 2D ensemble parameter estimates and marginal uncertainty distributions. [Work supported by NSERC.]

2:00

4pSP4. Using nested sampling with Galilean Monte Carlo for model comparison problems in acoustics. Paul Goggans, Wesley Henderson (Elec. Eng., Univ. of Mississippi, Anderson Hall Rm. 302, University, MS 38677, goggans@olemiss.edu), and Ning Xiang (Grad. Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Nested sampling is increasingly being used to calculate the evidence for competing models in Bayesian model comparison problems arising in acoustics applications. Use of nested sampling offers advantages in robustness over alternative methods of evidence calculation and enables evidence calculation for models with many parameters. The most challenging aspect of implementing nested sampling is sampling from the prior for the parameters constrained by a threshold likelihood value. For models with just a few parameters, sampling from the constrained prior can be accomplished with a simple Monte Carlo algorithm implementing a random walk, however, this simple method is inefficient and fails as the number of model parameters increases. John Skilling, the originator of nested sampling, has proposed the “Galilean” Monte Carlo method for efficiently sampling from the constrained prior when there are many parameters. Unlike the random walk method, the Galilean Monte Carlo method moves samples with a vector velocity, reflecting them from the likelihood constraint surface when necessary. This directed sampling gives the method its greater efficiency. In this paper we discuss our experience in implementing Galilean Monte Carlo in nested sampling and compare Galilean and random walk Monte Carlo for a model comparison problem in room acoustics.

2:20

4pSP5. Energy based Markov Chain Monte Carlo algorithms for Bayesian model selection. Tomislav Jasa (Thalgorithm Inc., 1688 Tarn Rd., Toronto, ON l4x1b1, Canada, jasa@ini.phys.ethz.ch), Jonathan Botts, and Xiang Ning (Grad. Program in Architectural Acoust., Rennselear Polytechnic Inst., Troy, NY)

Markov Chain Monte Carlo (MCMC) algorithms for Bayesian model selection have been increasingly applied to acoustics applications. One of challenging tasks required in Bayesian model selection is the exploration of high-dimensional multi-variate spaces such that a key quantity, termed the Bayesian evidence, can be estimated in order to rank a set of competing models. This work presents a class of energy-based MCMC algorithms specifically designed to estimate the Bayesian evidence. As illustrative examples, the energy-based MCMC algorithms are applied to the problem of filter design as used in human head-related transfer functions and in acoustic impedance boundaries within the finite-difference time-domain framework for room-acoustics simulations.

2:40

4pSP6. Nested sampling in practice. Jonathan Botts (Dept. of Media Technol., Aalto Univ., 110 8th St., Greene Bldg., Troy, New York 12180, botts.jonathan@gmail.com)

For problems requiring both model comparison and parameter estimation, nested sampling is an attractive choice because it provides an estimate of the normalized posterior. The critical assumption of nested sampling is that exploration proceeds regularly toward the region of maximum likelihood. However, in practice, achieving such regular compression is far from trivial, particularly for realistic, multi-modal problems. This paper offers a comparison of both systematic and random walk exploration for generating samples within a constrained prior. Ranges for size of ensemble and amount of exploration required for regular compression are also established for different acoustic data analysis problems.

Contributed Papers

3:00

4pSP7. Geoacoustic inversion via trans-dimensional sampling over seabed and error models. Gavin Steininger, Jan Dettmer, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. (Ring Rd.), Victoria, BC V8P 5C2, Canada, gavin.amw.steininger@gmail.com), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ. Univ., State College, PA)

This paper develops an efficient Bayesian sampling approach to geoacoustic scattering and reflection inversion based on trans-dimensional (trans-D) sampling over both the seabed model (number of sediment layers) and error model (autoregressive order to represent residual correlation). Sampling is carried out using a population of interacting Markov chains employing a range of sampling temperatures (parallel tempering). The approach is applied to both simulated and measured data. The advantages of trans-D autoregressive model sampling over alternative methods of error model selection is explored in terms of the reduction in posterior uncertainty of geoacoustic parameters and evaluation of residual correlation. The seabed is modeled as a stack of homogeneous fluid sediment layers overlying an elastic basement. Including elastic (shear) parameters in the basement makes this layer distinct from the overlying sediment layers and requires a novel formulation of the partition prior distribution for trans-D sampling. [Work supported by ONR.]

3:20

4pSP8. Bayesian-based estimation of acoustic surface impedance: finite difference frequency domain approach. Alexander Bockman (Massachusetts Inst. of Technol. Lincoln Lab., 244 Wood St., Lexington, MA 02420, alexander.bockman@ll.mit.edu), Cameron Fackler, and Ning Xiang (Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Design for acoustic performance in an interior fluid domain requires accurate description of boundary materials' specific acoustic impedance. The standard approach for the estimation of this material characteristic is the two-microphone, impedance-tube method. Modifications to the processing of the sampled acoustic field have been proposed to allow for more general test geometries. While analytical methods may be applied to a small class of ideal geometries, numerical methods provide greater geometric flexibility. In general, solutions to the wave equation forward problem are found from boundary element, finite element, or finite difference methods. The inverse problem of parameter estimation is solved by evaluating accuracy of prediction of the acoustic field for given distributions of the specific acoustic impedance parameter against observed data. In this presentation a Bayesian-network sampling approach is used to estimate specific acoustic impedance of a micro-perforated panel in an impedance tube test geometry. The choice of geometry and material allow for direct comparison to the two-microphone, impedance-tube method within the appropriate frequency range, and a theoretical model for the material beyond that frequency range. The potential to extend the frequency range of operation of the impedance tube is explored. Sensitivity of the method to nuisance parameters is discussed.

3:40

4pSP9. A Bayesian based equivalent sound source model for a military jet aircraft. David M. Hart (Physics, Brigham Young Univ., 363 N. 835 E., Lindon, UT 84042, dmh1993@studentbody.byu.edu), Tracianne B. Neilsen, Kent L. Gee (Physics, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

The two-source model for jet noise holds that turbulent mixing noise in jets is generated by uncorrelated, fine-scale (FSS) and partially correlated, large-scale (LSS) turbulent structures [Tam *et al*., J. Fluid Mech. **615**, 253–292, (2008)]. The noise from an F-22A Raptor is modeled with an equivalent source consisting of two line arrays of monopole sources. These arrays, one correlated and one uncorrelated, with Rayleigh distributed amplitudes, account for both FSS and LSS sound propagation [Morgan, J. Acoust. Soc. Am. **129**, 2442 (2011)]. The equivalent source parameters are selected based on Bayesian methods implemented with simulated annealing and fast Gibbs sampler algorithms. This method yields the best fit parameters, and the sensitivity of the solution is indicated by the generated posterior probability distributions. Analysis of the resulting equivalent sources shows that the directional, correlated line array has a greater effect on the near field sound, and the sensitivity of the array's parameters increases as the frequency increases. This equivalent source model can generate results up to 2500 Hz and accurately predict both near field and far field measurements. The analysis suggests that the shape of the source distribution changes as the frequency increases. [Work sponsored by the Office of Naval Research.]

4:00

4pSP10. Identification of acoustic sources with uncertain data. Vincent Martin and Frédéric Cohen-Tenoudji (Institut Jean Le Rond d'Alembert, UMR CNRS/UPMC 7190, 4 Place Jussieu, Paris 75252 Paris Cedex 05, France, vincent.martin@upmc.fr)

In inverse acoustic problems where attempting to identify the vibratory velocities of sources at the origin of an acoustic radiated field, we have the measured radiated field (called objective) on an antenna with numerous sensors and a propagation model. If both are erroneous, misidentification follows. Here, the problem is formulated in the frequency domain and solved in the least mean square sense. An impaired objective including an unstructured error has virtually no chance of satisfying the propagation equation. Accordingly, with an accurate radiation model, we cannot identify source velocities able to generating this objective. With the same model but now with unknown parameters (in the case of only one parameter it could be the speed of sound within the medium), it is expected intuitively that the parameter value aiming at the perturbed objective does not reach it but ultimately generates a pressure satisfying the wave equation, with a value near the correct pressure. The error in the model is structured in the sense that the model keeps a form satisfying the equations of physics. Currently, it is reported that this intuitive expectation is observed quantitatively through the geometric interpretation of over-determined inverse problems dealt with in L2.

4:20

4pSP11. Nested sampling-based design of multilayer microperforated panel sound absorbers. Cameron Fackler and Ning Xiang (Grad. Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, facklc@rpi.edu)

A model-based design approach for microperforated panel absorbers comprised of multiple panel layers is developed. Microperforated panels (MPPs) are becoming increasingly popular as sound absorbers, capable of providing broadband absorption with high absorption coefficients, without the use of traditional porous materials. To increase the bandwidth of the intrinsically peaked narrowband absorption of a single MPP, multiple such panels can be combined into composite sound absorbers. We propose a method based on Bayesian inference to design multilayered MPP absorbers capable of producing a user-specified absorption profile. Using nested sampling to accumulate Bayesian evidence and to implement Occam's razor, the method produces a design requiring the fewest number of MPP layers while meeting the specified design requirements.

4:40

4pSP12. Assessing model uncertainties for joint inversions of seismological data using a genetic algorithm. Priscilla Brownlow (Grad. Program in Acoust., Penn State Univ. Univ., 307B Dunham Hall, White Course Apartments, University Park, PA 16802, pdb153@psu.edu), Richard Brazier, Andrew Nyblade, Jordi Julia, and K. B. Boomer (Dept. of Geosciences, Penn State Univ., University Park, PA)

Error bars were generated for velocity models using receiver functions and surface wave dispersion curves for four seismic stations in southern Africa, with a genetic algorithm adapted from the code NSGA-II. Each receiver function and dispersion curve was originally created by Eldridge Kgaswane (2009). We examined these stations, and through a series of statistical resampling, we were able to place an uncertainty on each layer's velocity in the lithosphere. Each station was set to an initial model, which was perturbed to generate a series of best-fit models for the corresponding receiver functions and dispersion curves. For each layer of depth, a series of solutions evolve over a set number of generations using "survival of the fittest" to come up with these best-fit models. These were constrained to only consider geologically viable models, such as the velocity range in each layer and smoothing. Afterward, the error bounds on velocities were able to be placed on each layer. The velocity vs. depth plot gives the uncertainty from 1 to 2.5 km in depth. Now a better estimate of the velocities of the waves can be made, which leads to a better estimate of the composition of the lithosphere under southern Africa.

Session 4pUW

Underwater Acoustics and Signal Processing in Acoustics: Sparse Process Modeling Techniques for Acoustic Signal Processing

Paul J. Gendron, Cochair
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Chair's Introduction—12:55

Invited Papers

1:00

4pUW1. Reweighted sparse source-location acoustic mapping in shallow water. Pedro A. Forero and Paul A. Baxley (Maritime Systems Div., SSC Pacific, 53560 Hull St., BS/160/Rm. 190, San Diego, CA 92152, pedro.a.forero@navy.mil)

Various applications for monitoring and surveillance in littoral waters rely on passive sonar for localizing acoustic sources in shallow-water environments. Although adaptive matched-field processing (MFP) has been successfully used for localization, its performance is degraded when localizing multiple sources at low signal-to-noise ratios (SNRs) and in the presence of model mismatch. Robust MFP using, e.g., the white-noise constraint offers an alternative to cope with the mismatch issue but remains ineffective in the multisource and low SNR setup. This work capitalizes on sparsity for constructing a source location map for shallow water environments. Sparsity naturally arises since only locations corresponding to acoustic sources are expected to appear in the map (nonzero entries), while the remaining map locations are empty (zero entries). A high-resolution map is constructed via a two-step approach that capitalizes on a model for the acoustic propagation environment while being robust to model mismatch. During the first step the robust map is obtained by solving a regularized least-squares (LS) problem. Then, the map coefficients are used to devise a modified criterion with a weighted regularizer yielding a lower-ambiguity map, facilitating detection of quiet sources in the presence of loud interferers.

1:20

4pUW2. Application of statistical reduced isometry property to design of line arrays for compressive beamforming. Charles F. Gaumond and Geoffrey F. Edelmann (Acoust. Div., Naval Res. Lab., Code 7162, 4555 Overlook Ave. SW, Washington, DC 20375, charles.gaumond@nrl.navy.mil)

The Statistical Reduced Isometry Property (StRIP) and Statistical Null Space Property (SNSP) are presented and reduced to numerical algorithms. These properties are used to predict the utility of a specific subsampled array for use in compressive sensing. Three examples of subsampling an equally spaced array are presented: random, Golomb and Wichmann. The Golomb array uses a Golomb ruler that has no repeated sensor element spacings. The Wichmann array includes at least one of every possible interval of sensor element spacings. The SNSP is shown to be insensitive to subsampling in the type of cases shown. The Golomb array is predicted to have superior performance to the Wichmann for comparable subsampling. The use of these two subsamplings for beamforming using at-sea data from the Five Octave Research Array (FORA) is shown. [Research funded by the Office of Naval Research.]

1:40

4pUW3. Time-frequency localization issues in the context of sparse process modeling. Ananya Sen Gupta (Elec. and Comput. Eng., Univ. of Iowa, 4016 Seamans Ctr. for the Eng. Arts and Sciences, Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

Practical applications in acoustics such as shallow water acoustic communications often involve non-stationary processes that follow a time-varying sparse support. A classic example is acoustic scatter due to multiple reflections at the moving ocean surface and sea bottom, localized in the time-frequency domain as the sparsely distributed and time-varying Delay-Doppler spread function. In this work, we connect time-frequency localization of non-stationary processes with related issues in adaptive sparse sensing in the context of modeling and tracking the Delay-Doppler spread function. To this end, we provide an overview of methodological advances in adaptive sparse sensing techniques, and compare them over experimental field data collected as 15 m depth, 200 m range and moderate to rough sea conditions. We also comment on other adaptive techniques, such as least-squared error minimization algorithms, and discuss their extensions to the sparse sensing domain. We also explore the uncertainty principle underlying time-frequency representations, particularly in terms of how it influences the related and occasionally competing challenges to sparse process modeling: (i) restricted isometry criteria (RIP) for precise sparse reconstruction and (ii) choice of temporal window to localize the non-stationary Delay-Doppler spread function.

2:00

4pUW4. Performance limits of a compressive sensing application to beamforming on a line array. Jeffrey A. Ballard (Appl. Res. Labs., The Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, ballard@arlut.utexas.edu)

Compressive sensing is a sampling theorem that exploits the sparsity of a signal in a domain Ψ, while being spread out in a sensing domain Φ. For example, a sinusoid time domain signal in Φ can be represented by one non-zero coefficient in the frequency domain Ψ. The time-frequency relationship is similar to the space-angle relationship that exists in underwater acoustics for an array of hydrophones. Wavefront curvatures that are spread out in the space domain can be represented in the angle domain by a sparse vector. This work investigates the performance limits of using compressive sensing to resolve signals in the angle domain, a task usually accomplished by beamforming. For compressive sensing, it has been shown that the performance of recovering a signal is related to the number of measurements, the number of non-zero coefficients, and the dimension of Ψ [Candes and Wakin, IEEE Signal Process. Mag. 21-30 (March 2008)]. Typically, in underwater acoustics, the number of hydrophones and their locations are fixed, so that the performance is found to be dependent on the number of non-zero coefficients (signals in the water) and the dimension of the angle domain (beams). [Work supported by ARL:UT IRD.]

2:20

4pUW5. Sensitivity of co-prime arrays to shape perturbation. Andrew T. Pyzdek and R. Lee Culver (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, atp5120@psu.edu)

Co-prime arrays offer savings in both implementation and computation by reducing the number of array elements. For passive beamforming, a pair of specially-spaced sparse arrays organized as a co-prime array provides unambiguous source bearings through the cancellation of the grating lobes inherent in the pattern response of each array if processed by itself. In the ocean environment, however, towed line arrays are difficult to keep aligned and take on a time-varying shape. Hodgkiss [IEEE JOE (1983)] showed that array shape perturbation can lead to beam broadening, an effect which may interfere with the grating lobe cancellation of co-prime arrays. In the present paper, performance degradation of co-prime sparse arrays are examined under the condition of perturbed array shape. Simulations are used to compare a co-prime array of known element spacing and position to an array with small uncertainties in both element location and interelement spacing along the array. Possible correction methods are examined. [Work sponsored by ONR Undersea Signal Processing.]

2:40

4pUW6. Effects of multipath distortion on sparse signal parameter estimation. Sung-Hoon Byun, Sea-Moon Kim, and Hyun-Taek Choi (MOERI/KORDI, 171 Jang-dong Yeseong-gu, Daejeon 305-343, South Korea, byunsh@kiost.ac)

Shallow underwater acoustic channel is typically characterized as sparse channel and the sparsity has been actively exploited to estimate the channel accurately. However, distortion of the multipath signal components degrades the performance of sparse approximation and the amount of distortion is dependent on specific time-varying channel condition which each multipath encountered during transmission. In this research we measure the signal distortion of multipath components and analyze its impacts on the sparse channel estimation. Especially, we are interested in the effects of the spatial difference of the distortion on sparse approximation of the multichannel receiver data. To this end, we use variety of experimental data sets which have different characteristics of multipath signal distortion and analyze the relation between the amount of distortion and the signal residual obtained from sparse approximation.

3:00

4pUW7. Numerical simulations of compressive beamforming with a vertical line array in the deep ocean. Geoffrey F. Edelmann (Acoust. Div., U. S. Naval Res. Lab., 4555 Overlook Ave SW, Code 7145, Washington, DC 20375, edelmann@nrl.navy.mil), Ian Rooney (Dept. of Elec. Eng., Univ. of Massachusetts at Dartmouth, Dartmouth, MA), and Charles F. Gaumond (Acoust. Div., U. S. Naval Res. Lab., Washington, DC)

Vertical line arrays (VLA) have been previously deployed in the deep ocean to detect surface targets in the far-field. Conventional beamforming is based upon l2 minimization due to the ubiquitous nature of the fast Fourier transform, but l2 minimization is not a unique method for solving the beamforming problem. Using a basis pursuit algorithm, this research project applied l1 minimization to target detection via beamforming. The compressive beamforming technique was shown to produce narrower beams, comparable noise resistance, and a relaxed requirement on the number of elements required to produce bearing-time records. Compressive beamforming was applied to simulated data and successfully detected surface targets in deep water with 30% of the elements of the full array. [Work supported by the Office of Naval Research.]

3:20

4pUW8. A hierarchical mixture model for sparse broadband scattering functions between moving platforms. Paul J. Gendron (Maritime Systems Div., SSC Pacific, A460, Bldg. 1, Bayside Campus, 53560 Hull St., San Diego, CA 92152, paul.gendron@navy.mil)

A hierarchical mixture model is considered for sparse broadband acoustic Green's functions [J. Acoust. Soc. Am. **130**, 2346, Canadian Acoust. **40**(3)]. Such a mixture model can be employed to match arbitrary second order statistics of a channel over time-bandwidth and angle. The model matches these statistics while simultaneously admitting the degree of sparsity necessary to capture propagation between moving platforms through the ocean waveguide. The uppermost stage of the hierarchy is specified by a mean bulk relative platform speed. Conditioned on this is a structured field of beta distributions associated with the probabilities of ensonified paths over beam-Doppler and frequency. The mixture model of the response is built from Bernoulli indicator variables whose probabilities are drawn from the field of betas. Posterior mean and variance are reviewed and used in an underwater acoustic receiver structure to replace Kalman like estimators of response as well as phase looked loop structures for symbol timing. The performance in terms of mean squared error is 10 dB lower than conventional Wiener filtering schemes when the channel response is significantly sparse. Reduction of this margin occurs as either sparsity or SNR is degraded. This degradation in performance is quantified under a range of sparsity constraints associated with the beta variates. [Work supported by the Naval Innovative Science and Engineering Program and the Office of Naval Research.]

3:40

4pUW9. Bayesian sequential sparse sampling. Peter Gerstoft (Scripps Inst. of Oceanogr., UCSD, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu) and Christoph Mecklenbrauker (Tech. Univ. of Vienna, Vienna, Austria)

We consider the sequential reconstruction of source waveforms under a sparsity constraint from a Bayesian perspective. We assume that the wave field that is observed by a sensor array is from a spatially sparse source distribution. A spatially weighted Laplace-like prior is assumed for the source distribution and the corresponding weighted LASSO cost function is derived. We demonstrate the sequential sparse sampling using a line array and track the direction of arrival. In a real world example we track a source using a 2D array.

4:00

4pUW10. High resolution beamforming using sparse recovery from sensor array data. Ravi Menon and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr, MC-0238, UC San Diego, La Jolla, CA 92093, rmenon@ucsd.edu)

We consider the problem of adaptive beamforming using fewer snapshots than the number of sensors. Given an array of sensors in an environment and signals impinging on the array in the presence of background noise, it is of practical interest to be able to estimate the direction of arrival and power of the signals using as few snapshots as possible. In a sparse recovery framework, the signal vector is modeled as a sparse vector in the bearing domain. By casting the beamforming operation as an ℓ^1 minimization problem (as opposed to the convention ℓ^2 minimization), the signal vector can be recovered (the sensing matrix must satisfy the restricted isometry property). The results show an improvement over traditional beamforming methods and these are demonstrated using simulations.

THURSDAY EVENING, 6 JUNE 2013 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees on the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings beginning at 7:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Thursday are as follows:

Animal Bioacoustics	510b
Noise	511be
Speech Communication	515abc
Underwater Acoustics/Acoustical Oceanography	510d

4p THU. PM

FRIDAY MORNING, 7 JUNE 2013 517D, 7:55 A.M. TO 8:40 A.M.

Session 5aID

Interdisciplinary: Plenary Lecture: Public Space Acoustics for Information and Safety

Gilles A. Daigle, Chair
Inst. Microstructural Sci., National Res. Council, Ottawa, ON K1A 0R6, Canada

Chair's Introduction—7:55

Invited Paper

8:00

5aID1. Public space acoustics for information and safety. Hideki Tachibana (Chiba Inst. of Technol., Tsudanuma 2-17-1, Narashino, Chiba 275-0016, Japan, pon-t@iis.u-tokyo.ac.jp)

In such public-spaces as railway stations, airport terminal buildings, shopping arcade, etc., intelligible acoustic information is substantially needed as well as visual information. It is not only for announcements at ordinary times but for evacuation information at emergencies. In reality, however, there are many cases where the public address announcements are much deteriorated by reverberation and background noises in the spaces which results in poor aural intelligibility. In order to improve this kind of acoustic problems in public spaces, a comprehensive research is needed not only by building acoustics but also by speech science, electro-acoustics, signal-processing technology, and cognitive psychology. The acoustics research group in Chiba Institute of Technology, Japan, has been investigating this research topic these several years. In this presentation, the outline of this research and some of its outcome are introduced.

FRIDAY MORNING, 7 JUNE 2013 513ABC, 9:00 A.M. TO 12:00 NOON

Session 5aAAa

Architectural Acoustics: Room Acoustics Computer Simulation III

Diemer de Vries, Cochair
RWTH Aachen Univ., Inst fuer Technische Akustik, Aachen D-52056, Germany

Lauri Savioja, Cochair
Dept. of Media Technol., Aalto Univ., P.O. Box 15500, Aalto FI-00076, Finland

Contributed Papers

9:00

5aAAa1. Directional sound source modeling by using spherical harmonic functions for finite-difference time-domain analysis. Shinichi Sakamoto (5th Dept., Inst. of Industrial Sci., The Univ. of Tokyo, 4-6-1, Komaba, Meguro-ku, Tokyo 153-8505, Japan, sakamo@iis.u-tokyo.ac.jp) and Risa Takahashi (ONO SOKKI Co., Ltd., Yokohama, Japan)

Directivity of a sound source originates from the source's shape and its size relative to wavelength. Therefore, the directivity varies with a frequency of a sound. In order to precisely simulate a directivity of a source in the finite-difference time-domain analysis, basically, it is necessary to model the shape of the source geometrically in detail. In the case of sources with complex shapes, however, geometrically precise modeling of the source shape requires small size of spatial discretization, and such a fine mesh discretization results in huge computational costs. In this study, applying a basic theory of Fourier analysis in which arbitrary directivity can be constructed by linear combination of spherical harmonic function, the condition of the sound source for the finite-difference time-domain method to reduce computational cost and to enable efficient analysis against a source with complex directivity characteristics is investigated. Spatial distribution of sound pressure of initial condition for respective spherical harmonic function and correction method of spectral characteristics for the finite-difference time-domain analysis are described.

9:20

5aAAa2. The use of equivalent source models for reduced order simulation in room acoustics. Yangfan Liu and J. Stuart Bolton (Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2037, liu278@purdue.edu)

The sound field in a room can be predicted by using the Boundary Element Method if the motion of the source is known along with the impedance of the room surfaces and furnishings: however, these computations are very time consuming, especially at high frequencies. It is known that the total sound field in a room consists of three components: (1) free space source radiation; (2) the incoming sound field reflected from the room surfaces; and (3) the outgoing sound field scattered from the source surface, particularly if it is large: e.g., a flat screen television. For the purpose of fast computation, an equivalent source method (ESM) can be adopted in which two sets of equivalent sources represent the incoming and scattered sound field components, respectively. When the free space component is known, the parameters of the ESM can be estimated by least squares approximations of the impedance boundary conditions on the room and source surfaces. Accuracy and computational load can be balanced by the choice of source order and the constraints placed on equivalent source locations. Simulation results will be presented to compare the performances of different ESM configurations.

9:40

5aAAa3. New opera house in Astana: A recent opportunity to use a room acoustic scale model. Maria Cairoli, Enrico Moretti (Acoust. Lab, Biobyte, via ampere, 40, Milano, Milano 20131, Italy, mariacairoli@biobyte.net), and Anders Gade (akustik A/S, Gade & Mortensen, Charlottenlund, Denmark)

The acoustic design of the New Opera House in Astana (Kazakhstan) currently under construction was carried out by Biobyte in Milan, Italy (Enrico Moretti and Maria Cairoli) assisted by Gade & Mortensen Akustik, Denmark (Anders Gade). In order to predict the acoustic consequences of the room geometry and decide on details in the design such as diffusion treatment of curved surfaces, it was decided to build a 1:20 scale model, in which several room acoustical parameters were measured. However, the scale model also provided an opportunity to compare the performance of scale model testing and room acoustic predictions by two computer simulation programs, which were also used for predictions in the design process. Therefore, besides information about the adequacy of the diffusion treatment to avoid focusing from concave surfaces (an aspect which is not well described by computer simulation) we obtained data on measurement accuracy—or rather deviations—between the results from the scale model (DIRAC) and the two different room acoustic prediction programs (CATT and ODEON). These results will be presented and discussed in the paper.

10:00

5aAAa4. Analytic expression to derive the image source point between two symmetrical sloping planes. Santiago Ortiz (Noise Control, Ctr. for Appl. Acoust. and Non Destructive Evaluation (CAEND), CSIC, c/Serrano 144, Madrid 28006, Spain, santiago.ortiz@caend.upm-csic.es), U. Peter Svensson (Electron. and Telecommunications, NTNU-Norwegian Univ. of Sci. and Technol., Trondheim, Norway), and Pedro Cobo (Noise Control, Ctr. for Appl. Acoust. and Non Destructive Evaluation (CAEND), CSIC, Madrid, Spain)

The sound field in open cavities with sloping walls can be represented by image sources and edge diffraction components. Sloping walls will generate a systematic structure to the positions of the image source, and explicit expressions for those positions are presented here. Using such explicit expressions, rather than the general image source method, makes it straightforward to study arbitrarily high orders. When two opposite walls form parts of an infinite wedge, then their image sources will be positioned on a cylindrical surface. The addition of a floor is easy, while adding more walls requires an iterative procedure, with explicit visibility tests. First-order diffraction components can then be generated for each image source. Simulation results are compared with experimental measurements in a three dimensional open cavity with sloping walls.

10:20

5aAAa5. Acoustical archaeology—Recreating the soundscape of John Donne's 1622 gunpowder plot sermon at Paul's Cross. Benjamin Markham, Matthew Azevedo (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, mazevedo@acentech.com), and John Wall (Dept. of English, NC State Univ., Raleigh, NC)

Auralization has become a valuable tool to explore the acoustics of spaces and activities that no longer exist. Generally, acoustical archaeology has explored a fairly limited number of sources in a space to determine specific acoustical aspects of the sound of the spaces and to separate intentionally designed acoustical phenomena from the often unintended effects of the architecture. We have expanded this technique to recreate the entire soundscape of a specific event, in this case John Donne's 1622 Gunpowder Plot sermon at Paul's Cross, outside St. Paul's Cathedral in London as it was prior to the fire of 1666. This work augments ambisonic auralization techniques with techniques borrowed from computer-aided music composition and audio production to create an immersive acoustical environment for the purpose of exploring the experience of listeners at many positions in a crowd that can be varied in size in real time. The paper outlines the role of geometric acoustics modeling, real-time convolution, randomized and statistically derived sound event triggers, and other techniques employed to auralize a soundscape that includes the sermon, crowd response, and the ambient sounds of pre-Industrial London.

10:40

5aAAa6. Auralization of municipal public address announcements by applying geometrical sound simulation and multi-channel reproduction techniques. Junichi Mori (Grad. School of Eng., Chiba Inst. of Technol., 2-17-1 Tshudanuma, Narashino-city, Chiba 271-0016, Japan, junichi@acoust.cs.it-chiba.ac.jp), Sakae Yokoyama (Inst. of Industrial Sci., The Univ. of Tokyo, Tokyo, Japan), Fumiaki Satoh, and Hideki Tachibana (Chiba Inst. of Technol., Chiba, Japan)

Municipal public address system (M.P.A. system) for disaster prevention is an important system for supplying information outdoors. The speech intelligibility of the M.P.A. system, however, tends to be deteriorated by long-pass-echoes due to reflections from the surrounding buildings and the sounds from the loudspeakers covering other sub-areas. When designing such a M.P.A. system, effective tools for prediction of outdoor sound propagation are needed. For this aim, the authors have been investigating the application of the sound simulation technique based on geometrical acoustics. In addition, auralization is also desirable to subjectively assess the speech intelligibility of the M.P.A. system in acoustical design. Therefore, the authors are developing a simulation technique in which the sounds calculated by the geometrical simulation are reproduced through a 6-channel reproduction system, by which 3D information can be aurally realized. In this paper, the outline of this simulation method and some examples of its application are presented.

11:00

5aAAa7. Semi-anechoic music recording using multi-microphone technique to simulate orchestra directivity in room acoustic auralizations. Alban A. Bassuet, Terence Caulkins, Joseph Digerness (Acoust., Ove Arup and Partners, 77 Water St., New York, NY 10002, alban.bassuet@arup.com), and Glenn KnicKrehm (R&D, ConstellationCenter, Cambridge, MA)

Over the last two decades, computer generated auralizations have become increasingly relevant to the design process of new performance spaces. Auralizations are based on anechoic material, and simulation realism is often limited by a small repertoire of recordings and the challenges of simulating an orchestra's complex sound radiation characteristics. To address some of these issues, this paper describes a method to record—in semi-anechoic conditions—large musical ensembles for use in computer simulations. The method uses a multi-microphone hemispherical array distributed over an orchestra playing on a hard floor. A ray-tracing program models the orchestra as sectors of a hemisphere corresponding to each microphone location. Microphone signals are then mapped to their respective source sector. The method permits to record and simulate an entire orchestra with its directivity, including the movement of the musicians while playing. The paper describes the influence of the radius and angular resolution of the microphone array; it also investigates different source configurations in the computer model. The method has been implemented with renowned performers and orchestras and has been used for auralizations of several new concert hall design projects.

11:20

5aAAa8. Interoperability Building Information Modeling and acoustical analysis software—A demonstration of a performance hall design process. Sunyoung Kim, Robert C. Coffeen, and Paola Sanguinetti (Architecture, Univ. of Kansas, 1730 Bagley Dr. #6, Lawrence, KS 66044, sunyoungkim@ku.edu)

By sharing and managing the database for a building model, Building Information Modeling (BIM) facilitates the design process at less cost. Some of BIM software has capabilities for acoustical analysis, but it is limited to noise level demonstration of MEP system in a building. Sophisticated acoustical analysis programs, such as EASE and CATT, provide the useful parameters to determine the sound quality of a space with graphs, A/V systems, material database, and auralization. This research explores how BIM with discipline-specific data reinforced with acoustical information will be beneficial to both architects and acousticians, by allowing them to communicate efficiently. The design process for a performance hall is used to demonstrate how to interoperate between BIM system and acoustical analysis programs. Revit Architecture and EASE program are used as design and analysis tools for for the project. A

model is developed to improve the process of embedding acoustic information in BIM and sharing knowledge in an integrated delivery process.

11:40

5aAAa9. The system (software) for acoustic and lighting calculation (SCAL) program like support for acoustic and lighting conditioning in interior spaces. Beatriz S. Garzón (AAII, IAA, FAU-SeCyT, UNT-CONICET, Av. Roca 1900, Crisóstomo Alvarez 722, San Miguel de Tucumán, Tucumán 4000, Argentina, bgarzon@gmail.com), Carlos R. Coria, and Maria Josefina Manson (FAU-SeCyT, UNT, San Miguel de Tucumán, Tucumán, Argentina)

This work aims to show an alternative software tool for calculating the acoustic and lighting conditioning and comfort in interior spaces. This program is developed in a desktop version and it can run on different operating systems because it is multiplatform. It is easy to operate. Program tools allow a quick calculation because the system values are stored standards of acoustics and lighting through an intuitive interface. The calculations can be stored and open for future modification; also, the results can be summarized and printed. The activities for its development and validation were as follows: (1) Survey on the requirements and interest that the system should contain. (2) Implementation of the program: It was conducted jointly between subject architecture specialists and computer engineers. (3) Development of training materials for the understanding of the functioning of the application. 4) Testing and evaluation period: The functional and non-functional requirements were evaluated and verify if the obtained results would be the expected results.

FRIDAY MORNING, 7 JUNE 2013 513DEF, 9:00 A.M. TO 11:40 A.M.

Session 5aAAb

Architectural Acoustics: Effects of Room Boundaries on Diffusion and Reverberation

Isabelle Schmich-Yamane, Chair
CSTB, 24 rue Joseph Fourier, Grenoble 38000, France

Contributed Papers

9:00

5aAAb1. Energy decay analysis of non-diffuse sound fields in rectangular rooms. Tetsuya Sakuma and Kazushi Eda (The Univ. of Tokyo, 5-1-5 Kashiwanoha, Kashiwa 277-8563, Japan, sakuma@k.u-tokyo.ac.jp)

Rectangular rooms with irregular aspect ratio or nonuniform absorption distribution apt to have non-diffuse sound fields, where the curvature of energy decay curve occurs in the reverberation process. In general, this curvature leads to longer reverberation times than the estimates by the Sabine or Eyring formula; however, it can be suppressed to a certain extent with diffusive wall surfaces. Recently, the author has proposed a new approximate theory of reverberation in rectangular rooms with specular and diffuse reflections. In this paper, the validity of the theory is investigated by two case studies with geometrical and wave-based acoustic simulation. In the first study, energy decay in a variety of rectangular rooms with changing the aspect ratio, the absorption distribution and the scattering coefficient, is simulated with the image source method and the ray tracing method. In the second study, a two-dimensional FDTD analysis is performed to demonstrate the frequency dependence of energy decay in a variety of rectangular rooms with flat or corrugated walls. Finally, the simulated results are compared with the theoretical ones, and the validity and the limitation of the approximation are discussed.

9:20

5aAAb2. Mean-free-paths in concert and chamber-music halls and validation of the Sabine/Eyring equations for predicting their reverberation times. Leo L. Beranek (10 Longwood Dr., 265, Westwood, MA 02090, beranekleo@ieee.org) and Noriko Nishihara (Takenaka R & D Inst., Chiba, Japan)

The Eyring/Sabine equations assume that in a large irregular room a sound wave travels in straight lines from one surface to another. It is assumed that the surfaces have an average sound absorption coefficient αav and that the average distance between reflections, i.e., the mean-free-path MFP, is 4V/Stot where V is the volume of the room and Stot is the total area of all of its surfaces. No account is taken of diffusivity of the surfaces. The 4V/Stot relation is based on experimental determinations in 11 very differently shaped rooms made by Vern Knudsen [*Architectural Acoustics*, (Wiley, 1932), pp. 132–141]. This paper sets out to test this relation experimentally for a wide variety of unoccupied concert and chamber music halls with seating capacities ranging from 200 to 15,000. To determine the MFP's in them, the measured values of the sound strengths G and reverberation times RT were used. Computer simulations of the sound fields for several of these rooms were also made to determine the MFP's. For these rooms, 4V/Stot is found to be an acceptable relation for MFP in the Sabine/Eyring equations.

9:40

5aAAb3. Exploration of the differences between a pressure-velocity based in situ absorption measurement method and the standardized reverberant room method. Peter Cats (Fontis Hogeschool Eindhoven, Eindhoven, Netherlands), Emiel Tijs, and Daniel Fernandez Comesana (Microflown Technol., Tivolilaan 205, Arnhem 6824BV, Netherlands, tijs@microflown.com)

Several measurement techniques are available for the determination of the sound absorbing properties of material packages. The Kundt's method and the reverberant room method are the most commonly used techniques and they are standardized. However, both methods cannot be used *in situ*. In the past, it has been shown that the PU *in situ* method can be used in a broad frequency range (typically from 300 Hz up to 10 kHz), on small samples (typically 0.03 m^2 to 0.38 m^2 or larger), while hardly being affected by background noise and reflections. Several studies revealed that similar results can be obtained as with the Kundt's tube if the measurements are performed under certain circumstances. A thorough comparison with the reverberant room method has not been conducted yet. In this paper, preliminary results are presented of a comparison of the reverberant room method, the PU *in situ* method, and measurements with PU probes in a reverberant room. Several factors that may cause discrepancies amongst the methods are discussed. In addition, edge effects, which are experienced with the reverberant room method due to the finite size of the sample, are visualized with 3D intensity measurements that are performed in a reverberant room.

10:00

5aAAb4. The reverberation radius (rH) in an enclosure with asymmetrical absorption distribution. Higini Arau–Puchades (ArauAcustica, Barcelona, Spain) and Umberto Berardi (CEE, Worcester Polytechnic Inst., via Orabona 4, Bari 70125, Italy, u.berardi@poliba.it)

This paper reviews the concept of the reverberation radius (rH) from the viewpoint of the classic theories of Sabine and Eyring. These theories are only valid when the sound field is uniformly distributed around a room, or in other words, when the energy density throughout a room is constant. Nevertheless, these theories have been applied to any spatial sound diffusion situation, also where this is not diffuse. For example, they are currently used in rooms with asymmetric absorption distribution. The limits of this approach also characterize the revised theory of Barron and more recent investigations. This paper proposes a solution to calculate the reverberation radius in rooms with non-uniformly distributed sound absorption (rH-ND).

10:20–10:40 Break

10:40

5aAAb5. Optimal design of a sound reflector by particle swarm optimization. Yuuki Tachioka (College of Humanities and Sci., Nihon Univ., E612 3-6-13 Fujisawa, Fujisawa, Kanagawa 251-0052, Japan, yuuki_tachioka@yahoo.co.jp)

For room acoustic design, the shape of the room and the materials need to be well considered. Designing such rooms requires extensive experience. Numerical analyses can minimize designers' burdens by the automatic generation of many room candidates and the selection of an optimized one on the basis of objective measurements. Research on both numerical analysis methods and optimal design methods is required, but few studies are done regarding the latter. This paper proposes an optimal method by combining geometrical acoustic simulation with an optimization technique known as "particle swarm optimization." Although the final aim is to achieve a global acoustic design, this paper is limited to local shape design, which is still important, considering that local shapes affect the characteristics of the global sound field. The proposed method is used for designing sound reflectors, whose aim is to scatter first reflected sound waves across the entire field. Hence, the optimization variables are sound reflectors shapes and materials, and the objective function is to evaluate the uniformity of a sound field in terms of room acoustic indices (T20, C80, and Ts). Compared with the initial reflectors, the optimized sound reflectors exhibited a higher uniformity in the room acoustic indices.

11:00

5aAAb6. How absorptive can a diffuser be to accurately measure random incidence scattering coefficients? Isabelle Schmich-Yamane, Marina Malgrange, and Christophe Rougier (Département Acoustique et Eclairage, CSTB, 24 rue Joseph Fourier, Grenoble 38000, France, christophe.rougier@cstb.fr)

In auditorium design, absorptive surfaces are as important as diffusive ones. Diffuse surface elements are rarely completely reflective. Absorption and scattering coefficients are not independent from each other. A real-size diffuser designed to be installed in a new auditorium under construction in France has been measured according to the ISO 17497-1 standard. This diffuser shows higher absorptiveness than any previously tested sample in CSTB's laboratory. The calculation of the random incidence scattering coefficient in the ISO 17497-1 standard is based on the Sabine equation for elements with α_s less than 0.5. This paper presents how scattering coefficient calculations give incoherent results for more absorptive diffuse elements and proposes a new calculation method based on the Eyring equation. Measurement results obtained with both methods are presented and compared.

11:20

5aAAb7. Prediction and measurement of the random incidence scattering coefficient of periodic reflective rectangular diffuser profiles. Isabelle Schmich-Yamane (Département Acoustique et Eclairage, CSTB, 24 rue Joseph Fourier, Grenoble 38000, France, isabelle.schmich@cstb.fr), Jean-Jacques Embrechts (INTELSIG group - Département E.E.I., Université de Liège, Sart-Tilman (Liege 1), Belgium), Markus Müller-Trapet (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany), Christophe Rougier (Département Acoustique et Eclairage, CSTB, Saint MArtin d'Hères, France), and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Germany)

Periodic reflective rectangular diffuser profiles have been previously studied by Embrechts through an analytical method calculating their random incidence scattering coefficients. This paper presents the results of further investigations obtained by real-scale and 1:5th scale random incidence scattering coefficient measurements on these profiles. All measurements were performed according to the corresponding ISO 354 and ISO 17497-1 standards. The measurement results are compared with the calculation results derived by the analytical method. Although some periodic profiles could not be treated by the analytical model, fairly good agreements are obtained between the different approaches. The real scale and 1:5th scale measurement results are also presented and compared.

Session 5aBAa

Biomedical Acoustics: Acoustic Characterization of Biological Media

James C. Lacefield, Chair
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Contributed Papers

9:00

5aBAa1. Evaluation of tumor cell death response in locally advanced breast cancer patients to chemotherapy treatment by scattering property estimates using ultrasound backscatter. Lakshmanan Sannachi, Hadi Tadayyon, Ali Sadeghi-Nani, Omar Falou, Zahra Jahedmotlagh (Imaging Res., Radiation Oncology, Med. Biophys., SunnyBrook Health Sci. Ctr., Univ. of Toronto, 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, Lakshmanan.Sannachi@sunnybrook.ca), Michael Oelze (Bioacoustics Res. Lab., Dept. of Electron. and Comput. Eng., Univ. of Illinois, Urbana, IL), and Gregory Czarnota (Imaging Res., Radiation Oncology, Med. Biophys., SunnyBrook Health Sci. Ctr., Univ. of Toronto, Toronto, ON, Canada)

Breast cancer tumor response to chemotherapy in 29 patients was examined using quantitative ultrasound. Backscatter parameters, such as the average scatterer diameter (ASD) and average acoustic concentration (ACC), were estimated from regions-of-interest in tumors prior to treatment onset and at four times during neoadjuvant chemotherapy treatment (weeks 1, 4, 8, and prior to surgery). Gaussian and Anderson form factor models were used over an analysis bandwidth of 4 to 9 MHz to obtain ASD and ACC estimates. The Gaussian model did not fit with the measured data as well as Anderson model. Both ASD and ACC estimates yielded significant differences with therapy times in clinically treatment responded patients. Data indicated increases of approximately 3.5 μm in scatterer size and 6.6 dB/cm in acoustic concentration obtained maximum at week 8 in treatment responding tumors. Non-responding tumors did not show any significant difference in both the parameters. This study demonstrates that the scattering parameters have the potential to being used in quantifying the changes in tumors during treatment noninvasively and distinguishing treatment responders and non-responders.

9:20

5aBAa2. Characterizing tumor heterogeneous response to chemotherapy using low-frequency ultrasonic spectroscopy. Ali Sadeghi-Naini, Omar Falou, and Gregory J. Czarnota (Imaging Res., Radiation Oncology, Med. Biophys., Sunnybrook Health Sci. Ctr., Univ. of Toronto, Odette Cancer Ctr., TB064, 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, a.sadeghi.naini@utoronto.ca)

Textural analysis techniques in conjunction with low-frequency ultrasonic spectroscopy have been proposed for characterization of heterogeneous responses developed within tumors undergoing chemotherapy. Such characterization can be beneficial for early appraisal of tumor responses to the cancer therapy. Breast xenograft tumor-bearing animals were treated with chemotherapy. Animals were assessed with low-frequency ultrasound data acquired at different times after chemotherapy exposure. Following imaging, tumors underwent standard histological analysis for detecting cell-death effects. Several average and texture-based parameters were derived from normalized ultrasonic spectral parametric maps. Statistically significant differences were revealed between parameters extracted from treated and untreated tumors 12–24 h after exposure. Regression analyses were also performed in order to assess levels of correlation between non-invasive ultrasonic surrogates of therapy response and standard histological findings, where strong correlations were obtained with a maximum r^2 value of 0.98. Obtained results demonstrate that ultrasound-based textural properties of tissue can be used for characterizing the micro-structure alterations and heterogeneous responses within tumors early on during treatment. This is an important observation which can be applied clinically for treatment response monitoring and predicting patient responses to therapy early after therapy initiation.

9:40

5aBAa3. Ultrasonic assessment of the *in vitro* biomechanical stability of a dental implant. Romain Vayron (CNRS, Laboratoire MSME, 61, Ave. du Général de Gaulle, bât P2, Créteil 94010, France, romain.vayron@u-pec.fr), Patrick Karasinski (Université Paris Est, Laboratoire LISSI, Créteil, France), Domitille Loriot, and Guillaume Haiat (CNRS, Laboratoire MSME, Créteil, France)

Dental implants are widely used for oral rehabilitation. However, there remain risks of failure, which are difficult to anticipate. The objective is to investigate the potentiality of a quantitative ultrasound method to assess the biomechanical stability of a dental implant *in vitro*. Two experimental configurations were considered using a 10 MHz contact transducer located at the implant extremity. For each ultrasound measurement, a quantitative indicator I is derived based on the time variation of the amplitude of the rf signal. First, seven implants were embedded in tricalcium silicate-based cement. One implant was left without any mechanical solicitation and six implants were subjected to mechanical stresses during 24 h. The ultrasonic response of each implant was measured during 24 h. The results show no variation of I without mechanical solicitation, while I significantly increases as a function of fatigue time. Second, 10 implants were unscrewed from bone tissue and their ultrasonic response was measured after each turn. Analysis of variance tests revealed a significant effect of the amount of bone in contact with the implant on the distribution of I. The results show the feasibility of our QUS device to assess the biomechanical quality of the interface surrounding the implant.

10:00

5aBAa4. Axial transmission measurements of guided modes on bone phantoms and *in vitro* human bone specimen. Jean-Gabriel Minonzio, Josquin Foiret, Maryline Talmant, and Pascal Laugier (LIP (Laboratoire Imagerie Parametrique), UPMC - CNRS, 15 rue ecole de medecine, Centre des Cordeliers, Paris 75006, France, jean-gabriel.minonzio@upmc.fr)

Cortical bone porosity has been evidenced as being a major if not the major “footprint” of bone loss and fragility. Several studies report that cortical bone behaves like a waveguide. Measurements of guided mode wavenumbers together with appropriate waveguide modeling have therefore the potential for providing estimations of effective stiffness coefficients (which are largely determined by cortical porosity) and also cortical thickness. However, data interpretation is challenging due to the heterogeneous, dissipative, and irregular nature of the wave guide. Moreover, surrounding and internal tissues modify the guided modes. This paper presents measurement of the wavenumbers of guided wave modes, using a multi-transmitter multi-receiver axial transmission probe. The method has been previously validated on isotropic elastic and visco-elastic plates covered by a soft tissue-mimicking layer. As an extension of our previous works, we have conducted *in vitro* experiments on (i) bone-mimicking tubes without and with soft-tissue mimicking layers and on (ii) human radius and tibia specimens. Despite challenging measurement conditions owing to low signal-to-noise ratio,

guided mode wavenumbers have been successfully measured. The results were consistent with a free plate model. Data inversion led to an reasonably accurate estimate of the shell thickness and bulk velocities for all cases.

10:20

5aBAa5. Determination of bone properties from Lamb type of waves. Jean-Gabriel Minonzio, Josquin Foiret, Pascal Laugier, and Maryline Talmant (Laboratoire d'Imagerie Parametrique, UPMC Univ Paris 06, UMR 7623, 15 rue de l'Ecole de Medecine, Paris F-75006, France, maryline.talmant@upmc.fr)

Our domain of interest is cortical bone characterization, i.e., determination of structural and material properties by means of ultrasound waves. The frequency spectrum of Lamb type of waves are the input data of the method of parameters identification, based on a least mean square algorithm. Specificities of clinical measurements of long bones require first to address the issue of measurements with limited access [Minonzio *et al*., J. Acoust. Soc. Am. (2010)] and second, to develop methods for a combined determination of structural and elastic properties. The first physical model was the free plate under plane strain assumption for transverse isotropic materials with four elastic parameters and one structural (thickness). Experiments on flat plates and tubes of circular cross section are used as test case and the method was validated by comparison to independent techniques dedicated to material properties [Bernard *et al*. (2012)]. The same method was applied on *in vitro* experiments on cortical bone samples. Here, the bone thickness was measured independently by x-ray technique. This study opens the perspective of updating the physical model to determine additional relevant parameters, the tube diameter and/or the soft tissue properties in case of *in vivo* measurements.

10:40

5aBAa6. Development and validation of resonant ultrasound spectroscopy for the measurement of cortical bone elasticity on small cylindrical samples. Simon Bernard, Quentin Grimal, and Pascal Laugier (Laboratoire d'Imagerie Paramétrique, CNRS - UPMC Univ Paris 6, UMR 7623, 15 ru de l'école de médecine, escalier A, 3ème étage, Paris 75006, France, simon.bernard@upmc.Fr)

Documentation of cortical bone elastic properties is important for orthopedic applications and fracture risk prediction. Cortical bone is heterogeneous and anisotropic. Ideally, measurements should be performed on cylindrical samples of characteristic size of a few millimeters, which are adapted to the native geometry and size of bones. Our objective was to measure bone with resonant ultrasound spectroscopy (RUS), which is a powerful method to determine the elastic constants of a sample from a set of its resonant frequencies. Application of RUS to bone is difficult due to viscoelasticity, which causes resonance peaks to overlap. Some of the resonances in the investigated frequency band cannot be observed. The formulation of the inverse problem, which requires pairing measured and predicted frequencies, must be adapted in the case of bone. We developed dedicated signal processing methods to retrieve resonant frequencies from overlapping peaks. A probabilistic approach was used, which allows an automated pairing based on a Bayesian criterion. The method was validated on isotropic (PMMA) and anisotropic (bone phantom) calibrated materials. The adapted RUS method allows the automated assessment of bone elasticity with a precision of a few percents, which is a significant improvement over traditional methods based on velocity measurements.

11:00

5aBAa7. Ultrasound propagation through bone fractures with reamed intramedullary nailing: Results from numerical simulations. Fernanda Catelani (Biomed. Eng. Program - COPPE, Federal Univ. of Rio de Janeiro, Nova Friburgo, Brazil), Ana Paula M. Ribeiro (Biomed. Ultrasound Lab., Estácio de Sá Univ., Nova Friburgo, Brazil), Carlos Alberto V. Melo (Physical Therapy Dept., Estácio de Sá Univ., Nova Friburgo, Brazil), Wagner C. Pereira (Biomed. Eng. Program - COPPE, Federal Univ. of Rio de Janeiro, Rio de Janeiro, Brazil), and Christiano B. Machado (Biomed. Ultrasound Lab., Estácio de Sá Univ., Av. Ariosto Bento de Mello, 35/101, Nova Friburgo, Rio de Janeiro 28610100, Brazil, cbmfisio@gmail.com)

Low-intensity pulsed ultrasound stimulation (LIPUS) accelerates fracture healing, enhancing the release of inflammatory mediators and subsequent bone formation. The reamed intramedullary nailing (with the same diameter of the medullary cavity) is a surgical procedure widely used in Medicine. The aim of this work was to study ultrasound propagation inside fractures with and without reamed intramedullary nailing using 2D simulations. It was used a custom-made simulation code applied to numerical models (a 4-mm thick cortical plate, a medullary cavity with radius 4 mm with and without reamed nailing, and fracture gaps varying from 1 to 3 mm). A 1-MHz emitter was positioned above fracture center, and 14 receptor transducers were uniformly placed inside fracture gap. The acquired signals were used to estimate the time-of-flight of the first arriving signal (TOF) and the energy amplitude by means of the root mean square (RMS). TOF was slightly influenced by fracture gap variations. It was observed an increase in RMS values with the presence of metal nailing, due to the reflection in the interface water-metal. The receptors placed near cortical plates received more energy (constructive interference between the direct and lateral waves). For the case of reamed nailing, ultrasound stimulation may be intensified.

11:20

5aBAa8. Computational simulations of time of flight and attenuation of first arriving signal from healing process of diaphyseal femur fractures. Paulo Tadeu C. Rosa, Aldo José F. Pereira, and Wagner Coelho A. Pereira (Biomed. Eng., UFRJ, P.O. Box 68510, Univ. City, Rio de Janeiro, Rio de Janeiro 21941-972, Brazil, ptcardozo@gmail.com)

Quantitative ultrasound (QUS) has been proposed to evaluate structural conditions from bone tissue in a non-invasive way. These techniques are based on the fact that the ultrasound propagation depends directly of tissues structures, so they carry information about them. Based on these arguments, the aim of this work was to estimate the time-of-flight (TOF) and attenuation of the first arriving signal (FAS) using computational simulations (Wave2000® CyberLogic, Inc., NY, EUA) in an axial transmission model in different kinds of fracture, during the healing process. In this work, we used QUS techniques to analyze the fracture healing process. The FAS has been chosen because it does not suffer interference of other waves since it is the first signal to arrive at the receiver. TOF increases immediately after bone fracture. When bone tissue starts to consolidate, TOF decreases and stabilizes with the same value of the intact bone. Attenuation is bigger in oblique and spiculate fracture than transversal ones for the same stage, which suggests that attenuation is sensitive to the kind of fracture. Other studies are being conducted to clarify this point.

11:40

5aBAa9. Ultrasonic attenuation and speed in phantoms made of polyvinyl chloride-plastisol and graphite powder. Guillermo A. Cortela (Ultrasonic Lab., U de la República, Montevideo, Uruguay), Luis E. Maggi (Escola Superior de Educacao Fisica, Universidade Estadual de Goias, Goiania, Brazil), Marco Antonio v. Kruger (Programa de Engenharia Biomédica - COPPE, Universidade Federal do Rio de Janeiro, Rio de Janeiro, Brazil), Carlos A. Negreira (Ultrasonic Lab., Universidad de la República, Montevideo, Montevideo, Uruguay), and Wagner C A. Pereira (Programa de Engenharia Biomédica - COPPE, Universidade Federal do Rio de Janeiro, P.O. BOX 68510, Rio de Janeiro, Rio de Janeiro 21941972, Brazil, wagner.coelho@ufrj.br)

Biological phantoms are very useful for controlled experiments on biomedical ultrasound. Nevertheless they are normally made of organic materials with short time-duration. We have studied the ultrasonic properties of test-blocks made of polyvinyl chloride-plastisol (PVCP) that are very stable in time. In this work, we analyzed ultrasonic (US) attenuation and speed at 1 MHz, as a function of temperature (15–45°C) of five phantoms made with PVCP and different concentrations of graphite powder (0, 0.5, 1, 2, and 5%) using the classical transmission method. US speed diminishes almost linearly (from 1408 to 1333 $m.s^{-1}$) as temperature increases. In general attenuation lied between 0.73 and 0.09 $dB.cm^{-1}$, but presenting a more complex behavior. For graphite concentrations of 0.5 and 1%, attenuation was lower than for 0% and for the other two phantoms (2 and 5% concentrations) attenuation was higher. This behavior can be perhaps due to the fact that the fabrication temperature for 0.5 and 1% was 140°C and for the other was 170°C. Although the standard recipe is 170°C, we observed that smaller temperatures may add in adjusting the attenuation values and it is a very useful property to mimic different biological tissues. We are now working on multilayer phantoms of PVCP.

Session 5aBAb

Biomedical Acoustics: Imaging, Therapy, and Bubbles (Again)

Michel Versluis, Chair
Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands

Contributed Papers

9:00

5aBAb1. The effect of boundary proximity on the fundamental and subharmonic emissions from individual microbubbles at higher frequencies. Brandon Helfield (Med. Biophys., Univ. of Toronto, 2075 Bayview, Toronto, ON M4N 3M5, Canada, brandonhelfield@gmail.com), Ben Leung (Sunnybrook Res. Inst., Toronto, ON, Canada), and David E. Goertz (Med. Biophys., Univ. of Toronto, Toronto, ON, Canada)

It is recognized that the proximity of a boundary can influence the dynamic behavior of acoustically stimulated microbubbles. In a biomedical ultrasound context, this is relevant to molecular imaging with targeted microbubbles, and when microbubbles are near vessel walls or contained within microvessels. Theoretical models have recently been developed to examine these effects, but experimental work has been more limited and primarily focused on the assessment of resonant frequency effects rather than its impact on nonlinear behavior, which is perhaps more relevant to imaging applications. Understanding this behavior is important to improving microbubble detection and for the quantitative interpretation of contrast images. With the use of an optical trap, this study experimentally investigates the effect of boundary proximity (0 to 200 μm) and boundary stiffness (Opticell and agarose) on fundamental and subharmonic emissions from individual Definity and MicroMarker bubbles at 11 MHz. The scattered pressure dependence on proximity from an Opticell boundary resulted in an oscillatory dependence, while from an agarose boundary resulted in a decreasing fundamental and an increasing subharmonic response with increasing distance from the boundary. These experimental findings are not entirely captured by basic analytical simulations, likely suggesting that more complex numerical models may be required.

9:20

5aBAb2. Bifurcation structure of the ultrasonically excited microbubbles undergoing buckling and rupture. Amin Jafari Sojahrood, Raffi Karshafian, and Michael C. Kolios (Dept. of Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B2K3, Canada, amin.jafarisojahrood@ryerson.ca)

Bubbles exposed to ultrasound are long known to exhibit highly nonlinear and chaotic dynamics. Bubbles stabilized by a shell material (MBs) are widely used as contrast agents in diagnostic ultrasound. However, the nonlinear behavior of the shell significantly increases the complexity of the dynamics. In order to realize the full potential of the MBs, better understanding of the MB behavior is necessary. In this study, the bifurcation structure of the MB with nonlinear shell behavior is investigated for the first time. The Marmottant model was numerically solved, and the bifurcation diagrams of the radial oscillations of the MB were plotted versus the control parameters (e.g., buckling radius). In agreement with recent experimental observations, results predict the generation of subharmonics at very low acoustic pressures. In addition, the numerical simulations predict the generation of higher order subharmonics (e.g., period 3) at very low acoustic pressures (<300 kPa and 25 MHz), which contradicts the predictions by free bubble and viscoelastic shell MB models. Results revealed the strong influence of the buckling and rupture radius on the order of the subharmonics. The numerical results were verified by experimental observations of higher order subharmonics in the oscillations of Definity at 25 and 55 MHz.

9:40

5aBAb3. Ultrafast dynamics of the acoustic vaporization of phase-change microdroplets. Oleksandr Shpak, Laura Stricker (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), Tom Kokhuis, Ying Luan (Biomed. Eng., Erasmus MC, Rotterdam, Netherlands), Brian Fowlkes, Mario Fabiilli (Dept. of Radiology, Univ. of Michigan, Ann Arbor, MI), Detlef Lohse (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), Nico de Jong (Biomed. Eng., Erasmus MC, Rotterdam, Netherlands), and Michel Versluis (Phys. of Fluids Group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl)

Superheated emulsion droplets are a promising tool for localized drug delivery. The physical mechanisms underlying the ultrasound-triggered vaporization of phase-change emulsions are largely unexplored. Here we study the acoustic vaporization of individual micron-sized perfluoropentane droplets at a nanoseconds timescale. The nucleation and growth of the vapor bubbles was imaged at frame rates up to 20 Mfps. The droplet vaporization dynamics was observed to have three distinct regimes: (1) prior to nucleation, a regime of droplet deformation and oscillatory translations; (2) a rapid growth of a vapor bubble enhanced by ultrasound-driven rectified heat transfer; and (3) a final phase characterized by a relatively slow expansion that is fully dominated by heat transfer. A method to measure the moment of inception of the nucleation event with respect to the phase of the ultrasound wave is proposed. A simple physical model captures quantitatively all of the features of the subsequent vapor bubble growth. In addition, we study the role of gas through a model for a vapor-gas bubble, including thermal diffusion and gas diffusion inside the liquid and we find good agreement with the experimental data. We underline the fundamental role of gas diffusion to prevent total recondensation of the bubble at collapse.

10:00

5aBAb4. Dynamics of a constrained bubble. Alexey Maksimov and Yuri Polovinka (Phys. of the Ocean, Pacific Oceanological Inst. Far Eastern Branch of the Russian Acad. of Sci., 43, Baltic St., Vladivostok 690041, Russian Federation, maksimov@poi.dvo.ru)

In recent studies [Bostwick and Steen (2009), Prosperetti (2012), Ramalingam *et al*. (2012)], the shape oscillations of constrained drops and bubbles have been analyzed and the difficulty of possible singularity in the pressure and the curvature of the free surface at the location of the constraint has been identified. This result indicates the need for a small cutoff length scale, as well as some more input from the physics at smaller length scales. The method of restraining the bubble against rising by attaching it to a wire is a common procedure in conducting precision acoustic measurements. The dynamics of the tethered bubble differs from those of free bubble due to variation in inertial mass. The objective of this study is to obtain a closed-form, leading order solution for the dynamics of the constrained bubble. It was shown that, by using the invariance of the Laplace equation to conformal transformations and the geometry of the problem, the toroidal coordinates provide separation of variables and are most suitable for analysis of this problem. Thus, the dynamics of the constrained bubble in toroidal coordinates can be investigated by using analytical approach and by analogy to the dynamics of a free spherical bubble.

10:20

5aBAb5. Temporal and spatial characteristics of nonlinear acoustic field generated by an extracorporeal shockwave therapy device: Modeling and measurements. Maria Karzova (Dept. of Acoust., Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Lab. de Mécanique des Fluides et d'Acoustique, Ecole Centrale de Lyon, Leninskie Gory 1/2, Moscow 119991, Russian Federation, masha@acs366.phys.msu.ru), Vera Khokhlova (Dept. of Acoust., Phys. Faculty, Lomonosov Moscow State Univ., Moscow, Russian Federation and Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Camilo Perez, and Thomas Matula (Ctr. for Industrial and Med. Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Extracorporeal shock wave therapy (ESWT) refers to the use of focused shock pulses to treat certain musculoskeletal disorders. Although the technology is used clinically, acoustic characteristics of ESWT fields and their relation to the bioeffects induced are not fully understood. In the present work, the acoustic field of a clinical ESWT device (Duolith SD1) was characterized in water using a combined measurement and modeling approach. Simulation model was based on the nonlinear KZK equation; the boundary condition was based on the pressure waveforms measured in a plane 5 mm away from the therapy head. The model was used to simulate and to analyze pressure waveforms along the axis of the therapy head, 2D spatial distributions of peak pressures, and the shock structure. The modeling results were found in a good agreement with experimental data. Simulations performed for different initial source pressure amplitudes showed that a shock is not formed at the focus, even at the maximum operational level of the device. Predictions from the modeling at higher output settings suggest that a true shock would develop if the initial pressure output were doubled. [Work supported in parts by RFBR 12-02-31830 and 12-02-31418 grants and student stipend from the French Government.]

10:40

5aBAb6. Dynamic time reversal acoustic focusing of ultrasound for biomedical applications. Yegor D. Sinelnikov (SUNY at Stony Brook, 126 Liberty Ave., Port Jefferson, NY 11777, yegorasha@yahoo.com), Alexander M. Sutin (Stevens Inst. of Technol., Hoboken, NJ), Sergey Y. Tsyuryupa, and Armen P. Sarvazyan (Artann Labs., Ewing Township, NJ)

Time Reversal Acoustic (TRA) system provides effective focusing in inhomogeneous media that can be used in various biomedical applications including high intensity ultrasound treatment, ultrasound-assisted drug delivery, ultrasonic battery charging of implants, etc. In many cases, the TRA focusing is conducted in tissues with time varied properties and variation in propagating media degrades the TRA focused signal. Dynamic focusing is required to maintain the same acoustic intensity in the focus. We suggest adjustment to the radiated signal in order to maintain the maximal amplitude in the focus using the inverse filtering technique with Tikhonov regularization. The suggested algorithm enables to determine the radiated signal based on the changes in the focused signal received by a beacon. Developed algorithm of TRA refocusing was tested in the experiments with a TRA system with an aluminum reverberator comprising ten piezotransducers with resonance around 1 MHz, DAC control system and a multichannel binary amplifier. This system was built in Artann Laboratories for investigation of time reversal acoustic focusing for hyperthermia therapy and drug delivery. The dynamic TRA refocusing is shown to be effective even in the case of significant lateral shifts in the media and strong focused signal variation.

11:00

5aBAb7. Superresolution imaging in ultrasound B-scan imaging. Kevin J. Parker (Dept. of Elec. & Comput. Eng., Univ. of Rochester, Hopeman Eng. Bldg. 203, P.O. Box 270126, Rochester, NY 14627-0126, kevin.parker@rochester.edu)

A number of imaging systems exhibit speckle, which is caused by the interaction of a coherent pulse reflecting off of random reflectors. The limitations of these systems are quite serious since the speckle phenomenon creates a pattern of nulls and peaks from subresolvable scatterers or targets that are difficult to interpret. Another limitation of these pulse-echo imaging systems is that their resolution is dependent on the full spatial extent of the propagating pulse, usually several wavelengths in the axial or propagating dimension and typically longer in the transverse direction. This limits the spatial resolution to many multiples of the wavelength. This paper focuses on the particular case of ultrasound B-scan imaging and develop an inverse filter solution that eliminates both the speckle phenomenon, and the poor resolution dependency on the pulse length and width, to produce SURUS (super-resolution ultrasound) images. The key to the inverse filter is the creation of pulse shapes that have stable inverses. This is derived by use of the standard Z-transform and related properties. Although the focus of this paper is on examples from ultrasound imaging systems, the results are applicable to other pulse-echo imaging systems that also can exhibit speckle statistics.

11:20

5aBAb8. Wearable long duration ultrasound therapy in rotator cuff tendinopathy. George Lewis (ZetrOZ, 421 N. Aurrora St., Ithaca, NY 14850, george@zetroz.com), Lyndon Hernanez (Med. College of Wisconsin, Milwaukee, WI), George Lewis (Transducer Eng., Andover, MA), and Ralph Ortiz (Cayuga Med. Ctr., Ithaca, NY)

Approximately one-third of the westernized adult population will experience some type of shoulder pain. The purpose of this pilot study was to evaluate a novel self-applied wearable therapeutic ultrasound device in the management of shoulder pain from rotator cuff tendinopathy. The Institutional Review Board of Cayuga Medical Center (CMC) approved this study and informed consent for the study was obtained from all subjects. The wearable ultrasound device provides 90 mW/cm^2, 2.95 MHz, continuous-wave ultrasound for 5.5 h on a single charge. Four subjects meeting the studies inclusion criteria, presenting with rotator cuff tendinopathy, and demonstrating cognitive and functional ability to apply the pager-size device were enrolled at the outpatient physical therapy center of CMC. Subjects were instructed to wear the device for 3–4 h/day for 12 consecutive treatment sessions, and record their daily pain score on the visual analog scale (1 to 10) and global rate of health improvement scale (-7 to 7). Across the 12 treatments, subjects reported a 30% reduction in pain and 52% improvement in health compared to baseline scores ($p < 0.05$). The results of the pilot study indicate the device may be applied successfully and provides supportive evidence for a placebo controlled study.

11:40

5aBAb9. Ultrasound-equipped colonoscope for point-of-procedure colorectal preparation and examination. Lyndon Hernanez (Med. College of Wisconsin, 8701 Watertown Plank Rd., Milwaukee, WI 14850, lvhernan@hotmail.com), Martin Ton (Cornell Univ., Ithaca, NY), Shane Fleshman, and George Lewis (ZetrOZ, Ithaca, NY)

Colorectal cancer is the second leading cancer killer in the United States. Poor colon preparation occurs in 20–40% of colonoscopies in the community, which increases the duration of the colonoscopy by at least 10% and the cost of the procedure by up to 22% due to repeat visits. The goal of this research was to develop and preliminarily evaluate the first ultrasound-cavitation equipped colonoscope as an innovative approach to liquefy fecal matter with water/cavitation and improve colonoscope utility. Two ultrasound-equipped colonoscopes were developed. The first consisted of a 30 element 235 kHz array that was mounted as a cap on the tip of a commercial colonoscope (Olympus). The second consisted of a Time-Reversal Acoustic extroporeal 32 channel 100 kHz array that was electrically steered to the commercial colonoscope using PVDF detectors to acquire and monitor the TRA focusing routines. Both systems were evaluated in a series of bench tests for fecal liquification, as well as in the porcine cadaver. Results show that ultrasound exposure assists the liquification of fecal matter and 50 kPa exposure to ultrasound increases liquification by greater than 50 times. Blinded histological reports on excised tissues showed no significant different findings between control and ultrasound experiments.

Session 5aEA

Engineering Acoustics: Sound Emission from Vehicle and Rotating Machinery

Stephen Elliott, Chair
Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom

Contributed Papers

9:00

5aEA1. Effects of median barriers on highway noise levels. Jonghoon Kim (Civil Eng., Arizona State Univ., 472W. Carob Dr., Chandler, AZ 85248, ghim96@gmail.com), Louis Cohn (Civil and Environ. Eng., Univ. of Louisvile, Louisville, KY), and Ning Shu (Environ., AZTEC Eng., Phoenix, AZ)

Median barriers are widely used on roadways in the United States. The main purpose of this paper was to evaluate a median barrier performance in reducing traffic noise using the latest FHWA Traffic Noise Model, TNM version 2.5. For this study, median barriers were modeled on three different roadway configurations—at grade, depressed, and elevated. The analysis results indicated that the range of insertion loss for a median barrier at grade was less than 1.5 dBA with a barrier height of 2.5 to 10 ft. The range of insertion loss for a median barrier on a depressed roadway (5, 10, and 20 ft below grade) was 0 to 2.8 dBA with a barrier height of 2.5 to 4.5 ft and insertion loss increased up to 4.3 dBA with a taller barrier height of 6 to 10 ft. On an elevated roadway (5, 10, and 20 ft above grade), the range of insertion loss for a median barrier was 0 to 1.7 dBA with a barrier height of 2.5 to 10 ft. Given the results of this research, it is reasonable to conclude that a standard median barrier would not provide a significant level of noise reduction. Furthermore, taller median barriers do not alter this conclusion. When considering the construction of very tall median barriers for noise reduction purposes, the costs would far outweigh the relatively minimal benefits of this approach.

9:20

5aEA2. Multichannel feedback control of interior road noise. Jordan Cheer and Stephen J. Elliott (Inst. of Sound and Vib. Res., Univ. of Southampton, Univ. Rd., Highfield, Southampton, Hampshire SO17 2LG, United Kingdom, j.cheer@soton.ac.uk)

Active noise control systems offer a potential method of reducing the weight of passive acoustic treatments and, therefore, increasing a vehicles' fuel efficiency. The active control of engine noise can be implemented cost-effectively by using the car audio loudspeakers as control sources and an array of low-cost microphones as error sensors. Such systems have been commercially implemented, but without also controlling road noise their subjective benefits may be limited. The active control of road noise using a feedforward control strategy has also been practically demonstrated, but these systems require a number of accelerometers to be mounted to the vehicle's structure to obtain a coherent reference signal and, therefore, lead to a significant implementation cost. This paper proposes a multichannel feedback system for the active control of road noise, which uses an array of microphones and car audio loudspeakers, which is common to a feedforward engine noise control system. The design of the multichannel feedback controller is described and its performance is validated using offline simulations employing data measured in a small city car.

9:40

5aEA3. Determination of the transfer matrix of an automotive compressor under realistic flow conditions. Benoit Rousselet (Lab. PHASE, Université Paul Sabatier - Toulouse III, FR CTL L16 1 29, 1 allée Cornuel, Lardy 91510, France, rousseletbenoit@hotmail.com), Vincent Gibiat (Lab. PHASE, Université Paul Sabatier - Toulouse III, Toulouse, France), Alain Lefebvre, and Stephane Guilain (Powertrain Eng. Div., Renault S.A., Lardy, France)

Waves propagating into air intake pipes of automotive engines have been widely studied and are commonly used to obtain high volumetric efficiency. Nowadays, most of modern engines have a centrifugal compressor in their intake line. If the acoustic of intakes lines for naturally aspirated engines is well known, it is not the case for lines comprising a compressor, where both the geometry and the flow effects are of primary interest. As it is quite difficult to experimentally determine the acoustical behavior of a complete line comprising a compressor, a new system has been developed for determining acoustical transfer matrices of each separated element of the line with flow. This new method is based on the two loads method; a derivative of the Two Measurements Three Calibrations calibration method has been introduced. The basics of the determination of transfer matrices with flow with our method are presented. Experimental results on cylindrical tubes and parts of intake lines are then discussed. Finally, an example of a transfer matrix of centrifugal compressor is presented.

10:00

5aEA4. Acoustical absorption by materials in a nacelle of turbojet. Ibanez R. Carlos (Mech., Benemérita Universidad Autónoma de Puebla, Boulevard del Niño Poblano 2901, Puebla, Puebla 72197, Mexico, carlos.ibanez@iberopuebla.mx) and Panneton Raymond (Génie Mécanique, Université de Sherbrooke, Sherbrooke, QC, Canada)

The paper discusses the absorption of certain materials with different acoustic properties used in a nacelle of aircraft is investigated using numerical model. A model 2-D with the configuration of axisymmetrical and the differential equations of the Continuity and Helmholtz equation is realized, it couples a finite element description with a boundary conditions a rigid wall, impedance and a frequency range (low–high) in order to evaluate three treatments (resonator, foam, and the combination of both of materials) and his frequency behavior. Experimental results are presented to validate the model in the special case of insertion loss and the coefficient of sound absorption.

10:20

5aEA5. Investigation of hysteresis effect in the impinging jet using acoustic data. Abhijit Dhamanekar and K. Srinivasan (Mech. Eng., Indian Inst. of Technol., Madras, TDCE Lab., Mech. Dept. IITM, Chennai, Tamilnadu 600036, India, abhikd11@gmail.com)

Experimental acoustic investigation of underexpanded free and impinging jet is carried out for various nozzle-plate spacing. The reservoir pressure is slowly increased from atmospheric pressure to 6 bar and than decreased from 6 bar to atmospheric pressure. The free jet acoustic radiation remains same for both paths, but it is observed that for impinging jet the acoustic

radiations differ in some regions. The hysteresis effect observed in acoustic characteristics may be due to the presence of hysteresis effects in the recirculation zone of the impinging jet. This variation is significant for nozzle-plate spacing of 2 to 4 times jet diameter. It is also seen that the acoustic staging occurs for low pressure and high pressure for small and large nozzle-plate spacing, respectively.

10:40

5aEA6. Effects of scalloping depth on the sound generated by turbofan engine lobed mixers. Hao Gong, Kaveh Habibi, and Luc Mongeau (Dept. of Mech. Eng., McGill Univ., Rm. 364, McDonald Eng. Bldg., 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada, hao.gong@mail.mcgill.ca)

The growing stringency of community noise regulations for commercial turbo-fan engine requires the development of effective jet noise suppression configurations. The large number of geometrical design parameters for lobed mixers precludes trial and error experimental studies. In this study, a robust computational tool was used to investigate the effects of the scalloping depth on sound radiated from a lobed mixer. The near field sound and flow were simulated using a flow solver based on the Lattice Boltzmann Method (LBM). The far field radiated sound was predicted using the Ffwocs William-Hawkings (FWH) surface integral method. One baseline confluent nozzle and three nozzles with varying scalloping depth were considered. Low Mach number flow was assumed, with operating conditions selected to best approach conditions for actual engines. The effects of an outer mean flow to simulate forward flight were not included. The increased scalloping depth was found to enhance mixing and to reduce noise levels in the mid-to-high frequency range, as anticipated. Results were in qualitative agreement with available experimental results.

11:00

5aEA7. Analysis of simulated flyover Contra Rotating Open Rotors noise data by using beamforming techniques for moving sources. Vincent Fleury and Alain Chélius (Onera, 29 Ave. de la Division Leclerc, Chatillon 92322, France, vincent.fleury@onera.fr)

The relevance of beamforming techniques to analyze conventional airframe noise sources from flyover noise measurements is now well-known. With the development of Contra Rotating Open Rotors (CROR), the performance of such techniques needs to be assessed. To this aim, realistic ground CROR noise data are simulated. First, the flow is simulated by a U-Rans approach in the CROR frame. The front rotor consists of 11 blades and the rear rotor is composed of 9 blades. In addition, the incoming flow Mach number is $M = 0.2$. Then, the hydrodynamic blade pressure is propagated toward a ground microphone array by using the Ffowcs-Williams Hawkings' equation. The CROR follows an horizontal trajectory at constant speed ($M = 0.2$) and 150 m altitude. Finally, the microphone array data are analyzed by a conventional beamforming-based deconvolution technique for moving sources, DAMAS-MS. The results show that the amplitude of the harmonics of each blade passage frequency is correctly recovered. However, the amplitude of the interaction tones is badly estimated. To overcome this difficulty, the DAMAS-MS methodology, based on an acoustic source model constituted of uncorrelated monopoles, should be modified in order to introduce the correlation between the monopoles.

11:20

5aEA8. An experimental investigation on the near-field turbulence for an airfoil with trailing-edge serrations and an owl specimen. Kunbo Xu, Weiyang Qiao, Liang Ji, and Weijie Chen (School of Power and Energy, Northwestern Polytechnical Univ., No.127 Youyi Rd. Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

The ability to fly silently of most owl species has long been a source of inspiration for finding solutions for quieter aircraft and turbomachinery. This study concerns the mechanisms of the turbulence for an airfoil with trailing-edge serrations and a real owl specimen . the turbulence spatio-temporal information are measured with 3D hot-wire. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel on the SD2030 airfoil. Mach numbers range up to 0.3, with the Reynolds numbers from 1.9e+4 to 2.6e+5, the angle of attack α at -50, 00, and +50, the sawtooth of $\lambda/h = 0.2$. The individual trailing-edge serrations tips and valleys could be seen in the wake region. It is showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge. It is also found that the turbulence peak occurs further from the airfoil surface in the presence of the serrations, and the serrations generate additional horseshoe vortices shed in the near wake. However, the boundary layer statistics slightly upstream from the TE are not influenced by the serrations.

11:40

5aEA9. Acoustic analysis of turbine generator to predict overall noise using statistical energy analysis methodology. Vasudev S. Nilajkar (India Technol. Ctr., GE, JFWTC, #122, EPIP, Phase 2, Hoodi Village, Whitefield Rd., Bangalore, Karnataka 560066, India, vasudev.nilajkar@ge.com), Sadeo Ramtahal, and Anil K. Tolpadi (Generator and Elec. Technol., GE, Schenectady, NY)

In the growing regions across the globe, products are being differentiated not only by their performance but also on characteristics like noise, aesthetics, ergonomics, etc. Noise norms are getting more and more stringent and customers are looking for quieter products. Engineers and technicians now face the challenge of developing effective products for lower noise emissions in addition to enhanced performance. As generators are a major noise source in a power plant, it is essential to contain generator noise well within the limit to meet statutory requirements. The major contributors to generator noise include cooling fan, rotor jet, electromagnetic excitation, and vibrations. These sources should be configured properly for reducing the overall noise. Therefore, it is essential to establish a robust process to predict the generator noise accurately. This paper describes the process for predicting generator noise by using Statistical Energy Analysis methodology. Correlation of the predictions with test data is also discussed.

Session 5aMUa

Musical Acoustics: Acoustic Analysis of Musical Instruments

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Vincent Gibiat, Cochair
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Contributed Papers

9:00

5aMUa1. Real-time magnetic resonance imaging fo the upper airways during harmonica pitch bends. Peter R. Egbert, Lewis K. Shin, David Barrett (School of Med., Stanford Univ., Stanford, CA), Thomas D. Rossing (School of the Blues, Los Altos Hills, CA), and Andrew Holbrook (Ctr. for Res. in Comput. Music and Acoust., Stanford Univ., Stanford, CA 94305, aholbrook@stanford.edu)

Skilled harmonica players learn to bend the pitch of certain notes by a semitone or more, especially in blues playing, by adjusting the shape of their vocal tract [Bahnson *et al*., J. Acoust. Soc. **103**, 2134 (2008)]. The changes of the vocal tract have been partially viewed with endoscopy and ultrasound but are still incompletely understood. While in a magnetic resonance imaging (MRI) scanner, a professional harmonica player using nonmagnetic, MRI-compatible diatonic harmonicas played draw and blow notes in both unbent and bent positions. Three-dimensional static and two-dimensional real-time magnetic resonance images of the upper airway were recorded in the sagittal and coronal planes. We identified and characterized the static and dynamic changes that facilitated pitch bends for low and high notes with specific attention to tongue positioning, tongue morphology, and airway shape. Deliberate changes in the tongue shape are often accompanied by changes in other parts of the vocal tract such as the pharynx.

9:20

5aMUa2. An acoustic study of ceramic traditional whistles. Vincent Gibiat (PHASE-UPS, Toulouse Univ., 118 route de Narbonne, Toulouse 31400, France, vincent.gibiat@univ-tlse3.fr) and Marie-Barbara Le Gonidec (Music Dept., MuCEM, Paris, France)

Ceramic whistles are very common all along the old European traditions. Gifts, jars, or simple decorative objects these whistles may have extremely various shapes: animals, jars, pitchers and sometimes flutes. Even with this extremely variability, their interior shapes are very similar: a channel where to blow, an edge, and a small cavity. Some of them produce a single sound when others present one or more tone holes, the latter corresponding mainly to flute shaped whistles. Their acoustic behavior appears to be very simple: Helmholtz resonators or flutes driven by the common non linear excitation system. Nevertheless some are known as water whistles and sometimes as "nightingales." The acoustic study presented will first verify that the simplest whistles are really working as Helmholtz resonator with the noticeable exception of the flute shaped ones that remain whistles for the organologist but are a different object for the acoustician. Then, a particular attention will be given to the "nightingales" ones. Their sound production will be related to the level of water they contain and the modulation of the acoustic signal analyzed in terms of coupling between the oscillation of the water and the Helmholtz acoustic resonance.

9:40

5aMUa3. Sound analysis and synthesis of Marquis Yi of Zeng's chime-bell set. Chih-Wei Wu (Master Program of Sound and Music Innovative Technol., National Chiao Tung Univ., 1001 Ta Hsueh Rd., Hsinchu 30010, Taiwan, g9611504@alumni.nthu.edu.tw), Chih-Fang Huang (Dept. of Information Commun., Kainan Univ., Luzhu Shiang, Taoyuan, Taiwan), and Yi-Wen Liu (Dept. of Elec. Eng., National Tsing Hua Univ., Hsinchu, Taiwan)

In this paper, the analysis and synthesis results from a complete set of Chinese chime-bells (also known as Chinese two tone bells) are presented. Consisting of 65 bells with different sizes and tones, Marquis Yi of Zeng's set is an ancient musical instrument with fascinating acoustical features but scarcely appears in current music performances for being huge and inaccessible. To preserve this cultural legacy in digital form, sounds of a complete set of replicated Marquis Yi of Zeng's chime-bells were recorded and analyzed, and the pitch discrimination of fundamental frequencies between this replica and the original set has been evaluated. Sound synthesis models of chime-bells were constructed using multiple inharmonic digital waveguides, creating chime-bell like sounds. Quality of synthetic sounds was evaluated using both objective and subjective measures. Objectively, the similarity between synthetic and recorded sounds was compared in both the spectral and the temporal domains. The subjective measure was achieved through listening tests. Results show that sounds from different bells on the rack could be successfully generated from the proposed models, leading toward the realization of virtual chime-bell set in the future.

10:00

5aMUa4. Acoustic analysis from pentatonic Angklung. Anugrah S. Sudarsono and I Gde Nyoman Merthayasa (Eng. Phys., Institut Teknologi Bandung, Jalan Ganesha 10, Bandung, West Java 40151, Indonesia, anugrahsabdono@gmail.com)

Angklung is traditional music instruments from Indonesia made from bamboo. Pentatonic angklungs are angklung that only have five notes. The notes are from the traditional Sundanese scale. Experiment was done to analyze spectral, temporal, and spatial parameters from pentatonic angklung. Spectral analysis was done by analyze the pitch and timbre. Temporal analysis was done by analyze the sound envelope and from τe parameter from the music. Spatial analysis was done from measurement of the sound energy by multiple microphones in semi anechoic room. Pitch and timbre were analyzed with Fast Fourier Transform from 29 angklungs. It was found from spectral analysis that angklung has overtone 1.44 f0, 2 f0, 3.47 f0, 6.31 f0, and 7 f0 with f0 as the fundamental frequency. The scales of the notes are very different from western scale and pentatonic angklung has four different scale intervals. Sound envelope analysis shows that angklung has attack time 192–179 ms and release time 346–902 ms. τe is a parameter that correlated with bandwidth frequency, lower value of τe correlated with wider bandwidth frequency. The value of τe from pentatonic angklung music is between 17 and 50 ms. The value was measured from six songs played by pentatonic angklung. Spatial analysis shows that the direction of sound headed to the left front side of angklung.

Session 5aMUb

Musical Acoustics: Digital Libraries for Speech and Singing

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Steven R. Livingstone, Cochair
Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada

Contributed Papers

10:40

5aMUb1. A digital library to advance interdisciplinary research in singing. Annabel J. Cohen (Psych., Univ. of Prince Edward Island, 550 Univ. Ave., Charlottetown, PE C1A 4P3, Canada, acohen@upei.ca), Ichiro Fujinaga (Music, McGill Univ., Montreal, QC, Canada), Nyssim Lefford (Sound Music & Digital Media, Lulea Tech. Univ., Stockholm, Sweden), Theresa Leonard (Audio Recording, Banff Ctr., Banff, AB, Canada), George Tzanatakis (Comput. Sci., Univ. of Victoria, Victoria, BC, Canada), and Coralie Vincent (Developmental Phonet., CNRS, Paris, France)

In 2008, at the ASA/EAA symposium honouring pioneering scientist of singing, Johan Sundberg, the Advancing Interdisciplinary Research in Singing (AIRS) project was introduced as a major collaborative research initiative on singing [Cohen, Acoustics 08, Paris (2008), 3177–3182]. Over 70 collaborators around the world were to investigate singing from perspectives of development, education, and well-being. A digital library was to facilitate distant team members' work on the same data, such as examples from voice studios around the world, performance stages, playgrounds, public places, solos, groups, classrooms, intergenerational or multicultural choirs, therapeutic settings or new tests of singing skills [Vincent *et al.* PEVOC9 (2011)]. Plans also included tools for annotation and analysis along with relevant documents and images. The present progress report on this endeavor describes preliminary prototypes, stages of development, and the current functional implementation. It is noted that although singing is primarily an acoustic and auditory phenomenon, video records of the singer are highly valuable. Their benefit, however, must be weighed against challenges arising from ethical considerations and storage requirements. Issues of ownership and data sharing are also raised as are practical matters of choice of platform, storage, formats, backup, human resources, and long-term preservation.

11:00

5aMUb2. Acoustic differences in the speaking and singing voice. Steven R. Livingstone, Katlyn Peck, and Frank A. Russo (Psychology, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, steven.livingstone@ryerson.ca)

Speech and song are universal forms of vocal expression that reflect distinct channels of communication. While these two forms of expression share a common means of sound production, differences in the acoustic properties of speech and song have not received significant attention. Here, we present evidence of acoustic differences in the speaking and singing voice. Twenty-four actors were recorded while speaking and singing different statements with five emotions, two emotional intensities, and two repetitions. Acoustic differences of speech and song were reported in several acoustic parameters, including vocal loudness, spectral properties, and vocal quality. Interestingly, emotion was conveyed similarly in many acoustic features across speech and song. These results demonstrate the entwined nature of speech and song, and provide evidence in support of the shared emergence of speech and song as a form of early proto-language. These recordings form part of our new Ryerson Audio-Visual Database of Emotional Speech and Song (RAVDESS) that will be freely released in 2013.

11:20

5aMUb3. The diverse environments multi-channel acoustic noise database: A database of multichannel environmental noise recordings. Joachim Thiemann (METISS, INRIA/IRISA, Campus Beaulieu, Rennes 35042, France, joachim.thiemann@irisa.fr), Nobutaka Ito (NTT Commun. Sci. Labs., Rennes, France), and Emmanuel Vincent (METISS, INRIA/IRISA, Rennes, France)

Multi-microphone arrays allow for the use of spatial filtering techniques that can greatly improve noise reduction and source separation. However, for speech and audio data, work on noise reduction or separation has focused primarily on one- or two-channel systems. Because of this, databases of multichannel environmental noise are not widely available. DEMAND (Diverse Environments Multi-channel Acoustic Noise Database) addresses this problem by providing a set of 16-channel noise files recorded in a variety of indoor and outdoor settings. The data was recorded using a planar microphone array consisting of four staggered rows, with the smallest distance between microphones being 5 cm and the largest being 21.8 cm. DEMAND is freely available under a Creative Commons license to encourage research into algorithms beyond the stereo setup.

11:40

5aMUb4. Variation of rhythm metrics in regional varieties of Acadian French. Wladyslaw Cichocki (French, Univ. of New Brunswick, P.O. Box 4400, Fredericton, NB E3B5A3, Canada, cicho@unb.ca), Sid-Ahmed Selouani, Alaidine B. Ayed, Catherine Paulin, and Yves Perreault (Gestion de l'information, Université de Moncton, Shippagan, NB, Canada)

This preliminary investigation studies variation of rhythm metrics in dialects of Acadian French spoken in New Brunswick (Canada). The aim is to determine whether regional and social factors are significant sources of this variation. Data are recordings of 140 speakers who represent five geographic regions, both genders and two age groups (20–30 and 40–55 years of age). Sound files were segmented manually; eight interval-based metrics were calculated. Mean metric scores indicate that regional varieties of Acadian French pattern with other dialects of French; these scores are similar to those of other "syllable-timed" languages. Regional differences are found for several metrics (%V, Delta C, VarcoC, nPVI-C, rPVI-C), although the five geographic regions are not all clearly distinguished by these metrics. A major pattern that emerges is regional variation in high-vowel devoicing and/or deletion. Analyses also show that social factors are significant sources of interspeaker variability: gender (VarcoV, nPVI-C, rPVI-C) and age (DeltaV, VarcoV). These results suggest a certain amount of complementarity between regional and social factors in their effects on rhythm metrics.

Session 5aNSa

Noise: Urban Noise and Modeling

Fabian Probst, Chair
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Contributed Papers

9:00

5aNSa1. The Internet of sound observatories. Dick Botteldooren, Timothy Van Renterghem, Damiano Oldoni, Samuel Dauwe, Luc Dekoninck, Pieter Thomas, Weigang Wei, Michiel Boes, Ramanan Muthuraman, Bert De Coensel, Bernard De Baets, and Bart Dhoedt (Information Technol., Ghent Univ., St. Pietersnieuwstraat 41, Gent 9000, Belgium, dick.botteldooren@intec.ugent.be)

With the advance of electronics, sound level meters have become more powerful when it comes to analyzing and storing huge amount of measurements. In recent years, these devices have been hooked up to the internet and stream life data. In the IDEA project, the whole concept of a sound observatory is turned upside down by stripping the sensor nodes to their bare essential, and by migrating all logic and data storage to computing centers. This opens new opportunities in particular for long-term environmental sound monitoring and analysis. As unlimited computing power is available, more advanced analysis such as auditory scene analysis can be incorporated. In addition, new analysis methods and indicators can be deployed on the whole network of sound observatories using up-to-date software agent technology. As each observatory is a cheap plug-and-measure device without any buttons or display, participatory sensing becomes easy: citizens plug in their device and data streams to central servers and is displayed on a website of choice for the community. During the presentation, application cases in urban tranquil area, building site noise, wind turbine noise, and train noise monitoring, as well as noise mapping validation will be shown.

9:20

5aNSa2. Permanent noise and vibration monitoring as a valuable tool to the construction industry. Daniel Vaucher de la Croix and Christian Freneat (ACOEM, 200 Chemin des Ormeaux, Limonest 69578, France, daniel.vaucherdelacroix@acoemgroup.com)

The construction of large infrastructures in dense urban areas comes along with a number of environmental challenges. Roads, railways, subways, and large building constructions necessarily have a significant impact on residents as well as surrounding buildings. This is especially true when it comes to consider large projects duration, which generally counts in months and even years. In this context, noise and vibration induced by the construction activities are major sources of annoyance to the community and may also induce potential damages to the immediate surroundings. Both issues have thus to be properly monitored in order to reduce adverse effects on residents, help mitigating risks and prevent potential interruption of the construction site's activity which would increase the overall project cost. The proposed paper will focus on how available communication technologies as an essential added value to noise and vibrations measurements. Operational conditions and project managers' requirements for system deployment will be reviewed. Then, benefits to the different parties will be highlighted on the basis of recent practical situations where adequate measures could be taken in the right timing and kept the project running while minimizing its noise and vibration environmental impact.

9:40

5aNSa3. Modeling urban noise exposure and contribution of noise reflection against façades of buildings: Does correction matter? Quentin M. Tenailleau, Nadine Bernard, Sophie Pujol (Laboratoire chrono-environnement (UMR6249), Univ. of Franche-comté, CMC - Hopital St. Jacques, 2 place St. Jacques, Besançon 25030, France, quentin.tenailleau@univ-fcomte.fr), Daniel Joly, Hélène Houot (Laboratoire ThéMA (UMR6049), Univ. of Franche-Comté, Besançon, France), and Frédéric Mauny (Ctr. de Methodologie Clinique, Ctr. Hospitalier Universitaire, Besançon, France)

European noise directives advise to apply corrections when measuring and modeling noise levels close to a building in the aim of excluding the contribution of noise reflection against the façade. The advised +3 dB correction is still subject to discussion. In order to investigate the needed correction for an household exposure studies, a high definition noise model was used to estimate noise levels at 10,394 inhabitable buildings. Three buffers were used to sample area surrounding façades of buildings. The surfaces were defined between the following distances : (i) 0 and 2 m, (ii) 0 and 6 m, (iii) 2 and 6 m. No differences between the distribution structures were observed. Mean noise levels do not differ significantly between the buffers methods [respectively (i) 48.8$\pm$6.5 dB, (ii) 48.9$\pm$6.4 dB, (iii) 49.0$\pm$6.5 dB; $p > 0.05$]. Maximum noise levels differ significantly between the methods [respectively (i) 52.0$\pm$7.2 dB, (ii) 52.5$\pm$7.2 dB, (iii) 52.4$\pm$7.2 dB; $p < 0.05$]. These results show no or light differences between indices computed by the three sampling methods. They are in favour of no or at least a low correction value to deal with the contribution of noise reflection against the façade of a building.

10:00

5aNSa4. Urban traffic noise assessment by combining measurement and model results. Frits Van der Eerden, Freek Graafland, Peter Wessels, and Tom Basten (Acoust. and Sonar, TNO, Oude Waalsdorperweg 63, The Hague 2597 AK, Netherlands, frits.vandereerden@tno.nl)

A model based monitoring system is applied on a local scale in an urban area to obtain a better understanding of the traffic noise situation. The system consists of a scalable sensor network and an engineering model. A better understanding is needed to take appropriate and cost efficient measures, especially when changes to the local infrastructure are proposed. The monitoring system provides information about the sound level distribution in the area in time and place. This can be used to create dynamic noise maps or to characterize the soundscape in the area. Results of a field test of two weeks in an urban area of 400 by 200 m are used. Three different areas are considered: (1) the main road which is the major source for traffic noise, (2) a quiet street, and (3) a quiet courtyard. The noise level measurements near the main road are compared with the engineering model results. Next, with the use of actual source levels from the measurements, the sound levels in the quiet street and the quiet courtyard are calculated. By comparing the model results with measurements in these areas, the parameters in the model are updated to better reflect the actual situation.

10:20

5aNSa5. Outdoor sound propagation for high-speed moving sources. Didier Dragna, Philippe Blanc-Benon (Laboratoire de Mécanique des Fluides et d'Acoustique, Ctr. Acoustique, 36 Ave. Guy de Collongue, Ecully, France, philippe.blanc-benon@ec-lyon.fr), Estelle Bongini, and Franck Poisson (SNCF, Direction de l'Innovation et de la Recherche, Paris, France)

This paper deals with modeling of sources in motion in time-domain solvers. In the context of transportation noise, acoustic sources are complex. Indeed, they are in motion, and they are generally not compact. Equivalent point sources are often used to simplify the problem. Heuristic methods are then applied to handle acoustic propagation over complex sites. Besides, time-domain solutions of the linearized Euler equations have proved to be an attractive approach to study outdoor sound propagation, and can then be used to validate these models. However, point sources in arbitrary motion are difficult to account for in these approaches. Distributed volume sources can be used instead. First, influence of the spatial support of the source on the acoustic field is investigated. The case of a harmonic source moving at a constant speed is studied. Directivity of a non-compact source is shown to be dramatically different to the one of a point source. Then, simulations of a broadband moving source above an impedance ground surface in a three-dimensional geometry are presented, and ground effect is highlighted.

10:40

5aNSa6. The modeling and calculation of sound radiation from facilities with gas flowed pipes. Fabian Probst (Res. & Development, DataKustik GmbH, Gewerbering 5, Greifenberg 86926, Germany, info@datakustik.com)

Computer modeling of industrial facilities like chemical plants, refineries, or other production areas is the first and most important step in the calculation of sound exposure in the environment. The pipework with gas flows is often contributing relevant to the sound radiation of the complete facility. This radiation can be determined applying the methods described in technical papers like VDI 3733 and ISO 15664. On the basis of these descriptions a software tool was developed that allows to create pipework in 3D models with line sources and to calculate the sound propagation with methods like ISO 9613-2. The line sources are linked with the technical parameters like pipe cross section, flow rate, pressure, density, and temperature of the gas and material parameters of the pipe wall. The sound power emission from the pipe to the environment and the internal flow of sound power—linked to the next section of piping—is calculated on the basis of these parameters. The same technique is used to calculate the sound emission of cooling towers, electric and fuel driven motors, gears pumps, and other devices. This powerful technique allows creating sustainable models that can be adapted to different operation conditions with minimum time and effort.

11:00

5aNSa7. Reproduction of sound source directionality in reduced scale model. Aline Lisot (FEC - Engenharia Civil, UNICAMP, Rua Colombo, 5790, Bloco C67, Maringá, Paraná 87020900, Brazil, alinelisot@gmail.com) and Stelamaris R. Bertoli (FEC - Engenharia Civil, UNICAMP, Campinas, Brazil)

The study of acoustic phenomena through reduced scale models is a useful tool for predicting the acoustic performance of closed and outdoor environments. When using reduced scale models, begins a process of constructive adequacy details of the study environment and sound source characteristics, reproducing its level and directionality. It is presented in this paper, a study on the reproduction of sound source directionality with application in outdoor reduced scale model. The sound source is an electric power converting substation. The directionality of this source was calculated using sound pressure levels measured, in 1/3 octave frequency bands between 50 Hz and 12.5 kHz, in 24 points positioned on a circle located around of the sound source. The sound was recorded in a point outside the circle for later playback. For playback of the sound signal, the noise was emitted in each monitoring direction, considering in each one the directionality through the signal equalization. The reduced scale used in the model was 1:5 and, based on this, it was transposed the frequencies of the signal. It was concluded that the sound source used in the reduced scale can be an efficient tool for reproduction of the directionality characteristics of the source.

FRIDAY MORNING, 7 JUNE 2013 — 511CF, 9:00 A.M. TO 10:40 A.M.

Session 5aNSb

Noise and ASA Committee on Standards: Current U.S. and Canadian Noise Standards

Richard L. McKinley, Chair
Air Force Res. Lab., Wright-Patterson AFB, OH 45433-7901

Contributed Papers

9:00

5aNSb1. Evaluation of proposed ASTM standard to measure the normalized insertion loss of doors. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

There is currently no ASTM method for field measurement of the acoustical noise isolation specific to doors. Measurement of the overall noise reduction of the composite wall/door assembly or the Apparent Sound Transmission Class of the door can be attempted using the methods in ASTM E336, but these methods are not well suited to measuring doors. Doors typically have poorly defined source and receiving rooms, such as long and narrow corridors, stairwells, or outdoor spaces, which often do not meet the room requirements of ASTM E336. The sound fields are rarely diffuse, and the measurement locations are not well defined in the ASTM standard. An alternative insertion loss method (i.e., a comparison of the sound pressure levels with the door open and closed) was developed by MJM Acoustical Consultants (Michel Morin, "Research project on the noise isolation provided by access doors in multi-dwelling buildings," 1993), and a draft ASTM standard has been developed based on this method. A laboratory testing program has been designed evaluate the proposed method and investigate variations in the test method. The results of the laboratory testing program are presented.

9:20

5aNSb2. Validation of the CSA Z107,56 standard method for the measurement of noise exposure from headsets. Alberto Behar, Gabe Nespoli, Tristan Defrancesco-Loria, and Frank Russo (Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, albehar31@gmail.com)

The last section of the CSA Standard CAN/CSA-Z107.56-06 (R2011)—Procedures for the Measurement of Occupational Noise Exposure—deals with communication headsets. This section is typically applied in retail, call centers, airport control towers, and other settings where an operator is exposed to background noise while communicating with a headset. Because the user adjusts the volume to hear speech comfortably, the total equivalent sound level is a function of the level of the background noise minus the attenuation provided by the headset, plus a comfortable signal/noise ratio, estimated to be 13.5 dBA. Despite ample research demonstrating the role of spectral and temporal factors in speech masking, the standard is applied consistently regardless of the type of background.

9:40

5aNSb3. Comparison of direct measurement methods for headset noise exposure in the workplace. Flora Nassrallah (Population Health, Univ. of Ottawa, 1 Stewart St., Ottawa, ON K1N 6N5, Canada, fnass039@uottawa.ca), Christian Giguère (Audiol./SLP Program, Univ. of Ottawa, Ottawa, ON, Canada), and Hilmi R. Dajani (School of Elec. Eng. and Comput. Sci., Univ. of Ottawa, Ottawa, ON, Canada)

Specialized equipment and techniques are required to carry out direct sound measurements under occluded ears for the purpose of assessing the noise exposure from communication headsets. Standard ISO 11904 describes two procedures: (1) the microphone in a real ear (MIRE) and (2) the acoustic manikin technique using an occluded ear simulator IEC 60318-4. Methods using simpler artificial ears, such as IEC 60318-1, have also been proposed in occupational noise measurement standards such as CAN/CSA Z107.56. Such devices are more practical to use and more easily accessible. However, they have not been designed specifically for noise measurements under communication headsets and there is little comparative data to the manikin technique, which is considered the gold standard for simulated in-situ acoustic measurements. Furthermore, little is known about measurement reliability for the purpose of standardization. Fit-refit measurements were obtained under laboratory conditions with four different types of artificial ears (type 1, type 2, type 3.3, manikin), three communication headset types (circum-aural, supra-aural, intra-aural) and six different communication signals. Data was transformed into equivalent-diffuse sound levels using third-octave procedures as well as single number corrections. Results illustrate that methods using single number corrections as well as fit-refit standard deviations vary according to measurement conditions.

10:00

5aNSb4. Evaluating harmful acoustic impulses. Gerald Fleischer (Justus-Liebig-Univ., Hoehenstr. 18, Giessen 35466, Germany, gerald.fleischer@gmx.net)

Noise-induced auditory impairment is widespread, despite many successful efforts to reduce sound emitted by technical equipment. Using accumulated acoustic energy as an indicator for health hazard, as in ISO1999, is insufficient to describe the hazard for impulses. While stimulating a system with many vibrating parts, like the ear, it is more important how the energy is fed into the various components than how much is applied. The Human Ear Model (AHAAH) uses the pressure-time-history of impulses, which is much better. It is based on theoretical assumptions, but it has serious deficits that lead to results that can be extremely wrong. Within another long-term approach the effects on hearing of accidents caused by impulses have been collected and documented. The harmful situations were reenacted thoroughly, to obtain acoustic measurements that can be correlated with the auditory damage, if present. All told, impulses cause three types of notches, at different frequency ranges. Hence, harmful impulses show characteristic footprints in carefully recorded audiograms, and the type of notch depends upon details of the causing impulse. This EPI-approach (Effects of Powerful Impulses) applies to most cases of auditory damage, and it is useful to prevent and understand such injuries.

10:20

5aNSb5. Experience with developing a room to meet the requirements of the joint Australian/New Zealand standard 1270. Ben Scott and John Pearse (Mech. Eng., Univ. of Canterbury, Ilam, Christchurch 8041, New Zealand, ben.scott@pg.canterbury.ac.nz)

Hearing protectors are used as a "last line of defence" to protect people in high noise exposure environments from noise induced hearing loss. The assessment of hearing protector performance has been standardized to ensure a minimum level of performance for the end user. In New Zealand, the standard AS/NZS 1270 specifies a REAT based test method to rate a hearing protector's performance. The rating is used to calculate appropriate exposure times for end users. This paper describes the modification of an existing audiology booth, to meet the sound field requirements specified in AS/NZS 1270. Current status of the facility will be reported; followed by a general discussion on future uses of the facility and related projects at the University of Canterbury.

FRIDAY MORNING, 7 JUNE 2013 — 519B, 8:55 A.M. TO 11:40 A.M.

Session 5aPA

Physical Acoustics: Chemical and Non-Medical Biological Effects of Ultrasound

Kenneth Suslick, Cochair
Chemistry, Univ. of Illinois, 600 S. Mathews Av., Urbana, IL 61801

Hao Feng, Cochair
Univ. of Illinois at Urbana-Champaign, 1304 W Pennsylvania Ave., Urbana, IL 61801

Chair's Introduction—8:55

Invited Papers

9:00

5aPA1. Sonofragmentation of molecular crystals. Kenneth Suslick and Brad W. Zieger (Chemistry, Univ. of Illinois, 600 S. Mathews Av., Urbana, IL 61801, ksuslick@uiuc.edu)

Developing processes for the production of active pharmaceutical ingredients (APIs) with a specific crystal size or polymorph distribution is critical for improved drug delivery by aerosolization, injection or ingestion, for control of bioavailability, and for economy of preparation. The use of ultrasound for the crystallization of APIs has attracted substantial recent attention due to (1) its influence on particle size and size distribution, (2) reduction of metastable zone-width, induction time, and supersaturation levels required for nucleation,

(3) improved reproducibility of crystallization, (4) control of polymorphism, and (5) reduction or elimination of the need for seed crystals or other foreign materials. Possible mechanisms for the breakage of molecular crystals under high-intensity ultrasound were investigated, using acetylsalicylic acid (aspirin) crystals as a model compound for active pharmaceutical ingredients (APIs). Surprisingly, kinetics experiments rule out particle-particle collisions as a viable mechanism for sonofragmentation. Two other possible mechanisms — particle-horn or particle-wall collisions — were dismissed based on decoupling experiments. Direct particleshockwave interactions are therefore indicated as the primary mechanism of sonofragmentation of molecular crystals.

9:20

5aPA2. Sonochemical synthesis of nano-cocrystals. Leonard R. MacGillivray, Dejan-Kresimir Bucar, John R. Sander, and Elizabeth Elacqua (Chemistry, Univ. of Iowa, E555 Chem. Bldg., Iowa CIty, IA 52242, len-macgillivray@uiowa.edu)

Cocrystals are multicomponent solids with organic molecules assembled in combination to form a crystalline solid with properties different than the individual components. A cocrystal typically consists of a target molecule crystallized with a second molecule, or cocrystal former, employed to influence properties of the target (e.g., solubility). The conformer interacts with the target via intermolecular forces (e.g., hydrogen bonds) that hold the components together. The modularity of a cocrystal makes such solids attractive for applications where fine-tuning of properties is important (e.g., optical). In this presentation, we describe the use of sonochemistry to form cocrystals of nanoscale dimensions. In contrast to single-component solids, cocrystals present a fundamentally different challenge with respect to those reprecipitation methods used to form nanocrystals since the components of a cocrystal will tend to exhibit different solubilities. We show that sonochemistry affords nano-cocrystals with properties (e.g., reactivity) that contrast solids of macroscale dimensions. Related applications of sonochemistry to afford single-component nanocrystals will also be presented.

9:40

5aPA3. Enhancing biofuel production by ultrasonics. David Grewell, Melissa Montalbo-Lomboy, and Priyanka Chand (Iowa State Univ., 100 Davidson Hall, Ames, IA 50011, dgrewell@iastate.edu)

This work evaluated the use of high-powered ultrasonics to enhance biofuel production in terms of efficiency and costs. A wide range of feed stocks, including switch grass, corn stover, and soft wood, were studied. The effect of ultrasonic pretreatment on the removal of lignin for hydrolysis of starches and cellulose to fermentable sugars was studied. It was found that many of the pretreatments were very successful in enhancing lignin removal. For example, time of dissolution of lingo-cellulosic biomass in ionic liquids was reduced from hours to minutes accompanied by a significant decrease in energy consumption compared to mechanical stirring. In addition, it was found that hydrolysis of corn starch could be greatly accelerated utilizing ultrasonics. Economic models showed that the technology, once implemented, would have a payback period of less than one year. The work also focused on biodiesel production. It was seen that ultrasonics accelerated the transesterification process so that soy bean oil could be converted to biodiesel in less than a minute, compared to 45 min using traditional methods. It was shown that yeast grown from glycerin, a co-product of biodiesel production, could be extracted and simultaneously converted to biodiesel with ultrasonics in less than a minute, compared to traditional techniques that require multiple processes and relatively long cycle times (+1 h).

10:00

5aPA4. The influence of ultrasound on the structure, rheological properties, and degradation path of citrus pectin. Donghong Liu (Fuli Inst. of Food Sci., Zhejiang Univ., No. 866 Yuhangtang Rd., Hangzhou 310016, China, dhliu@zju.edu.cn) and Lifeng Zhang (Food Sci. and Nutrition, Zhejiang Univ., Hangzhou, China)

The effects of ultrasound on the molecular weight, structure, and rheological properties of citrus pectin were investigated. The degradation path of citrus pectin by ultrasound was also studied. The structure and rheological properties of the degradation products were identified by high performance liquid chromatography-photodiode array detector (HPLC-PAD), Fourier transform infrared spectroscopy (FTIR), nuclear magnetic resonance spectroscopy (NMR), atomic force microscope (AFM), and rheometer. The results indicated that the average molecular weight of citrus pectin decreased rapidly after ultrasound treatment and reduced to one third of the initial pectin after treated for 90 min. The polydispersity reduced from 2.30 to 1.59. The degradation products had a uniform and narrow distribution of molecular weight. The main chain composition and monosaccharide constituents of citrus pectin remained unchanged after ultrasound treatment. The reduction ratio of (Gal+Ara): Rha suggested a decrease in neutral sugar side chain size of citrus pectin after ultrasonication. FT-IR and NMR results approved that the main chain of citrus pectin was not changed by ultrasound treatment. Together with the AFM results indicated that ultrasound could reduce the branched structure of citrus pectin. The viscosity of citrus pectin decreased after ultrasound treatment. Meanwhile, the ultrasound-treated citrus pectin showed predominantly viscous responses ($G' < G''$) over the same frequency range.

10:20

5aPA5. Free radical formation and scavenging by solutes in the sonolysis of aqueous solutions. Franz Grieser (Chemistry, Univ. of Melbourne, Grattan st, Parkville, VIC 3010, Australia, franz@unimelb.edu.au)

It has long been known that the primary radicals generated in water (H and OH), on the collapse of acoustic bubbles, largely recombine. It has been estimated that as much as 90% react within the bubble to produce molecular hydrogen, hydrogen peroxide and water [Henglein, Ultrason. Sonochem.**2**, S115–S121 (1995)]. This high recombination efficiency has been likened to radicals reacting within "spurs" produced by ionizing radiation in water. Several studies have shown that by using high concentrations (100s of mM) of primary radical scavengers, e.g., aliphatic alcohols, iodide, etc., a large number of the primary radicals are able to be captured in acoustically produced hot spot spurs. What is less well examined is the effect the scavengers themselves have on the production of the primary radicals and hence on the radical yields measured. The talk will consider the effect that typical radical scavengers have on active bubble populations in aqueous solutions, and on the production of primary radicals in the presence of added solutes.

10:40

5aPA6. Microbial inactivation by ultrasound for enhanced food safety. Hao Feng, Bin Zhou, Hyoungill Lee, Yanfang Li, Hee Kyung Park, Sindy Palma, and Arne Pearlstein (Univ. of Illinois at Urbana-Champaign, 1304 W Pennsylvania Ave., Urbana, IL 61801, haofeng@illinois.edu)

Inactivation of foodborne pathogens by power ultrasound provides an alternative to traditional thermal processing modalities, with potential for minimizing food-quality degradation. To enhance efficacy, ultrasonic treatment is often combined with other physical or chemical lethal factors, which serve to shorten treatment time and improve quality retention. The inactivation mechanisms, thermodynamic aspects, and kinetic modeling of ultrasonic microbial inactivation will be discussed. The critical issue of how to achieve a relatively uniform acoustic field distribution during treatment will be investigated by computer simulation and verified with microbial inactivation tests. Inactivation of foodborne pathogens in liquid foods, and surface decontamination of fresh produce and nuts, will be used as examples demonstrating the potential of ultrasound-assisted processes. Lastly, the effect of sonication treatment on food product quality and quality retention will be examined.

Contributed Papers

11:00

5aPA7. Effects of frequency and initial concentration of methylene blue on rate constants of ultrasonic degradation. Chiemi Honma (Dept. of Chemical Sci. and Technol., Tokyo Univ. of Sci., Shinjyukuku, Tokyo 162-8601, Japan, jb112864@ed.tus.ac.jp), Daisuke Kobayashi (Dept. of Industrial Chemistry, Tokyo Univ. of Sci., Tokyo, Japan), Hideyuki Matsumoto (Dept. of Chemical Eng., Tokyo Inst. of Technol., Tokyo, Japan), Tomoki Takahashi (Dept. of Industrial Chemistry, Tokyo Univ. of Sci., Tokyo, Japan), Chiaki Kuroda (Dept. of Chemical Eng., Tokyo Inst. of Technol., Tokyo, Japan), Katsuto Otake, and Atsushi Shono (Dept. of Industrial Chemistry, Tokyo Univ. of Sci., Tokyo, Japan)

Ultrasound has been found to be an attractive advanced technology for the degradation of hazardous organic compounds in water. The ultrasonic degradation of dyes has been investigated by many researches, but the effects of ultrasonic frequency on degradation rate were not investigated quantitatively. In our previous study, we proposed a simple model for estimating the apparent degradation rate constant of methylene blue based on ultrasonic power and the SE value in the range of frequency between 20 and 500 kHz. However, we have not investigated the ultrasonic degradation of methylene blue in high frequency region around 1 MHz. The effects of initial concentration of methylene blue have not been investigated yet, either. In this study, the degradation process using methylene blue as model hazardous organic compounds by ultrasonic irradiation was investigated. The ultrasonic frequency was operated in the range from 20 kHz to 1.6 MHz, and the initial concentration of methylene blue was in the range from 0.005 to 0.04 mM. Our proposed model can apply to this study which extended the frequency range. The effects of initial concentration can also be estimated in the range of 0.01 to 0.04 mM using this model.

11:20

5aPA8. Effect of gases on radical production rates during single-bubble cavitation. Shin-ichi Hatanaka (Dept. of Eng. Sci., The Univ. of Electro-Communications, 1-5-1 Chofugaoka, Chofu, Tokyo 182-8585, Japan, hatanaka@pc.uec.ac.jp)

The yields of hydroxyl radicals and nitrite ions produced from single-bubble cavitation were quantified while the corresponding dynamics of the single bubble was observed by stroboscopic and laser-light scattering methods. The numbers of both hydroxyl radicals and nitrite ions per cycle were proportional to pressure amplitude at 25 kHz under the conditions of stable sonoluminescing single-bubbles. Under the dancing single-bubbles without sonoluminescence below sonoluminescence threshold for air bubbles, however, the number of hydroxyl radicals was larger than that for the stable single-bubble and nitrite ions were not detected in contrast to hydroxyl radicals. For argon bubbles, the numbers of hydroxyl radicals were larger than those for air bubbles and the number of hydroxyl radical was significantly increased with the dancing bubble by argon bubbling. The results imply that the shape instability may promote the dissolution and the diffusion of hydroxyl radicals from the bubble into the liquid and the part of hydroxyl radicals may convert into nitrite ions in the case of the air bubble above the threshold of sonoluminescence.

Session 5aPP

Psychological and Physiological Acoustics: Recent Trends in Psychoacoustics I

Hugo Fastl, Cochair
AG Technische Akustik, TU München, Arcisstr.21, München 80333, Germany

Sonoko Kuwano, Cochair
Osaka Univ., 2-24-1-1107 Shinsenri-Nishimachi, Toyonaka, Osaka 560-0083, Japan

Chair's Introduction—8:55

Invited Papers

9:00

5aPP1. Evaluation of the loudness of stationary and non-stationary complex sounds. Sonoko Kuwano (Osaka Univ., 2-24-1-1107 Shinsenri-Nishimachi, Toyonaka, Osaka 560-0083, Japan, kuwano@see.eng.osaka-u.ac.jp), Tadasu Hatoh (Toho Gakuen, College of Drama and Music, Yokohama, Japan), Tohru Kato (Otemon Gakuen Univ., Osaka, Japan), and Seiichiro Namba (Osaka Univ., Osaka, Japan)

In our sound environment, there are various complex sounds including both stationary and non-stationary sounds. It is important to evaluate the loudness of these sounds, to clarify the relation between the loudness and the physical metrics and to estimate the loudness by the physical metrics in order to evaluate and control the sound environment. The temporal patterns of non-stationary sounds in daily life environment, such as road traffic noise, construction noise, speech and music, are different from each other. It would be desirable to evaluate the loudness of both stationary and non-stationary sounds with various temporal and spectral characteristics by the same metric in the evaluation of the loudness. It is needless to say that the metric should have a good relation with the subjective impression. From physical viewpoint, sounds with various temporal patterns including stationary sounds can be measured by a single scale on the basis of energy. In this paper, a single common metric for the evaluation of the loudness of both stationary and non-stationary sounds is examined by conducting psychological experiments.

9:20

5aPP2. Loudness of complex time-varying sounds – A challenge for current Loudness models. Jan Rennies (Hearing, Speech and Audio Technol., Fraunhofer IDMT, Marie-Curie-Str. 2, Oldenburg 26129, Germany, jan.rennies@idmt.fraunhofer.de), Jesko L. Verhey (Experimental Audiol., Otto von Guericke Universität, Magdeburg, Germany), Jens E. Appell (Hearing, Speech and Audio Technol., Fraunhofer IDMT, Oldenburg, Germany), and Birger Kollmeier (Med. Phys., Carl-von-Ossietzky Universität, Oldenburg, Germany)

The calculation of perceived loudness is an important factor in many applications such as the assessment of noise emissions. Generally, loudness of stationary sounds can be accurately predicted by existing models. For sounds with time-varying characteristics, however, there are still discrepancies between experimental data and model predictions, even with the most recent loudness models. This contribution presents a series of experiments in which loudness was measured in normal-hearing subjects with different types of realistic signals using an adaptive loudness matching procedure and categorical loudness scaling. The results of both methods indicate that loudness of speech-like signals is largely determined by the long-term spectrum, while other speech-related properties (particularly temporal modulations) play only a minor role. Loudness of speech appears to be quite robust towards even severe signal modifications, as long as the long-term spectrum is similar. In contrast, loudness of technical, strongly impulsive signals is considerably influenced by temporal modulations. For some of the signals, loudness could not be predicted by current models. Since the perceived loudness was underestimated by the models for some signals, but overestimated for other signals, a simple adjustment of the employed time constants in the temporal integration stage could not eliminate the discrepancies.

9:40

5aPP3. Differentiating between loudness and preference in the case of multi-tone stimuli. Stephan Toepken and Reinhard Weber (Acoust. Group, Oldenburg Univ., Carl-von-Ossietzky-Str.9-11, Oldenburg 26129, Germany, stephan.toepken@uni-oldenburg.de)

When exploring sound quality, often a high correlation between pleasantness and loudness can be observed. However, sometimes it is desirable to know to which extent other sound characteristics than loudness are responsible for a preference evaluation. In this respect multi-tone sounds with rich perceptual aspects are interesting test sounds. This talk will present a separate determination of preference and loudness by comparing a test sound of interest with a reference sound. Using an adaptive paired comparison the points of subjective equality (PSEs) for preference and loudness between test and reference sound are separately measured—"Which sound is louder?" and "Which sound do you prefer?"—by varying the test sound level in an adaptive staircase manner. The level changes affect both loudness and preference evaluation of the test sound. The results of these experiments are level differences ΔL between the test and the reference sound at which equal preference and equal loudness are reached between them. (Similar procedures have been employed to determine equal loudness contours.) It will be shown, with multi-tone sounds as examples, how this method reliably differentiates between loudness and preference.

10:00–10:20 Break

10:20

5aPP4. Suprathreshold perception under a masking release condition using categorical scaling. Jesko L. Verhey and Wiebke Heeren (Dept. of Experimental Audiol., Otto von Guericke Univ. Madgeburg, Leipziger Str. 44, Magdeburg 39120, Germany, jesko.verhey@med.ovgu.de)

Masking experiments provide important information on how the auditory system processes sounds. For example, a tone is less masked by a modulated sound than by an unmodulated sound with the same long-term spectrum, indicating the ability of the auditory system to use modulation as a cue. In general, studies on this modulated-unmodulated difference (MUD) focus on the thresholds, whereas little is known about suprathreshold perception under these conditions of masking release. In the present study, loudness growth functions of a masked 1000-Hz tone are measured for two different masker types: (i) amplitude modulated broadband noise with a square-wave modulator and (ii) an unmodulated noise with the same spectral content and level. A categorical loudness scaling procedure (ISO 16832) is used to measure loudness of the masked tone over a large level range. The accuracy of the procedure is quantified by comparing the scaling results with loudness matching data for the same masker types. It is investigated (i) up to which suprathreshold level a masking release is still observed and (ii) whether the effect of the reduced masking for the modulated masker is equivalent to a condition where the unmodulated masker is reduced in level by the magnitude of the MUD.

10:40

5aPP5. Comparison of detection threshold measurements and modeling for approaching electric cars and conventional cars presented in traffic and pink noise. Julian Grosse, Reinhard Weber, and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Carl-von-Ossietzky-strasse 9-11, Oldenburg 26129, Germany, steven.van.de.par@uni-oldenburg.de)

This study investigates the difference in audibility of an approaching conventional car with internal combustion engine and an electric car at various velocities. The goal was to compare the risk that pedestrians do not hear the approaching car in time. Binaural recordings of each of these approaching cars were presented together with either a traffic noise masker or a pink noise masker. In the first detection experiment, the threshold level was determined for which the cars could just be detected. In a second reaction time experiment, the moment was determined at which the approaching car was first detectable. This measured reaction time should give an indication about how much time a person has to evade an impending collision. Results indicated that slowly approaching electric cars where less audible than cars with a conventional engine. The results also showed that the decrement of reaction times as a function of SNR was halved when pink noise was used instead of traffic noise. A psycho-acoustic masking model [Dau *et al*., J. Acoust. Soc. Am. **99**, 3615–3622 (1996)] was applied to predict detection thresholds and showed good correspondence with the subjective data.

11:00

5aPP6. Rating the dieselness of vehicle noise using different psychoacoustic methods. Jakob Putner and Hugo Fastl (AG Technische Akustik, MMK, Technische Universität München, Arcisstraße 21, Munich 80333, Germany, putner@tum.de)

Modern diesel engines meet the demand for high power-engines while strict emission regulations have to be fulfilled. Therefore, diesel engines entered vehicle segments where the expectations on the sound quality are exceptionally high. Sound quality and fuel efficiency are often conflicting goals during the development of a diesel engine. The typical sound character of diesel engines, the so called Dieselness, is an indicator for the overall sound quality of the vehicle noise. Hence, it is desirable to rate the Dieselness of engine sounds. Sounds emitted by gasoline- and diesel-powered vehicles in idle condition were rated in psychoacoustic experiments using different methods. First, the method of line length was used as direct scaling procedure to get ratio ratings of the relative Dieselness of the vehicle noises. Second, a direct ranking of the noises has been done with the Random Access method where subjects had to rank the sounds according to their Dieselness. Third, in a paired-comparison test the participants had to judge which of two sounds had more Dieselness, resulting in an indirect scaling. These methods are compared regarding the time the experiments took and the resulting ranking respectively scaling. In addition, a semantic differential test with general adjective pairs was conducted.

11:20

5aPP7. Fluctuation strength on real sound: Motorbike exhaust and marimba tremolo. Masanobu Miura (Dept. of Media Informatics, Ryukoku Univ., 1-5, Seta, Oe-cho, Yokotani, Otsu 5202194, Japan, miura@mail.ryukoku.ac.jp) and Nozomiko Yasui (Dept. of Information Eng., Matsue College of Technol., Matsue, Japan)

Psychoacoustics research has been contributed on evaluating the timbre of acoustic signals. The fluctuation strength (FS) has been respected as the important index which describes the sensory fluctuation of the acoustic signal based on its amplitude, frequency and the combination of them as well. Although FS has been calculated by tremor components on acoustic signals, here points out not only the inappropriateness of the calculation when simply based on literatures but also a method which focuses on the shape of the waveform in order to extract parameters from it, so that the FS is newly calculated by our original method with estimation results of the subjective scores of sensory fluctuation. Signals dealt with are motorbike exhaust sounds and tremolo played by marimba. The developed method is then applied to the design of an electric vehicle approaching sound for pedestrians in order to let pedestrians notice the approaching car without rising up the sensory loudness, on which designed sounds have a fluctuation with irregular pulses proven to give us the sensory fluctuation so that it is expected to be noticeable for pedestrians. This paper discusses the possibility to apply the prestigious psychoacoustic indexes to industrial and artistic sounds.

11:40

5aPP8. Perception of roughness of time-variant sounds. Roland Sottek and Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

Besides loudness, other psychoacoustic parameters like sharpness and roughness, are important for sound quality evaluation. Sharpness considers the relationship between the loudness of high frequency components to total loudness, and roughness evaluates modulation characteristics. While loudness of stationary sounds has been standardized for decades, standards for sharpness of stationary sounds and for loudness of time-varying sounds have been published in 2009 (DIN 45692:2009-08) and 2010 (DIN 45631/A1:2010-03), respectively. In addition, there are several roughness models available, performing more or less well for synthetic and selected technical signals. Currently, a roughness standard is under discussion in a DIN working group. In recent listening tests, the subjects showed consistent overall roughness evaluation of synthetic signals but heterogeneous judgments concerning the more complex technical sounds containing different rough components. The results of the listening test will be discussed and compared with the evaluation by several models like the roughness calculation using the hearing model of Sottek and another approach based on a psychoacoustically weighted modulation spectrum.

FRIDAY MORNING, 7 JUNE 2013 512CG, 9:00 A.M. TO 12:00 NOON

Session 5aSA

Structural Acoustics and Vibration: Applications in Structural Acoustics and Vibration IV

Kuangcheng Wu, Cochair
Ship Survivability, Newport News Shipbuild, 4101 Washington Ave., Newport News, VA

Vyacheslav Ryaboy, Cochair
Newport Corp., 1791 Deere Ave., Irvine, CA 92606

Contributed Papers

9:00

5aSA1. Splitting up resonances of elastic elliptical disc. Emmanuelle Bazzali, Stéphane Ancey, Paul Gabrielli, and Michaël Mercier-Finidori (Physics, UMR CNRS 6134 SPE Université de Corse, Campus Grimaldi, Bâtiment Alfonsi, Corte, Corsica 20250, France, ebazzali@gmail.com)

The elastodynamic resonances of two-dimensional elliptical objects are studied from a modal formalism by emphasizing the role of the symmetries as the circumference is deformed from circular to elliptical geometry. More precisely, as the symmetry is broken in the transition from the circular disc to the ellipse, resonance splittings and level crossings are observed. This observation can be mathematically explained by the broken invariance of the continuous O(2) symmetry group of the circular disc. However, the ellipse remains invariant under the finite C2v group. The main difficulty comes from the application of the group theory in elastodynamics since the vectorial formalism is used to express the physical quantities involved in the boundary conditions. This method significantly simplifies numerical computations and provides a full classification of the resonances. The vibrational normal modes are computed. We focus on the resonance splittings in the transition from the circular disc to the elliptical one. Then, the resonances are tagged and tracked as the eccentricity of the ellipse increases. A series of experiment on three-dimensional objects are also carried out to emphasize the physical effects described above, although no quantitative comparison can be done between theory and experiment. We expect that those effects in 2D appear also in 3D when the sphere is deformed to the spheroid.

9:20

5aSA2. Extension of SmEdA to non-resonant transmission. Laurent Maxit, Kerem Ege, Nicolas Totaro, and Jean-Louis Guyader (INSA LYON, 25 bis av. Jean Capelle, Villeurbanne 69621, France, laurent.maxit@insa-lyon.fr)

Statistical modal Energy distribution Analysis (SmEdA) may be used as alternative to Statistical Energy Analysis for describing subsystems with low modal overlap. In its original form, SmEdA predicts the power flow exchanged between the resonant modes of different subsystems. In the case of the sound transmission through a thin light structure, it is well-known than the non-resonant response of the structure may have a significant role on the transmission below the critical frequency. In this paper, one presents an extension of SmEdA taking into account the contributions of the non resonant modes of the thin structure. The dual modal formulation (DMF) is used for describing the behavior of two acoustic cavities separated by a thin structure knowing their subsystem modes. A condensation in the DMF equations is achieved on the amplitudes of the non-resonant modes. Using some simplifications, a new coupling scheme between the resonant modes of the three subsystems is obtained. It shows direct couplings of the cavity modes through stiffness elements characterized by the modes shapes of the cavities and the structure, both. Comparisons with reference results show the ability and the interest of the present approach for representing the non-resonant contributions of the structure.

9:40

5aSA3. On the energy finite element method for the acoustic design of ships. Bernd Stritzelberger, Martin Abele, and Otto von Estorff (Inst. of Modeling and Computation, Hamburg Univ. of Technol., Denickestr. 17, Hamburg 21073, Germany, bernd.stritzelberger@tuhh.de)

To ensure dynamic requirements of technical systems, methods like the finite element method (FEM) are successfully applied. For large structures as ship geometries, such analyses in the acoustic-relevant frequency range are usually not used productively. Highly time consuming investigations are incompatible to the generally single-unit production and short conception phases in ship design. The energy finite element method (EFEM) is a grid-based approach, which has the potential to provide a technique for the evaluation of acoustic characteristics even for major and complex structures at high frequencies. The less time consuming calculations generally result from a smaller number of degrees of freedom at the nodes and, in particular, it is feasible to use coarser grids than in the FEM. The governing equations are similar to that of the static heat conduction. State variables are the time and locally space averaged energy densities of the different wave types. The main focus is on the coupling— not only between the structure and the fluid, but also at junctions within the structure. Preliminary investigations on the reliability of EFEM results will be presented, questioning if the approach is applicable to operative ship design. [This work was done within the collaborative research project EPES.]

10:00

5aSA4. Simulating sound radiation using the energy-finite-element-method. Marius Karger, Otto von Estorff, and Olgierd Zaleski (Novicos GmbH, Kasernenstrasse 12, Hamburg 21073, Germany, karger@novicos.de)

The most established numerical methods for calculation of sound radiation are the boundary-element-method (BEM) and the finite-element-method (FEM). For large-scale geometries and high-frequency ranges these methods are limited by enormous numerical costs. The applicability of the

energy-finite-element-method (EFEM) in these cases is analyzed within the research project EPES, sponsored by the Federal Ministry of Economy and Technology. Under certain assumptions, the equations describing structure-borne sound and sound radiation can be condensed to the static heat conduction equation, transforming the pressure and velocities in energy densities. Using EFEM, the structure geometry and acoustic cavities are separately modeled and coupled by transmission coefficients for energy flow interactions. An important value calculating the coefficients is the radiation efficiency. This paper focuses on the analysis of the radiation efficiency for EFEM calculations. This contribution presents the EFEM approach, calculations of radiation efficiency, transmission coefficients and energy densities of different fluid-structure interactions. Based on those calculations, the applicability of the EFEM is discussed.

10:20

5aSA5. Validity of transfer matrix method for prediction of the transmission loss of curved panels. Mejdi Abderrazak, Sgrad Franck (bruit et Vibration, Institue de recherche de robert sauvé en santé et sécurité de travail (IRSST), 505, boul. De Maisonneuve Ouest, Montréal, QC H3A 3C2, Canada, abderrazak.mejdi@usherbrooke.ca), and Atalla Noureddine (Mechanical, Université de Sherbrooke, Sherbrooke, QC, Canada)

This paper discusses the modeling of the transmission loss of curved panels with attached sound absorbing materials (foam or fiber) using both analytical and numerical methods. Special attention is devoted to the validity of modeling the problem using the transfer matrix method (TMM) and statistical energy analysis (SEA). Classically, in SEA models the sound package is unwrapped and the TMM is used to calculate its effects in terms of added damping, absorption and insertion loss. A systematic comparison with an efficient FEM/VBEM formulation of the problem is presented to examine validity of this practice and demonstrate its range of applicability and usefulness.

10:40

5aSA6. The analytical model of rheological fluid for vibration and noise control. Marek L. Szary (College of Eng., Southern Illinois Univ., Carbondale, IL 62901, szary@engr.siu.edu)

Rheological fluids (RF) also known as a controllable viscosity fluids (CVF) introduced in the area of vibration control of mechanical systems made possible a more efficient control of both: transient and continuous vibration. They are also used in design of sound barriers to control noise transmission loss and diaphragms for modification of noise absorption characteristics of sound absorbing materials. Their apparent viscosity is controllable by the use of external (electrical in electro-rheological ER or electromagnetic in magneto-rheological MR fluids) field. In the absence of applied external field, the RF exhibits Newtonian-like behavior. Applied external field changes this behavior and the RF shows in addition a yield shear stress which depends on strength of this field. In proposed model, the shear stress is expressed as a superposition of two components. One of them is proportional to the viscosity and relative velocity of a base fluid, and the second one, which depends on strength of applied external field. Modification of external field strength according to selected design parameter (velocity or time or distance—combination of them) allows to develop family of vibration or sound attenuating devices, which were not achievable before.

11:00

5aSA7. Multi-component power transmission from structure-borne sound sources into lightweight structures. Sebastian Mathiowetz and Hannes A. Bonhoff (Tech. Univ. of Berlin, Einsteinufer 25, Berlin 10587, Germany, s.mathiowetz@tu-berlin.de)

The power transmission between structure-borne sound sources and adjacent structures is generally of complex nature. For an accurate description, the interaction between multiple contact points and several directional components must be taken into account. While calculation methods to predict the transmitted power are generally available, the main problem is the acquisition of extensive source and receiver data. This is especially true with regard to lightweight structures where source and receiver mobilities exhibit matched conditions and when rotational components of motion are involved. Therefore, the description needs to be simplified while at the same time a sufficient accuracy has to be retained. This work investigates the power transmission of a fan unit source that is mounted to a rib-stiffened aluminium plate at several contact points. Full data sets of source and receiver have been measured using a finite difference technique, including translational motion perpendicular to the structure as well as moment excitation around the in-plane axes of the plate. Both a rigid connection as well as a connection using resilient mounts is considered. Contributions of different components of motion are discussed and possible simplifications are deduced.

11:20

5aSA8. A low density, high stiffness flat loudspeaker with combined feedback-feedforward response correction. Jen-Hsuan Ho (Signals and Systems Group, Faculty of EEMCS, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, j.ho@ewi.utwente.nl) and Arthur Berkhoff (Acoust. and Sonar, TNO Tech. Sci., Den Haag, Netherlands)

This paper presents a novel perforated flat loudspeaker with improved sound frequency response by applying velocity feedback control. Flat loudspeakers provide advantages of compact dimensions and high durability. Known flat loudspeaker technology is based on high model density. However, the resonances in the panel are complex and difficult to control, which often leads to complicated computations and insufficient low frequency response. The flat loudspeaker in this paper comprises a novel panel structure, which offers low density, high stiffness, and efficient space utilization. Direct velocity feedback control provides a simple and stable control loop. We apply the multiple direct velocity feedback control method to obtain flat sound frequency response. Furthermore, a Linkwitz filter is applied to our system to increase response at very low frequencies. Experimental results show that a multiple combined feedback-feedforward control method effectively improves the performance of the flat loudspeaker with extended low-frequency response.

11:40

5aSA9. Ultrasonic transducers with directional converters of vibration of longitudinal-longitudinal type and longitudinal-longitudinal-longitudinal type intended to work in gaseous media. Tadeusz Gudra, Lukasz Palasz, and Krzysztof J. Opielinski (Electronics, Wroclaw Univ. of Technol., Wybrzeze Wyspianskiego 27, Wroclaw 50-370, Poland, tadeusz.gudra@pwr.wroc.pl)

The study presents a realized concept of an ultrasonic transducer intended to work in a gaseous medium, which radiates a focused ultrasonic beam in many directions simultaneously. The use of a longitudinal-longitudinal (L-L) type vibration direction converter makes it possible to radiate ultrasonic energy in a required direction without changing the location of the activation source. The transducer consists of four resonant elements: an ultrasonic sandwich type transducer, vibration amplitude transformer, L-L type converter, and axisymmetrical radiating plates. It is also acceptable to use a longitudinal-longitudinal-longitudinal (L-L-L) type vibration direction converter. The benefit of such a solution is that it is possible to use one ultrasound source for simultaneous activation of several radiating plates. The study presents conductance and susceptance characteristics of a complex resonance system, amplitude frequency characteristics of the level of acoustic pressure and sample characteristics of the directivity of a transducer with two independent radiating plates. The suggested solution creates new possibilities for applications of this type of resonance systems.

Session 5aSCa

Speech Communication: Flow, Structure, and Acoustic Interactions During Voice Production I

Scott L. Thomson, Chair
Mech. Eng., Brigham Young Univ., 435 CTB, Provo, UT 84602

Chair's Introduction—8:55

Invited Papers

9:00

5aSCa1. The role of the thyroarytenoid muscle in the regulation of prephonatory glottal opening. Zhaoyan Zhang and Jun Yin (UCLA School of Med., 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Previous research has demonstrated a positive correlation between the thyroarytenoid muscle activity and the closed quotient, particularly for chest voice production. To understand the physical mechanism behind this correlation, the interaction between the thyroarytenoid and the cricothyroid muscles and the subglottal pressure was investigated using a muscularly controlled vocal fold model. The results showed that activation of the thyroarytenoid muscle stiffened the vocal folds and caused the vocal fold to bulge toward the glottal midline, which makes it possible to maintain intended degree of glottal closure, from fully open to fully closed to tightly compressed, against the varying subglottal pressure. For a constant flow rate, increasing contraction of the thyroarytenoid muscle also led to increased subglottal pressure, thus increasing voice intensity. This function of the thyroarytenoid muscle as a fine regulator of glottal closure is most effective at conditions of low levels of cricothyroid muscle contraction and low-to-middle range of subglottal pressure. At strong cricothyroid muscle contraction or extremely high subglottal pressures, a full-closed prephonatory glottis become impossible. Vibration at this condition is likely to have a small closed quotient and the effects of thyroarytenoid muscle contraction on closed quotient and subglottal pressure are minimal. [Work supported by NIH.]

9:20

5aSCa2. Relationship between divergence angle and skewing of the volumetric flow rate in an excised canine larynx model without a vocal tract. Sid M. Khosla, Liran Oren (Otolaryngology, Univ. of Cincinnati Academic Health Ctr., 231 Albert Sabin Way, 0528, Cincinnati, OH 45208, khoslasm@uc.edu), and Ephraim Gutmark (Aerosp. Eng., Univ. of Cincinnati, Cincinnati, OH)

Until now, skewing of the volumetric flow rate (Q) curve was thought to be due to the inertance effects produced by the vocal tract. The goal of these experiments is to measure the volumetric flow rate at the entrance and exit of the glottis during the closing phase in a 1 mm thick coronal section halfway between the anterior commissure and the vocal process. The velocity fields and the intraglottal geometry are measured using modified particle imagining velocimetry (PIV) methodology. In these experiments, five excised canine larynges were used, and in all of them it is shown that the flow rate is greater at the glottal exit than at the glottal entrance when the glottis is divergent and an intraglottal vortex is formed at the superior aspect of the fold. In addition, flow skewing is seen at the glottal exit but not at the glottal entrance. In this talk, we will show that this flow skewing without a vocal tract is due to the entrainment effects of the intraglottal vortex.

9:40

5aSCa3. Determination of the stresses and strain on the superior surface of excised porcine larynges during phonation using digital image correlation. Hani Backshaei, Chan Woo Yang, Amir K. Miri, and Luc Mongeau (Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada, luc.mongeau@mcgill.ca)

The stresses and strains acting over the superior surface of excised porcine vocal folds during self-excited, flow-induced vibrations were investigated. Digital image correlation analysis of stereoscopic high speed video data was used. The pre-strain resulting from the initial deformation of the tissue before phonation onset was estimated in four larynges. The kinematics of the vocal folds were also measured with a high temporal resolution using a laser Doppler vibrometer. Chaotic jumps between different vibration modes was observed. Impact stresses were estimated from a model based on Hertzian theory, and isotropic constitutive laws. The results yielded values of compressional stresses near the vocal fold edges that were larger than previously reported data. Comparisons were made with similar data previously obtained for a synthetic silicone replica of the human larynx.

10:00

5aSCa4. Modeling incomplete glottal closure due to a posterior glottal opening and its effects on the dynamics of the vocal folds. Matías Zañartu (Dept. of Electron. Eng., Universidad Técnica Federico Santa María, Av España 1680, Valparaíso, Valparaíso 2390123, Chile, matias.zanartu@usm.cl), Byron D. Erath (Dept. of Mech. & Aeronautical Eng., Clarkson Univ., Potsdam, NY), Sean D. Peterson (Dept. of Mech. and Mechatronics Eng., Univ. of Waterloo, Waterloo, ON, Canada), Robert E. Hillman (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA), and George R. Wodicka (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Even though incomplete glottal closure is present in normal and pathological voices, it has received little attention in self-sustained models of phonation. The effects of acoustic interaction due to a posterior glottal gap on the tissue dynamics, energy transfer, and glottal aerodynamics were numerically investigated. The domain was prescribed as flow through two separate orifices (posterior gap and

membranous vocal folds) that merge in the supraglottal tract, with the governing flow equations determined from a control volume analysis based on conservation of mass and linear momentum. The equations of motion remained unaffected, although the driving forces were indirectly altered through the acoustic interaction. The method was implemented using the body-cover model, wave-reflection-analog sound propagation, and a boundary-layer asymmetric flow solver. The inclusion of a gap area of 0.03 cm^2 reduced the RMS energy transfer from the fluid to the vocal folds by 20 % and radiated SPL by 5 dB. When compensating for this reduction with an increased subglottal pressure to match the same SPL, a significant increase in MFDR and AC flow was noted, thus mimicking vocal hyperfunction. In addition, larger gap areas yielded less glottal pulse skewing and a glottal airflow proportional to the transglottal pressure drop.

10:20

5aSCa5. Modeling flow through the posterior glottal gap. Ronald Scherer, Brittany Frazer, and Guangnian Zhai (Commun. Sci. and Disord., Bowling Green State Univ., 200 Health Center, Bowling Green, OH 43403, ronalds@bgsu.edu)

The BGSU three mass model of phonation incorporates a posterior glottal gap. The airflow through the gap is governed by a simple hydraulic formula. The current study compares predictions of the gap size to the DC flow of three human subjects over two pitches at normal loudness. This work also explores the importance of the posterior gap by comparing extensive human mean subglottal pressure—mean flow—vocal process adduction data (from one subject) with the behavior of the model. The relation between glottal gap size to the DC flow for the three subjects was exceptionally strong ($R^2 = 0.99$). The relatively linear pressure-flow relationships at specific adduction values for the human data are hypothesized to be mimicked by the model, with a reasonable relationship to the size of the posterior glottal gap. [Work support from NIH.]

10:40

5aSCa6. Acoustic coupling during incomplete glottal closure and its effect on the inverse filtering of oral airflow. Matías Zañartu (Dept. of Electron. Eng., Universidad Técnica Federico Santa María, Av España 1680, Valparaíso, Valparaíso 2390123, Chile, matias.zanartu@usm.cl), Julio C. Ho (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN), Daryush D. Mehta, Robert E. Hillman (Ctr. for Laryngeal Surgery and Voice Rehabilitation, Massachusetts General Hospital, Boston, MA), and George R. Wodicka (Weldon School of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Inverse filtering of oral airflow using closed-phase linear prediction is expected to preserve the effects of source-filter interactions in the glottal airflow pulse. Under incomplete glottal closure, the glottal airflow estimation is more challenging due to a lowered glottal impedance, increased subglottal coupling, and violated all-pole assumption. To account for these effects, a model-based inverse filtering scheme allowing for coupling between glottis and upper and lower airways was developed. Acoustic transmission in the tracts used a frequency-domain transmission line. A linearized, time-varying expression was used for the glottal impedance, along with a dipole representation. Synthetic vowels sounds and actual recordings were used to evaluate the proposed scheme. Subject-specific model parameters were obtained from simultaneous aerodynamic, acoustic, and high-speed videoendoscopic recordings of normal subjects uttering vowels with various degrees of glottal closure. Results illustrated that, even under incomplete glottal closure, the airflow entering the vocal tract preserved source-filter interactions and was comparable to that obtained using closed-phase linear prediction. The scheme also yielded an uncoupled glottal airflow that exhibited a clear pulse de-skewing, making it proportional to the glottal area. Cases with larger glottal gaps exhibited lower mean impedances and less pulse skewing, with airflow estimates proportional to the transglottal pressure drop.

11:00

5aSCa7. Liquid dynamics in a self-oscillating poroelastic model of the vocal fold. Chao Tao (Inst. of Acoust., Nanjing Univ., 22 Hankou Rd., Nanjing, Jiangsu 210093, China, taochao@nju.edu.cn), Jack J. Jiang (Dept. of Surgery, Univ. of Wisconsin Med. School, Madison, WI), and Xiaojun Liu (Inst. of Acoust., Nanjing Univ., Nanjing, China)

The vocal fold tissue is considered as the composite of a porous elastic frame (composed of specialized proteins, carbohydrates, lipids, collagen fibers, and elastin fibers) filled with liquid. The properties of tissue are codetermined by the porous solid, the fluid, and their interaction. The poroelastic description of the vocal fold tissue could improve our knowledge about the mechanical properties and microstructures of these biological tissues. In this study, a self-oscillating poroelastic model is proposed to study the liquid dynamics in the vibrating vocal folds, where the vocal-fold tissue is treated as a transversally isotropic fluid-saturated porous material. Rich liquid dynamics have been found in this model. In the vertical direction, the liquid is transported from the inferior side to the superior side due to the propagation of the mucosal wave. In the longitudinal direction, the liquid is accumulated at the anterior-posterior midpoint. However, the strong collision between two vocal folds forces the accumulated liquid out there in a very short duration. These findings of the liquid dynamics could be helpful for exploring etiology of some laryngeal pathology, optimizing laryngeal disease treatment, understanding hemodynamics in the vocal folds, etc.

11:20

5aSCa8. Nonlinearities in block-type reduced-order vocal fold models with asymmetric tissue properties. Byron D. Erath (Mech. and Aeronautical Eng., Clarkson Univ., 8 Clarkson Ave., Box 5725, Potsdam, NY 13699, berath@clarkson.edu), David E. Sommer (Mech. and Mechatronics Eng., Univ. of Waterloo, Waterloo, ON, Canada), Matías Zañartu (Deparment of Electron. Eng., Universidad Técnica Federico Santa María, Valparaíso, Chile), and Sean D. Peterson (Mech. and Mechatronics Eng., Univ. of Waterloo, Waterloo, ON, Canada)

Modeling the vocal fold structure as a reduced-order system is an attractive approach for exploring the dynamics of both normal and pathological phonation. This approach has been used ubiquitously in scientific speech investigations due to its relatively high order of accuracy and low computational cost. In addition, good agreement can also be found between model and clinical data. In the case of pathological speech complex vocal fold dynamics may exist, exhibiting phenomenon such as bifurcation and chaos. The ability to capture these features in reduced-order vocal fold models is a much celebrated feature. However, the question has arisen whether these nonlinearities arise due to the physics, or if they are merely an artifact of the model and its sensitivity to initial and boundary conditions. We explore the sensitivity of commonly-employed reduced-order vocal fold models to both contact mechanics, and the geometric prescription of the vocal fold model. Nonlinearities arising from asymmetric vocal fold tensioning are investigated. Nonlinearity in the vocal fold dynamics is identified by determining the predictive capability of linear and nonlinear Volterra-Weiner-Korenberg series. Nonlinearities in the vocal fold oscillations are shown to be highly dependent upon model formulation and implementation, as opposed to physical features of speech.

11:40

5aSCa9. High-dimensional and low-dimensional models of tissue and fluid movement in airways: How will they converge? Ingo R. Titze (National Ctr. for Voice and Speech, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@ncvs2.org), Ingo R. Titze (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, Utah), and Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT)

Airways surrounded with soft tissues are used by many species to produce sound. The structure of the airway can become unstable and oscillate at constricted regions such as the vocal folds, the ventricular folds, the aryepiglottic folds, the tongue dorsum, the velum, the tongue tip, or the lips. In modeling these fluid-structure interactions, approaches ranging from simple analytical models to low- and high-dimensional computational models have been used. For voice and speech scientists and practitioners who use these models, major issues are (1) what phenomenon do they explain, (2) how valid are model predictions, (3) what level of accuracy is needed to make a prediction, and (4) how user-friendly are the models? Some criteria will be given for making choices on degrees of freedom, tissue and flow characterization, and benchmarking with measurement to converge on utility and complexity of models.

FRIDAY MORNING, 7 JUNE 2013 — 516, 9:00 A.M. TO 12:00 NOON

Session 5aSCb

Speech Communication: Production and Perception II: The Speech Segment (Poster Session)

Michael Kiefte, Chair

Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

5aSCb1. Acoustic and articulatory evidence for the phonological status of liaison consonants. Marie-Josee L'Esperance and Sam Tilsen (Linguistics, Cornell Univ., 522, West Seneca St., Apt A, Ithaca, NY 14850, ml978@cornell.edu)

French liaison consonants (LC) are a special class of word-final segments whose realization depends on a combination of phonological, lexical, and syntactic factors. Most previous analyses viewed LCs as coda consonants realized only before vowel-initial words. Because of their special status as syntactically and lexically conditioned, an interesting question is whether LCs exhibit typical acoustic and articulatory characteristics of word-final consonants. This paper presents the results of an experimental investigation of LCs in adjective+noun pairings in Quebec French, using electromagnetic articulography to collect kinematic data. Compared to typical coda and onset consonants, liaison consonants were found to exhibit smaller magnitude release gestures, and in some cases, LC closure gestures were more similar to those of onsets than codas. Unlike coda consonants, LCs did not induce gestural shortening or laxing (F1 raising) of the preceding vowel. Hence our results indicate that, although LCs have been analyzed as word-final consonants, they exhibit neither typical coda- or onset-like acoustic and articulatory properties. These results are important because they show that syntactically conditioned lexical phonology can result in non-canonical articulatory patterns, and hence speak to the need for models of production to incorporate both lexical representations and syntactic context as factors.

5aSCb2. Comparison of native and non-native consonant articulation with real-time magnetic resonance imaging of the vocal tract. Sam Tilsen (Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, tilsen@cornell.edu), Bo Xu, Pascal Spincemaille (Weill Cornell Medical College, New York, NY), Madhur Srivastava, Peter Doershuck (Cornell Univ., Ithaca, NY), and Yi Wang (Weill Cornell Medical College, New York, NY)

This study examines the effects of vocalic context and stress on consonantal articulation using real-time magnetic resonance imaging (rtMRI) of the vocal tract. A native speaker of English and an L2 English speaker of Mandarin produced eight repetitions of a set of vowel-consonant-vowel sequences in which the target consonants—voiceless stops and nasals—occurred before or after a stressed vowel. Images of the mid-sagittal plane of the vocal tract were acquired with a sampling rate of 8.1 ms and were reconstructed using a variable density golden angle ordered spiral algorithm. Vocal tract variable time-series for each token were extracted from the images by taking the average pixel intensity for each frame in hand-labeled regions of interest. Analyses of variance were conducted on kinematic variables (movement range, velocity, and duration of consonantal closure and release movements) and relative timing of consonantal and vocalic gestural landmarks. The results showed greater effects of stress and vocalic context on articulatory kinematics and timing for the native speaker compared to the non-native speaker. This study demonstrates that rtMRI can be used to assess fine-grained differences in articulation that are likely attributable to language background.

5aSCb3. Effects of phonemic variability and language dominance on Canadian French-English bilinguals' perception of French vowels in various phonological contexts. Allison A. Johnson (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, 2121 Univ. Ave., apt 28, Madison, WI, aajohnson4@wisc.edu) and Franzo Law (Psychology, Univ. of Wisconsin-Madison, Madison, WI)

Phonemic variability can cause productions within a category of one language to be mapped onto multiple categories in a different language [Escudero (2009)]. For example, French productions of /ɛ/ are labeled both as /e/ and /ɛ/ by monolingual English listeners, likely due to spectral variability in production of French /ɛ/ [Strange *et al.* (2009)]. The present study examines how phonemic variability in Canadian French (CF) affects the perceptual tendencies of bilingual Canadian French-English listeners with varying levels of language dominance. Vowel productions by monolingual CF speakers were used in a modified identification task [Law (2011)]. The vowels were word-final in several phonological contexts (preceded by labial, coronal, and back consonants; followed by labial and coronal consonants) in real and nonwords embedded in carrier phrases. A subset of these

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vowels /e-ɛ-i-y/ was analyzed for duration, vocalic midpoint, and formant trajectories to examine the relationship between phonemic variation (as a function of phonological environment) and the perception results by bilinguals dominant in either CF or Canadian English (CE). We predict that the performance of CE-dominant listeners will vary in terms of speed and accuracy based on how similar each token is to CE vowel category expectations.

5aSCb4. Listeners' sensitivity to talker differences in voice-onset-time: Phonetic boundaries and internal category structure. Rachel M. Theodore, Emily B. Myers, and Janice Lomibao (Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

Recent findings indicate that listeners are sensitive to talker differences in phonetic properties of speech, including voice-onset-time (VOT) in word-initial stop consonants. The current work extends our earlier research by examining the degree to which listeners adjust the initial mapping from acoustic signal to segmental representation on a talker-specific basis. Two groups of listeners are exposed to a talker producing "cane." Word-initial VOTs are manipulated such that one group hears "cane" produced with short VOTs and the other group hears "cane" produced with relatively longer VOTs. Following training, listeners' voicing boundary for a /g/-/k/ continuum is tested. In addition, listeners are tested on phonetic category space by rating members of the continuum for typicality as /k/. If listeners adjust segmental mapping to accommodate talker differences in phonetic properties of speech, then we expect to observe a displacement in the voicing boundaries in line with earlier exposure. Moreover, if this adjustment entails a comprehensive reorganization of phonetic category space, then the /k/ exemplars rated most prototypical will also be displaced for the two listener groups. These data will be discussed in terms of potential constraints on talker-specificity in spoken language processing.

5aSCb5. The Nez Perce vowel system: A phonetic analysis. Katherine Nelson (Dept. of Linguist, MS 23 Rice Univ., P.O. Box 1892, Houston, TX 77251-1892, katienelson@rice.edu)

The Nez Perce language, a highly endangered American Indigenous language, has been of great interest in phonology over the years due to its unusual vowel system and vowel harmony process. Nez Perce has five monophthongs and seven diphthongs, all with phonemic length. This system is unusual because rather than /i, e, a, o, u/ as is common, the Nez Perce inventory is /i, æ, a, o, u/. This uncommon inventory leads to two seemingly unrelated dominant, /i, a, o/, and recessive, /i, æ, u/, vowel harmony groups. To date there has been no phonetic analysis of the vowel system. This paper provides an acoustic analysis of the vowels as well as the vowel harmony system. Five native speakers (two males and three females) were used to analyze the vowels and the three female native speakers were used for the vowel harmony study. Results support the current vowel system analysis for Nez Perce. The vowel harmony data lend support to the current advanced tongue root analysis; however, it also poses questions for future research.

5aSCb6. Perception of Canadian French rhotic vowels. Jeffrey Lamontagne (Linguistics, Univ. of Ottawa, Ottawa, ON, Canada) and Jeff Mielke (English, North Carolina State Univ., 221 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695-8105, jimielke@ncsu.edu)

Some speakers of Canadian French produce words such as *pneu*, *un*, and *coeur* with rhotic-sounding vowels similar to English /ɹ/ (Dumas 1972). Articulatory imaging [Mielke (2011)] shows that they are produced with bunched and retroflex tongue postures and low F3, much like English /ɹ/. Nevertheless, native speakers typically are completely unaware of the difference, even when it is pointed out to them. We report a preliminary perception study of rhotic vowels. 7735 words with mid front round vowels were coded as "rhotic," "non-rhotic," or "ambiguous" by two listeners: a French-English bilingual from eastern Ontario and an American English speaker. The bilingual coded 0.3% as rhotic (vs. 10.0% for the anglophone) and 7.3% as ambiguous (vs. 8.9%). Logistic regressions show that the anglophone relied on F3 to distinguish rhotic+ambiguous tokens from non-rhotic tokens, while the bilingual weighted several cues about equally, including F1 cues to diphthongization, which can co-occur with rhoticity. Results will be presented from an ongoing AX discrimination task experiment involving rhotic, non-rhotic, and ambiguous vowel tokens, with francophone, bilingual, and anglophone listeners from the Ottawa-Gatineau region, Paris, France, and Raleigh, North Carolina.

5aSCb7. Describing alternative articulations of the Spanish trill /r/ by ultrasound technology. Ahmed Rivera-Campos and Suzanne E. Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3407 Clifton Ave., Apt 27, Cincinnati, OH 45220, riveraam@mail.uc.edu)

The Spanish trill /r/ is typically described as having a single realization—that is, as the result of aerodynamic forces operating on a tongue tip/blade constriction placed midsagittally in the dental, alveolar, or postalveolar place of articulation, in such a way that air channels along the sides of the tongue open and close for multiple cycles of vibration. As with American English /ɹ/, this sound is acquired late by typically developing children and is frequently an element in articulatory disorders. As with American English /ɹ/, perceptually equivalent "correct" trill /r/'s may be realized differently by different speakers. Knowledge of these alternate "correct" realizations would clearly be helpful to clinicians and learners of Spanish. In this preliminary study, we report data from ultrasound images of individuals who speak different dialects of Spanish. Preliminary data suggests there are at least two different articulatory postures used when producing the Spanish trill /r/, one of which involves lateralization. These articulatory differences do not affect what native listeners categorize as perceptually correct Spanish trills.

5aSCb8. Producing whole speech events: Anticipatory lip compression in bilabial stops. Chenhao Chiu and Bryan Gick (Linguistics, Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, chenhao@alumni.ubc.ca)

Bilabial stops /b/, /p/, and /m/ ostensibly share a common lip constriction. Recent evidence shows that different bilabial stops involve distinct facial muscle activations, suggesting that oral speech movements anticipate aerodynamic conditions [Gick *et al*. Proc. Acoust. (2012) 2pSC1]. The present study investigates how the lips themselves behave in whole speech events. Existing models of speech production governing only articulatory motions predict that lip compression would respond to changes in aerodynamic conditions rather than anticipating such changes; a model that includes whole events predicts anticipatory activation of lip muscles with concomitant kinematic lip compression, but only in cases where a real increase in air pressure is expected. Lip kinematics were recorded using OptoTrak to trace lip movements of bilabial stops in response to imperative acoustic stimuli. Results show consistent anticipatory lip compression in spoken /b/ and /p/, but not in non-speech jaw opening movements and only sporadic compression in mouthed /b/ and /p/, where air pressure is not expected to increase. Biomechanical simulation using an orofacial model developed within the Artisynth simulation toolkit (www.artisynth.org) confirms anticipatory muscle activations. These findings support a model of speech tasks wherein coordinated body-level muscular systems govern whole speech events.

5aSCb9. Intrinsic variations in jaw deviations in English vowels. Caroline Menezes (Health and Human Sci., Univ. of Toledo, 2801 W. Bancroft St., Toledo, OH 43606, caroline.menezes@utoledo.edu) and Donna Erickson (Showa Music Univ., Kawasaki, Japan)

Research shows that jaw deviations follow lexical and phrasal stress patterns reflecting the rhythmic structure of English [Erickson (2002), (2010), Erickson *et al*. (in press), Menezes (2003), Menezes *et al*. (2003)]. Syllable strength thus can be determined by the amount of jaw displacement regardless of the target vowel. This study systematically analyzes jaw displacement based on the intrinsic variations in vowel height. EMA recordings were analyzed of one speaker producing a short phrase wherein 11 English vowels were spoken in a controlled phonetic environment. The phrase used was "Type X first" where, X was a closed monosyllabic word containing the target vowel surrounded by stop consonants. Stop consonants allow the jaw to start from the bite plane and return to the bite plane therefore all deviations of the jaw are attributed to the articulation of the vowel. The target word was produced in phrase initial, middle and final positions. Comparisons were made across vowel height, tongue root advancement and phrase position. Preliminary findings reveal that jaw displacement was significantly

different at the level of $p = 0.05$ for vowels based on all three parameters: Vowel height, vowel tense/lax, and phrasal positioning.

5aSCb10. Multidimensional scaling of English fricatives using the acoustic change complex of electroencephalogram recordings. Paul Iverson, Marta Mulyak, and Anita Wagner (Univ. College London, 2 Wakefield St, London WC1N 1PF, United Kingdom, p.iverson@ucl.ac.uk)

In electroencephalogram (EEG) recordings, there is a characteristic P1-N1-P2 complex after the onset of a sound, and a related complex, called the Acoustic Change Complex (ACC), when there is a change within a sound (e.g., a formant transition between two vowels). In the present study, the ACC was measured for all possible pairs of eight sustained voiced and voiceless English fricatives, in EEG recordings from native speakers of British English. The magnitude of the ACC was used as a similarity measure for multidimensional scaling (MDS), producing a two-dimensional perceptual space that related to both voicing and place of articulation. The results thus demonstrate that this combination of ACC and MDS can be effective for mapping multidimensional phonetic spaces at relatively early levels of auditory processing, which may be useful for evaluating the effects of language experience in adults and infants.

5aSCb11. An acoustic study of the Mary-merry-marry vowels in the Mid-Atlantic United States. Carina Bauman (Linguistics, New York Univ., 10 Washington Place, New York, NY 10003, cb1864@nyu.edu)

This paper examines the acoustic properties of a subset of American English vowels before /r/–specifically, the mid and low front vowels in the MARY, MERRY, and MARRY classes. While the dialectal variation associated with the MARY, MERRY, and MARRY classes is well known in the sociolinguistics literature, the precise phonetic nature of these vowels has not been well studied. Many scholars [e.g., Labov *et al*. (2006)] transcribe the vowels as /eɪ/, /ɛ/, and /æ/, respectively, but these labels are largely impressionistic. The present study samples five speakers from the Mid-Atlantic region of the United States who maintain a three-way distinction between MARY, MERRY, and MARRY. The speakers' productions of MARY, MERRY, and MARRY vowels in both sentence and word list tasks were analyzed in Praat, and the resultant formant values were submitted to one-way ANOVAs, followed by pairwise comparisons (Tukey's HSD). The primary finding of this study is that the MARY vowel is acoustically closest to a tense (raised and fronted) variant of /æ/, similar to that which appears before nasals in many American English dialects. Contrary to previous descriptions, the MARY vowel shows little overlap with /eɪ/, suggesting that the conventional transcription should be revised.

5aSCb12. Effects of variation on processing of word-medial consonants. Benjamin V. Tucker (Dept. of Linguist, Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca), Kaitlin Mackie (Dept. of Speech Pathol. and Audiol., Univ. of AB, Edmonton, AB, Canada), and Tatiana Kryuchkova (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada)

The present study investigates processing of variation in word-medial stops and the role of this variation in phoneme recognition. Variation in word-medial stops has been shown to influence lexical access, but it is unclear whether this variation also affects recognition at the phoneme level [Tucker, J. Phonet., **39**, (2011), 312–318]. A phoneme monitoring task is used to investigate the role of production variation in the identification of word-medial stop consonants. Following Warner and Tucker [J. Acoust. Soc. Am. **130**, (2012), 1606–1617], the stimuli comprise English nonwords with target consonants [b], [p], [d], [t], [g], [k] in an intervocalic post-stress position produced with a range of production variation. This variation was grouped into three sets including groups at each extremes and a group in the middle of the range. The acoustic characteristics of the stops, which reflect this variation (e.g., consonant duration and intensity difference) were extracted and used to model the reaction times collected during the phoneme monitoring experiment. The reaction times were statistically modeled using a linear mixed-effects regression. Models of spoken word recognition and models of lexical storage are used to interpret the results and the contribution of the results to our understanding of these models is discussed.

5aSCb13. Assimilation of word-final nasals to following word-initial place of articulation in United Kingdom English. Margaret E. Renwick, Ladan Baghai-Ravary, Rosalind Temple, and John S. Coleman (Phonet. Lab., Univ. of Oxford, 41 Wellington Square, Oxford, United Kingdom, margaret.renwick@phon.ox.ac.uk)

Using very large speech corpora, we can study rare but systematic pronunciation patterns in spontaneous speech. Previous studies have established that word-final alveolar consonants in English (/t/, /d/, /n/, /s/ and /z/) vary their place of articulation to match a following word-initial consonant, e.g., "ran quickly" → "ra[ŋ] quickly." Assimilation of bilabial or velar nasals, e.g., "alar[ŋ] clock" for "alarm clock," is unexpected according to linguistic frameworks such as underspecification theory. The existence of systematic counterexamples would challenge that theory, but these might have been previously overlooked because they are infrequent. From the c. 8-million word Audio BNC (http://www.phon.ox.ac.uk/AudioBNC) we extracted more than 4,000 tokens of relevant word pairs, to determine whether non-alveolar assimilations occur and with what distribution. Word and segment boundaries were obtained by forced alignment, and F1–F3 formant frequencies were estimated using Praat. Formant frequencies in assimilation environments were compared to non-assimilating controls (e.g., the**m d**own vs. the**m b**ack/the**n d**own). We also examined patterns of variability in different contexts. We will present evidence that velar and bilabial nasals sometimes *do* assimilate, though less frequently than alveolars.

5aSCb14. Nasality effects in word-final nasal clusters. Emily Nguyen (Linguistics, New York Univ., 10 Washington Place, New York, NY 10003, emily.nguyen@nyu.edu)

This study presents an acoustic analysis of the proposed phonological process of nasal deletion in English [Cohn (1993), Bybee (2001)] in which vowel-nasal consonant-oral consonant (VNC) sequences differ in the realization of the nucleus based on the voicing feature of the final oral consonant. Previous work has proposed that the nasal consonant in VNC[-voice] sequences, e.g., tent, is deleted resulting in the realization of a fully nasalized vowel followed by a voiceless oral consonant, ṼC[-voice]. This phonological process of nasal deletion is said not to take place in VNC[+voice] sequences, e.g., tend, resulting in the full realization of all segments. Data from 15 speakers of American English were analyzed, and nasal deletion rates for VNC[-voice] sequences were low ($<12\%$). A further complication for this proposed phonological process is that vowel nasality differences between VNC[-voice] and VNC[+voice] sequences are small. A measure of vowel nasality, A1-P0, shows that while nasality is predicted to be constant from the onset of vowel production in VNC[-voice] sequences, statistical results show that vowels in both VNC[-voice] and VNC[+voice] sequences are progressively nasalized throughout production. Taken together, these findings suggest that nasal deletion is occasional and occurs due to phonetic implementation rather than a phonological process.

5aSCb15. Articulatory overlap in English syllables with postvocalic /ɹ/. Rachel Walker (Linguistics, Univ. of Southern California, GFS 301, 3601 Watt Way, USC, Los Angeles, CA 90089-1693, rwalker@usc.edu) and Michael Proctor (Linguist, Univ. of Western Sydney, Sydney, NSW, Australia)

In General American English (GAE), only two full vowels [ɑ̃, ɔ] occur in syllables ending in [ɹ] plus a non-coronal consonant, e.g., <harp>, <pork>. An articulatory study of rhotic production by three speakers of GAE was conducted using real-time structural magnetic resonance imaging (rtMRI) [Narayanan *et al*. (2004)]. Subjects produced /ɹ/ in simple and complex syllable codas in a range of vocalic environments. Results show that the tongue dorsum shows the least movement in [-ɑ̃ɹ-] and [-ɔɹ-] sequences. This dorsal stability sheds light on why [ɑ̃] and [ɔ] are the only full vowels occurring before codas with /ɹ/ and a non-coronal consonant. English syllable rimes have been analyzed as maximally three timing units in length [Hammond (1999)]. Long vowels occupy two units, and most coda consonants occupy one unit, rendering rimes with long [ɑ̃]/[ɔ] followed by [ɹp] or [ɹk] problematic. We hypothesize that the high degree of overlap in the dorsal posture in [ɑ̃ɹ] and [ɔɹ] sequences allows their gestures to be partially blended and function like a diphthong that occupies two units in the rime. This study supports a view of maximal constituency in rimes with [ɹ] that takes articulatory overlap into account.

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5aSCb16. Acoustic correlates of flaps in North American English. Donald Derrick (NZILBB, NZILBB, U Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, donald.derrick@gmail.com) and Ben Schultz (MARCS Inst., Univ. of Western Sydney, Penrith, NSW, Australia)

Using Brightness and Motion mode ultrasound, Derrick and Gick (2011, CJL) identified four categorical variations of flap (rapid “d”-like) tongue movements produced in North American English (that is, up-flaps, down-flaps, alveolar taps, and postalveolar taps). These variants can be used to test hypotheses about constraints on speech articulation, such as local context, gravity and elasticity, speech rate, and longer distance anticipatory coarticulation. The present study examines acoustic correlates of flap and rhotic (“r”-like) vowel variations in order to facilitate the understanding of articulatory mechanisms that underlie acoustic outputs. Understanding the relationship between articulatory mechanisms and acoustic outputs may allow us to draw inferences about articulatory mechanisms from pre-existing and future acoustic databases. Preliminary results identify significant differences in f0, F1, F2, F3, F4, and F5 values between these flap variants. In addition, we used supervised hierarchical clustering to aid in identification of flap variants based on both vocalic context and acoustic parameters. Unsupervised hierarchical clusters will also be used to identify whether the four flap variants that were previously identified in our articulatory studies are enough to capture the actual categorical variation that occurs and, if not, to identify currently unknown categories of flap variation in North American English.

5aSCb17. Cortical hemodynamic response patterns to normal and whispered speech. Gerard B. Remijn (Int. Education Ctr., Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8545, Japan, remijn@design.kyushu-u.ac.jp), Mitsuru Kikuchi, Yuko Yoshimura (Res. Ctr. for Child Mental Development, Kanazawa Univ., Kanazawa, Japan), Sanae Ueno, Kiyomi Shitamichi, and Yoshio Minabe (Dept. of Psychiatry and Neurobiology, Kanazawa Univ., Kanazawa, Japan)

Whispered speech is often used in direct person-to-person communication as a means to confidentiality. Compared with normally vocalized speech, whispered speech is predominantly unvoiced, i.e., produced without vocal fold vibration, and has no clear fundamental frequency. By using near-infrared spectroscopy (NIRS), we assessed cortical hemodynamic response patterns to normally vocalized and whispered speech in adult listeners ($n = 13$). Stimuli consisted of 20-s strings of Japanese word associations spoken by a female voice. Average oxygenated hemoglobin values (oxy-Hb) were obtained over two regions of interest (ROIs). Results showed that oxy-Hb values during the perception of normally vocalized speech were highest over the left temporal ROI, but not significantly different from values measured over other ROIs. Oxy-Hb values during whispered speech were highest over the right temporal ROI and significantly higher ($p < 0.05$) than those obtained over the left temporal ROI. No significant differences, however, were found in oxy-Hb comparisons between normally vocalized and whispered speech, although the right temporal ROI comparison bordered on significance, with whisper inducing the higher value. Together, the results seem to suggest that whispered speech is a potent catalyst of cortical hemodynamic activity, especially over the right temporal cortex, in spite of its relatively modest sound level as compared to normal speech.

5aSCb18. Spectral discrimination of consonants for English and Chinese Listeners. Tatum M. Fritz and Chang Liu (Commun. Sci. and Disord., Univ. of Texas at Austin, 1 Univ. Station A1100, Austin, TX 78712, tatumfritz@utexas.edu)

Previous work in our laboratory showed that Chinese listeners had significantly higher thresholds for vowel formant discrimination than American English listeners, possibly due to a sparse vowel system in Chinese but a crowded vowel space in American English. Considering that the two languages have similar consonant densities (i.e., similar numbers of consonants), we hypothesized that English and Chinese listeners might have similar thresholds of spectral discrimination of consonant stimuli. Thresholds of spectral shift in the English consonant, /s/, were measured in an isolated form and in a VCV context for both English and Chinese listeners. Preliminary results showed that there was no significant difference in consonant discrimination thresholds between the two groups of listeners for either isolated or VCV consonants. These findings support that phonemic density may play an important role in spectral discrimination of speech sounds.

5aSCb19. Three-dimensional vocal tract modeling of fricatives /s/ and /sh/ for post-glossectomy speakers. Xinhui Zhou (Elec. Eng., Univ. of Maryland, College Park, 4325 Rowalt Dr., apt 101, College Park, MD 20740, zxinhui2001@gmail.com), Jonghye Woo (Dept. of Neural and Pain Sci. and Orthodontics, Univ. of Maryland Dental School, Baltimore, MD), Maureen Stone (Dept. of Neural and Pain Sci. and Orthodontics, Univ. of Maryland Dental School, College Park, MD), and Carol Espy-Wilson (Elec. Eng., Univ. of Maryland, College Park, College Park, MD)

Production of fricatives involves a narrow supraglottal constriction along the vocal tract. Air flows through the constriction, and generates turbulent noise source(s) by impinging on some obstacles downstream. In post-glossectomy speakers, the production of /s/ and /sh/ is often problematic. It is mainly caused by the tongue surgery, which changes tongue properties such as volume, motility, and symmetry, preventing the tongue from creating proper constrictions. The purpose of this study was to gain some insights on how the vocal tracts of abnormal /s/ and /sh/ are shaped and what are their corresponding acoustic consequences. Based on cine magnetic resonance images, we built 3-D vocal tract models for /s/ and /sh/ from two post-glossectomy speakers (one with abnormal /s/ and the other with abnormal /sh/). Due to the missing part of the tongue, the reconstructed vocal tracts are asymmetric with either an air-flow bypass or a side branch formed near the constrictions. Two coupled physics submodels are included in the 3-D FEM acoustic simulation: incompressible potential flow for the mean air flow and aeroacoustics for the distributed noise sources. The resulting acoustic spectra and acoustic roles of air flow bypass or side branch will be discussed. [This study was supported by NIH R01CA133015.]

5aSCb20. A study of data normalization measured by an electro-magnetic articulograph. Seiya Funatsu (Sci. Information Ctr., Prefectural Univ. of Hiroshima, 1-1-71 Ujinahigashi Minami-ku, Hiroshima 734-8558, Japan, funatsu@pu-hiroshima.ac.jp) and Masako Fujimoto (Ctr. for Corpus Development, National Inst. for Japanese Lang. and Linguist., Tachikawa, Japan)

We investigated the normalization of the data measured using an electro-magnetic articulograph (EMA). The data normalization was needed because the size of the articulator of each subject is different from the sex and/or body size. Moreover, tongue movement range depends upon the speaking styles, clear or unclear. Our experiments were as follows: tongue tip movements during articulation of non-native consonant clusters were measured. Speakers were 2 Japanese and 2 Germans. Speech samples were four nonsense words, bnaht, pnaht, gnaht, knaht. Tongue tip displacement D (mm) and moving time T (ms) between first and second consonant in consonant clusters were measured. D_x (X component of D) was normalized by the difference between maximum value of X (X_{max}) and minimum value of X (X_{min}) in each utterance, i.e., $D_{nx} = D_x/(X_{max} - X_{min})$. Also, D_y (Y component of D) was normalized, i.e., $D_{ny} = D_y/(Y_{max} - Y_{min})$. Hence, D_n (normalized D) was $D_n = (D_{nx}^2 + D_{ny}^2)^{1/2}$. T was normalized by word length L (ms), i.e., $T_n = T/L$. Before the normalization, the measured data was localized by each speaker on the T-D plane, while the normalized data were not localized on the $T_n - D_n$ plane. Accordingly, it was suggested that this simple normalization method would be effective in this experiment.

5aSCb21. The fluctuating-masker benefit for normal-hearing and hearing-impaired listeners with equal audibility at a fixed signal-to-noise ratio. Kenneth K. Jensen and Joshua G. Bernstein (Audiol. and Speech Ctr., Walter Reed National Military Medical Ctr., 8901 Rockville Pike, Bethesda, MD 20889, kkj@jensenkk.net)

While normal-hearing (NH) listeners demonstrate better speech intelligibility for fluctuating-masker than for stationary-noise conditions, hearing-impaired (HI) listeners generally show little or no fluctuating-masker benefit (FMB). This result has been interpreted in terms of suprathreshold deficits (e.g., reduced spectral or temporal resolution or distorted stream-segregation cues) that limit “dip-listening.” However, reduced FMB for HI listeners might instead be attributable to audibility limitations or to differences between the signal-to-noise ratios (SNRs) at which NH and HI listeners are tested. This study examined this issue by equalizing stationary-noise performance to allow measurements at a common SNR, equalizing audibility, and presenting identical signals to pairs of NH and HI listeners. Audibility was equalized using linear gain, low-pass filtering (4 kHz) and intensity filtering to remove speech-signal elements below the HI audiometric

threshold. Nonsense-syllable identification performance in stationary noise was equalized by adjusting the response set size. Stationary-noise trials (adapting set size) were interleaved with fluctuating-masker trials (adapting SNR), ensuring stable stationary-noise performance throughout the test. Fluctuating maskers included low- and high-rate modulated noise, speech-modulated noise, and an interfering-talker condition. Results were assessed to determine whether and under which conditions the HI listeners demonstrated reduced FMB not attributable to SNR or audibility effects.

5aSCb22. Phonetic properties of [v] in Russian, Serbian, and Greek. Christina Bjorndahl (Linguistics, Cornell Univ., 407 Lake St., B24, Ithaca, NY 14850, cjm295@cornell.edu)

This study examines the phonetic properties of the segment [v] in Greek, Serbian, and Russian. [v] patterns phonologically like an obstruent in Greek, but like a sonorant in Serbian; in Russian, it patterns with both obstruents and sonorants. We test the hypothesis that cross-linguistic differences in the phonological status of [v] correlate with phonetic differences. We report on spectral and durational measures of [v] in four environments: word-initial stressed, word-initial unstressed, word-medial stressed, and word-medial unstressed. Our results show an association between phonological patterning and phonetic realization. Greek tokens of [v] are produced with significantly more high-frequency spectral energy than those in Serbian, suggesting a relation between phonological status and phonetic realization in these two languages. Tokens of Russian [v] exhibit the same relationship to tokens of Serbian [v] in word-initial stressed position; elsewhere, they are produced with relatively little high-frequency spectral energy. Furthermore, the effects of word position and syllable stress are found to be additive in Russian. These results are important because they support the notion that there exist interactions between the phonological status of a segment and its phonetic realization.

5aSCb23. Phonological structure, non-native phoneme discrimination, working memory, and word learning. Noah H. Silbert, Benjamin K. Smith, and Scott R. Jackson (Ctr. for Adv. Study of Lang., Univ. of Maryland, 7005 52nd Ave., College Park, MD 20742, nsilbert@umd.edu)

It is well known that perception of non-native speech sounds is influenced by exposure and the mapping between non-native and native phonological categories. However, very little is known about the relationships between phonological structure, individual differences in non-native phoneme discrimination ability, and non-native word learning. These relationships are important in the design of tests for personnel selection for second language training. Two experiments were conducted to probe the generality of phoneme discrimination ability and the role of phonological structure and discrimination ability in word learning. In one experiment, 169 participants discriminated non-native contrasts from nine languages—three voicing/laryngeal contrasts, three place contrasts, and three tone/intonation contrasts. Confirmatory factor analysis model comparisons show that correlations between discrimination accuracies across contrasts are driven by low-level phonological structure (featural and segmental/super-segmental properties). In a second experiment, phonological working memory and voicing, place, and tone discrimination were measured for 167 participants and used to predict learning of pairs of non-native words differing in voicing, place, and tone. Consistent with the results from the first experiment, discrimination ability predicts accuracy in word learning above and beyond the ability of phonological working memory and according to feature-specific differences.

5aSCb24. Neutralizing differences in jaw displacement for English vowels. J. C. Williams (Independent, Kamuela, HI), Donna Erickson (Showa Univ. of Music, 1-11-1 Kamiasao, Asao-ku, Kawasaki 215-8558, Japan, EricksonDonna2000@gmail.com), Yusuke Ozaki (Univ. of Tokyo, Tokyo, Japan), Atsuo Suemitsu (JAIST, Ishikawa prefecture, Japan), Nobuaki Minematsu (Univ. of Tokyo, Tokyo, Japan), and Osamu Fujimura (The Ohio State Univ., Columbus, OH)

Maximum jaw displacement in the syllable varies primarily by vowel quality, syllable position in the phrase, lexical and phrasal stress, prosodic conditions, and the syllable consonantal periphery. EMA recordings were made of CVC syllables in 3-word phrases uttered by an American English speaker, where three target CVC words occurred in phrase initial, middle, and final positions, in order to ascertain the effect of vowel quality and syllable phrase position on jaw displacement, independent of other factors. Eleven English vowels, omitting diphthongs, formed the syllable nuclei, voiceless obstruents /p, t, k/ formed the syllable periphery, and the intonation pattern was kept constant for each phrase. Jaw displacement was measured by coil placement at the midline of the base of the lower incisors. The maximum vertical mandibular displacement on the vertical axis (z-axis for 3D EMA) was measured for each target CVC word. For each of the 11 vowels, an algorithm was developed to neutralize differences in the contribution of the mandibular vertical excursion in each of the three phrasal positions, i.e., 33 neutralization measures. These results indicate that this method may neutralize the mandibular contribution to differences in phonological vowel quality and phrasal position.

5aSCb25. Effect of syllable onset, coda, and nucleus on degree of skin stretching over the mandible. Ian L. Wilson (CLR Phonet. Lab., Univ. of Aizu, Tsuruga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, wilson@u-aizu.ac.jp) and Donna Erickson (Showa Music Univ., Kawasaki, Japan)

Movements of the mandible have been shown to correlate with English speech rhythm, and significant differences have been found between native speakers' mandible movements and those of second-language speakers. A simple, inexpensive method of inferring movements of the mandible is to use video tracking of a chin marker during speech. However, since the skin is free to stretch over the mandible, inferences using the chin marker may not always be accurate. This study examines the degree of skin stretching during onset stop consonant, coda stop consonant, and vowel in CVC syllables spoken as the middle word in a 3-word utterance. We made electromagnetic articulometer (EMA) recordings of two North American English speakers (1 male, 1 female). Measurements were made from coils placed on the lower incisor (LI) and on the skin of the mental protruberance (chin). Preliminary results show that both speakers have significant differences during the syllable nucleus between the LI and chin coils due to onset consonant, but not coda consonant. These results need to be taken into account as we continue to develop a method for video recording jaw displacement patterns in running speech.

5aSCb26. Anatomical considerations on the extrinsic tongue muscles for articulatory modeling. Kiyoshi Honda (School of Comput. Sci. and Technol., Tianjin Univ., 92, Weijin, Nankai, Tianjin 300072, China, khonda@sannet.ne.jp), Emi Z. Murano (Dept. of Otolaryngol., Johns Hopkins Univ., Baltimore, MD), Sayoko Takano (Dept. of Physiol., Univ. of Arizona, Tucson, AZ), Shinobu Masaki (ATR-Promotion, Inc., Kyoto, Japan), and Jianwu Dang (School of Comput. Sci. and Technol., Tianjin Univ., Tianjin, China)

Physiological articulatory models have evolved from simpler forms to complex ones, while recent models preserve traits of oversimplification and anatomical unreality. This work combines MRI observations at ATR-BAIC and Johns Hopkins University to point to the issue for advancing extrinsic tongue muscle modeling. The genioglossus, previously thought to arise from the genial tubercle of the mandible, has direct fiber attachments on the short tendon of the tongue. The posterior genioglossus, being regarded as a functional division of the muscle, can be given anatomical definition to the horizontal bundle arising from the inferior aspect of the short tendon. The styloglossus has been modeled as linear strings traveling "free in air" before inserting into the tongue, but the extralingual part is actually restrained by the surrounding soft tissues to lack mobility. The intralingual styloglossus forms anterior and posterior slings in the tongue tissue, possibly with the distal fibers of the hyoglossus. Combined styloglossus and hyoglossus shortening via the slings may be a factor shaping the tongue into various forms. [Work supported by WQ20111200010, 2013CB329301, and NIDCD K99/R00-DC009279.]

5aSCb27. Pharyngeal constriction in English diphthong production. Fang-Ying Hsieh, Louis Goldstein, Dani Byrd (Linguistics, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, fangyinh@usc.edu), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

This study tests the hypothesis that the acoustic difference between [a] in English diphthongs (e.g., [a] in "pie'd") and its corresponding monophthong (e.g., [a] in "pod") results from the same pharyngeal gesture being truncated by the following palatal glide in the diphthongal environment. Production data were collected with real-time MRI and have been analyzed using the direct image analysis (DIA) technique, which infers tissue

movement by tracking pixel intensity change over time in regions of interest. Preliminary results show that (1) DIA is capable of capturing the timing and magnitude of the pharyngeal constriction gesture that produces [a], and (2) the proposed hypothesis is supported; the formation time of the pharyngeal constriction in diphthongs is generally shorter than that in like monophthongs. Further, the pharyngeal component of the diphthong is shortened in phrase-medial as opposed to phrase-final position or when followed by a voiceless as opposed to a voiced coda consonant, and the duration of this interval strongly correlates with the resulting constriction degree as predicted by the truncation analysis. [Work supported by NIH.]

5aSCb28. Does articulatory setting provide some mechanical advantage for speech motor action? Vikram Ramanarayanan, Adam Lammert (Elec. Eng., Univ. of Southern California, 3740 McClintock Ave., EEB421, Los Angeles, CA 90089-2564, vramanar@usc.edu), Louis Goldstein (Linguistics, Univ. of Southern California, Los Angeles, CA), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Articulatory setting postures adopted during speech production are examined with the goal of determining whether setting postures are more mechanically advantageous than rest positions in facilitating motion of vocal tract articulators toward task goals. Articulatory simulations using the Task Dynamics Application (TADA) suggest that setting postures afford large changes with respect to speech tasks for relatively small changes in low-level speech articulators, thus affording greater mechanical advantage as compared to absolute rest postures. This study investigates this hypothesis using real-time Magnetic Resonance Imaging (rtMRI) data of read and spontaneous speech elicited from five healthy speakers of American English. Frames corresponding to inter-speech pauses, speech-ready intervals, and absolute rest intervals were identified and image features were automatically extracted to quantify the vocal tract postures in terms of both task-level constriction variables and articulatory variables. Locally Weighted Regression is then used to estimate the ratio of task velocities to articulator velocities (i.e., the lever or speed ratio) at postures corresponding to the different intervals of interest. Results show substantially higher speed ratios at inter-speech and ready postures as compared to absolute rest postures. [Work supported by NIH.]

5aSCb29. An electropalatography study of nasal-trill/lateral sequences in Spanish. Laura Colantoni (Spanish and Portuguese, Univ. of Toronto, Toronto, ON, Canada) and Alexei Kochetov (Dept. of Linguist, Univ. of Toronto, 100 St. George St., Rm. 4076, Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca)

Trills and laterals require relatively precise articulatory and aerodynamic settings that are at least partly incompatible with setting necessary to produce nasal stops. Historically, this incompatibility has often been resolved through assimilation, deletion, or epenthesis in within-word [n+r] and [n+l] clusters (e.g., Romance [nr] > [rr] or [ndr]). It is expected that similar, yet gradient effects would be observed in across-word or hetero-morphemic sequences of nasals and liquids. This study examines the production of Spanish nasal-liquid sequences using electropalatography (EPG). Linguopalatal contact data were collected from nine native speakers of Spanish (representing three dialects) producing various utterances with nasals before /r/ and /l/ (as well as before /n/). The analysis of C1 and C2 using standard EPG indices of constriction location and degree showed that nasals had a more retracted and partly deocclusivized constriction before /r/, and a lowered tongue dorsum before both /r/ and /l/. These differences, clearly reflecting anticipatory coarticulatory effects, were similar across speakers and the three dialects, which differ in the default realization of the final nasal (alveolar or velar). The findings provide evidence for articulatory incompatibility as a source of historical development of combinations of nasals and liquids.

5aSCb30. Perception of non-native consonant length in naïve English listeners. Vincent Porretta and Benjamin V. Tucker (Univ. of Alberta, 2-40 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, porretta@ualberta.ca)

English speakers are sensitive to phonetic length [Pickett and Decker (1960)] but do not maintain a phonemic length contrast [Hayes (2002)]. This study examines English speakers' ability to discriminate and identify intervocalic consonant length in Finnish non-words. The consonants were manipulated for length and half of the participants were given brief written instruction regarding the Finnish length contrast. In an AX discrimination task, participants responded to increasing contrast ratio gradiently; however, the instructed group performed significantly better than the uninstructed group. Proficiency in any second language aids contrast detection in those receiving no instruction. In a forced-choice identification task, participants showed no evidence of boundary effects; however, the instructed group performed significantly more like native Finnish controls. Again, second language proficiency aids in consonant length detection. The present results indicate that information about and attention to a novel contrast, along with second language experience, aid processing in novice listeners. Given that learners can eventually maintain native-like contrasts, these factors may be influential in the initial formation of L2 phonological representations and support phonetic level of processing [Werker and Tees (1984)], intermediate to non-linguistic acoustic processing and phonemic processing, at which acoustic duration becomes a phonetically relevant cue.

5aSCb31. A following sibilant increases the ambiguity of a sibilant continuum. David Fleischer (Linguistics, McGill Univ., 1085 Ave. Dr. Penfield, Montreal, QC, Canada, david.fleischer@mail.mcgill.ca), Meghan Clayards (School of Commun. Sci. and Disord. & Linguist, McGill Univ., Montreal, QC, Canada), and Michael Wagner (Linguistics, McGill Univ., Montreal, QC, Canada)

We examined the effect of three following contexts: /s/, /ʃ/, and vowel, on the categorization of a /s/-/ʃ/ continuum. Unlike previous findings of a shift in category boundary due to context [Mann and Repp (1980)], we found that in the context of a following sibilant, listeners found the target sibilant to be more ambiguous (shallower categorization slopes and responses closer to chance) than when followed by a vowel ($p < 0.001$). There was also a tendency for the /ʃ/ context (which affects pronunciation) to create more ambiguity than the /s/ context (which does not) ($p = 0.057$). In a second experiment, half the participants heard the following context as part of the same syntactic phrase as the target (e.g., "Whenever they fra? Shelly gets upset") and half heard it as part of a different phrase (e.g., "Whenever they fra? Shelly, John gets upset"). Pronunciation usually is more affected when target and context are in the same phrase (Holst and Nolan (1995). Although our stimuli were acoustically identical, listeners tended to perceive target sibilants as more ambiguous when the following /ʃ/ was part of the same phrase ($p = 0.059$), suggesting a role for top-down knowledge in interpreting segmental information.

5aSCb32. Exploring vowel mergers in northeast Ohio. Anna M. Schmidt and Kristin M. Weaver (Speech Pathol. & Audiol., Kent State Univ., A104 MSP, Kent, OH 444242, aschmidt@kent.edu)

The northeast corner of Ohio, as listed on dialect area maps, contains the boundaries of several dialect regions. This study compared perception and production for speakers from northeast Ohio and surrounding areas who merge and do not merge back vowels before /l/ in words such as "pool, pull, pole, dull." Acoustic formant analysis indicated a lowered and fronted vowel, as expected, for those who merge some or all of these types of words but different patterns of merging were seen, especially when a final syllable was added as in "pulling." Formant values were compared with values in "Luke" and "look." Perceptually, preliminary results indicate that the merged vowel is often labeled as /u/ unlike findings for other US dialect regions such as Utah. Productions from over 100 speakers were situated on detailed dialect maps.

5aSCb33. Automating phonetic measurement: The case of voice onset time. Neville Ryant, Jiahong Yuan, and Mark Liberman (Linguist. Data Consortium, Univ. of Pennsylvania, 3600 Market St., Ste. 810, Philadelphia, PA 19104, nryant@gmail.com)

Of 58 papers published so far this year in Journal of Phonetics, 16 (28%) feature Voice Onset Time (VOT) or related measurements, confirming that VOT remains a central concern in the field. However, phoneticians' VOT measurements generally continue to rely on human judgment, which requires significant labor, makes even large laboratory experiments onerous, and prevents the field from taking full advantage of the millions of hours of digital speech now becoming available. We present an algorithm for accurate automatic measurement of VOT, combining HMM forced alignment for determining approximate stop boundaries with paired burst and voicing onset detectors. Each detector is a frame-level max margin classifier operating on the scale-space projection of a small number of relevant acoustic features. On a large set of clean lab speech, this system has a mean absolute error (relative to human annotation) of only 2.8 ms, with 98% of errors <10 ms. On a subcorpus independently annotated by two of the authors, the system agreed with the two human annotators as well as they agreed with one another (1.49 vs 1.50 ms). Promising results on other datasets will be reported. The system will be released as open-source software.

5aSCb34. An electromagnetic articulography investigation of the Czech trill-fricative. Phil Howson (The Dept. of Linguist, The Univ. of Toronto, Sidney Smith Hall, 4th Fl. 100 St. George St., Toronto, ON M5S 3G3, Canada, phil.howson@mail.utoronto.ca), Chris Neufeld (The Dept. of Speech-Language Pathol., The Univ. of Toronto, Toronto, BC, Canada), and Alexei Kochetov (The Dept. of Linguist, The Univ. of Toronto, Toronto, British Columbia, Canada)

Previous studies have found that the degree of tongue grooving during production of fricatives correlates with their place of articulation. Czech has a cross-linguistically rare alveolar trill-fricative <ř>, which would be expected to pattern with alveolar fricatives in terms of the tongue grooving. The current study employs electromagnetic articulography (EMA) to investigate differences in tongue grooving between the trill-fricative and alveolar/post-alveolar fricatives. Czech native speakers produced words with target consonants in word-initial, intervocalic, and word-final positions, with sensors being attached to both the midline and the sides of the tongue. An angle between these sensors was calculated, and taken as a measure of the degree of grooving. The results (currently based on the data obtained from one speaker) showed that, contrary to the prediction, the degree of grooving for the trill-fricative was closer to the post-alveolar fricatives than to the alveolar fricatives, yet unique in its distribution. While the degree of grooving remained relatively stable temporally for the fricatives, it tended to gradually decrease for the trill-fricative. Furthermore, <ř> differed from the post-alveolar fricatives in having a significantly lower tongue body. The results thus suggest a unique articulatory configuration for the Czech trill-fricative. [Work supported by SSHRC.]

5aSCb35. Perception of vowel-inherent spectral change. Kathleen Chiddenton and Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, kathleen.chiddenton@dal.ca)

One family of theories regarding vowel perception suggests that onset and offset formant-frequencies are important for identification but that the shape of the transitions themselves is not perceptually important [e.g., Morrison and Nearey, J. Acoust. Soc. Am. **122**, EL15–EL22 (2007)]. The present study determined just-noticeable-differences in deviations for linear formant trajectories. Diphthong-like stimuli were manipulated by inserting a point of inflection into the otherwise linear transition. Several parameters were manipulated including vowel duration, location of the inflection point in time, direction of formant change, and fundamental frequency. Data from the first experiment indicate that listeners are largely insensitive to deviations from linearity of formant trajectory but that deviations that fell outside the range of frequencies spanned by the onset and offset were detectable. However, a second experiment in which only the first half of stimuli were presented gave different results. Results from these experiments along with several hypotheses are presented. [Work supported by SSHRC.]

5aSCb36. Modeling of vowel-inherent spectral change in spontaneous and elicited speech. Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., 1256 Barrington St., Halifax, NS B3J 1Y6, Canada, mkiefte@dal.ca) and Terrance M. Nearey (Dept. of Linguist, Univ. of Alberta, Edmonton, AB, Canada)

A database of speech samples was collected to examining vowel formant patterns in spontaneous and read speech. Recordings were made from 220 long-time residents of Nova Scotia and Prince Edward Island in two separate tasks: Participants read a series of sentences in several phonetic contexts and also provided a monologue of at least 10 min duration resulting in approximately 36 h of spontaneous speech. Vowel formant frequencies were measured for all stressed vowels in non-function words surrounded by obstruents. Speakers were divided among eight geographic regions. Consonant contexts were modelled via nonlinear regression based on a model inspired by Broad and Clermont (J. Acoust. Soc. Am. **81**,155). Spontaneous speech was characterized by retraction of front vowels, fronting of /u/, backing of /æ/, and greater diphthongization of /ɒ/. Older speakers were less likely to show a merger between /ɒ/ and /ɔ/. There was also some evidence of a partial Canadian Shift whereby /æ/ and /ɛ/ are lowered following merger of /ɒ/ and /ɔ/. Results from alternate models currently under development, which may better accommodate vowel-inherent formant movement, will be discussed. [Work supported by SSHRC.]

5aSCb37. Development of a vocal tract design tool based on a growth curve of the vocal tract length. Daisuke Ito and Kohichi Ogata (Ogata Lab., Graduate School of Sci. and Technol., Kumamoto Univ., 2-39-1 Kurokami, Chuo-ku, Kumamoto 860-8555, Japan, ito@st.cs.kumamoto-u.ac.jp)

One of the advantages of speech synthesis based on the vocal tract shapes is flexibility in speech sounds depending on the articulatory parameters. In order to take advantage of the flexibility, preparing various vocal tract shapes is important. To achieve this, this paper describes the development of a vocal tract design tool based on a growth curve of the vocal tract length reported by literature. In this tool, the growth curve is used to calculate the size of the vocal tract. A user can obtain a vocal tract shape depending on the age and sex using a slide button on the interface window. The parameters that describe vocal tract shape and vocal folds are used to produce vowel sounds and their formant frequencies are obtained. According to literature, there is a relationship between the fundamental frequency and formant frequencies through the age. We compared the formant frequencies calculated from the tool with those estimated from the literature. The comparison showed that the calculated formant frequencies were in relatively good agreement with the estimated ones. In addition, the distribution of the first and second formant frequencies showed a typical pentagonal distribution.

5aSCb38. Formant-based articulatory normalization and its application to vowel restoration. Yuichi Ueda, Kosuke Tominaga, and Tadashi Sakata (Graduate School of Sci. and Technol., Kumamoto Univ., 2-39-1 kurokami, chuoh-ku, kumamoto-shi, kumamoto 860-8555, Japan, ueda@cs.kumamoto-u.ac.jp)

Visual feedback of spontaneous speech is effective for articulatory training of deaf children and for speech rehabilitation of dysarthric patients. Especially, the visual representations of vowel formant frequencies have been used directly or indirectly for those purposes, because those acoustical parameters reflect the articulatory behavior. However, since not only the shape of the vocal tract but its size also affect the formant frequencies, minimization of the effect due to the differences in size is required. In such a speaker normalization, we defined a color space consisting of three circular ratios of formant frequencies and applied it to the color visualization of vowel sound. In this paper, we proposed a normalized articulation space as an expansion of the color space, where we assumed that neutral vowels of any speaker are mapped into a unique point. In addition, since the proposed articulatory space was regarded as the speaker independent representation of vocal tract shape, we also proposed a method to convert the modified articulation shape into the real formant space and applied it to the vowel restorations of disarthric speech.

5aSCb39. On distinguishing articulatory configurations and articulatory tasks: Tamil retroflex consonants. Caitlin Smith (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, smithcm@usc.edu), Michael Proctor (Linguist, Univ. of Western Sydney, Sydney, NSW, Australia), Khalil Iskarous, Louis Goldstein (Linguistics, Univ. of Southern California, Los Angeles, CA), and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, Los Angeles, CA)

Speech production can be described in multiple coordinate frames: articulatory configurations, gestural tasks, and acoustic patterns. Examination of the achievement of retroflex stops and liquids in Tamil suggests that we must consider separately the gestural task of apical pre-palatal constriction and the articulatory maneuver to achieve the task. The maneuver of the tongue during retroflex consonants varies across vowel contexts. Specifically, in the symmetrical intervocalic contexts between back vowels /a/ and /u/, an apical pre-palatal constriction is achieved by curling back the tongue. In the context of high front vowel /i/, a laminal pre-palatal constriction is achieved by bunching the tongue. However, the location of retroflex consonant constriction within the vocal tract is consistent across all of these vowel contexts, suggesting that the constriction task remains the same. Variation in the articulatory configuration of the retroflex in the two contexts was quantified through Gaussian curvature functions at fourteen points along the tongue, sampled at evenly spaced points throughout the vocal tract, on every other gridline of a polar-rectangular grid in every frame in each utterance. The empirical results support the notion that the articulatory configuration coordinate frame and the gestural task frame provide separate, but related, descriptions of speech production.

5aSCb40. C-V coarticulation in velar plosives. Vicki L. Krebs (Linguistics, The Ohio State Univ., 301 W. Euclid Ave., Springfield, OH 45506, krebs.86@buckeyemail.osu.edu), Yourdanis E. Sedarous (Linguistics, The Ohio State Univ., Grove City, OH), and Amanda L. Miller (Linguistics, The Ohio State Univ., Columbus, OH)

We present 114 fps lingual ultrasound data of three speakers' productions of words containing the initial velar plosive, and the following [a] and [i] vowels, in Mangetti Dune !Xung (N = 95 for [k] in the [a] context, N = 36 for [k] in the [i] context). We traced the midsaggital tongue edge at the frame just prior to the [k] release, and measured the tongue dorsum (TD) and tongue root (TR) constriction locations (CL's). The TRCL was measured 1 cm below the [k] peak. Results show that [k] in the [i] context has a 1.1 cm further forward TDCL than [k] in the [a] context for one speaker, and 1.2 cm further forward for the second speaker. The results are similar to those found for English by Stevens and House (1963). The TRCL is retracted 0.3 cm in [k] in the [a] context compared with [k] in the [i] context for the first speaker, and 0.6 cm more retracted for the second speaker. A third speaker had a retracted TDCL and TRCL in both vowel contexts. These results confirm that [k] is less resistant to coarticulation. Results show that the tongue root is involved in dorsal-front vowel coarticulation.

5aSCb41. The "panphonic" text of "The North Wind and the Sun" for the illustration of the International Phonetic Alphabet of Japanese consonants and its use in the phonetic analysis of Japanese speech. Shizuo Hiki (Waseda Univ., 5246-90-1-402 Yamaguchi, Tokorozawa 359-1145, Japan, hiki@waseda.jp) and Kuniko Kakita (Toyama Prefectural Univ., Izumi, Toyama, Japan)

A "panphonic" version of the text of "The North Wind and the Sun" for the illustration of the IPA of Japanese (Tokyo dialect) consonants has previously been devised by the present authors [Hiki *et al.*, Proc. 17th Int'l. Cong. on Phonetic Sciences, Hong Kong, 871–873 (2011)]. The present paper describes the main characteristics of this panphonic text in relation to its use in the phonetic analysis of Japanese speech utterances, providing examples of the result of the analysis of sample recitations. The panphonic text embraces all 16 consonant phonemes, their 11 major positional allophones, and 5 free variants in a short, simple text that consists of 8 sentences (113 words, 294 syllables). The relation among consonant phonemes and their allophones is shown effectively by a new arrangement of rows and columns in the IPA consonant chart. Possible pause locations are systematically indicated using appropriate pause symbols. The text is useful in examining the phonetic properties of Japanese utterances, for example, the effect of consonants on vowel devoicing, and large-scale segmentation of utterances by pauses of different durations.

5aSCb42. Perception and production in non-native speech: Russian palatalization. Leandro Bolanos (Linguistics, Yale Univ., 188 1/2 Willow St., Apt B, New Haven, CT 06511, leandro.bolanos@yale.edu)

It is well known that adults struggle in perceiving and producing certain phonological contrasts not present in their native language. Adults also find difficulty in learning the specific timing of non-native articulatory gestures and contextual differences present in the language. The present study investigates English speakers' perception and production of Russian contrasts involving palatalized consonants in varying contexts. Of interest are the effects of syllable position and palatalization on speakers' performance in perception and production. The framework of Articulatory Phonology [e.g., Browman and Goldstein (1986), (1992)] and the Perceptual Assimilation Model [e.g., Best *et al.* (2001)] are adopted to account for differences in timing between English and Russian with respect to palatalization, and to subsequently make predictions on English speakers' perception as well as their production of the different timing property present in Russian palatalization. Speakers of American English lacking any previous exposure to Russian participated in a series of perception and production experiments involving Russian palatalized stops, which vary in place of articulation (labials, coronals) and syllable position (onset, coda). Preliminary results indicate diminished performance for some contrasts in syllable coda position as well as correlations between the perception and production of palatalized consonants by English speakers.

5aSCb43. Analysis of stop consonants in Devanagari alphabet. Kushagra Singh and Nachiketa Tiwari (Mech. Eng., Indian Inst. of Technol. Kanpur, C110/9, IIT Kanpur, Kanpur, Uttar Pradesh 208016, India, kushagrs@iitk.ac.in)

The Devanagari alphabet, which is used by several Indian languages including Sanskrit and Hindi, has vowels and consonants are placed in tabular format, which are arranged according to how they originate. A part of this table is a 5 x 5 matrix and comprises of stop consonants, where different rows corresponding to velar, palatal, retro-flex, dental and labial consonants. In this paper, we have explored patterns that exist between different consonant sounds belonging to different rows and columns of this table. Toward this end, four sound samples from individuals have been recorded, and analyzed. Our analysis shows the existence of many interesting relationships, which exist between sounds populating different rows and columns of 5 X 5 matrix. One interesting observation which has been made is that the fundamental differences between 1st and 2nd/ 3rd or 4th member of each row are essentially the same in all the rows, but a few exceptions in the 5th row. For example, in all rows the 3rd member is a combination of a short-duration signal and the 1st member. Similarly, the 2nd member is 1st member with a nonzero mean pressure line. Further, the 4th member is found to be the combination of the other two (1–2 and 1–3) variations. In this paper, we present several such interesting relationships. These relationships may be potentially useful in several sound processing algorithms.

5aSCb44. Estimated relative vocal tract lengths from vowel spectra based on fundamental frequency adaptive analyses and their relations to relevant physical data of speakers. Mayuko Kobayashi, Ryuichi Nisimura, Toshio Irino, and Hideki Kawahara (Design Information Sci., Wakayama Univ., 930 Sakaedani, Wakayama, Japan, s130043@center.wakayama-u.ac.jp)

A Japanese vowel database of males, females and children speakers (385 speakers in total) along with relevant physical data [Deguchi *et al.* (2011)] was analyzed using a set of F0 adaptive procedures, which were developed for a speech analysis, modification and synthesis framework TANDEM-STRAIGHT [Kawahara *et al.* (2008)] and its extensions. By restricting spectral region in estimating ratios for vocal tract length normalization (VTLN), cepstrally weighted distance measure yielded estimates with 2% standard error, by assuming relative vocal tract lengths estimated using regression analysis as the ground truth. A set of regression analyses of the estimated relative vocal tract lengths, average F0s, ages, gender, and physical data (speakers' height, weight) were conducted. The results suggest that the proposed analysis procedures applied to the Japanese five vowels may provide sufficient information for estimating speakers' physical data. Possible applications of the proposed estimation procedures will also be discussed.

5aSCb45. Perception of /ra/-/la/ contrast in different contexts: monosyllable vs. sentence. Kanako Tomaru and Takayuki Arai (Sophia Univ., 7-1 Kioi-cho, Chiyodaku 1028554, Japan, himawari.kanako@gmail.com)

A strict assumption that underlies categorical perception hypotheses is that two speech sounds are discriminable only when they cross a categorical boundary emerging from an identification function. A number of researchers have attempted to show the categorical perception of the sounds in question (particularly consonants) through discrimination and identification tests. However, these tests are usually used for mono-syllables or mono-syllabic words. In this study, we first investigated whether the perception of /ra/-/la/ contrast indicate any categorical perception in a mono-syllabic CV context, using synthesized syllables. Next, we tested whether categorical perception can be also observed in a sentence through perceptual discrimination and identification experiments. Results showed that (1) a discrimination peak

predicted by the identification function was obtained only for the mono-syllabic context, and (2) discrimination accuracy in the sentence condition was consistently low. These results suggest that categorical perception in a strict sense may not be evident in the perception of a sentence.

5aSCb46. Pronunciation of German suffixes by Japanese native speakers of different proficiency levels. Marino Kasuya and Takayuki Arai (Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku Tokyo 102-8554, Japan, mar3576ino_7@hotmail.co.jp)

This study investigates an aspect of speech rhythm in German spoken by Japanese native speakers of different proficiency levels. Previous studies on the production of vowel reduction have indicated that this is an area of difficulty for non-native speakers. One study, working on the assumption that second language (L2) speech production is affected by first language (L1), suggested that Japanese native speakers tend to fail at producing the required vowel reductions in unstressed syllables. The present study further investigated this issue by dividing Japanese native speakers into two groups: advanced and elementary learners. The aim of the present study was to investigate acoustic properties of vowel quality (first and second formants) and quantity (durational ratio) of unstressed syllables in German suffixes on the basis of German proficiency levels and the influence of L1. From results, main effect was obtained for the proficiency levels; acoustic analysis showed significant differences between first two formants and durational ratio same to be the factors that caused the difference among the levels. This suggests that L2 learning process may accompany the acquisition of L2 sounds even when rhythmic structures differ between L1 and L2. [Work supported by JSPS.]

5aSCb47. A comparative cross-linguistic study of vocal tract shaping in sibilant fricatives in English, Serbian and Mandarin using real-time magnetic resonance imaging. Li Hsuan Lu, Adam Lammert, Vikram Ramanarayanan, and Shrikanth Narayanan (Univ. of Southern California, 3771 McClintock Ave., 4013A, Los Angeles, CA 90089, lihsuanl@usc.edu)

An articulatory study of sibilant fricatives is described, with the goal of describing variability in lingual articulation across languages. Real-time magnetic resonance imaging (rtMRI) data were collected from three speakers each of English and Mandarin and two speakers of Serbian and reconstructed at a rate of 22.4 frames per second. Parallel acoustic data were also collected and subsequently denoised. Subjects spoke the segments /s/ and /S/ in symmetrical vowel contexts (e.g., "pa sap" for English and "asa" for Mandarin). Articulation was analyzed using a semi-polar grid overlaid on the image plane and midsagittal distance functions were obtained by measuring cross distances at ~0.5 cm intervals from the glottis to the lips. Analysis shows that place of articulation for /s/ is more anterior compared to /S/ across languages. Apical articulation is observed for /s/ across languages, while /S/ is produced laminally in English and apically in Serbian and Mandarin. Patterns of tongue shaping variability differ, as well, across languages. For instance, higher standard deviation is observed anterior to the place of articulation for /S/ in Mandarin, compared to Serbian and English. [Work supported by NIH.]

5aSCb48. Acoustic characteristics of glottalized obstruents in Gitksan. Michael D. Schwan (Dept. of Linguist, The Univ. of British Columbia, 2329 West Mall, Vancouver, BC V6T 1Z4, Canada, michael.d.schwan@gmail.com)

Glottalized obstruents are a defining feature in the phonetic inventory of languages of the Pacific Northwest. Gitksan (Tsimshianic), an endangered and understudied language in this region, is no exception. However, these segments, which have typically been labelled as ejectives by fieldworkers, have also been variously described as implosives or even as voiced ejectives. Evidently, the ability of fieldworkers to perceive these segments has proven difficult, even by those who have worked on the language for many years. This project seeks to describe some of the salient acoustic cues associated with glottalized obstruents in Gitksan by comparing glottalized and plain stops. While previous work has examined these stops only in word-initial position, the present study compares stops across positions within the word and across stressed and non-stressed environments. Speech tokens were collected from three fluent native speakers of three dialects of Gitksan in order to describe the prominent acoustic cues which characterize glottalized obstruents in Gitksan.

5aSCb49. Simulation of neural mechanism for Chinese vowel perception with neural network model. Chao-Min Wu, Ming-Hong Li, and Tao-Wei Wang (National Central Univ., #300, Chung-Da Rd., Chung-Li 32001, Taiwan, wucm@ee.ncu.edu.tw)

Based on the results of psycholinguistic experiments, the perceptual magnet effect is the important factor in speech development. This effect produced a warped auditory space to the corresponding phoneme. The purpose of this study was to develop a neural network model in simulation of speech perception. The neural network model with unsupervised learning was used to determine the phonetic categories of phoneme according to the formant frequencies of the vowels. The modified self-organizing map (SOM) algorithm was proposed to produce the auditory perceptual space of English vowels. Simulated results were compared with findings from psycholinguistic experiments, such as categorization of English /r/ and /l/ and prototype and non-prototype vowels, to indicate the model's ability to produce auditory perception space. In addition, this speech perception model was combined with the neural network model (Directions into Velocities Articulator, DIVA) to simulate categorization of ten English vowels and their productions to show the learning capability of speech perception and production. We further extended this modified DIVA model to show its capability to categorize six Chinese vowels (/a/, /i/, /u/, /e/, /o/, /y/) and their productions. Finally, this study proposed further development and related discussions for this speech perception model and its clinical application.

5aSCb50. Vowel onset marker based objective evaluation of Japanese phonemic length contrast produced by non-native speakers. Mee Sonu (Faculty of Sci. and Technol., Sophia Univ., 7-1 Kioi-Cho, Chiyoda-ku, Tokyo 102-8554, Japan, sonumeephonetic@gma1l.com), Yanlong Zhang (Global Information and Telecommunication Inst., Waseda Univ., Tokyo, Japan), Hiroaki Kato (National Inst. of Information and Commun. Technol., Kyoto, Japan), and Yoshinori Sagisaka (GITI/LASS Lab., Waseda Univ., Tokyo, Japan)

Aiming at building up an objective evaluation of second language (L2) learner's Japanese timing control characteristics, this study propose an objective measure to simulate a subjective measure of L2 speakers' production given by Japanese native evaluators from the view point of goodness of production. The focus here is on phonemic length contrast, e.g., /kako/ "the past" versus /kakko/ "parentheses" and /kaze/ "wind" versus /kaze:/ "taxation" which is difficult for L2 learners particularly when incorporated with speaking rates. The proposed objective measure uses a vowel onset time marker as a key perceptual and psychoacoustic marker to normalize speaking rate variations. The proposed new measure reflects tempo normalization between L2 learners by dividing the proficiency of mora-timing control in production. Results show that both vowel length contrast and consonant length contrast have a significantly higher correlation with the subjective evaluation score in which the coefficient was stable than using a simple duration difference measure. These results suggest that applying the psychoacoustic parameters would be effective to build up an objective evaluation of L2 learners. [Work supported by JSPS.]

5aSCb51. Canadian oats and Canadian goats: Comparing distal cues to segmentation and segments. Christopher C. Heffner (Neurosci. and Cognit. Sci., Univ. of Maryland, 1401 Marie Mount Hall, College Park, MD 20742, heffner@umd.edu) and Rochelle S. Newman (Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Distal timing cues (specifically, the speech rate of sentences temporally removed from a point of ambiguity in speech) have been shown to weakly modulate segmental perception [i.e., the perception of basic speech sounds, like "p" and "b"; Newman and Sawusch (1996)] but strongly affect the perception of word boundaries [i.e., the perception of separation between words in fluent speech; Dilley and Pitt (2010)]. However, no study as of yet has directly compared the two classes of percept using identical manipulations. In this study, we will examine the role of distal timing cues to segmental perception and word segmentation using the same distal contexts (e.g., "The merchant sold Canadian oats/notes," a word boundary distinction, or "The merchant sold Canadian coats/goats," a voicing-based segmental distinction). Distal speech rate will be artificially slowed to an identical extent for both types of contrast. We predict that categorical perception leads distal context effects to be much stronger on word segmentation ("…Canadian

oats/notes") than on segmental perception ("…Canadian coats/goats"). This research may help bridge speech perception theories that have been developed for each class of percept, and further clarify the role of distal cues in speech perception.

5aSCb52. The role of duration in regional U.S. vowel shifts. Valerie Fridland (English Dept., Univ. of Nevada, Reno, English Dept, UNR, MS 0098, Reno, NV 89557, fridland@unr.edu), Tyler Kendall, and Charlie Farrington (Dept. of Linguist, Univ. of Oregon, Eugene, OR)

Contemporary American dialect research has produced a rich body of literature on phonetic differences in vowel production. Many studies have focused on contrasting vowel shifts occurring in the speech of Northerners and Southerners. While such research clearly shows formant differences across dialects, the investigation of regionally variable phonetic cues beyond F1 and F2 has been under-addressed. The limited work that exists on sociolinguistic aspects of duration suggests that speakers do show durational differences across regions [Clopper *et al*. (2005), Jacewicz *et al*. (2007), Labov and Baranowski (2006), Tauberer and Evanini (2009)]. However, beyond these studies, there has been little research on the regional variability of vowel duration and on how phonemically non-distinctive durational differences found across regions may be related to the degree and type of spectral shift present in individuals' systems. In this paper we consider the extent to which duration corresponds to spectral differences in the major U.S. vowel shifts. In addition to looking at overall regional patterns, we examine whether durational differences co-vary with degree of individuals' participation in spectral shift. We discuss how duration, particularly of lax vowels, is related to the advancement of vowel shift features and whether such a relationship supports a physiological or grammatical explanation for duration.

5aSCb53. One small step for (a) man: Function word reduction and acoustic ambiguity. Laura Dilley, Melissa M. Baese-Berk, Stephanie Schmidt, Jesse Nagel (Dept. of Communicative Sci. and Disord., Michigan State Univ., Oyer Center, Michigan State Univ., East Lansing, MI 48824, mbaese@msu.edu), Tuuli Morrill (Dept. of Psych., Michigan State Univ., East Lansing, MI), and Mark Pitt (Dept. of Psych., The Ohio State Univ., Columbus, OH)

"That's one small step for man, one giant leap for mankind." Neil Armstrong insisted for years that his famous quote upon landing on the moon was misheard, and that he had said "one small step for a man." This controversy has continued, as examinations of the sound files of his transmission have yielded mixed opinions about whether he produced a. The disagreement stems partly from the fact that function words like a can be acoustically fleeting in casual speech, consist of just a few pitch periods, and be spectrally indistinguishable from the preceding context. As a result, they can be perceptually fragile, and easily disappear if the rate of surrounding speech varies [Dilley and Pitt, Psychological Science (2010)]. Here, we examine naturally produced, reduced tokens of for (spoken as "fer"), which were or were not followed by the word a from the Buckeye Speech Corpus, which consists of speakers from Mr. Armstrong's home state of Ohio. Comparison of the acoustic properties of the two sets of tokens will provide an indication of how similar they can be. Inclusion of Mr. Armstrong's production will assist in evaluating the likelihood of the function word being spoken. [Work supported by NSF Grant No. BCS-0847653.]

5aSCb54. Manipulating phonological boundaries using distributional cues. Eric Schreiber (Psychology, McGill Univ., 3701 Blackthorn Ct, Chevy Chase, Maryland 20815, eric.schreiber@mail.mcgill.ca), Kristine H. Onishi (Psychology, McGill Univ., Montreal, QC, Canada), and Meghan Clayards (School of Commun. Sci. and Disord. & Linguist, McGill Univ., Montreal, QC, Canada)

This study investigates the role of phonetic variability in category formation. The category boundary between /n/ and /m/ was manipulated in English-speaking adults. In Experiment 1 ($n = 24$), categories were indicated by lexical (novel word-object pairings) and distributional (within-category variability, category mean) information. During pre- and post-tests, listeners categorized stimuli from a 10-step re-synthesized continuum from "nado" to "mado." During training participants heard a bimodal distribution of tokens with one mode spanning 7 steps (wide) and one spanning 3 (narrow). For half the participants the wide category was "nado" and for the other half it was "mado." During training, tokens from each category were paired with different objects (socks or balls). A repeated-measures ANOVA showed that after training, categorization shifted toward the wide distribution [$F(1,22) = 10.259, p < 0.01$] and more so for ambiguous steps [$F(9,198) = 5.283$, $p < 0.001$]. In Experiment 2 ($n = 12$), auditory information was identical to experiment 1, but lexical information about the categories was removed (all training tokens referred to the same object). Listeners again shifted categorization toward the wide distribution [$F(1,10) = 7.686$, $p < 0.05$], indicating that distributional information alone was sufficient to change categorization behavior.

5aSCb55. C-V coarticulation in consonants with multiple lingual constrictions. Amanda L. Miller (Linguistics, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, amiller@ling.osu.edu)

C-V coarticulation in monosyllabic words containing initial click consonants and /i:/ vowels is investigated in Mangetti Dune !Xung with 114 fps lingual ultrasound and acoustic data collected using the CHAUSA method [Miller and Finch (2011)]. The 114 fps rate yields an image of the tongue every 9 ms (+/−4.5 ms). Vowels following clicks have three lingual gestures involving the tongue tip/blade (TT), tongue body (TB), and tongue root (TR). TT and TB constrictions carried over from the clicks merge into a single vowel constriction at consonant specific rates. The second formant (F2) distinguishes each word type through the vowel midpoint. In regression analyses, TBCL and TRCL best predict F2 for alveolar click initial words, while TTCL best predicts F2 for dental/palatal click initial words. The more open constriction is acoustically inert. In the palatal click initial word, both constrictions are equally close for some speakers, and the gestures undergo blending [Browman and Goldstein (1990)]. I argue that these patterns are prosodically controlled. Dental and palatal clicks have TB and TR gestures associated with the syllable onset. In alveolar clicks, the right edge of TB and TR gestures are aligned to the right edge of the first mora.

5aSCb56. Examining the extent of anticipatory coronal coarticulation: A long-term average spectrum analysis. Alexei Kochetov (Linguistics, Univ. of Toronto, 100 St George St., 4th Fl., Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca) and Chris Neufeld (Speech-Language Pathol., Univ. of Toronto, Toronto, ON, Canada)

Phonetic studies of English liquids /r/ and /l/ have shown these consonants can exert strong coarticulatory effects on both adjacent and non-adjacent vowels. The studies of long-range coarticulation, however, have so far been limited to British English and have not explicitly compared liquids to other coronal consonants. The current study investigated local and long-range effects of coronals /l/, /r/, and /d/ in Canadian English. 14 speakers were recorded reading the sentences "We thought it might be(a) ram/lamb/dam/ham" repeatedly. Long-term average spectra of five vowels preceding the target consonants were calculated and compared to baseline values. The results revealed significant differences between the coronal consonants and the control (/h/) in up to five preceding syllables. Significant differences among the three coronals, however, were limited to the immediately preceding vowel. The other vowels showed some differences between /r/ and /d/, but not between /l/ and the other coronals. The results show that long-range coarticulation can spread for up to five syllables and involve both liquid and non-liquid coronals. The spectral differences between the two liquids in Canadian English, however, may not be as robust as have been reported for British English.

5aSCb57. A preliminary ultrasound study of Nepali lingual articulations. Alexei Kochetov (Linguistics, Univ. of Toronto, 100 St. George St., Sidney Smith 4076, Toronto, ON M5S 3G3, Canada, al.kochetov@utoronto.ca), Marianne Pouplier (Institut für Phonetik und Sprachverarbeitung, Ludwig-Maximilians-Universität, Munich, Germany), and Sarah Truong (Linguistics, Univ. of Toronto, Toronto, ON, Canada)

Previous descriptive and phonetic works on Nepali provided conflicting accounts of place contrasts in coronal consonants. Specifically, apical stops were characterized as either retroflex or alveolar, while laminal affricates were described as either alveolar or palatal. Some of these works used static palatography, which shows the contact between the tongue and the palate,

but provides no information about the tongue shape for a given consonant or its dynamic properties. In this study we used ultrasound to image tongue shapes for various Nepali lingual consonants produced by a single native speaker of Brahmin dialect. The results showed that the speaker's apical stops were produced with a substantially raised tongue front and retracted tongue tip, as would be expected of retroflex articulations. Laminal affricates had the tongue shape similar to dental stops, yet with a somewhat retracted tongue tip, indicative of the alveolar constriction. Apicals that differed in laryngeal features (voiceless, voiced, aspirated, breathy) did not show systematic differences in the tongue shape, except for the voiced stop, which was somewhat less retracted. While limited to the single speaker, the results confirm and extend some previous observations about Nepali coronals as showing a 3-way place contrast among dentals, alveolars, and retroflexes.

FRIDAY MORNING, 7 JUNE 2013 — 511AD, 9:00 A.M. TO 12:20 P.M.

Session 5aUW

Underwater Acoustics and Signal Processing in Acoustics: Using Graphic Processing Units for Computationally Intensive Applications in Acoustic Modeling and Signal Processing

Paul Hursky, Cochair
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Lauri Savioja, Cochair
Dept. of Media Technol., Aalto Univ., P.O. Box 15500, Aalto FI-00076, Finland

Stan E. Dosso, Cochair
School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada

Invited Papers

9:00

5aUW1. Large-scale virtual acoustics simulation at audio rates using three dimensional finite difference time domain and multiple graphics processing units. Craig Webb and Alan Gray (Univ. of Edinburgh, 1/3 Flat 1, Drummond St., Edinburgh EH8 9TT, United Kingdom, C.J.Webb-2@sms.ed.ac.uk)

The computation of large-scale virtual acoustics using the 3D finite difference time domain (FDTD) is prohibitively computationally expensive, especially at high audio sample rates, when using traditional CPUs. In recent years the computer gaming industry has driven the development of extremely powerful graphics processing units (GPUs). Through specialized development and tuning we can exploit the highly parallel GPU architecture to make such FDTD computations feasible.This paper describes the simultaneous use of multiple NVIDIA GPUs to compute schemes containing over a billion grid points. We examine the use of asynchronous halo transfers between cards, to hide the latency involved in transferring data, and overall computation time is considered with respect to variation in the size of the partition layers. As hardware memory poses limitations on the size of the room to be rendered, we also investigate the use of single precision arithmetic.This allows twice the domain space, compared with double precision, but results in phase shifting of the output with possible audible artifact. Using these techniques, large-scale spaces of several thousand cubic meters can be computed at 44.1 kHz in a usable time frame, making their use in room acoustics rendering and auralization applications possible in the near future.

9:20

5aUW2. Adapting the minimum variance beamformer to a graphics processing unit for active sonar imaging systems. Jo Inge Buskenes (Dept. of Informatics, Univ. of Oslo, Søndre Mohagen 17, Frogner 2016, Norway, joibu@ifi.uio.no), Jon Petter Åsen (MI-Lab, Norwegian Univ. of Sci. and Technol., Oslo, Norway), Carl-Inge C. Nilsen, and Andreas Austeng (Dept. of Informatics, Univ. of Oslo, Oslo, Norway)

The MVDR beamformer has been shown to improve active sonar image quality compared to conventional methods. Unfortunately, it is also significantly more computationally expensive because a spatial covariance matrix must be estimated and inverted for each image pixel. We target this challenge by altering and mapping the MVDR beamformer to a GPU, and suggest three different solutions depending on the system size. For systems with relatively few channels, we suggest arithmetic optimizations for the estimation step, and show how a GPU can be used to yield image creation rates of more than 1 Mpx/s. For larger systems we show that frequency domain processing is preferable, as this promotes high processing rates at a negligible reduction in image quality. These GPU implementations consistently reduced the runtime by 2–3 orders of magnitude compared to our reference C and Matlab implementations. For even larger systems we suggest employing the LCA beamformer. It does not calculate a weightset, but merely computes the beamformer output for each of a predefined set of weights, and selects the one that best fulfils the MVDR criterion. The LCA creates images with a quality comparable to MVDR, and it is perfectly suited for a GPU.

9:40

5aUW3. Exploiting data parallelism and population Monte Carlo on massively-parallel architectures for geoacoustic inversion. Jan Dettmer, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3P6, Canada, jand@uvic.ca), and Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA)

Bayesian inference algorithms in geoacoustic inversion have high computational requirements on multiple computational scales. Predicting (modeling) data to match observations represents fine-grained computations which often cannot be implemented efficiently on CPU clusters since high latency and communication overhead outweigh parallelization gains. However, GPUs, which operate efficiently on 100,000s of parallel threads with low latency and high bandwidth, can provide significant performance gains. Bayesian sampling is generally coarse-grained, and can be implemented efficiently in parallel on multi-core/cluster architectures. For example, population Monte Carlo simulates many Markov chains in parallel, with chains running independently between interactions (at predefined intervals) which exchange information throughout the population, substantially increasing sampling efficiency. This paper combines fine- and coarse-grained parallelization to profoundly improve the efficiency of geoacoustic inversion of seabed reflection data. Spherical-wave reflection-coefficient predictions, which require solving the Sommerfeld integral for a large number of grazing angles and frequencies, constitute fine-grained, data-parallel computations which are implemented efficiently on a GPU. Sampling is based on population Monte Carlo simulation with chain interactions as exchange and crossover moves.The algorithm is applied to Malta Plateau data to study frequency dependence of velocity and attenuation in marine sediments. [Work supported by ONR.]

Contributed Papers

10:00

5aUW4. Efficient Bayesian multi-source localization using a graphics processing unit. Stan E. Dosso and Jan Dettmer (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper presents a highly efficient approach to matched-field localization of an unknown number of ocean acoustic sources employing a graphics processing unit (GPU) for massively parallel computations. A Bayesian formulation is developed in which the number, locations, and complex spectra (amplitudes and phases) of multiple sources, as well as noise variance at each frequency, are considered unknown random variables constrained by acoustic data and prior information. The number of sources is determined during an initial burn-in stage by minimizing the Bayesian information criterion using an efficient birth/death scheme. Marginal posterior probability distributions for source locations are then computed using Gibbs sampling. Source and noise spectra are sampled implicitly by applying analytic maximum-likelihood solutions in terms of the source locations (explicit parameters). This greatly reduces the dimensionality of the inversion, but requires solving a very large number (order 10^5) of complex matrix inversions for each sample of the explicit parameters. These inversions can be solved in parallel on a GPU, increasing efficiency by a factor of ~100. Examples are given of localizing a large number of sources (up to 10) in near real time.

10:20

5aUW5. Computing the singular value decomposition in parallel on graphics processing units using a one-sided Jacobi method. Michael V. Romer (Appl. Res. Labs. at the Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, romer@arlut.utexas.edu)

The singular value decomposition (SVD) provides a robust means of determining the dominant modes in a collection of signals that can then be used in adaptive beamforming to suppress loud interferers that would otherwise cover signals of interest. However, on serial architectures this decomposition can quickly become a bottleneck, hindering the real-time performance of the beamformer. Commercial off-the-shelf graphics processing units (GPUs) are an inexpensive means of adding parallel processing capabilities to a system and can reduce computation time by several orders of magnitude. The following work presents an algorithm that computes the full SVD of a dense matrix in parallel using a one-sided Jacobi method on a standard GPU. Both the runtime performance and relative accuracy of the algorithm are compared to the Intel Math Kernel Library (MKL) LAPACK implementation run on a CPU. Potential limitations to this approach due to restrictions imposed by the hardware are also discussed.

10:40

5aUW6. Interactive gpu-based sound auralization in dynamic scenes. Qi Mo, Micah Taylor, Anish Chandak, Christian Lauterbach, Carl Schissler, and Dinesh Manocha (Comput. Sci., Univ. of North Carolina at Chapel Hill, 201 S Columbia St., Chapel Hill, NC 27599, qmo@cs.unc.edu)

We present an auralization algorithm for interactive virtual environments with dynamic objects, sources, and listener. Our approach uses a modified image source method that computes propagation paths combining direct transmission, specular reflections, and edge diffractions up to a specified order. We use a novel multi-view raycasting algorithm for parallel computation of image sources on GPUs. Rays that intersect near diffracting edges are detected using barycentric coordinates and further propagated. In order to reduce the artifacts in audio rendering of dynamic scenes, we use a high order interpolation scheme that takes into account attenuation, cross-fading, and delay. The resulting system can perform auralization at interactive rates on a high-end PC with NVIDIA GTX 280 GPU with 2–3 orders of reflections and 1 order of diffraction. Overall, our approach can generate plausible sound rendering for game-like scenes with tens of thousands of triangles. We observe more than an order of magnitude improvement in computing propagation paths over prior techniques.

11:00

5aUW7. Some comments about graphic processing unit architectures applied to finite-difference time-domain room acoustics simulation: Present and future trends. Jose J. Lopez (ITEAM, Tech. Univ. of Valencia, Camino de Vera s/n, Valencia, Valencia 46022, Spain, jjlopez@dcom.upv.es), Juan M. Navarro (Telecommunications, Universidad Catolica de San Antonio, Guadalupe, Murcia, Spain), Diego Carnicero, and Jose Escolano (ITEAM, Tech. Univ. of Valencia, Valencia, Valencia, Spain)

The parallelization of the finite-difference time-domain (FDTD) method for room acoustic simulation using graphic processing units (GPUs) has been subject of study even prior to the introduction of general-purpose computing environments such as the CUDA architecture. Nowadays CUDA offers enough flexibility and processing power to obtain performance gains higher than 200 times compared to single-threaded CPU codes. In this paper, different aspects related to the implementation of FDTD in CUDA are analyzed; first, how the evolution of the different CUDA architectures affects implementations is inquired, paying special attention to the Kepler architecture, the latest available. Also performance increasing by using the different memory subsystems the GPU offers is discussed. Moreover, the performance in the use of the available computing power in the GPU is also analyzed together with the limiting factors such as memory consumption and computing time that prevent the simulation of large rooms at very high frequencies. Finally, some comments and ideas about the possible evolution

of the computing power of GPU in the next years are evaluated together with future possibilities that GPU architectures might bring to FDTD applied to room acoustics.

11:20

5aUW8. Efficient implementation of power-law attenuation of elastic waves in time-domain numerical simulations. David C. Calvo (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC, david.calvo@nrl.navy.mil) and Gaetano Canepa (Ctr. for Maritime Res. and Eng., La Spezia, Italy)

Numerical simulation of wave propagation in the time domain is easily parallelizable on high performance computing systems due to the spatially local nature of the governing equations. The disadvantage of working in the time domain arises when lossy media must be modeled which generally gives rise to convolution-type loss terms in the governing time-domain equations. Computation of these convolutions usually requires the storage of several solution fields at thousands of previous time steps. This requirement can be memory prohibitive in three-dimensions. In this talk we present a recursive convolution approach to computing lossy (power-law) elastic wave propagation that is an extension of the one-way, one-dimensional acoustic wave equation work done by Liebler [Liebler *et al*., J. Acoust. Soc. Am. **116** (2004)] in order to handle multiple dimensions and shear waves. Convolutions are computed recursively by first using a nonlinear least-squares technique to fit the kernel of the convolution with a series of decaying exponentials. We demonstrate how graphical processing units (GPUs) can be used to obtain speed-up factors as high as 35 on a test computation of time-domain scattering from a highly resonant but lossy elastic cylinder. [Work sponsored by the Office of Naval Research.]

11:40

5aUW9. The use of graphical processing unit processing in rough surface scattering. Ahmad T. Abawi and Paul Hursky (HLS Res., 3366 North Torrey Pines Court, La Jolla, CA 92037, abawi@hlsresearch.com)

The use of graphical processing units (GPU's) in scientific computation has drawn significant interest in recent years. In this paper we use GPU processing to evaluate the performance of a number of approximate techniques in computing scattering from two-dimensional rough surfaces by comparing their results with those obtained using the boundary element technique, which produces a numerically exact solution of the problem. To compute scattering from a two-dimensional surface, we use a technique that we developed for computing scattering from compact objects, which uses an analytical expression for scattering from a single, flat triangle. In this technique the surface is meshed using triangular patches and the scattering is computed as a coherent sum of scattering from individual triangles. This technique not only provides accurate evaluation of the surface integrals that appear in scattering theory, but it also lends itself easily to the benefits of GPU processing. We apply this technique to the Kirchhoff approximation, the small slope approximation and to a rather less familiar technique based on the work of Dashen *et al*. [J. Math. Phys.**32**, 986–996 (1991)].

12:00

5aUW10. Reverberation modeling on graphics processing units. Paul Hursky and Ahmad T. Abawi (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

We will discuss modeling reverberation on GPUs. An accompanying talk will discuss using GPUs to model scattering from rough surfaces. Here we discuss using GPUs to model the two sets of propagation paths, one set from the source to scatterer, the other set from the scatterer to the receiver, and to combine these two sets via a scattering operation and assemble the reverberation waveform at the receiver. We will discuss how we have adapted various aspects of our modeling to a GPU platform. GPUs provide several differentiated memory architectures that an application can exploit. For example, the texture memory provides hardware-assisted interpolation—as a result, we can load a 3D environment into texture memory, sparsely sampled, and then reconstruct interpolated slices as needed the modeling task. GPU architectures have been evolving for many years to meet the demands of computer gaming and rendering, applications most ambitiously served by ray tracing (of light). As a result, NVIDIA provides a ray tracing framework called OptiX, sufficiently general-purpose, that it can be linked with externally provided functions to specialize the mathematics implemented during the ray trace process. We will describe our work on adapting OptiX to underwater acoustic ray tracing.

FRIDAY AFTERNOON, 7 JUNE 2013 514ABC, 1:00 P.M. TO 2:40 P.M.

Session 5pPP

Psychological and Physiological Acoustics: Recent Trends in Psychoacoustics II

Hugo Fastl, Cochair
AG Technische Akustik, TU München, Arcisstr.21, München 80333, Germany

Sonoko Kuwano, Cochair
Osaka Univ., 2-24-1-1107 Shinsenri-Nishimachi, Toyonaka, Osaka 560-0083, Japan

Invited Papers

1:00

5pPP1. What can we learn from simulated acoustic environments? Bernhard U. Seeber (Audio Information Process., Technische Universität München, Arcisstrasse 21, Munich 80333, Germany, seeber@tum.de)

Psychoacoustic research has often used headphones to reproduce sound stimuli. Recently, the spatial dimension has regained attention in basic research and the talk will make a case for the importance of binaural hearing when assessing sound quality. Technological advances made it possible to accurately reproduce real and artificial sound stimuli with high spatial fidelity for their assessment. The Simulated Open Field Environment (SOFE) is a laboratory setup to reproduce sounds from multiple loudspeakers in an anechoic chamber. The free-field presentation allows participants to interact with sound stimuli in a natural way using head turns and movements - important when working with participants inexperienced with laboratory procedures. In connection with room simulation software the SOFE can also create acoustic scenes with multiple sources and sound reflections—thereby increasing the realism. I will give an overview of recent findings gained with the SOFE and their relation to evaluating sound quality.

1:20

5pPP2. A cognitive approach for binaural models using a top-down feedback structure. Jonas Braasch, Anthony Parks (Ctr. for Cognit., Commun., and Culture/Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, braasj@rpi.edu), and Jens Blauert (Inst. of Commun. Acoust., Ruhr-Univ. Bochum, Bochum, Germany)

Current models to explain binaural hearing generally focus on bottom-up processes of the auditory periphery and subcortical brain functions to simulate sound localization and other binaural tasks. While these models have been very successful in explaining a number of psychoacoustic phenomena, their architecture is not suitable to explain experiments that involve cognition. The project presented here seeks to close the gap between functional binaural models and research in applied robotics. A software architecture that was originally designed to simulate the process of music improvisation using a combination of Computational Auditory Scene Analysis, machine learning and logic-based reasoning, the Creative Artificially Intuitive and Reasoning Agent (CAIRA) was extended to simulate a number of basic binaural phenomena including sound localization of multiple-sources, resolving front/back confusions through strategic head movements, and adapting inhibitory parameters to the presented signals to evoke localization dominance. [Work supported by the National Science Foundation, No. 1002851.]

1:40

5pPP3. Continuous features of emotion in classical, popular, and game music. Masashi Yamada (Dept. of Media Informatics, Kanazawa Inst. of Technol., 3-1 Yatsukaho, Hakusan, Ishikawa 924-0838, Japan, m-yamada@neptune.kanazawa-it.ac.jp)

In the present study, musical emotion was measured continuously for classical, popular and game music. Eight pieces for each genre were used as stimuli. The following computer system was prepared: The screen showed a cursor on a two-dimensional plane spanned by valence (pleasant-unpleasant) and arousal axes. The position of the cursor was manipulated by a mouse and recorded every 0.5 s. In one session, the system continuously recorded the emotional responses of 20 listeners while listening to each piece. In the other session, the listeners listened to each piece without manipulating the mouse, and then pointed out the overall emotional feature of the piece on the two-dimensional plane. The results showed that the valence did not vary largely for any genre. The arousal varied largely for classical music, but not for game music. For popular music, it varied in between classical and game music. For all genres, the overall emotional values of valence and arousal showed almost equal to the maximum or minimum values in the continuous variation, respectively. This implies that the overall emotional features of a musical piece are almost entirely determined by the part which shows the highest degree of emotion in the piece.

2:00

5pPP4. Perceptual outcomes by rapid alternation of the resonant scaling and its relation to the fundamental frequency. Minoru Tsuzaki (Faculty of Music, Kyoto City Univ. of Arts, 13-6 Kutsukake-cho, Oe, Nishikyo-ku, Kyoto 610-1197, Japan, minoru.tsuzaki@kcua.ac.jp), Chihiro Takeshima (J. F. Oberlin Univ., Machida, Japan), Toshie Matsui (Dept. of Otorhinolaryngol., Nara Med. Univ., Kashihara, Japan), and Toshio Irino (Faculty of System Eng., Wakayama Univ., Wakayama, Japan)

Timbre provided by the resonant characteristics of the vibrating body can be represented as spectral envelope patterns and can contribute as one of the important cues for sound source identification. However, its concept is not strictly established while that of loudness, and of pitch are well known. Recently, the fact that the spectral pattern can be decomposed into two factors, i.e., the shape and size of the resonant body, has been reconsidered. Several psychophysical findings have successfully suggested that a "bottom-up" perceptual mechanism of the decomposition might be implemented. Manipulating the scaling factor of resonance can change the perceptual size of the sound source. By concatenating synthesized vowel segments whose resonant scale (RS) alternates between two values in an "ABA-ABA-" fashion, one can generate series of test stimuli for stream segregation with the galloping rhythm paradigm. The experimental results revealed that the RS factor could provide a reliable cue for streaming. As an extreme variation of this RS alternation, scale alternating wavelet sequences (SAWSs) have been proposed. In the SAWS, the RS alternates at every regular time grid. When the difference between the two RS factors exceeded a certain limit, perceived pitch shifted downwards by an octave.

Contributed Paper

2:20

5pPP5. Neuroacoustics: Study on the perception of stereo reverberant sound field at cortical level. Alejandro Bidondo (Ciencia y Tecnología, Universidad Nacional de Tres de Febrero, Av. De Los Constituyentes 1426, Villa Maipú. San Martín. Buenos Aires 1650, Argentina, abidondo@untref.edu.ar)

On the basis of the Ando's brain hemispheric specialization auditory model, spatial information is processed in the right hemisphere. When hearing a complex sound stimuli, like a monaural sound source reproduced in a reverberant sound field, several independent acoustic cues are processed in both hemispheres simultaneously. To study the brain specialization perceiving these types of sounds, it was developed the Auditory Evoked Potentials analysis for 2000 ms after the first 80 ms from the sound onset, even though the first 300 ms is normally analyzed, and Cortical Activity descriptors, which were applied to mismatch negativity electro-physiological signals taken from left and right hemispheres. It was possible to measure the specialization of hemispheres by using two different monaural and anechoic sound sources, one with a minimum effective duration of its autocorrelation function (tau e) as low as 0.2 ms and another with minimum tau e of 190 ms, both embedded into the same reverberant sound field and reproduced through headphones. This study opens the possibility to measure the perception of listener envelopment to further develop a subjective descriptor.

Session 5pSC

Speech Communication: Flow, Structure, and Acoustic Interactions During Voice Production II

Scott L. Thomson, Chair
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Contributed Papers

1:00

5pSC1. Quantification of the false vocal-folds effects on the intra-glottal pressures using large eddy simulation. Mihai Mihaescu (KTH Mekanik, Linné FLOW Ctr., Royal Inst. of Technol. (KTH), Osquars Backe 18, Stockholm 10044, Sweden, mihai@mech.kth.se), Sid M. Khosla (Otolaryngol. -Head and Neck Surgery, Univ. of Cincinnati-Medical Ctr., Cincinnati, OH), and Ephraim J. Gutmark (Aerosp. Systems, Univ. of Cincinnati, Cincinnati, OH)

During the closing phase of the phonation cycle the true vocal-folds (TVF) have a convergent-divergent shape. The negative pressures generated by the flow through the glottal passage are producing closing forces acting on the TVFs towards the center of the larynx. Intra-glottal pressures can affect both vocal-fold vibration and voice production, since they can accelerate the closing phase. This has a positive impact on the voice quality. Large Eddy Simulation approach is used to investigate the intra-glottal forces generated solely by the flow during the closing phase. The influence of the gap between the false vocal-folds (FVFs) and the location of FVFs with respect to the TVFs are analyzed. Based on anatomical measurements, four different gaps between the FVFs and two different distances between the true and false vocal-folds are investigated for the same trans-laryngeal pressure. The TVFs gap is kept constant. All cases exhibit a non-symmetric flow behavior in the mid-frontal plane. As compared with the baseline, significant negative pressures were found acting on the TVFs when the false glottal width is below a certain threshold value. The closing forces are increasing when the FVFs are located at larger distances further downstream from the TVFs.

1:20

5pSC2. Production of child-like vowels with nonlinear interaction of glottal flow and vocal tract resonances. Brad H. Story and Kate Bunton (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

Acoustically, the mechanisms of vocal sound production may be considered to exist along a continuum. At one end, the glottal flow wave is weakly coupled to the resonances of the vocal tract such that the output is a linear combination of their respective acoustic characteristics, whereas at the other end there is strong nonlinear coupling of the flow source to the vocal tract resonances. To express phonetic properties in the output, such as formants, the linear case requires that the source produce sound that is rich in harmonic or broadband energy. In contrast, the nonlinear case allows for the possibility of an harmonically-rich source signal to be generated even when the glottal area variation is so simple that it may contain only one harmonic (i.e., a sinusoid) [Titze, J. Acoust. Soc. Am. **123** (2008)]. The latter case is most likely to occur when the fundamental frequency is relatively high, such as in children's speech. The purpose of this study was to investigate the nonlinear end of the continuum with respect to the harmonic content of the glottal flow and pressure waveforms for vowels generated with a model of a child-like speech production system. [Research supported by NIH R01-DC011275, NSF BCS-1145011.]

1:40

5pSC3. Non-invasive *in vivo* measurement of the mechanical properties of human vocal fold tissue. Siavash Kazemirad, Hani Bakhshaei, Luc Mongeau (Dept. of Mech. Eng., McGill Univ., 817 Sherbrooke St. West, Montreal, QC H3A 0C3, Canada, siavash.kazemirad@mail.mcgill.ca), and Karen Kost (Dept. of Laryngology, McGill Univ., Montreal, QC, Canada)

The mechanical properties of the vocal fold mucosa have a great effect on the vocal folds oscillations, and voice quality. A non-invasive method was developed and examined to obtain the mechanical properties of human vocal fold tissue *in vivo* via measurements of the mucosal wave propagation speed during phonation. High speed and MRI images of three subject's vocal folds were captured while phonating at different pitches. The images obtained from these two techniques were matched and the dimensions of the vocal folds were obtained. The mucosal wave propagation speed was determined for the three subjects at different pitches through an automatic image processing procedure. The shear modulus of the subjects' vocal fold mucosa was then calculated using a surface (Rayleigh) wave propagation model and the measured wave speeds. This is revealed that the mucosal wave propagation speed and the shear modulus of the vocal fold tissue increased with the pitch. The results were in good agreement with those from other studies obtained via *in vitro* measurements, thereby supporting the validity of the proposed measurement method. This method offers the potential for *in vivo* clinical assessments.

2:00

5pSC4. Quantification of porcine vocal fold geometry in three dimensions. Kimberly A. Stevens (Mech. Eng., Brigham Young Univ., 435 Crabtree Bldg., Provo, UT 84602, kimst12@gmail.com), Marie E. Jette, Susan L. Thibeault (Div. of Otolaryngol. - Head and Neck Surgery, Univ. of Wisconsin–Madison, Madison, WI), and Scott L. Thomson (Mech. Eng., Brigham Young Univ., Provo, UT)

The geometry of the vocal folds, including the internal spatial distribution of the various tissue types, plays a central role in governing vocal fold flow-induced vibration. Quantifying the tissue geometries is therefore important for voice production research. In this presentation, quantitative measures of porcine vocal fold geometry obtained using three different techniques are reported and compared. MicroCT scans of three fresh porcine larynges were obtained with the folds in abducted and adducted positions. Adducted larynges were frozen, rescanned, cut in the midsagittal plane, and rescanned while frozen. The larynges were then thawed and the vocal folds were dissected. One vocal fold was frozen in optimal cutting temperature medium, serial-sectioned in the coronal plane, and imaged, and the other vocal fold was fixed in 10% formalin and embedded in paraffin before being sectioned and imaged. Geometrical data was obtained and compared between the different sets of histological images and the microCT images. This study provides quantitative measures of porcine vocal fold geometry and characterizes the influence of formalin fixation and histological processing on tissue deformation tissue across the three-dimensional vocal fold, leading the way for similar studies using human vocal folds.

2:20

5pSC5. Evaluation of contact pressure in human vocal folds during phonation using high-speed videoendoscopy, electroglottography, and magnetic resonance imaging. Zhe Li, Hani Bakhshaee (Mech. Eng., McGill Univ., #305 3650 de la montagne, Montreal, QC H3G2A8, Canada, zhe.li2@mail.mcgill.ca), Leah Helou (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA), Luc Mongeau (Mech. Eng., McGill Univ., Montreal, QC, Canada), Karen Kost (Otolaryngol. - Head and Neck Surgery, McGill Univ., Montreal, QC, Canada), Clark Rosen (Voice Ctr., Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA), and Katherine Verdolini (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

Mechanical stresses on the vocal folds surface during high pitch or amplitude phonation have been postulated to cause vocal fold damage. Models for the quantitative estimate of the contact pressure may be valuable for prevention and treatment. The objective of this study was to non-invasively estimate the contact pressure for different phonation types, amplitudes, and pitch in human subjects using concurrent high-speed videoendoscopy and eletroglottography. The edge velocities before and after contact were estimated from the analysis of consecutive digital images. Instantaneous contact areas were determined from electroglottography along with magnetic resonance image (MRI). The contact pressure was assessed using the impulse momentum form of Newton's second law. Quantitative verifications using silicone models were made. Investigations were then carried out in quantitative human subjects to compare contact pressures for three different voice types, pitches, and loudnesses. Contact pressures for breathy, normal, and pressed voice were obtained for fifteen female subjects. The results were verified through comparisons with values measured directly using a probe microphone. The proposed methods appear to be robust and accurate for contact pressure estimation.

2:40

5pSC6. Development of a small film sensor for the estimation of the contact pressure of artificial vocal folds. Sandra Weiss, Alexander Sutor, Stefan J. Rupitsch (Chair of Sensor Technol., Friedrich-Alexander-University Erlangen-Nuremberg, Paul-Gordan-Str. 3/5, Erlangen 91052, Germany, sandra.weiss@lse.e-technik.uni-erlangen.de), Stefan Kniesburges (Inst. of Process Machinery and Systems Eng., Friedrich-Alexander-Univ. Erlangen-Nuremberg, Erlangen, Germany), Michael Doellinger (Dept. of Phoniatrics and Paediatric Audiol., Univ. Hospital Erlangen, Erlangen, Germany), and Reinhard Lerch (Chair of Sensor Technol., Friedrich-Alexander-Univ. Erlangen-Nuremberg, Erlangen, Germany)

Since voice production is the precondition for speech and, therefore, verbal communication, people suffering from voice disorders are limited in their everyday life. Physiological voice disorders, like vocal fold nodules are understood to result from increased contact pressure of the vocal folds during vibration. Thus, measurement of arising contact pressure during vocal fold oscillation helps quantify the collision forces and to substantiate previous assumptions. Due to limited access to the vocal folds, there is need for a small sensor that minimally influences air flow. For that purpose, we developed a small film sensor which is capable of measuring the contact pressure along the vocal fold contact areas. To investigate the suitability of the sensor, a hemi-larynx flow channel experiment with silicone replicas of the vocal folds was performed. Replicas of different elasticities were investigated. The contact sensor signals as well as the subglottal pressure were acquired. High-speed recordings of the vocal fold oscillation were synchronously performed to accurately relate the opening, closing and contact phases to the time signals. The measured contact pressures will be reported and compared to previous results. The advantages and limitations of the developed sensor will be discussed.

3:00

5pSC7. Dynamics of supraglottal flow structures and sound generation in phonation. Stefan Kniesburges, Christina Hesselmann, Stefan Becker (Inst. of Process Machinery and Systems Eng., Univ. of Erlangen-Nuremberg, Cauerstrasse 4, Erlangen, Bavaria 91058, Germany, kn@ipat.uni-erlangen.de), and Michael Döllinger (Phoniatrics and Pediatric Audiol., Univ. Hospital Erlangen, Erlangen, Germany)

Within the fully coupled multi-physics process of phonation the fluid flow plays an important role in the sound production. Therefore, the study addresses phenomena in the flow downstream of synthetic self-oscillating vocal folds and their influence on the sound production. A test setup consisting of devices for producing and conditioning the flow including a test section was used. The supraglottal channel was developed to prevent acoustic coupling to the vocal folds. Hence, the oscillations were aerodynamically driven. The vocal folds consist of silicone rubber having homogeneous material distribution. The flow was visualized in the immediate supraglottal region using a laser-sheet technique and a highspeed camera. The flow shows asymmetric behavior in cases with channel. The glottal jet is bend to one side depending on the turbulent flow conditions in the channel. In cases without channel, the jet was stabilized by the constant ambient pressure being symmetrical at each instance during an oscillation cycle. Additionally, the acoustic response of the studied supraglottal cases was investigated in an anechoic chamber. The spectral analysis of the data turns out differences in tonal and broadband parts of the acoustic spectrum. The context to the visualized flow structures will be discussed.

3:20

5pSC8. Acoustic perturbation equations and Lighthill's acoustic analogy for the human phonation. Stefan Zoerner (Inst. of Mech. and Mechatronics, Vienna Univ. of Technol., Wiedner Hauptstrasse 8-10, Vienna 1040, Austria, stefan.zoerner@tuwien.ac.at), Petr Šidlof (Inst. of Thermomechanics, Acad. of Sci. of the Czech Republic, Liberec, Czech Republic), Andreas Hüppe, and Manfred Kaltenbacher (Inst. of Mech. and Mechatronics, Vienna Univ. of Technol., Vienna, Austria)

In speech, air is driven through the larynx by compression of the lungs. Thereby, air flows through the glottis which forces the vocal folds to oscillate which in turn results in a pulsating air flow. This air flow is the main source of the generated sound—the phonation. The acoustic wave then passes through the vocal tract, which acts as a filter modulating the propagated sound leaving the mouth. We model the fluid-structure-acoustic interaction with a so called hybrid approach. The air flow in the larynx, together with a prescribed vocal fold motion, is simulated with help of the open source solver OpenFOAM. Based on the resulting fluid field, acoustic source terms and the wave propagation is calculated within the finite element solver CFS++. Two methods are available to choose from, Lighthill's acoustic analogy and an aeroacoustic analogy based on a perturbation ansatz. Additionally, the simulation domain is extended by a realistic but geometrical fixed vocal tract and connected to a propagation region. The different acoustic approaches are compared, by analyzing the acoustic pressure in the glottis (source region) and outside the vocal tract. Moreover, to illustrate the effects of the vocal tract an alternative geometry is used for comparison.