# JASA

THE JOURNAL OF THE
ACOUSTICAL SOCIETY OF AMERICA


## 183rd Meeting
of the Acoustical Society of America

Vol. 152 • No. 4 • Pt. 2 of 2 • 10.2022

**GRAND HYATT NASHVILLE**

NASHVILLE, TENNESSEE
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183rd Meeting: Acoustical Society of America

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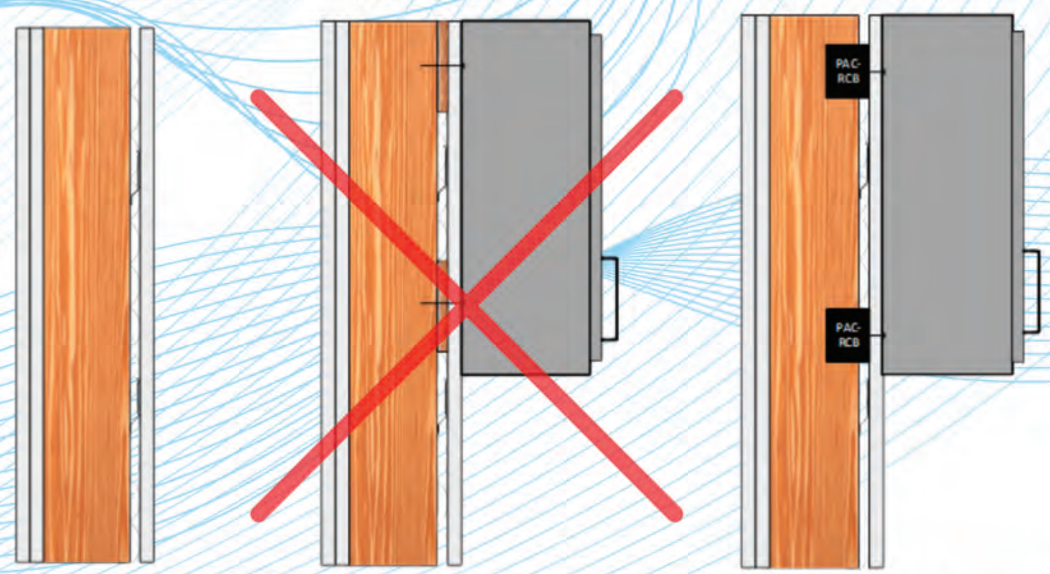


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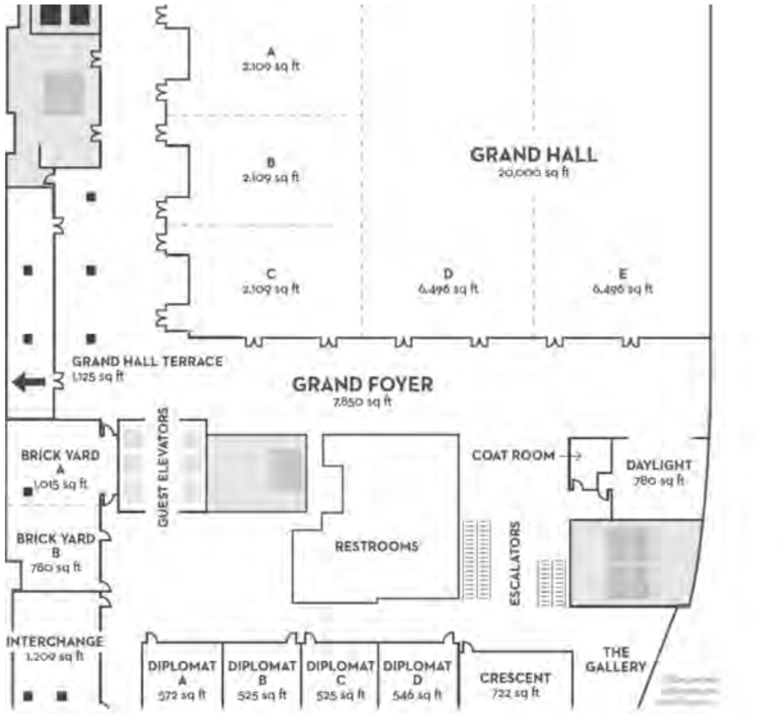


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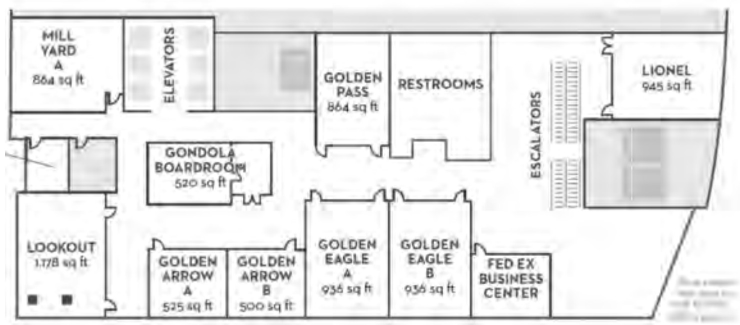


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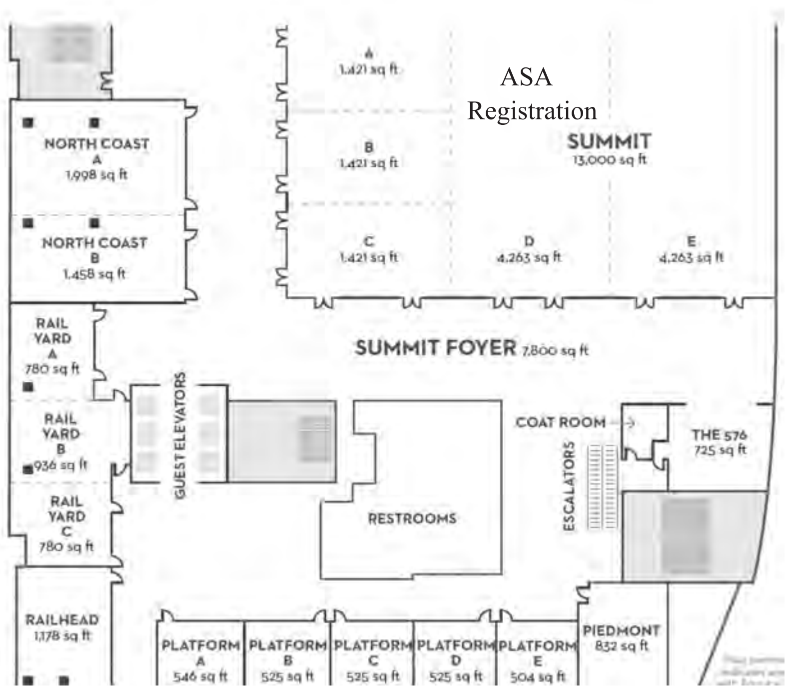


Level 4

# TECHNICAL PROGRAM CALENDAR
## 183rd Meeting of the Acoustical Society of America
## 5–9 December 2022

**Please refer to the Itinerary Planner and Mobile App for Updated Information**

**Monday Morning**

8:20 1aAA **Architectural Acoustics**: Acoustical Challenges in Small Rooms I. Summit A

9:00 1aAB **Animal Bioacoustics and Psychological and Physiological Acoustics**: Effects of COVID-19 on Animals and Soundscapes. Grand Hall A

8:00 1aAO **Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics**: Shelfbreak Acoustics I. North Coast A

8:45 1aBA **Biomedical Acoustics**: 150th Anniversary Celebration of Paul Langevin: Inventor of Modern Ultrasound (Hybrid Session). Mill Yard A

9:00 1aNS **Noise and Psychological and Physiological Acoustics**: Community Impacts Associated with Entertainment Sound. Summit B

8:00 1aPA **Physical Acoustics and Biomedical Acoustics**: Acoustofluidics. Golden  ass

9:00 1aSA **Structural Acoustics and Vibration**: Tunable Metamaterials. Golden Eagle B

9:00 1aSP **Signal Processing in Acoustics**: Machine Learning in Signal Processing. Rail Yard

**Monday Afternoon**

1:20 1pAA **Architectural Acoustics**: Acoustical Challenges in Small Rooms II. Summit A

1:00 1pAO **Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics**: Shelfbreak Acoustics II. North Coast A

1:00 1pBA **Biomedical Acoustics**: General Topics in Biomedical Acoustics I: Assessment of Tissue Material Properties. Mill Yard A

1:00 1pCA **Computational Acoustics, Biomedical Acoustics, and Signal Processing in Acoustics:** Physics-Informed Artificial Intelligence/Machine Learning for Acoustics (Hybrid Session). Summit C

1:00 1pEA **Engineering Acoustics**: General Topics in Engineering Acoustics. Lionel

1:00 1pMU **Musical Acoustics**: General Topics in Musical Acoustics I–Perception and Psychoacoustics. Rail Head

1:00 1pPA **Physical Acoustics**: General Topics Physical Acoustics I: Acoustic Manipulation and Atmospheric Propagation. Golden Pass

1:00 1pSC **Speech Communication**: Methods and Instrumentation in Speech (Poster Session). Summit E

1:30 1pSP **Signal Processing in Acoustics:** Signal Processing in Acoustics Poster Session. Summit E

4:00 1eID **Interdisciplinary**: Keynote Lecture. Grand Hall A

**Tuesday Morning**

8:00 2aAAa **Architectural Acoustics, Noise, and ASA Committee on Standards:** Sound Transmission and Impact Noise in Buildings. Summit A

9:00 2aAAb **Architectural Acoustics:** Student Design Competition (Poster Session). Summit E

8:30 2aAB **Animal Bioacoustics:** General Topics in Animal Bioacoustics I - Terrestrial. Grand Hall A

8:30 2aAO **Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics:** Acoustical Remote Sensing, Navigation, and Passive Monitoring in the Polar Ocean I North Coast A

8:30 2aBAa **Biomedical Acoustics, Computational Acoustics, and Signal Processing in Acoustics:** Deep Learning in Ultrasound Imaging and Tissue Characterization I. Lookout

8:30 2aBAb **Biomedical Acoustics:** Look How Big You've Gotten! A Story of Droplets and Ultrasound I. Mill Yard A

9:00 2aCA **Computational Acoustics and Biomedical Acoustics:** Numerical Approaches for Complex Media and Geometries I. Summit C

9:00 2aMU **Musical Acoustics:** General Topics in Musical Acoustics II–Sound Production and Radiation. Rail Head

8:00 2aNS **Noise and Psychological and Physiological Acoustics:** Methods for Community Noise Testing and Analysis I. Summit B

8:00 2aPAa **Physical Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics:** Effective Medium Theories in Acoustics. Golden Pass

9:15 2aPAb **Physical Acoustics and Structural Acoustics and Vibration:** Frontiers of Resonant Ultrasound Spectroscopy and Its Applications I. Golden Pass

9:00 2aPP **Psychological and Physiological Acoustics:** Cochlear Implants and Computation Grand Hall C

8:30 2aSA **Structural Acoustics and Vibration and Engineering Acoustics:** Additive Manufacturing and Acoustics. Golden Eagle B

9:00 2aSC **Speech Communication, Architectural Acoustics, Noise, and Psychological and Physiological Acoustics:** Acoustics and Communication in Healthcare Settings (Hybrid Session). Grand Hall B

8:00 2aSP **Signal Processing in Acoustics and Structural Acoustics and Vibration:** Active Control of Sound and Vibration (Hybrid Session). Rail Yard

8:00 2aUW **Underwater Acoustics and Acoustical Oceanography:** Mud Acoustics I. North Coast B

**Tuesday Afternoon**

1:00 2pAA **Architectural Acoustics, Musical Acoustics, and Signal Processing in Acoustics:** Recording Studios (Hybrid Session). Summit A

1:00 2pAB **Animal Bioacoustics:** General Topics in Animal Bioacoustics II – Marine. Grand Hall A

1:00 2pAO **Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics:** Acoustical Remote Sensing, Navigation, and Passive Monitoring in the Polar Ocean II. North Coast A

1:00 2pBAa **Biomedical Acoustics, Computational Acoustics, and Signal Processing in Acoustics:** Deep Learning in Ultrasound Imaging and Tissue Characterization II. Lookout

1:15 2pBAb **Biomedical Acoustics:** Look How Big You've Gotten! A Story of Droplets and Ultrasound II. Mill Yard A

3:40 2pBAc **Biomedical Acoustics:** Tribute to Fellows and Award Winners of the BATC–2022. Mill Yard A

1:00 2pCA **Computational Acoustics and Biomedical Acoustics:** Numerical Approaches for Complex Media Geometries II. Summit C

1:00 2pEA **Engineering Acoustics, Computational Acoustics, and Structural Acoustics and Vibration:** Automotive Acoustics (Hybrid Session). Lionel

1:00 2pID **Interdisciplinary and Student Council:** Graduate Programs in Acoustics Poster Session. Summit E

1:00 2pNS **Noise and Psychological and Physiological Acoustics:** Methods for Community Noise Testing and Analysis II. Summit B

1:00 2pPA **Physical Acoustics and Structural Acoustics and Vibration:** Frontiers of Resonant Ultrasound Spectroscopy and Its Applications II. Golden Pass

1:00 2pPP **Psychological and Physiological Acoustics:** Speech and Pitch Perception. Grand Hall C

1:30 2pSA **Structural Acoustics and Vibration and Computational Acoustics:** Surrogate and Reduced-Order Modeling for Structural Acoustics Applications. Golden Eagle B

1:00 2pSC **Speech Communication:** Healthcare Settings and Clinical Population (Poster Session) Summit E

1:00 2pSP **Signal Processing in Acoustics:** General Topics in Signal Processing. Rail Yard

1:00 2pUW **Underwater Acoustics and Acoustical Oceanography:** Mud Acoustics II. North Coast B

**Wednesday Morning**

9:00 3aAA **Architectural Acoustics, Noise, and Psychological and Physiological Acoustics:** Architectural Acoustics and Audio – Even Better Than the Real Thing I. Summit A

8:30 3aAO **Acoustical Oceanography:** Acoustical Sensing of Ocean Turbulence, Mixing, and Stratification (Hybrid Session). North Coast A

8:30 3aBA **Biomedical Acoustics:** Ultrasound Therapy in the Brain. Mill Yard A

8:00 3aCA **Computational Acoustics:** Learning and Stochastic Modeling in Computational Acoustics I. Summit C

8:15 3aNS **Noise and Psychological and Physiological Acoustics:** Acoustics Design of Indoor and Outdoor Firing Ranges, and Protection from High Level Impulse Noise. Summit B

8:00 3aPAa **Physical Acoustics and Computational Acoustics:** Infrasound I. Golden Pass

8:20 3aPAb **Physical Acoustics and Education in Acoustics:** My Favorite Homework Problems (Based on Measurements, Demonstrations, or Experimental Data). Lionel

8:00 3aSA **Structural Acoustics and Vibration and Computational Acoustics:** Acoustic Metamaterials. Golden Eagle B

8:00 3aSC **Speech Communication:** Speech Perception (Poster Session). Summit E

9:00 3aSP **Signal Processing in Acoustics:** Signal Processing for Musical Audio Production. Rail Yard

**Wednesday Afternoon**

1:00 3pAA **Architectural Acoustics, Noise, and Psychological and Physiological Acoustics**: Architectural Acoustics and Audio – Even Better Than the Real Thing II. Summit A

1:00 3pBA **Biomedical Acoustics**: Modern Image Quality Assessment. Mill Yard A

1:00 3pCA **Computational Acoustics**: Learning and Stochastic Modeling in Computational Acoustics II. Summit C

1:00 3pED **Education in Acoustics**: Acoustics Education Prize Lecture. Grand Hall C

2:15 3pID **Interdisciplinary**: Hot Topics in Acoustics. Grand Hall C

1:00 3pNS **Noise**: Topics on Noise: Noise Induced Hearing Loss and Community Noise. Summit B

1:00 3pPAa **Physical Acoustics and Computational Acoustics**: Infrasound II. Golden Pass

1:00 3pPAb **Physical Acoustics and Signal Processing in Acoustics**: Particle Velocity Sensing and Associated Signal Processing. Lionel

1:00 3pPP **Psychological and Physiological Acoustics**: Psychological and Physiological Acoustics Poster Session I. Summit E

1:00 3pSC **Speech Communication**: Topics in Speech Production. Grand Hall B

1:00 3pUW **Underwater Acoustics, Acoustical Oceanography, and Computational Acoustics**: Updating Ocean Acoustic Situational Awareness with In-situ Measurements. North Coast B

**Thursday Morning**

8:30 4aAA **Architectural Acoustics and ASA Committee on Standards:** Show Your Data: Architectural Acoustics Metrics. Summit A

8:00 4aAO **Acoustical Oceanography:** Topics in Acoustical Oceanography. North Coast A

9:00 4aBAa **Biomedical Acoustics:** Detection and Quantification of Bubble Act vity in Therapeutic Ultrasound I. Mill Yard A

9:00 4aBAb **Biomedical Acoustics and Signal Processing in Acoustics:** Novel Ultrasound Beamforming Techniques and Their Applications I. Lookout

9:00 4aMU **Musical Acoustics:** Modeling and Simulation of Physical Effects in Sound Reproduction. Rail Head

8:30 4aNS **Noise, Computational Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration:** Jet and Launch Vehicle Noise I (Hybrid Session). Summit B

8:00 4aPAa **Physical Acoustics and Biomedical Acoustics:** Interaction of Light and Sound I (Hybrid Session). Lionel

9:00 4aPAb **Physical Acoustics:** General Topics in Physical Acoustics II. Summit C

10:00 4aPP **Psychological and Physiological Acoustics:** Psychological and Physiological Acoustics Poster Session II. Summit E

9:00 4aSAa **Structural Acoustics and Vibration:** Fluid Structure Interaction. Golden Eagle B

10:30 4aSAb **Structural Acoustics and Vibration:** Analysis of Vibratory Systems. Golden Eagle B

8:40 4aSC **Speech Communication and Psychological and Physiological Acoustics:** Bilingualism and the Brain. Grand Hall B

8:30 4aSP **Signal Processing in Acoustics:** Dispersive Wave Signal Processing I. Rail Yard

8:00 4aUW **Underwater Acoustics and Acoustical Oceanography:** Memorial Session for Lisa Zurk I (Hybrid Session). North Coast B

**Thursday Afternoon**

1:30 4pBAa **Biomedical Acoustics and Signal Processing in Acoustics**: Novel Ultrasound Beamforming Techniques and Their Applications II. Lookout

1:00 4pBAb **Biomedical Acoustics**: Detection and Quantification of Bubble Act vity in Therapeutic Ultrasound II. Mill Yard A

1:00 4pCA **Computational Acoustics and Biomedical Acoustics**: Finite Difference Time Domain Methods Across Acoustics. Summit C

1:00 4pED **Education in Acoustics**: Connecting Industry and Education (Hybrid Session). Summit A

1:00 4pNS **Noise, Computational Acoustics, Physical Acoustics, Signal Processing in Acoustics, and Structural Acoustics and Vibration:** Jet and Launch Vehicle Noise II (Hybrid Session). Summit B

1:00 4pPAa **Physical Acoustics**: Physical Acoustics Best Student Paper Competition. Summit E

1:30 4pPAb **Physical Acoustics and Biomedical Acoustics**: Interaction of Light and Sound II (Hybrid Session). Lionel

1:00 4pSC **Speech Communication**: Bilingualism and Second Language Acquisition (Poster Session). Summit E

1:00 4pSP **Signal Processing in Acoustics**: Dispersive Wave Signal Processing II. Rail Yard

1:00 4pUW **Underwater Acoustics and Acoustical Oceanography**: Memorial Session for Lisa Zurk II (Hybrid Session). North Coast B

**Friday Morning**

9:00 5aAA **Architectural Acoustics and Noise:** Sound with Context: Cultural Heritage Acoustics Summit A

8:30 5aAB **Animal Bioacoustics**: Mapping Acoustic Features to Production Mechanisms in Speech and Animal Communication. Grand Hall A

9:15 5aBA **Biomedical Acoustics**: General Topics in Biomedical Acoustics II: Imaging and Therapeutics. Mill Yard A

8:00 5aPA **Physical Acoustics and Biomedical Acoustics**: Ultrasonic Assessment of Properties in Complex Materials I. Golden Eagle B

8:00 5aSC **Speech Communication**: Speech Production (Poster Session). Summit E

9:00 5aUW **Underwater Acoustics**: General Topics in Underwater Acoustics I. North Coast B

**Friday Afternoon**

1:00 5pPA **Physical Acoustics and Biomedical Acoustics**: Ultrasonic Assessment of Properties in Complex Materials II. Golden Eagle B

1:00 5pUW **Underwater Acoustics**: General Topics in Underwater Acoustics II. North Coast B

# SCHEDULE OF COMMITTEEE MEETINGS AND OTHER EVENTS

## ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

| | | |
|---|---|---|
| Sun, 4 Dec, 6:00 p.m. | Executive Council Dinner | Brickyard B |
| Mon, 5 Dec, 9:00 a.m. | Executive Council | Lookout |
| Mon, 5 Dec, 1:00 p.m. | Technical Council | Lookout |
| Tue, 6 Dec, 1:00 p.m. | Member Engagement | Diplomat C |
| Tue, 6 Dec, 7:00 a.m. | Revenue Reimagined | Diplomat B |
| Tue, 6 Dec, 5:00 p.m. | Women in Acoustics | Interchange |
| Wed, 7 Dec, 7:00 a.m. | Regional and Student Chapters | Piedmont |
| Wed, 7 Dec, 9:30 a.m. | Foundation Board | Diplomat C |
| Wed, 7 Dec, 11:30 a.m. | Public Relations | Piedmont |
| Thu, 8 Dec, 7:30 a.m. | Editorial Board | Interchange |
| Thu, 8 Dec, 1:00 p.m. | Investments | Piedmont |
| Fri, 9 Dec, 8:00 a.m. | Technical Council | Lookout |
| Fri, 9 Dec, 12:00 noon | Executive Council | Lookout |

## TECHNICAL COMMITTEEE OPEN MEETINGS

| | | |
|---|---|---|
| Tue, 6 Dec, 4:30 p.m. | Signal Processing in Acoustics | Rail Yard A |
| Tue, 6 Dec, 4:45 p.m. | Engineering Acoustics | Lionel |
| Tue, 6 Dec, 7:30 p.m. | Acoustical Oceanography | North Coast A |
| Tue, 6 Dec, 7:30 p.m. | Animal Bioacoustics | Grand Hall A |
| Tue, 6 Dec, 7:30 p.m. | Architectural Acoustics | Summit |
| Tue, 6 Dec, 7:30 p.m. | Physical Acoustics | Golden Pass |
| Tue, 6 Dec, 7:30 p.m. | Psychological and Physiological Acoustics | Grand Hall C |
| Tue, 6 Dec, 7:30 p.m. | Structural Acoustics and Vibration | Golden Eagle B |
| Wed, 7 Dec, 7:30 p.m. | Biomedical Acoustics | Mill Yard A |
| Thu, 8 Dec, 4:30 p.m. | Computational Acoustics | Summit C |
| Thu, 8 Dec, 7:30 p.m. | Musical Acoustics | Rail Head |
| Thu, 8 Dec, 7:30 p.m. | Noise | Summit B |
| Thu, 8 Dec, 7:30 p.m. | Speech Communication | Grand Hall B |
| Thu, 8 Dec, 7:30 p.m. | Underwater Acoustics | North Coast B |

## STANDARDS COMMITTEES AND WORKING GROUPS

| | | |
|---|---|---|
| Mon, 5 Dec, 1:00 p.m. | S12/WG11 Hearing Protector Attenuation and Performance | Piedmont |

## MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

| | | |
|---|---|---|
| Mon, 5 Dec 7:00 a.m.–5:00 p.m. Tue-Thu, 6–8 Dec 7:30 a.m.–5:00 p.m. Fri, 9 Dec 7:30 a.m.–1:00 p.m. | Registration | Summit D |
| Mon-Thu, 5–8 Dec 7:30 a.m.–5:00 p.m. Fri, 9 Dec, 7:30 a.m.–1:00 p.m. | Internet Zone | Summit D |
| Mon-Thu, 5–8 Dec, 7:00 a.m.–5:00 p.m. Fri, 9 Dec 7:00 a.m.–3:00 p.m. | A/V Preview | Diplomat A |
| Mon-Thu, 5–8 Dec, 8:00 a.m.–5:00 p.m. Fri, 9 Dec, 8:00 a.m.–3:00 p.m. | Mother's Room | Platform E |
| Mon-Fri, 5–9 Dec, 9:30 a.m.–11:00 a.m. | Coffee Breaks | Summit Foyer |
| Mon-Thu, 5–8 Dec, 8:00 a.m.–10:00 a.m. | Accompanying Persons | Platform D |
| Tue, 6-Dec, 4:00 p.m.–5:00 p.m. Wed, 7 Dec, 5:00 p.m.–7:00 p.m. | Listening Room | Platform D |
| Mon, 5 Dec. 4:00 p.m.–5:00 p.m. | Keynote Lecture | Grand A/B/C |
| Mon, 5 Dec, 5:30 p.m.–7:00 p.m. | Exhibit Opening Reception | Summit Foyer |
| Mon, 5 Dec, 5:00 p.m.–5:30 p.m. | First-Time Attendee Orientation | Grand C |
| Mon, 5 Dec, 5:30 p.m.–6:45 p.m. | Student Meet and Greet | North Coast B |
| Mon, 5 Dec. 7:00 p.m.–9:00 p.m. | Tutorial | Summit A |
| Tue, 6 Dec., 9:00 a.m.–5:00 p.m. | Exhibit | Summit Foyer |
| Tue, 6 Dec, 6:00 p.m.–7:30 p.m. | Social Hour | Grand Hall D/E |
| Wed, 7 Dec, 11:45 a.m.–1:30 p.m. | Women in Acoustics Luncheon | Grand Hall A |
| Wed, 7 Dec, 3:45 p.m.–5:00 p.m. | Plenary Session/Awards Ceremony | Grand Hall D |
| Wed, 7 Dec, 6:00 p.m.–8:00 p.m. | Student Reception | Grand Hall A |
| Wed, 7 Dec, 8:00 p.m.–12:00 midnight | ASA Jam | Grand Hall D |
| Thu, 8 Dec, 12:00 noon–2:00 p.m. | Society Luncheon and Lecture | Grand Hall D |
| Thu, 8 Dec, 6:00 p.m.–7:30 p.m. | Social Hour | Grand Hall D/E |

# 183rd Meeting of the Acoustical Society of America

The 183rd meeting of the Acoustical Society of America will be held Monday through Friday, 5–9 December 2022 at the Grand Hyatt Nashville Hotel, Nashville, Tennessee, USA.

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2. TRANSPORTATION AND TRAVEL
3. REGISTRATION
4. ACCESSIBLITY
5. TECHNICAL SESSIONS
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35. DATES OF FUTURE ASA MEETINGS

## 1. HOTEL INFORMATION

The Grand Hyatt Nashville is the headquarters hotel where all meeting events will be held.

The cut-off date for reserving rooms at special rates has passed. Please contact the Grand Hyatt (Grand Hyatt Nashville Hotel, 1000 Broadway, Nashville, Tennessee, United States, 37203; +1 615 622 1234; https://www.hyatt.com/en-US/hotel/tennessee/grand-hyatt-nashville/bnarn)

## 2. TRANSPORTATION AND TRAVEL

Ground transportation options include Taxis (From the airport, the meter starts at $9 and the rate is $2.50 per mile. There is a flat rate of $30 to the downtown area), Uber, Rental Cars, and Airport Shuttles & Ride Shares.

## 3. REGISTRATION

Registration is required for all attendees and accompanying persons. Registration badges must be worn in order to participate in technical sessions and other meeting activities.

Registration will open on Monday, 5 December, at 7:00 a.m. in Summit D (see floor plan on page A9).

Visa, MasterCard and American Express credit cards and checks in US dollars drawn on a bank in the US will be accepted for payment of registration. Meeting attendees who have pre-registered may pick up their badges and registration materials at the pre-registration desk.

The registration fees (in USD) are $775 for members of the Acoustical Society of America; $925 for non-members, $250 for Emeritus members (Emeritus status pre-approved by ASA), $425 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $200 for ASA Student members, $300 for students who are not members of ASA, $25 for Undergraduate Students, and $250 for accompanying persons.

One-day registration is available at $425 for members and $500 for nonmembers (one-day means attending the meeting on only one day either to present a paper and/or to attend sessions). A nonmember who pays the $925 nonmember registration fee and simultaneously applies for Associate Membership in the Acoustical Society of America will be given a $50 discount off their dues payment for 2023 dues.

Invited speakers who are members of the Acoustical Society of America are expected to pay the Member full-week or one-day registration fees. Nonmember invited speakers who participate in the meeting only on the day of their presentation may register without charge. The registration fee for nonmember invited speakers who wish to participate for more than one day is $400 and includes a one-year Associate Membership in the ASA upon completion of an application form.

**Special note to students who pre-registered online: You will also be required to show your student id card when picking-up your registration materials at the meeting.**

## 4. ACCESSIBLITY

If you have special accessibility requirements, please indicate this by informing ASA (1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300; asa@acousticalsociety.org) at a minimum of forty-fi e days in advance of the meeting. Please provide a cell phone number, email address, and detailed information including the nature of the special accessibility so that we may contact you directly.

## 5. TECHNICAL SESSIONS

The technical program includes 103 sessions with over 925 abstracts scheduled for presentation during the meeting.

A floor plan of the Grand Hyatt Nashville appears on page A9. Session Chairs have been instructed to adhere strictly to the printed time schedule, both to be fair to all speakers and to permit attendees to schedule moving from one session to another to hear specific papers. If an author is not present to deliver a lecture-style paper, the Session Chairs have been instructed either to call for additional discussion of papers already given or to declare a short recess so that subsequent papers are not given ahead of the designated times.

Several sessions are scheduled in poster format, with the display times indicated in the program schedule.

## 6. TECHNICAL SESSION DESIGNATIONS

**Abstract code examples: 1aAA1, 2pBAb4, 1eID1**

The first character is a number indicating the day the session will be held, as follows:

1-Monday, 5 December
2-Tuesday, 6 December
3-Wednesday, 7 December
4-Thursday, 8 December
5-Friday, 9 December

The second character is a lower case “a” for a.m., “p” for p.m., or “e” for evening corresponding to the time of day the session will take place. The third and fourth characters are capital letters indicating the primary Technical Committee that organized the session using the following abbreviations or codes:

AA Architectural Acoustics
AB Animal Bioacoustics
AO Acoustical Oceanography
BA Biomedical Acoustics
CA Computational Acoustics
EA Engineering Acoustics
ED Education in Acoustics
ID Interdisciplinary
MU Musical Acoustics
NS Noise
PA Physical Acoustics
PP Psychological and Physiological Acoustics
SA Structural Acoustics and Vibration
SC Speech Communication
SP Signal Processing in Acoustics
UW Underwater Acoustics

In sessions where the same group is the primary organizer of more than one session scheduled in the same morning or afternoon, a fifth character, either lower-case “a,” or “b,” is used to distinguish the sessions. Each paper within a session is identified by a paper number following the session-designating characters, in conventional manner. As hypothetical examples: paper 2pEA3 would be the third paper in a session on Tuesday afternoon organized by the Engineering Acoustics Technical Committee; 3pSAb5 would be the fifth paper in the second of two sessions on Wednesday afternoon sponsored by the Structural Acoustics and Vibration Technical Committee.

Note that technical sessions are listed both in the calendar and the body of the program in the numerical and alphabetical order of the session designations rather than the order of their starting times. For example, session 3aAA would be listed ahead of session 3aAO even if the latter session begins earlier in the same morning.

## 7. SOLUTIONS SHOWCASE

In an effort to provide industry members and supporting companies better visibility at the meetings, ASA is hosting the second Solutions Showcase in Nashville on Monday, 5 December, 1:00 p.m. to 5:30 p.m. in Summit B. This will be an opportunity to present a product, service, or solution in a setting similar to a technical session, but without restricting the commercial character of the talk.

## 8. EXHIBIT AND EXHIBIT OPENING RECEPTION

An instrument and equipment exhibition will be located in the Summit foyer on the 4th floor and will open on Monday, 5 December, with an evening reception serving a complimentary drink. Exhibit hours are Monday, 5 December, 5:30 p.m. to 7:00 p.m., Tuesday, 6 December, 9:00 a.m. to 5:00 p.m., and Wednesday, 7 December, 9:00 a.m. to 12:00 noon.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

## 9. TUTORIAL: EFFECTIVE MEDIA INTERACTIONS TRAINING WORKSHOP

The Public Relations Committee and the AIP Media Services team present this hands-on workshop for meeting attendees who are interested in effectively communicating scientific work to the public. The tutorial will be held on Monday evening, 5 December, 7:00 p.m. to 9:00 p.m. in Summit A. This workshop counts as the required media training for anyone interested in serving as a media liaison for their Technical Committee.

The workshop will consist of presentations by media professionals to provide a toolkit of techniques for engaging the media. The registration fee is USD $25 (USD $12 for students with current student IDs).

## 10. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 7 December, at 2:15 p.m. in Grand Hall C.

## 11. TECHNICAL TOUR TO BELMONT UNIVERSITY

A technical tour of Belmont University is scheduled to be held on Thursday, 8 December, 3:30 p.m. to 6:15 p.m. (including travel time). The tour participation fee is $10.00.

Located in the capital of music, Belmont University offers an impressive list of unique programs only available at a handful of colleges, including songwriting, music business, audio engineering, motion pictures, commercial music composition, and more. This technical tour will explore some of the world-class facilities at Belmont University.

## 12. ROSSING PRIZE IN ACOUSTICS EDUCATION AND ACOUSTICS EDUCATION PRIZE LECTURE

The 2022 Rossing Prize in Acoustics Education will be presented at the Nashville meeting during the Plenary Session on Wednesday, 7 December, in Grand Hall D.

Dr. Kathleen Wage will present the 2022 Acoustics Education Prize Lecture on Wednesday, 7 December, in session 3pED at 1:00 p.m. in Grand Hall C.

**13. TECHNICAL COMMITTEE OPEN MEETINGS**

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday. The schedule and rooms for each Committee meeting are given on page A14.

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 discussions.

**14. PLENARY SESSION AND AWARDS CEREMONY**

A plenary session will be held Wednesday, 7 December, at 3:30 p.m. in Grand Hall D.

ASA scholarship recipients will be introduced. The Rossing Prize in Acoustics Education, and the Gold Medal will be presented, Certificates will be presented to Fellows elected at the Denver meeting. See page A202 for a list of fellows and award recipients.

All attendees are welcome and encouraged to attend. Please join us to honor and congratulate these medalists and other award recipients.

**15. ANSI STANDARDS COMMITTEES**

Meetings of ANSI Accredited Standards Committees will not be held at the Nashville meeting.

People interested in attending and in becoming involved in working group activities should contact the ASA Standards Manager for further information about these groups, or about the ASA Standards Program in general, at the following address: Nancy Blair-DeLeon, ASA Standards Manager, Standards Secretariat, Acoustical Society of America, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300; T: 631-390-0215; E: asastds@acousticalsociety.org

**16. COFFEE BREAKS**

Morning coffee breaks will be held daily from 9:30 a.m. to 11:00 a.m. and an afternoon break will be held on Wednesday from 2:00 p.m. to 3:00 p.m. in the Summit Foyer.

**17. A/V PREVIEW ROOM**

Diplomat A will be set up as an A/V preview room for authors' convenience and will be available on Monday through Thursday from 7:00 a.m. to 5:00 p.m. and Friday from 7:00 a.m. to 3:00 p.m.

**18. INTERNET ZONE**

Wi-Fi will be available in all ASA meeting rooms and spaces.

Tables with power cords will be set up in the Summit D for attendees to gather and to power-up their electronic devices.

**19. MOTHERS ROOM**

A Mothers Toom for ASA meeting attendees will be available Monday to Friday, 5-9 December in Platform E. The hours are Monday to Thursday, 8:00 a.m. to 5:00 p.m. and Friday, 8:00 a.m. to 12:00 noon.

**20. SOCIALS**

Socials will be held on Tuesday and Thursday evenings, 6:00 p.m. to 7:30 p.m. in Grand Hall D/E.

The ASA hosts these social hours to provide a relaxing setting for meeting attendees to meet and mingle with their friends and colleagues as well as an opportunity for new members and first-time attendees to meet and introduce themselves to others in the field. A second goal of the socials is to provide a sufficient meal so that meeting attendees can attend the open meetings of Technical Committees that begin immediately after the socials.

**21. SOCIETY LUNCHEON AND LECTURE**

The Society Luncheon and Lecture, sponsored by the College of Fellows, will be held Thursday, 8 December, at 12:00 noon in Grand Hall D.

This luncheon is open to all attendees and their guests Purchase your tickets at the Registration Desk before 10:00 a.m. on Thursday, 8 December. The cost is USD $30.00 per ticket.

**22. STUDENT EVENTS: NEW STUDENTS/ FIRST-TIME ATTENDEE ORIENTATION, MEET AND GREET, STUDENT RECEPTION**

Follow the student twitter throughout the meeting @ ASAStudents.

A New Students/First-Time Attendee Orientation will be held on Monday, 5 December, from 5:00 p.m. to 5:30 p.m. in Grand Hall C. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in North Coast B where refreshments and a cash bar will be available.

The Students' Reception will be held on Wednesday, 7 December, from 6:00 p.m. to 8:00 p.m. in Grand Hall B. This reception, sponsored by Reality Labs Research, will provide an opportunity for students to meet informally with fellow students and other members of the Acoustical Society. All students are encouraged to attend, especially students who are first time attendees or those from smaller universities.

To encourage student participation, limited funds are available to defray partially the cost of travel expenses of students to attend Acoustical Society meetings. Instructions for applying for travel subsidies are given in the Call for Papers which can be found online at http://acousticalsociety.org. The deadline for the present meeting has passed but this information may be useful in the future.

**23. WOMEN IN ACOUSTICS LUNCHEON**

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 7 December, in Grand Hall A. Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Wednesday, 7 December. The fee is USD $35 for non-students and USD $15 for students.

**24. JAM SESSION**

You are invited to The JAM (at a location to be announced) on Wednesday night, 8 December, from 8:00 p.m. to midnight for the ASA Jam. Bring your axe, horn, sticks, voice, or anything else that makes music. Musicians and non-musicians are all welcome to attend. A full PA system, backline equipment, guitars, bass, keyboard, and drum set will be provided. All attendees will enjoy live music, a cash bar with snacks, and all-around good times. Don't miss out.

## 25. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Nashville meeting. The on-site registration fee for accompanying persons is USD $250. A hospitality room for accompanying persons will be open in Platform D, 8:00 a.m. to 10:00 a.m. Monday through Thursday. This entitles you access to the accompanying persons room, social events on Tuesday and Thursday, the Jam Session, and the Plenary Session on Wednesday afternoon.

## 26. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Nashville meeting will have a published proceedings, and submission is optional. The proceedings will be a separate volume of the online journal, "Proceedings of Meetings on Acoustics" (POMA). This is an open access journal, so that its articles are available in pdf format for downloading without charge to anyone in the world. Authors who are scheduled to present papers at the meeting are encouraged to prepare a suitable version in pdf format that will appear in POMA. It is not necessary to wait until after the meeting to submit one's paper to POMA. Further information regarding POMA can be found at the site http://acousticsauthors.org. Published papers from previous meeting can be seen at the site http://asadl/poma.

## 27. WEATHER

In December Nashville has an average high of 50°F and an average low of 33°F. There might also be a small chance of rain. Attendees should pack a warm jacket and umbrella.

## 28. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Michael Haberman, Technical Program Chair; Christopher Bassett, Acoustical Oceanography; Rolf Mueller, Animal Bioacoustics; Brandon Cuedquest, David Manley, Architectural Acoustics; Kang Kim, Libertario Demi, Biomedical Acoustics; Amanda Hanford, Computational Acoustics; Daniel Russell, Education in Acoustics; Thomas Blanford, Engineering Acoustics; Kurt Hoffman, Taffeta Elliott, Musical Acoustics; Aaron Vaughn, James Phillips, Hales Swift; Noise; Ralph Herman, Samuel Wallen, Physical Acoustics; Gregory Ellis, Psychological and Physiological Acoustics; Kai Gemba, Trevor Jerome, Signal Processing in Acoustics; Pasquale Bottalico, Matthew Masapollo, Benjamin Tucker, Speech Communication; Anthony Bonomo, Stephanie Konarski, Structural Acoustics and Vibration; David Dall'Osto, Underwater Acoustics; Zane Rusk, Student Council

## 29. MEETING ORGANIZING COMMITTEE

Veerle M. Keppens, Chair; Michael Haberman, Technical Program Chair

## 30. PHOTOGRAPHING AND RECORDING

Photographing and recording during regular sessions are not permitted without prior permission from the Acoustical Society.

## 31. ABSTRACT ERRATA

This meeting program is Part 2 of the November 2022 issue of *The Journal of the Acoustical Society of America*. Corrections, for printer's errors only, may be submitted for publication in the Errata section of the *Journal*.

## 32. GUIDELINES FOR ORAL PRESENTATIONS

**Preparation of Visual Aids**

- See the guidelines for computer projection in section 41 below.
- Allow at least one minute of your talk for each slide (e.g., PowerPoint). No more than 12 slides for a 15-minute talk (with 3 minutes for questions and answers).
- Minimize the number of lines of text on one visual aid. 12 lines of text should be a maximum. Include no more than 2 graphs/plots/figures on a single slide. Too little information is better than too much.
- Presentations should contain simple, legible text that is readable from the back of the room.
- Characters should be at least 0.25 inches (6.5 mm) in height to be legible when projected. A good rule of thumb is that text should be 20 point or larger (including labels in inserted graphics). Anything smaller is difficult to read.
- Make symbols at least 1/3 the height of a capital letter.
- For computer presentations, use all of the available screen area using landscape orientation with very thin margins. If your institutions logo must be included, place it at the bottom of the slide.
- Sans serif fonts (e.g., Arial, Calibri, and Helvetica) are much easier to read than serif fonts (e.g., Times New Roman) especially from afar. Avoid thin fonts (e.g., the horizontal bar of an e may be lost at low resolution thereby registering as a c.)
- Do not use underlining to emphasize text. It makes the text harder to read.
- All axes on figures should be labeled.
- No more than 3–5 major points per slide.
- Consistency across slides is desirable. Use the same background, font, font size, etc. across all slides.
- Use appropriate colors. Avoid complicated backgrounds and do not exceed four colors per slide. Backgrounds that change from dark to light and back again are difficult to read. Keep it simple.
- If using a dark background (dark blue works best), use white or yellow lettering. If you are preparing slides that may be printed to paper, a dark background is not appropriate.
- If using light backgrounds (white, off-white), use dark blue, dark brown or black lettering.
- DVDs should be in standard format.

**Presentation**

- Organize your talk with introduction, body, and summary or conclusion. Include only ideas, results, and concepts that can be explained in the allotted time. Four elements to include are:
- Statement of research problem
- Research methodology
- Review of results
- Conclusions
- No more than 3–5 key points can be covered adequately in a 15-minute talk so keep it concise.

- Rehearse your talk so you can confidently deliver it in the allotted time. Session Chairs have been instructed to adhere to the time schedule and to stop your presentation if you run over.
- An A/V preview room will be available for viewing computer presentations before your session starts. It is advisable to preview your presentation because in most cases you will be asked to load your presentation onto a computer which may have different software or a different configuration from your own computer.
- Arrive early enough so that you can meet the session chair, load your presentation on the computer provided, and familiarize yourself with the microphone, computer slide controls, laser pointer, and other equipment that you will use during your presentation. There will be many presenters loading their materials just prior to the session so it is very important that you check that all multi-media elements (e.g., sounds or videos) play accurately prior to the day of your session.
- Each time you display a visual aid the audience needs time to interpret it. Describe the abscissa, ordinate, units, and the legend for each figure. If the shape of a curve or some other feature is important, tell the audience what they should observe to grasp the point. They won't have time to figure it out for themselves. A popular myth is that a technical audience requires a lot of technical details. Less can be more.
- Turn off your cell phone prior to your talk and put it away from your body. Cell phones can interfere with the speakers and the wireless microphone.

## 33. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS

### Content

The poster should be centered around two or three key points supported by the title, figures, and text. The poster should be able to "stand alone." That is, it should be understandable even when you are not present to explain, discuss, and answer questions. This quality is highly desirable since you may not be present the entire time posters are on display, and when you are engaged in discussion with one person, others may want to study the poster without interrupting an ongoing dialogue.

- To meet the "stand alone" criteria, it is suggested that the poster include the following elements, as appropriate:
- Background
- Objective, purpose, or goal
- Hypotheses
- Methodology
- Results (including data, figures, or tables)
- Discussion
- Implications and future research
- References and Acknowledgment

### Design and layout

- A board approximately 8 ft. wide × 4 ft. high will be provided for the display of each poster. Supplies will be available for attaching the poster to the display board. Each board will be marked with an abstract number.
- Typically, posters are arranged from left to right and top to bottom. Numbering sections or placing arrows between sections can help guide the viewer through the poster.
- Centered at the top of the poster, include a section with the abstract number, paper title, and author names and affiliations. An institutional logo may be added. Keep the design simple and uncluttered. Avoid glossy paper.

### Lettering and text

- Font size for the title should be large (e.g., 70-point font)
- Font size for the main elements should be large enough to facilitate readability from 2 yards away (e.g., 32-point font). The font size for other elements, such as references, may be smaller (e.g., 20–24 point font).
- Sans serif fonts (e.g., Arial, Calibri, Helvetica) are much easier to read than serif fonts (e.g., Times New Roman).
- Text should be brief and presented in a bullet-point list as much as possible. Long paragraphs are difficult to read in a poster presentation setting.

### Visuals

- Graphs, photographs, and schematics should be large enough to see from 2 yards (e.g., 8 × 10 inches).
- Figure captions or bulleted annotation of major findings next to figures are essential. To ensure that all visual elements are "stand alone," axes should be labeled and all symbols should be explained.
- Tables should be used sparingly and presented in a simplified format.

### Presentation

- Prepare a brief oral summary of your poster and short answers to questions in advance.
- The presentation should cover the key points of the poster so that the audience can understand the main findings. Further details of the work should be left for discussion after the initial poster presentation.
- It is recommended that authors practice their poster presentation in front of colleagues before the meeting. Authors should request feedback about the oral presentation as well as poster content and layout.

### Other suggestions

- You may wish to prepare reduced-size copies of the poster (e.g., 8 1/2 × 11 sheets) to distribute to interested audience members.

## 34. GUIDELINES FOR USE OF COMPUTER PROJECTION

A PC computer with monaural audio playback capability and projector will be provided in each meeting room on which all authors who plan to use computer projection should load their presentations. Authors should bring computer presentations on a CD or USB drive to load onto the provided computer and should arrive at the meeting rooms at least 30 minutes before the start of their sessions. Assistance in loading presentations onto the computers will be provided. Note that only PC format will be supported so authors using Macs must save their presentations for projection in PC format. Also, authors who plan to play audio during their presentations should ensure that their sound files are also saved on the CD or USB drive.

**Introduction**

It is essential that each speaker who plans to use his/her own laptop connect to the computer projection system in the A/V preview room prior to session start time to verify that the presentation will work properly. Technical assistance is available in the A/V preview room at the meeting, but not in session rooms. Presenters whose computers fail to project for any reason will not be granted extra time.

**Guidelines**

- Set your computer's screen resolution to 1024x768 pixels or to the resolution indicated by the AV technical support. If it looks OK, it will look OK to your audience during your presentation.
- Remember that graphics can be animated or quickly toggled among several options: Comparisons between fi - ures may be made temporally rather than spatially.
- Animations often run more slowly on laptops connected to computer video projectors than when not so connected. Test the effectiveness of your animations before your assigned presentation time on a similar projection system (e.g., in the A/V preview room). Avoid real-time calculations in favor of pre-calculation and saving of images.
- If you will use your own laptop instead of the computer provided, connect your laptop to the projector during the question/answer period of the previous speaker. It is good protocol to initiate your slide show (e.g., run PowerPoint) immediately once connected, so the audience doesn't have to wait. If there are any problems, the session chair will endeavor to assist you, but it is your responsibility to ensure that the technical details have been worked out ahead of time.
- During the presentation have your laptop running with main power instead of using battery power to ensure that the laptop is running at full CPU speed. This will also guarantee that your laptop does not run out of power during your presentation.

**Specific Har ware Configuration**

**Macintosh**

Older Macs require a special adapter to connect the video output port to the standard 15-pin male DIN connector. Make sure you have one with you.

- Hook everything up before powering anything on. (Connect the computer to the RGB input on the projector).
- Turn the projector on and boot up the Macintosh. If this doesn't work immediately, you should make sure that your monitor resolution is set to 1024x768 for an XGA projector or at least 640x480 for an older VGA projector. (1024x768 will most always work.). You should also make sure that your monitor controls are set to mirroring.
- If it's an older PowerBook, it may not have video mirroring, but something called simulscan, which is essentially the same.
- Depending upon the vintage of your Mac, you may have to reboot once it is connected to the computer projector or switcher. Hint: you can reboot while connected to the computer projector in the A/V preview room in advance of your presentation, then put your computer to sleep. Macs thus booted will retain the memory of this connection when awakened from sleep.
- Depending upon the vintage of your system software, you may find that the default video mode is a side-by-side configuration of monitor windows (the test for this will be that you see no menus or cursor on your desktop; the cursor will slide from the projected image onto your laptop's screen as it is moved). Go to Control Panels, Monitors, configur - tion, and drag the larger window onto the smaller one. This produces a mirror-image of the projected image on your laptop's screen.
- Also depending upon your system software, either the Control Panels will automatically detect the video projector's resolution and frame rate, or you will have to set it manually. If it is not set at a commensurable resolution, the projector may not show an image. Experiment ahead of time with resolution and color depth settings in the A/V preview room (please don't waste valuable time adjusting the Control Panel settings during your allotted session time).

**PC**

- Make sure your computer has the standard female 15-pin DE-15 video output connector. Some computers require an adaptor.
- Once your computer is physically connected, you will need to toggle the video display on. Most PCS use either ALT-F5 or F6, as indicated by a little video monitor icon on the appropriate key. Some systems require more elaborate keystroke combinations to activate this feature. Verify your laptop's compatibility with the projector in the A/V preview room. Likewise, you may have to set your laptop's resolution and color depth via the monitor's Control Panel to match that of the projector, which settings you should verify prior to your session.

**Linux**

- Most Linux laptops have a function key marked CRT/LCD or two symbols representing computer versus projector. Often that key toggles on and off the VGA output of the computer, but in some cases, doing so will cause the computer to crash. One fix for this is to boot up the BIOS and look for a field marked CRT/LCD (or similar). This field can be set to Both, in which case the signal to the laptop is always presented to the VGA output jack on the back of the computer. Once connected to a computer projector, the signal will appear automatically, without toggling the function key. Once you get it working, don't touch it and it should continue to work, even after reboot.

## 35. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, or to obtain instructions for the preparation and submission of meeting abstracts, contact the Acoustical Society of America, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300; Telephone: 516-576-2360; E-mail: asa@acousticalsociety.org

184th Meeting, Chicago Illinois, 8-12 May 2023

185th Meeting, joint meeting with the Australian Acoustical Society, WESPAC, PRUAC, Sydney, Australia, 4-8 December 2023

189th Meeting, joint with the International Commission for Acoustics, New Orleans, Louisiana.

## Session 1aAA

## Architectural Acoustics: Acoustical Challenges in Small Rooms I

Joseph Keefe, Chair
*Ostergaard Acoustical Associates, 1460 US Highway 9 North, STE 209, Woodbridge, NJ 07095*

**Chair's Introduction—8:20**

### *Invited Papers*

**8:25**

**1aAA1. Recital Hall into existing structure limited height space.** David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Logan Pippitt (DLR Group, New York, NY)

Renovations for a college Fine Arts program desired to reincorporate multiple art departments that had become scattered over various locations throughout campus into an existing mid-century architecture legacy building. Included in the renovation program is a 130 seat Recital Hall, Instrumental Music Labs, and Commercial Music Recording program. The particular small rooms challenge presented is that the 130 seat Recital Hall must fit within an existing concrete waffle slab structure with a height of 17 feet from floor to bottom of structure and a total room volume of only 30,000 cubic feet. This presentation will walk through the challenges of providing an acceptable room acoustic environment for recitals amongst the other design challenges of ADA accessibility, sound separation to other arts programs, modern MEP services and requirements, and limitations of existing structural capacity.

**8:45**

**1aAA2. Podcast recording room design considerations and best practices.** Madeline Didier (Jaffe Holden, 114-A Washington St., Norwalk, CT 06854, mdidier@jaffeholden.com)

Podcast recording and listening has seen significant growth over recent years and continues to expand. The spaces used to record podcasts include everything from small closets to more sophisticated recording studios. There are not presently any known acoustic standards aimed specifically at the design of podcast studios. This paper discusses the question of what is required acoustically for a room to function successfully as a space for podcast recording. This is accomplished by examining an assortment of Jaffe Holden podcast studio projects, as well as existing standards for similar room types. Design considerations include background noise targets, sound isolation systems and performance targets, and room finishes and reverberation time targets. This paper also includes a discussion of how podcast studio acoustic design targets should be selected to optimize the functionality without overdesigning a space.

**9:05**

**1aAA3. Adjustable acoustics for imperfect rooms.** Richard L. Lenz (RealAcoustix LLC, 2361 B Ave., Ogden, UT 84401, RL@RealAcoustix.com)

Very often, acousticians are faced with room dimensions that are fixed and cannot be adjusted. This dilemma is exascerbated when the room is to be used for purposes of music performance. It is further complicated when the goal of the room is to present a broad spectrum of music types from classical symphony to modern jazz or rock. An example will be presented of an essentially square room space that was designed and treated to be adaptable to meet the requirements of all types of performances. Tabor College in Hillsboro, KS had space on its campus to build one concert that had to serve all of its performance needs. The physical space available, in order to maximize the size of the hall, also turned out to be a nearly perfect square in the audience area. Tom Ryan of The Hallani Group, working with the author and RealAcoustix, designed an acoustical system that allows the room to perform well for all types of music performances while being easy to change and architecturally beautiful. The design criteria, auralizations, and other information will be presented showing how this was accomplished.

**9:25**

**1aAA4. Is that all the space you've got?** Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, STE 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

A few case studies regarding small room architectural acoustics concerns are presented. These address problematic and unanticipated low-frequency room modes, constraints regarding vertical sound isolation in a tenant-space theater, and challenges pertaining to control of a chiller plant in commercial office space. Our approach to criteria, analyses, noise control recommendations, and lessons learned will be presented.

**9:45**

**1aAA5. Small band rooms for large bands: Balancing loudness and musicality.** Laura C. Brill (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, lbrill@thresholdacoustics.com) and Carl P. Giegold (Threshold Acoust. LLC, Evanston, IL)

When band rooms are smaller than ideal for the ensembles they support, there are safety concerns due to loudness. In the case study presented, students have experienced excessive loudness and difficult ensemble conditions due to inadequate ceiling height, absorption, and volume. This paper will discuss the strategies explored to increase acoustic volume to the extent possible within floorplate boundaries while finishing the space with a combination of absorption and diffusion. The goal of these interventions is to preserve ensemble and a sense of musicality to assure students and faculty they are working in a space designed for music rather than simply the control of loudness.

**10:05–10:20 Break**

**10:20**

**1aAA6. Two case studies in approach to the renovation of volumetrically challenged music rehearsal spaces.** Kaitlyn Hunt (Kirkegaard Assoc., 2101 CityWest Blvd., Ste 204, Houston, TX 77042-2830, khunt@kirkegaard.com) and Anthony Shou (Kirkegaard Assoc., Houston, TX)

A music rehearsal room renovation can be acoustically challenging in that there is often limited room volume, lack of loudness control, and budget constraints. Two case studies detail these challenges. (1) The University of North Texas' choral rehearsal room was due for renovation and with a volume of only 44,000 cubic feet, the room had to accommodate choral ensembles varying from 20–85 people for daily use, over 100 voices for an annual festival, and an added recital use. At an appropriate budget, the renovation used room shaping to simultaneously break up parallel surfaces and support voices while also adding adjustable acoustics to allow for tuning of the room for each rehearsal or performance. (2) The University of Texas at San Antonio's large band, orchestra, and jazz rehearsal room lacked loudness control even with an approximate volume of 73,500 cubic feet to serve 30 – 70 + musicians. At a much more limited budget, the solution was to flip the orientation of the musician layout in the room to take advantage of the existing room geometry and to introduce variable absorption stored exposed to the room which increased the overall absorption while also giving the ensembles an opportunity to tune the room.

**10:40**

**1aAA7. Undersized music ensemble rooms: Challenges and case study design approaches.** Benjamin E. Markham (Acentech, 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Music ensemble rooms require sufficient volume to develop adequate and appropriate reverberation, facilitate ensemble balance, and avoid excess loudness. Typical design practice begins with establishing minimum volume requirements based on expected ensemble sizes and types. When faced with inadequate room volume (e.g., due to existing building limitations, or otherwise), creative design solutions must be employed to achieve the best possible hearing conditions—and to avoid excess loudness in particular, which can be an occupational hazard for musicians. Case studies will illustrate a variety of design approaches: adding volume to existing spaces at the Crane School of Music (SUNY Potsdam), electronic reverberation enhancement (also at Crane), adding absorptive materials to an existing space and master-planning for larger spaces at Arkansas Tech, creative diffusive and absorptive material for the chorus rehearsal room at Boston Symphony Hall, and programming work to match ensembles to the most appropriate rooms available at a variety of educational institutions.

**11:00**

**1aAA8. Architectural improvements to increase reverberation and reduce flutter echo in two music rehearsal rooms.** Kent McKelvie (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Sudbury, MA 01776, kmckelvie@cavtocci.com) and Aaron Kanapesky (Cavanaugh Tocci Assoc., Sudbury, MA)

New music rehearsal rooms at a small college suffered from flutter echo and very short reverberation times. The architecture of these rooms included visually elegant designs which the college wanted to retain if feasible. We coordinated with the college, the architect, and a wood worker to develop concept ideas for improvement which would be cost effective and retain the elegant architectural. By using finite difference time domain (FDTD) simulation and mock-up listening tests, we evaluated the acoustic properties of these concept options. After completion of these tests and simulations, a design was implemented in each room based on acoustic performance and visual impact. We will present measurement results pre- and post-renovation, along with our methodology used for the FDTD analysis and mockup testing.

**11:20**

**1aAA9. Loudness control in a small volume Recital Hall.** Justin Yau (Kirkegaard, 2101 City West Blvd., Houston, TX 77042, justin.yh.yau@gmail.com) and Brian Corry (Kirkegaard Assoc., St. Louis, MO)

Mississippi State University's new music building features a 173-seat Lecture Hall that also serves as a recital hall. The limited volume presents an acoustical challenge to accommodate a variety of uses—most concerning are the louder and larger ensembles with up to 25 musicians and 50 singers sharing the stage. Due to project-specific constraints, the 24 foot stage-to-ceiling height and the 60,000 cubic foot volume are less than the early design-phase recommendations for the expected uses, which raises concerns about loudness buildup and inability to generate desired reverberation. Our goal is to implement variable loudness control systems that provide a useful reverberation range with good clarity across different settings. We identify and justify four loudness control strategies—maintain early, supportive reflections; shape the room to form patterns of sustained sound in areas with adjustable absorption; reduce high-frequency sustain in areas near audiences; avoid flutter echoes—and illustrate how those strategies were integrated into the architectural design

through room shaping, fixed wall treatments, and deployable curtains and banners. We discuss subjective and instrumented observations about the loudness (measured in Sound Strength G), reverberation time (measured in T30), clarity (measured in Clarity Index C80), and overall character of the room response.



**11:40**

**1aAA10. The Georgia Tech West Village Building Multipurpose Space: A case study of a higher education multipurpose space with conflicting programming to find a practical and cost-effective solution for end users.** Joseph F. Hackman (Special Technologies Group, Newcomb & Boyd, LLP, 800 Peachtree St. NE, Apt 8404, Atlanta, GA 30308, jhackman@newcomb-boyd.com)

This case study examines the challenges in architectural acoustics pertaining to small multipurpose rooms used for musical and instructional purposes as exhibited in the West Village Building on the Georgia Institute of Technology Atlanta Campus. The multipurpose space, while designed with variable acoustical elements, was challenging for the various user groups. This study aimed to find a solution to balance the acoustic, budgetary, and practical needs of this space. The goal of this study was to determine what configurations of existing and new material would improve functionality of the space. Solutions were created based on measured reverberation time of the space, and calculations based on properties of known materials and potential configurations. The findings were used to create a user-friendly guide for those who utilize the space, as well as an auditory demonstration. Most significantly, this study arrived at a solution that employed both existing and readily available materials, a modest budget, and maintained flexibility within the space.

MONDAY MORNING, 5 DECEMBER 2022 GRAND HALL A, 9:00 A.M. TO 10:05 A.M.

## Session 1aAB

## Animal Bioacoustics and Psychological and Physiological Acoustics: Effects of COVID-19 on Animals and Soundscapes

Kaitlin Palmer, Chair
*SMRU Consulting, 55 Water St., Vacouver, V6B 1A, Canada*

**Chair's Introduction—9:00**

### *Invited Papers*

**9:05**

**1aAB1. Mapping the intersection of helicopter routes and public lands in Northeast Scotland.** Nathan Wolek (Creative Arts, Stetson Univ., 421 N Woodland Blvd., Unit 8252, DeLand, FL 32723, nwolek@stetson.edu)

During the various COVID lockdowns, public narratives about the changing soundscape emphasized the reduction in commercial air traffic. However, other forms of air traffic were not reduced. Due to the need to service North Sea oil and gas operations, helicopter overflights from Aberdeen Airport (ABZ) remained a prominent feature of the local soundscape in Northeast Scotland. The primary route used by these helicopters (HMR WHISKEY) crosses public lands at several points, making the sound of overflights a prominent source of noise at these locations. These public lands include locations controlled by the city, like Seaton Park and Donmouth Local Nature Reserve, and others controlled by national entities, like the Forvie National Nature Reserve. This presentation will map several public lands in Northeast Scotland where noise from frequent helicopter overflights impacts the soundscape. The author will present field recordings and other documentation of helicopter overflights from February to June 2021, both during and immediately after the United Kingdom's third COVID lockdown, and invite discussion about potential for further study of noise pollution in this coastal region.

9:25

**1aAB2. Comparison of the marine soundscape before and during the COVID-19 pandemic in dolphin habitat in Sarasota Bay, FL.** Emma G. Longden (Sea Mammal Res. Unit, Univ. of St. Andrews, Scottish Oceans Inst., East Sands, St. Andrews, Fife KY16 8LB, United Kingdom, egl5@st-andrews.ac.uk), Douglas Gillespie (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, Scotland, United Kingdom), David Mann (Loggerhead Instruments, Inc., Sarasota, FL), Katherine A. McHugh (Chicago Zoological Society's Sarasota Dolphin Res. Program, Sarasota, FL), Athena M. Rycyk (Div. of Natural Sci., New College of Florida, Sarasota, FL), Randall Wells (Chicago Zoological Society's Sarasota Dolphin Res. Program, Sarasota, FL), and Peter L. Tyack (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, Fife, United Kingdom)

During the COVID-19 pandemic, decreases in large vessel activity and low-frequency noise have been reported globally. Sarasota Bay is home to a large and increasing number of recreational vessels, as well as a long-term resident community of bottlenose dolphins, *Tursiops truncatus*. We analyzed data from two hydrophones to compare the soundscape during the COVID-19 pandemic to previous years (March–May 2020 and 2018/2019). Hourly metrics were calculated: vessel passes, 95th percentile noise levels (125 and 16 kHz Third Octave Bands (TOBs) and two broadbands: 88–1122 Hz, 1781–17959 Hz), and dolphin whistle detection, to understand changes in vessel activity and the effect on wildlife. Vessel activity increased during COVID-19 restrictions by almost 80% at one site and remained the same at the other. Changes in noise levels varied between sites. Only the 125 Hz TOB and 88–1122 Hz band increased with vessel activity at both sites, suggesting this may be an appropriate measure of noise from small vessels in very shallow (<10 m) habitats. Dolphin whistle detection decreased during COVID-19 restrictions at one site but remained the same at the site that experienced increased vessel activity. Our results suggest that pandemic effects on wildlife should not be considered to be homogeneous globally.

9:45

**1aAB3. MonitorMyOcean.com: Measuring the acoustic impact of the COVID-19 lockdown on underwater ocean noise.** Artash Nath (MonitorMyOcean.com, King St., Toronto, ON, Canada, artash.nath@gmail.com)

Low-frequency noise from marine shipping is an underwater acoustic pollutant. The noise spectrum overlaps with frequencies marine mammals use to communicate, leading to stress and behavioural disruptions. This research established a model to measure the effects of anthropogenic activities on underwater noise. The COVID-19 lockdown led to a decline in marine traffic. The model quantified the reduction in noise levels before and during the lockdown in the Arctic, Atlantic, Pacific Oceans, and the Mediterranean Sea. Underwater ocean sound peaks between 10 and 100 Hz and is dominated by noise from shipping traffic. Hydrophones data from seven ocean observatories were analyzed at 1-Hz spectral and 1-minute temporal resolution. Power spectral densities were calculated, aggregated into monthly long-term spectral averages, and noise levels in the 63 Hz third-octave band were compared. The analysis revealed that global oceans quietened by an average of 4.5 dB during the lockdown period. The maximum decrease was at locations close to major shipping channels and cruise tourism destinations. The study proved that strategic "anthropauses" could reduce underwater noise levels. A web application MonitorMyOcean.com endorsed as a UN Ocean Decade Activity, was created to provide updated anthropogenic ocean noise levels.

## Session 1aAO

# Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics: Shelfbreak Acoustics I

Ying-Tsong Lin, Cochair
*Woods Hole Oceanographic Institution, 266 Woods Hole Road, Woods Hole, MA 02543*

Martin Siderius, Cochair
*Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201*

Andone C. Lavery, Cochair
*AOPE, Woods Hole Oceanographic Institution, 98 Water Street, Woods Hole, MA 02543*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**1aAO1. Between the devil and the deep blue sea—A journey towards shelfbreak acoustics (1982–2016).** James Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 86 Water St., Falmouth, MA 02543, jlynch@whoi.edu), Arthur E. Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), Glen Gawarkiewicz (Woods Hole Oceanogr. Inst., Woods Hole, MA), Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), Ying-Tsong Lin (Woods Hole Oceanogr. Inst., Woods Hole, MA), Keith von der Heydt, and John Kemp (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Falmouth, MA)

In this talk, we present a retrospective of the journey that both we and the community have taken in going from the more well-trodden areas of deep and shallow water acoustics to the arena of sloping shelves and the continental shelfbreak. We discuss the bathymetry, geology, oceanography, biology, and acoustics of these regions, and also the technology and science advances made during that time. A number of the major experiments that we participated in will be highlighted. A quick look at how things have evolved since 2016, and where they might be going will also be included.

**8:25**

**1aAO2. Shelfbreak processes affecting acoustic propagation in a changing ocean.** Glen Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanogr. Inst., M.S. #21, WHOI, Woods Hole, MA 02543, gleng@whoi.edu)

Fundamental physical oceanographic processes affecting sound speed fields and acoustic propagation have been studied in shelfbreak regions for some time. Integrated observations extending back to the 1990s have related acoustic propagation variability to basic thermohaline structure as well as physical oceanographic processes causing significant variability in sound speed fields. Key shelfbreak processes such as frontal wave propagation, eddy-Shelfbreak Jet interactions, and wind-driven motions will be described. The influence of ocean warming and climate change impacts on these processes will be outlined with emphasis on the Middle Atlantic Bight in the northeastern United States. The impacts of changes in both the atmospheric Jet Stream as well as the Gulf Stream on the Shelfbreak Front and Jet and implications for acoustic propagation will be presented.

**8:45**

**1aAO3. Geologic features at the continental shelf edge and their influence on acoustic propagation and scattering.** Jason Chaytor (U.S. Geological Survey, Woods Hole Coastal and Marine Sci. Ctr., Woods Hole, MA, jchaytor@usgs.gov)

The continental shelf edge, encompassing the seaward edge of the continental shelf and the upper continental slope, is a region of significant variation in the physical, chemical, and biological character of the ocean and seabed. The surficial and shallow sub-surface geologic features across this transition from the seaward-dipping shelf to the steeper upper slope reflect a spectrum of modern and antecedent constructional and destructional processes, each affecting acoustic propagation and scattering. Geologic features across the shelf edge are created by eustatic and regional sea level change, glacial processes, development and abandonment of fluvial pathways, and variations in slope stability. These geologic features are further modified by seasonal and persistent water column and bottom-water boundary layer physical processes, diagenetic modification, and biological activity supported by nutrient rich upwelling. Using examples from the southern New England shelf edge (SNESE), the scale, morphology, formative processes of shelf edge features and their impact on acoustic propagation and scattering will be presented. The SNESE has been shaped by a combination of glacial and interglacial processes resulting in a region with a complex seabed environment characterized by features such as variable surficial sediment, pockmarks, landslide scars, and submarine canyons. [Work supported by the Office of Naval Research.]

9:05

**1aAO4. Acoustic observations of nekton and zooplankton along the Northeast continental shelf break.** J. Michael Jech (NEFSC, 166 Water St., Woods Hole, MA 02543, michael.jech@noaa.gov)

Observations of fish and squid (nekton) and crustacean and gelatinous animals (zooplankton) at the shelf break using active acoustic systems (e.g., echosounders) have been made along the United States shelf break from the mid-Atlantic to New England since the early 2000s as part of a biennial bottom trawl survey (NOAA's Northeast Fisheries Science Center), since 2012 as part of the Atlantic Marine Assessment Program for Protected Species (AMAPPS), and recently as part of the Task Force Ocean New England Shelf Break Acoustic project. The shelf break can be a boundary between seabed-oriented species on the continental shelf and pelagic species in the open ocean, as well as transitional habitat for some species to exploit. These data have been collected using multifrequency, narrowband (e.g., 18, 38, 70, 120, and 200 kHz) and wideband (18–250 kHz) scientific echosounders with hull-mounted transducers to primarily map the prey species of mid- to upper-trophic level foragers and predators. Classification of acoustic data to biologically meaningful metrics and how these can be integrated with oceanographic features and spatial and temporal distributions of predators will be explored in relation to the broader goals of developing ocean acoustic networks. [Work supported by the Office of Naval Research.]

9:25

**1aAO5. Application of long-term passive acoustic monitoring to evaluate spatial and temporal patterns of multi-species acoustic assemblages of marine mammals in the western North Atlantic Ocean.** Samara Haver (Dept. of Fisheries, Wildlife, and Conservation Sci., Cooperative Inst. for Marine Ecosystem and Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, samara.haver@oregonstate.edu), Peter Corkeron (Anderson Cabot Ctr. for Ocean Life, New England Aquarium, Boston, MA), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Danielle Cholewiak, Genevieve Davis, Annamaria DeAngelis (Northeast Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, Woods Hole, MA), Kait Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Molly Martin, Nicole Pegg (Northeast Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, Woods Hole, MA), Natalie Posdaljian, Macey Rafter, Clara Schoenbeck, Alba Solsona Berga (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), Annabel Westell (Integrated Statistics, Inc., under contract to Northeast Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, Woods Hole, MA), and Sofie Van Parijs (Northeast Fisheries Sci. Ctr., National Marine Fisheries Service, National Oceanic and Atmospheric Administration, Woods Hole, MA)

Passive acoustic monitoring (PAM) is an efficient method for large-scale marine mammal monitoring. PAM technologies concurrently sample multiple soniferous marine mammal species, and when coupled with verified detectors, provide information that can be used to evaluate community composition. This analysis used data collected by an array of high-frequency PAM recorders at eight sites between 30°0'N and 42°0'N along the shelf-break in the western North Atlantic Ocean. Daily acoustic presence of 13 known marine mammal species and grouped unknown odontocetes was determined for all months between June 2016 and May 2019. Classification trees were generated from monthly summaries of daily detections to identify community composition dissimilarities. Detections of Gervais's beaked whale (*Mesoplodon europaeus*) represented the root node and split sites by latitude, grouping the three sites south of 34°0'N and the five sites north of 38°0'N. Tree nodes were further divided by other odontocetes. The presence of mysticetes also varied by site and season. Distinctive communities were identified for each site, with odontocetes being resident and mysticetes more migratory. The root and secondary splits were all driven by beaked whale species, demonstrating the importance of identifying these whales to species instead of aggregating them as is common for visual survey data.

## *Contributed Papers*

9:45

**1aAO6. Overview of the New England shelf break acoustics (NESBA) experiment.** Ying-Tsong Lin (Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, ytlin@whoi.edu), Jason Chaytor (U.S. Geological Survey, Woods Hole, MA), Brendan J. DeCourcy, Glen Gawarkiewicz (Woods Hole Oceanogr. Inst., Woods Hole, MA), J Michael Jech (NEFSC, Woods Hole, MA), Andone C. Lavery, Arthur E. Newhall (Woods Hole Oceanogr. Inst., Woods Hole, MA), Martin Siderius (Portland State Univ., Portland, OR), William L. Siegmann (Math Sci., RPI, Troy, NY), and Weifeng G. Zhang (Woods Hole Oceanogr. Inst., Woods Hole, MA)

A joint ocean acoustics experiment was conducting at the southern edge of the New England Shelf in spring 2021 to investigate the sound propagation and scattering effects of physical oceanographic (PO) processes, marine geological (GEO) features, and biology (BIO) in the shelfbreak area. The experiment consisted of a network of sound sources, hydrophone arrays, and physical oceanography, and biological survey equipment was established. The environmental processes of particular interest include shelfbreak fronts, thermohaline intrusions, shelf water streamers, and internal waves, along with seafloor depressions and slopes, submarine canyons, and variable seabed properties (GEO), as well as fish schooling and shoaling (BIO). With fixed propagation paths, acoustic fluctuations can be correlated with environmental variations within the observation network with a temporal and spatial scales ranging from minutes to weeks and up to 30 km. One of the primary experiment objectives was on adaptive sampling and tracking of acoustic sensitivity "hot spots" to improve the real-time joint ocean acoustics and circulation modeling that was conducted on a research vessel in the field. This talk will review the design concept of this ocean acoustic network and provide an overview the experiment results. [Work supported by the Office of Naval Research.]

**10:00–10:15 Break**

10:15

**1aAO7. Variations of acoustic ducting in the presence of gulf stream warm core rings at the New England shelfbreak.** Jennifer J. Johnson (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543, jjjohnson@whoi.edu), Ying-Tsong Lin, Glen Gawarkiewicz (Woods Hole Oceanogr. Inst., Woods Hole, MA), Brendan J. DeCourcy (Woods Hole Oceanogr. Inst., Falmouth, MA), and Arthur E. Newhall (Woods Hole Oceanogr. Inst., Woods Hole, MA)

The physical oceanographic (PO) conditions at the New England shelfbreak have become more dynamic with increased warming and meandering of the gulf stream. Consequently, underwater sound propagation in the area has also become more variable. The most profound PO feature that breaks off from Gulf Stream instabilities is warm core rings, which have complex interactions with bathymetry and seasonal shelf water masses. Warm core rings can cause shelf water streamers to extend offshore, resulting in surface and/or subsurface acoustic ducting conditions. This study observed acoustic surface ducting and breakdown conditions induced by a shelf water streamer at the New England shelfbreak. Ducting conditions were variable when a subsequent warm core ring intruded the acoustic source and receiver network, mixing with the shelf water streamer and influencing the sound propagation during and after the acoustic duct breakdown period. The field observations were also compared with a broadband numerical model. With increasing gulf stream warm core ring formations, streamer induced surface ducts are becoming more relevant in shelf break acoustics. [Work supported by the Office of Naval Research.]

10:30

**1aAO8. Sound propagation measurements using an autonomous underwater vehicle acoustic array in the New England shelf break acoustics network.** Natalie Kukshtel (Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543, nkukshtel@whoi.edu), Ying-Tsong Lin, Andone C. Lavery, Scott Loranger (Woods Hole Oceanogr. Inst., Woods Hole, MA), Jason Chaytor (U.S. Geological Survey, Woods Hole, MA), and Glen Gawarkiewicz (Woods Hole Oceanogr. Inst., Woods Hole, MA)

The New England shelf break is a highly dynamic region, which experiences complex spatial and temporal water-column variations due to interactions with warm core rings originating from the gulf stream. This widely varying ocean environment leads to changes in sound speed and acoustic propagation. Acoustic payload-equipped autonomous underwater vehicles (AUVs) are advantageous for sound propagation measurements in such environments due to their ability to detect changes with greater spatial resolution compared to fixed moorings. An AUV-towed acoustic array was tested and deployed in the New England Shelf Break Acoustics (NESBA) experiment in May 2021. The acoustic AUV system was comprised of a modified REMUS 600 vehicle, a hull-mounted 3.5 kHz transducer, and a towed multi-channel linear hydrophone array. The AUV sound source was tested at the Dodge Pond Naval Facility to characterize the effect of AUV body resonance, and the resulting calibration was incorporated into the data processing. Propagation paths between the AUV, acoustic moorings, and a ship-towed sound source were studied to investigate the acoustic effects of varying physical oceanographic conditions and biological scattering layers. These measurements also enabled investigation of the local seabed conditions and sub-bottom layering structure. [Work supported by the Office of Naval Research.]

10:45

**1aAO9. New England shelf break acoustic (NESBA) experiment: Seabed analysis.** Martin Siderius (Portland State Univ., 1600 SW 4th Ave., Ste. 160-14, Portland, OR 97201, siderius@pdx.edu), Bill Stevens (Portland State Univ., San Diego, CA), Alex Higgins, Drew Wendeborn (Portland State Univ., Portland, OR), Ying-Tsong Lin (Woods Hole Oceanogr. Inst., Woods Hole, MA), Brendan J. DeCourcy (Woods Hole Oceanogr. Inst., Falmouth, MA), and Jason Chaytor (U.S. Geological Survey, Woods Hole, MA)

The NESBA signals and noise experiment was conducted in April–May, 2021. The goal of the experiment was to assess the potential for improving sonar prediction through enhanced environmental awareness. In this talk, an overview of the experiment will be presented along with a description of the environmental conditions. Extensive measurements and modeling were used to help characterize the complex oceanography and seabed. The study region had water depths that varied from 70 m to 2000 m and included a diversity of seabed types and sub-bottom layering. Along the shelf break, there are regions where the seabed sub-bottom layering is particularly intricate. This talk will focus on the NESBA sub-goal of determining the properties of the seabed using passive sensing from two drifting, vertical hydrophone arrays. One array had 16 hydrophones spaced at 1 m and the other had 32 hydrophones with spacing of 0.1875 m. The seabed characterization is based on array processing techniques using the wind generated surface noise. Results will be presented for seabed characterization from several sites in the NESBA region with discussion on the potential impact on acoustic propagation. [Work supported by the Office of Naval Research.]

## *Invited Papers*

11:00

**1aAO10. Influence of stratified sub-bottom sediment layers on acoustic simulations in the New England shelf break environment.** Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu), Ying-Tsong Lin (Woods Hole Oceanogr. Inst., Woods Hole, MA), and Jason Chaytor (U.S. Geological Survey, Woods Hole, MA)

During the 2021 New England Shelf Break Acoustics (NESBA) Experiment, a large number of acoustic signal transmissions along and across the continental shelf were compared with combined physical oceanographic (PO) and ocean acoustic (OA) models. This experiment aimed to examine the influence of environmental changes in the water column on acoustic propagation in the shelf break environment. Sub-bottom surveys carried out by a combination of autonomous underwater vehicles (AUVs) and ship-board instruments revealed distinctly stratified sediment layers. Accounting for the properties of sub-bottom sediment layers is an integral component of diagnosing mismatch between experimental acoustic data and simulations, as these properties can influence the attenuation and scattering of received signals. A comparison of homogeneous half-space bottom models calculated at sea in May of 2021 and improved models which incorporate the results of sub-bottom surveys is presented, and the importance of this modeling consideration is examined for the purposes of data and model comparisons. [This work was supported by the Office of Naval Research.]

11:20

**1aAO11. High resolution mapping of sound speed and density with broadband echo sounders.** Scott Loranger (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 86 Water St., Woods Hole, MA 02543, sloranger@whoi.edu), Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., Falmouth, MA), Ying-Tsong Lin (Woods Hole Oceanogr. Inst., Woods Hole, MA), and Andone C. Lavery (AOPE, Woods Hole Oceanogr. Inst., Woods Hole, MA)

Advances in acoustic propagation and physical oceanographic modeling necessitate improved spatiotemporal resolution of water column properties (temperature, salinity, sound speed, and density) to inform and validate models. Shipboard profiling provides adequate vertical resolution, but high-resolution horizontal sampling is expensive and time consuming. Stationary arrays generate critical long-term data sets and high vertical resolution, however, there is a trade-off between dense horizontal sampling and total spatial coverage. Towed systems provide dense coverage horizontally but lack vertical resolution in a single location. During the ONR funded New England Shelf Break Acoustics experiments in 2021, sparse shipboard CTD and expendable bathythermograph (XBT) casts were combined with backscatter measurements by shipboard broadband echo sounders to map water column physical properties of the New England Shelf Break front with high resolution vertically and horizontally. The shipboard echo sounders detected backscatter from the strong gradients of the shelf break front, physical microstructure, and internal waves present at the frontal boundary. Acoustically tracking the front between CTD casts resulted in high resolution profiles of water column properties. Acoustic propagation modeling at 500 Hz and 4 kHz was compared when using the acoustically determined water column profile and when using profiles from CTD and XBT casts alone.

## *Contributed Paper*

11:40

**1aAO12. Assessment of passive bottom loss estimation methods in the New England shelf break area.** Tyler J. Flynn (The Johns Hopkins Univ. Appl. Phys. Lab., 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu), Gabriel P. Kniffin, Patrick Ferat, and Michael Mandelberg (The Johns Hopkins Univ. Appl. Phys. Lab., Laurel, MD)

Proper knowledge of geoacoustic bottom properties can be critical for accurate acoustic propagation predictions in the ocean, particularly in shallow environments. Unfortunately, these properties are unknown in much of the ocean. This knowledge gap presents a need for measurement techniques that enable the inference of critical bottom properties in a timely, cost effective manner. One such approach, proposed by Harrison and Simons, exploits the spatial structure of surface-generated, mid-frequency ambient ocean noise to infer bottom loss in a region as a function of bottom grazing angle. This method, known as the Up-Down Ratio, is fully passive and necessitates only a vertical line array with sufficient resolution for the frequencies of interest. A second approach, introduced by Buckingham, utilizes passive noise correlation between two vertically-separated hydrophones for similar purposes. In this presentation, these passive methods will be employed to estimate bottom parameters from mid-frequency (2–4 kHz) MACOS glider measurements collected within the New England Shelf Break Area as part of the Seabed 2021 joint experiment. In addition to comparisons of the inferred bottom parameters, discussion will include the ease of implementation of the methods as well as robustness against real data-collection challenges such as array tilt and strong surface interferers.

## Session 1aBA

## Biomedical Acoustics: 150th Anniversary Celebration of Paul Langevin: Inventor of Modern Ultrasound (Hybrid Session)

Thomas L. Szabo, Cochair
*Biomedical Engineering, Boston University, 44 Cummington Mall, Boston, MA 02215*

Francis Duck, Cochair
*None, No institutional address, Bath, BA1 3QF, United Kingdom*

**Chair's Introduction—8:45**

### *Invited Papers*

**8:50**

**1aBA1. Paul Langevin: His life and family.** Francis Duck (None, No institutional address, Bath BA1 3QF, United Kingdom, bathduckf@gmail.com) and Paul-Éric Langevin (None, Paris, France)

2022 marks the $150^{th}$ Anniversary of the birth of Paul Langevin, the originator of ultrasonics. He was born, lived, and died in Paris. His parents were of modest means. He was a humanist and a rationalist. Scientifically precocious, he was taught by Pierre Curie and spent a year at the Cavendish Laboratory, Cambridge. He gained a professorship at the ESPCI eventually becoming Director there. He also was Professor at the *Collége de France*, where his talent as a teacher gained wide recognition. He married Jeanne Desfosses in1898 and his children were as important to him as his science. They had two sons, Jean and André, both physicists, and two daughters Madeleine and Hélène. Paul-Gilbert, son of his scientific co-worker Eliane Montel, became a physicist and musicologist. In the interwar years he was an anti-Fascist activist. Imprisoned in 1940, he was then placed under house arrest in Troyes. After escaping to Switzerland in 1944, he returned to huge acclaim in liberated Paris. Elected as a Councilor for the Communist Party, he led the Committee for the Reorganization of Education in France, in spite of failing health. He died in 1946 and is interred in the Pantheon.

**9:10**

**1aBA2. Paul Langevin the Scientist.** Stefan Catheline (INSERM U1032, 151 Cours Albert Thomas, Lyon 69003, France, stefan.catheline@inserm.fr)

"Summarizing Paul Langevin's work implies revising the whole history of physics (…), this is a difficult but exciting task" Louis de Broglie said during Langevin's funeral. Paul Langevin was a universal mind, and he had worked in all branches of physics: gaseous ions, Brownian motion, electromagnetism, dia and para magnetism, birefringence, special relativity, ultrasonic waves, and neutron collisions. Very rare are those who can pretend to such a scientific impact. Yet at the same time, during a time span that included two world wars, Paul Langevin directed an engineering school ESPCI (Ecole Supérieure de Physique et Chimie Industrielle), gave courses at collège de France, ENS (Ecole Normale Supérieure), and was actively involved in the Solvay Conferences that he organized when Hendrick Lorentz passed away. The last part of the presentation is devoted to the present Paul Langevin legacy. As illustrations, a membrane wave experiment that revisits the 1919 Eddington observation of Albert Einstein prediction is followed by a walking liquid droplet set-up by Yves Couder. Open questions about the unfinished work that Paul Langevin and his scientific Solvay group left us will conclude this presentation.

**9:30**

**1aBA3. Langevin's ultrasonics.** Francis Duck (None, No institutional address, Bath BA1 3QF, United Kingdom, bathduckf@gmail.com)

Langevin's work in ultrasonics, which resulted in about twenty publications and patents, will be reviewed. At the beginning of WWI, encouraged by Marie Curie, he started work on a directional ultrasonic system to detect submarines and for underwater communications, based on Chilowski's proposal. Rejecting whistles, sirens and magnetic sources, he used a large mica 'singing' condenser for transmission, and proposed a large area carbon granule microphone as receiver. Noise problems with carbon receivers led Langevin, in early 1917, to test a single large slice of quartz, cut perpendicularly to one of its three electrical axes. This orientation differed from that used by Jacques and Pierre Curie's *quartz piezo-électrique*, in which the stress and electrical axes are perpendicular. Later that year he demonstrated quartz as a transmitter, creating the enabling technology for later ultrasonic developments. The knowledge was freely disseminated to other Allied laboratories. After the war, Langevin gave the first ever course on theoretical and practical ultrasonics. He facilitated the technological transfer of pulse-echo technology from military to civil sectors and the establishment of a laboratory in

Toulon for the certification of ultrasonic transducers. His co-workers went on to study magneto-strictive and high-frequency quartz transducers, therapeutic applications, and finite-amplitude propagation.

**9:50**

**1aBA4. Langevin's contributions to pulse echo piezoelectric transducers.** Thomas L. Szabo (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

Paul Langevin made fundamental discoveries which laid the foundations of modern pulse echo ultrasound over a hundred years ago. He was the first to employ piezoelectric crystals, X-cut quartz, with the propagation and electric axes aligned. In his design, the crystal thickness was a half-wavelength longitudinal wave resonator for both a transmitter and receiver which is the most popular configuration in use today. He invented the Langevin transducer which added two quarter wave layers on either side of the piezoelectric crystal to increase overall efficiency. A mosaic pattern embodied smaller pieces of scarce quartz. His original designs will be demonstrated through simulator models. By selecting ultrasound frequency of 40 KHz, he was able to make more directive beams and extend range for pulse echo detection. By adding triode amplifiers, the sensitivity of reception was increased greatly, and transmission powers of a kilowatt were achieved. With special circuits for making short pulses, long range, high resolution pulse echo ranging became a practical reality. Langevin's innovations led to sonar and eventually to ultrasound imaging.

**10:10–10:30 Break**

**10:30**

**1aBA5. Paul A. Langevin's contributions to nonlinear acoustics.** Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Paul A. Langevin made several important contributions to nonlinear acoustics, but our knowledge of these contributions is mainly due to accounts given by others, most notably Biquard (Ann. Phys., 1936). At the end of Chapter 1 in *Nonlinear Acoustics* (ASA, 2008), Blackstock summarizes Biquard's account, which is in turn based on lectures delivered by Langevin at the Collège de France in 1923. Langevin derived the coefficient of nonlinearity for liquids, predating the quantity subsequently expressed as $\beta = 1 + B/2A$, and provided measured values for several liquids. He also obtained an expression for the profile of a steady shock wave and described its subsequent decay, predating the classical Fay solution published in 1931. However, Langevin is most well known in nonlinear acoustics for a quantity that bears his name, the Langevin radiation pressure, and for what distinguishes it from the Rayleigh radiation pressure [Beyer (JASA, 1978)]. Rayleigh's result applies to a constrained field such as a standing wave in a closed tube, and it depends on the nonlinearity parameter $B/A$, whereas Langevin's result applies to an unconstrained field such as a sound beam, which is more common in practice, and it is independent of $B/A$.

**10:50**

**1aBA6. From sonar to medical ultrasound—The impact of Paul Langevin.** J. Brian Fowlkes (Radiology, Univ. of Michigan, 3226C Medical Sci. Bldg. I, 1301 Catherine St., Ann Arbor, MI 48109-5667, fowlkes@umich.edu)

The technical developments and discoveries of Paul Langevin figuratively and literally powered an incredible spectrum of scientific achievement. The quartz transducer technology of his 1918 patent (*Procédé et appareils d'émission et de réception des ondes élastiques sous-marines à l'aide des propriétés piézo-électriques du quartz. – French Patent No. 505,703)* enabled a level of acoustic output power generally unavailable at ultrasonic frequencies along with an increase in detection sensitivity. While there were disputes as to who should have been included on this patent, its effect on the field of ultrasonics is clear. The impetus for this work was largely high-power sonar for military application but his reported observations of effects on fish and even humans ushered in a new era of ultrasound bioeffects investigations including those of the famous Loomis Laboratory and numerous others. This is arguably the genesis of the rich tradition of research in ultrasound bioeffects and helped drive the pursuit of diagnostic and therapeutic applications of ultrasound including work by such famous researchers as the Fry Brothers. This presentation will review the evolution of bioeffects from the Langevin's work and its impact on the field of medical ultrasound.

**11:10**

**1aBA7. From Paul Langevin's quartz sensor to ultrasound metrology of today.** Peter A. Lewin (School of Biomedical Eng., Drexel Univ., 3141 Chestnut St., Philadelphia, PA 19104, lewinpa@drexel.edu)

This talk focuses on the remarkable impact of Paul Langevin's research since he (in 1917) proposed to use the piezoelectric properties of quartz for underwater detection of ultrasound signals. Not only has his pioneering idea opened the field of quantitative measurements of acoustic waves, but it also had far-reaching implications for the development of modern ultrasound metrology, driven by the need to ensure the safety of ultrasound used in visualizing of internal human organs for diagnostic applications and maximizing the efficacy and precision of treatment in therapeutic application of ultrasound waves. In summarizing the current advances in ultrasound metrology, a succinct background explaining why, initially, both the scientific community and industry were skeptical about the existence of the nonlinear (NL) wave propagation in tissue will be given and the design of an adequately wideband piezoelectric polymer hydrophone probe that was eventually used to verify that the 1–5 MHz probing wave then used in diagnostic ultrasound imaging was undergoing nonlinear distortion and generated harmonics in tissue will be discussed. [ *Work supported by the NIHNINR through Grant No. 5R01NR015995. The contents of this presentation are solely the responsibility of the authors and do not necessarily represent the official views of the NIH.]*

11:30

**1aBA8. Ultrasound in oceanography and seabed mapping.** Philippe Blondel (Phys., Univ. of Bath, Claverton Down, Bath, Avon and NE Somerset BA2 7AY, United Kingdom, pyspb@bath.ac.uk)

The scientific achievements of Paul Langevin are numerous and far-reaching. In the 150$^{th}$ anniversary of his birth, we look at one of his early contributions, forged in the heat and pressure of World War I. For most historians of science, Paul Langevin is the creator of sonars, and this opened a totally new window onto our world. In their first century of use, sonars quickly spread throughout the world, revealing the depths of the oceans, the complexities of marine life and the richness of oceanography. In the last decades, their capabilities were greatly augmented with the use of new materials and new techniques, from photographic techniques to computers and Artificial Intelligence. The early innovations of Paul Langevin enabled new fields of science, and this talk will explore the physics as well as the links between research, industry and governments that made these advances possible. François Rabelais, the French Renaissance humanist, wrote that "Science without conscience is only ruin of the soul," and the life of Paul Langevin fully illustrated this sentence. This first century of ultrasound in oceanography and seabed mapping has lessons for future efforts in research efforts and its applications throughout the world, especially as oceans are crucial to climate changes.

MONDAY MORNING, 5 DECEMBER 2022 SUMMIT B, 9:00 A.M. TO 12:00 NOON

## Session 1aNS

## Noise, Psychological and Physiological Acoustics and Architectural Acoustics: Community Impacts Associated with Entertainment Sound

Brandon Cudequest, Cochair
*Threshold Acoustics, 141 W Jackson Blvd. Suite 2080, Chicago, IL 60604*

David S. Woolworth, Cochair
*Roland, Woolworth & Associates, 356 County Road 102, Oxford, MS 38655-8604*

### *Invited Papers*

9:00

**1aNS1. A raceway's outdoor concerts get shut down by new residents.** Sarah Taubitz (45dB Acoust., LLC, PO BOX 1717, Buellton, CA 93427, st@45dB.com)

An existing raceway near a state highway desired to add concerts to its schedule. The oval racetrack (cars, motorcycles, etc.) has been in the community for decades, and more recently houses have been built within 700 feet. 45dB Acoustics acoustically modeled the concert situation, reporting that sound levels from concerts would be not unlike those from existing racetrack events. Nearby residents had signed their acknowledgement of potential noise from the racetrack, but some complained to the County about live music being "a new project and not an approved use of the facility." A Commercial Outdoor Entertainment License application was then required and granted by County Board of Supervisors, which included a public hearing at which the author was present to answer any Board questions. Although the License was granted, COVID put everything on hold, and some residents filed a lawsuit citing that the Board improperly granted an exemption of an Environmental Impact Review and California Environmental Quality Act review. After COVID restrictions lifted, the Board required the applicant to repeat the application process. Raceway ownership has not only abandoned the concerts, but also completely shut down the business, citing legal and consultant fees.

9:20

**1aNS2. Advocating the use of C weighted measurements in sound ordinances to manage modern music entertainment sound.** David S. Woolworth (Roland, Woolworth & Assoc., 356 County Rd. 102, Oxford, MS 38655-8604, dwoolworth@rwaconsultants.net)

Indoor and outdoor modern music entertainment since the 1980s has had incrementally more low frequency content with the advent of inexpensive, efficient, and portable subwoofers. Commercial sound reinforcement systems are being developed to reproduce lower frequencies and higher sound levels. Once sound leaves the venue, the low frequencies can travel the furthest, diffract around obstacles, and more easily penetrate building envelopes; in addition, it often comes in the form of impulses which are more easily detectable by the listener. Many ordinances only utilize A weighted sound metrics, which leaves enforcement at a loss to remedy low frequency

complaints using sound level violations. This paper will examine the sound spectra of different musical genres, modern sound systems, building envelope sound transmission, frequency based propagation, and sound measurement metrics, and the resulting need for and advantages of the use of C weighted measurements. The material is intended to provide a comprehensive way to explain the benefits of C weighted limits and measurements to stakeholders in ordinance development and enforcement.

**9:40**

**1aNS3. Can rock and roll be a good neighbor: Auralizing the thump next door.** John T. Strong (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, jstrong@thresholdacoustics.com), Brandon Cudequest, and Dawn Schuette (Threshold Acoust., Chicago, IL)

Downtown Omaha is undergoing significant revitalization to include a 3,000-capacity live music venue. While myriad event types are expected in this new venue, a substantial portion are expected to be high-energy, high-volume musical performances such as rock/metal, country, hip hop, and electronic dance music. This new building is sited within one hundred feet of condominiums, apartments, and a hotel. The overall area is zoned as a central business district and has a permissive policy for decibel limits at neighboring properties. This paper will discuss how the design team helped project stakeholders create a good neighbor policy through the use of auralizations and how those criteria informed the enclosure design and sound isolation strategies for the venue.

**10:00**

**1aNS4. Preventing, predicting, and simulating the noise impact of one indoor music venue on another through a large glass wall.** Adam Paiva (Jaffe Holden, 41 Lafayette Ave., Kingston, NY 12401, adam.paiva@gmail.com)

Jaffe Holden has worked on a confidential new project in Cambridge, MA in a proposed new building which will feature a 2-venue performing arts center in the lower floors of the building. The PAC includes a multi-use theater for music, speech, and dance with a large upstage wall made predominantly of a glass curtain-wall system. On the opposite side of the glass is an atrium with a second smaller venue that will also feature loud music programming. The client would like to allow as much flexibility in programming for both spaces as possible, while avoiding acoustic impact between the two spaces. As acoustical consultants on the project, Jaffe Holden modeled different glazing systems and provided recommendations to achieve the project goals. We used our acoustic lab to simulate the effect of different curtain-wall designs and played back the demo to the client to provide a subjective listening evaluation in addition to computational decibel predictions. This paper will study the acoustical challenges with this pairing, the challenges in acoustically modeling the performance of the curtain-wall, the architectural/constructability/budget challenges associated with the curtain-wall, and the lab demo simulation that was used to help the client make design decisions.

**10:20–10:35 Break**

**10:35**

**1aNS5. Soundscape planning for mixed use urban communities.** Gary W. Siebein (Siebein Assoc.,Inc., 625 NW 60TH St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Keely M. Siebein, Gary Siebein Jr., Marylin Roa, Jennifer Miller, and Matthew Vetterick (Siebein Assoc., Inc., Gainesville, FL)

Mixed use urban communities with a variety of retail, dining, entertainment, commercial, and residential occupants on one building or within several blocks of each other often bring a vitality to cities and towns. Careful planning and design are needed for the different uses to maintain sonic compatibility with each other over a 24-hour day. There are acoustical challenges in measuring amplified entertainment sounds due to the rapid succession of individual notes and words in pieces of music or trains of speech. Sounds of vehicles with straight pipes revving engines as they drive through a main street is another acoustical challenge. The enclosed walls of glass, metal and masonry of high-rise building create sound reflections and reverberation similar to what occurs in closed rooms reducing the decay of sounds with distance. Soundscape measurements, computer model studies of urban sound flows, implementing acoustical strategies as part of a design and planning process and addressing the challenges of rapidly time varying sounds measured by LAeq's are among the methods that are often implemented to address these issues. A case study of acoustical design for a revitalizing urban area is presented with its challenges, design processes, and initial strategies discussed.

## *Contributed Papers*

**10:55**

**1aNS6. Design and optimization of sound reinforcement systems for outdoor venues to minimize community noise impact.** Gino Pellicano (none, 853 Warwick Dr., Thousand Oaks, CA 91360, gino.pellicano@gmail.com) and Scott Sugden (none, Westlake Village, CA)

Outdoor entertainment venues are often constructed adjacent to residential communities, resulting in disruptive noise pollution during live events. This presentation offers an overview of the environmental noise impact of reinforcement systems used in outdoor venues and explores solutions for reducing noise impact to surrounding communities. We will discuss how cardioid and end-fire techniques, coupled with mechanical and electronic optimization can deliver significantly improved broadband pattern control and reduce the environmental noise impact of sound reinforcement systems.

**11:10**

**1aNS7. Sound intensity and directivity measurements for a loudspeaker canopy in a restrictive noise environment.** Francisco J. Irarrazabal (MD Acoust., 170 S William Dillard Dr., Ste 103, Gilbert, AZ 85233, francisco@mdacoustics.com), Robert M. Pearson, Michael L. Dickerson, and Samuel K. Hord (MD Acoust., Gilbert, AZ)

It is common to have noise complaints from outdoor event and entertainment venues in urban and non-urban environments due to the loudness of the reinforcement system, rhythmic nature of the music, acoustical spectrum, and the geometric spreading of said system. Complaints about noise from events, bars, and restaurants are on the rise. Increasing traffic, construction, and commercial activities tend to increase the ambient noise level in urban environments, creating noise creep. The pandemic restrictions helped reduce the ambient levels in the urban and non-urban areas.

However, after the reopening, more and more noise complaints started to be issued since residents were more sensitive to noise pollution. This paper presents a practical application of the source directivity for a loudspeaker system. The loudspeaker canopy is designed to produce a radiation pattern directing the main beam lobe to the front of the array and reduce the noise propagation to other directions. This device can be implemented in entertainment and outdoor events (such as wedding parties) to reduce the noise impact on sensitive receivers and meet the jurisdictional noise regulations. Real-world acoustic intensity measurements were made to obtain the system's sound power and assess the noise impact at distances up to 35 feet away from the source. The viability of using the system for outdoor activities within locations restricted by the City's ordinances is discussed.

**11:25**

**1aNS8. The New Rady Shell at Jacobs Park in San Diego.** Jason Duty (Salter, San Francisco, CA), Valerie Smith (Salter, 1100 Dexter Ave. N, Ste. 100, Seattle, WA 98107, valerie.smith@cmsalter.com), Felipe Tavera, Tom Schindler, and David Schwind (Salter, San Francisco, CA)

Located along the waterfront at Jacobs Park, the Rady Shell is the summer home for the San Diego Symphony. The shell opened in summer 2021 as a unique outdoor venue in downtown San Diego. When not used for performances, the lawn is open to the public. Salter provided input on the physical acoustics and technology components within the shell. This presentation focuses on the interior acoustics of the shell. The shell utilizes the first Meyer Sound Constellation system designed for stage acoustics in an outdoor venue. To enhance the experience for the symphony on stage, a mixture of reflective and diffusive panels were used to provide some natural early reflections. During design, the ODEON Room Acoustics software was used to predict the acoustical response on the stage. Above the stage, the shell features a checkerboard pattern of absorptive and reflective panels. At the lawn, Salter worked with L'Acoustics to design a sound system that focused sound on the audience to reduce impact to the neighboring properties.



## *Invited Paper*

**11:40**

**1aNS9. A game theory model of the Lombard effect in public spaces.** Braxton Boren (Performing Arts, American Univ., 4400 Massachusetts Ave. NW, Washington, DC 20016, boren@american.edu)

The Lombard effect is the tendency for humans and other animals to raise their vocal intensity in the presence of background noise. As a result, the aggregate effect of many people talking in a public space can lead to multiple equilibria for speech intelligibility, as the incentive for an individual conversation is dependent on the actions of all the other speakers in the space. This system is modeled using game theory, in particular, an n-person Prisoner's Dilemma model, otherwise known as the Unscrupulous Diner's Dilemma. It is suggested that conscious knowledge of the dynamics of the system can affect the system's behavior up to a point, but that gradually the subconscious process of the Lombard Effect will cause the system to increase in background noise, only limited by the comfort of each speaker's voice or an exogenous interrupting event.

## Session 1aPA

## Physical Acoustics and Biomedical Acoustics: Acoustofluidics

James Friend, Cochair
*Mechanical and Aerospace Engineering, University of California San Diego, 9500 Gilman Dr MC0411, MADLab SME344K, La Jolla, CA 92093*

Charles Thompson, Cochair
*Electrical and Computer Eng., UMASS Lowell, 1 Univ. Ave., Lowell, MA 01854*

J. Mark Meacham, Cochair
*Mechanical Engineering & Materials Science, Washington University in Saint Louis, 1 Brookings Dr., Jubel Hall, Rm 203K, Saint Louis, MO 63130*

Kedar Chitale, Cochair
*Flodesign Sonics, Inc., 380 Main Street, Wilbraham, MA 01095*

Max Denis, Cochair
*University of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008*

### *Invited Papers*

**8:00**

**1aPA1. Acoustic radiation torque on an interface reflecting acoustic vortex beams.** Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., University, MS 38655, zhang@olemiss.edu)

Torque exerted by acoustic waves to rotate objects can result from asymmetry such as the case of an averaged torque on a Rayleigh disk exerted by an ordinary acoustic field. In the absence of asymmetry, acoustic torque can occur by viscous absorption either in the object or in the adjacent boundary layer of the object when using specific types of acoustic fields such as acoustic vortex beams carrying twisted wave fronts [Zhang and Marston, Phys. Rev. E 84, 065601 (2011)] or two orthogonal standing wave fields [Zhang and Marston, J. Acoust. Soc. Am. 136, 2917 (2014)]. Recent study suggested a torque on a flat water-air interface exerted by an ultrasonic vortex beam obliquely incident and reflecting off the interface [Zou, Lirette, and Zhang, Phys. Rev. Lett. 125, 074301 (2020)]. This torque is not associated with viscous absorption and instead results from variation of orbital angular momentum carried by the vortex beam during the reflection. The torque would otherwise disappear in normal incidence or using ordinary beams. The investigations advance understanding of radiation torque physics and approaches for non-contact manipulations using acoustic waves.

**8:30**

**1aPA2. Microscopic rogue waves in strongly nonlinear capillary wave turbulence.** Jeremy Orosco (Mech. and Aerosp. Eng., Univ. of California San Diego, 13754 Mango Dr. Unit 306, Del Mar, CA 92014, jrorosco@ucsd.edu), William Connacher, Kha Nguyen (Mech. and Aerosp. Eng., Univ. of California San Diego, San Diego, CA), and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA)

When an otherwise quiescent minuscule ($O(\leq 10^{-6})\mu$l) basin of water is driven by ultrasonic vibrations at high frequencies ($O(\geq 10^{6})$Hz) and with nanoscale ($O(\leq 10^{-9})$m) amplitudes, turbulent capillary waves that are visible by eye ($O(10^{-2})$m amplitudes and $O(10^{-1})$s periods) form at the air-water interface. Classical mechanisms typically attributed to such instabilities—such as Faraday wave theory—are absent. Contemporary wave turbulence studies have been mainly limited to weakly nonlinear regimes. In this talk, we present detailed measurements of strongly nonlinear, microscopic capillary wave turbulence. We show that this regime is reliably statistically characterized as an alpha-stable Lévy flight with a varying tail parameter. Our results demonstrate that as input power is increased, the heaviness of distribution tails also increases so that rogue events play an increasingly prevalent role in the overall wave system. Implications for future study and potential applications within the area of controlled atomization are discussed.

9:00

**1aPA3. Acoustically driven shear mechanoporation for efficient intracellular delivery and transfection.** J. Mark Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Jubel Hall, Rm 203K, Saint Louis, MO 63130, meachamjm@wustl.edu)

Delivery of large and structurally complex molecules into cells is useful in numerous biomedical applications. Furthermore, this capability is critical to the emerging area of cell-based therapeutics. Extant *in vivo* (viral) and *in vitro* (chemical, electrical) methods are often inadequate for protein, nucleic acid, and synthetic nanomaterial delivery, leading researchers to develop alternative physical approaches. Over the past two decades, our lab has established a micromachined ultrasonic droplet generator that uses geometric focusing of acoustic waves to break the liquid surface and form extremely fine droplets. More recently, we have found that these focused mechanical forces generated on a microsecond time scale can be applied to reversibly disrupt the membranes of cells. Acoustic shear poration (ASP) relies on cell ejection from microscale orifices only slightly larger than the cells to open transient pores large enough for passage of small to large macromolecules at greater than 75% efficiency. I will discuss the principles of device operation, the connection between operating parameters and treatment outcomes, and the limitations of diffusive delivery following mechanoporation. I will also introduce a two-stage approach that combines mechanoporation and an electrophoretic action to improve transfection and delivery of complex assemblies that carry multiple payloads.

## *Contributed Papers*

9:30

**1aPA4. An investigation of the maximum particle velocity as an invariant in acoustofluidic applications and more.** Arik Singh (Mech. and Aerosp. Eng., Univ. of California San Diego, San Diego, CA), Naiqing Zhang (Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA), and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego, 9500 Gilman Dr. MC0411, MADLab SME344K, La Jolla, CA 92093, jfriend@ucsd.edu)

Vibrating materials experience internal stress waves that can cause material failure or energy loss due to inelastic vibration. Failure is traditionally defined in terms of acceleration, yet this approach has many drawbacks, principally because it is not invariant with respect to scale, type of vibration, or material choice. Here, the likelihood of failure is instead defined in terms of the maximum particle velocity. While the exact dependence of the internal stress on the particle velocity may be determined for specific cases, here we take a statistical approach in seeking a universal maximum particle velocity correlated to a specific risk of material failure. Our Monte Carlo-based analysis on a variety of materials, vibration types and frequencies, structures with flaws, and vibration velocities produced results in support of the notion that a maximum particle velocity on the order of 1 m/s is a universal and critical limit. Upon exceeding this limit, we find the probability of failure or excessive acoustic loss in suppression of the vibration to become significant, regardless of the details of the material, geometry, or vibration. We illustrate this in a specific example relevant to acoustofluidics, a simple surface acoustic wave device with fluid sample, and point out the value of using the maximum particle velocity as a design invariant.

9:45

**1aPA5. Acoustic streaming in a channel.** Charles Thompson (Elec. and Comput. eng, UMASS Lowell, 1 Univ. Ave. Lowell, MA 01854, charles_thompson@uml.edu), Max Denis (Univ. of the District of Columbia, Washington, DC), and Kavitha Chandra (Elec. and Comput. eng, UMASS Lowell, Lowell, MA)

This paper presents an analysis of acoustic streaming in a 2D channel. The unsteady flow resulting from acoustic excitation of the enclosed fluid is of particular interest. The magnitude of the streaming velocity will depend on the Strouhal, S, and oscillatory Reynolds numbers, R. For the frequencies of interest, 1/R and 1/S are much less than one. Results are given for the streaming Reynolds number $R/S^2 = O(1)$. Pressure correction approaches used to satisfy the no-slip boundary condition are explored.

**10:00–10:15 Break**

10:15

**1aPA6. Interactions between epithelial cells after their exposure to ultrasounds.** Jon Luzuriaga (Inst. of Physical Techs, Dept. of Ultrasounds, Consejo Superior de Investigaciones Científicas CSIC, Madrid, Spain), Iciar Gonzalez (Inst. of Physical Techs, Dept. of Ultrasounds, Consejo Superior de Investigaciones Científicas CSIC, Serrano 144, Madrid 28006, Spain, iciar.gonzalez@csic.es), and Manuel Candil (Inst. of Physical Techs, Dept. of Ultrasounds, Consejo Superior de Investigaciones Científicas CSIC, Madrid, Spain)

Motions of a pair of epithelial cells over a substrate become altered after their exposure to ultrasonic waves at a frequency close to 1 MHz. An irradiation of 20 min on the cell samples produces on them long-term effects, which experience alternating slow attraction repulsion processes in direction perpendicular the acoustic vector direction. One cell approaches toward the other one, which shows retractile motion of its fillopodia, acquiring a rounded shape in self-protection mode. after some hours, the situation is reversed: the retracted cell ceases to be so and begins to advance towards the “invading” cell, reversing the roles. These approaching-repulsion processes take several hours after the ultrasonic irradiation during the culture. These experiments evidence singular long-term effects of the ultrasounds on the biodynamics of epithelial cells at certain acoustic conditions.

10:30

**1aPA7. Using motile cells to characterize surface acoustic wave-based acoustofluidic devices.** Advaith Narayan (Mech. Eng. and Mater. Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Saint Louis, MO 63130, narayan@wustl.edu), Mingyang Cui, and J. Mark Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, Saint Louis, MO)

Acoustic microfluidics is a robust and powerful method to manipulate cells and cell-like particles on chip, having good biocompatibility and ease of incorporation into multioperation microfluidic devices compared to optical manipulation. However, the use of acoustic microfluidics is largely confined to research settings. The primary barrier to translation of this technology toward clinical and industrial uses is the inability to experimentally determine the pressure field (shape and amplitude) and associated acoustophoretic forces in real time as device conditions vary. Despite the multitude of previous characterization methods, none provide the flexibility of motile cells (e.g., the unicellular alga *Chlamydomonas reinhardtii*) as probes to map evolving pressure fields on chip. We have previously developed this approach for use with bulk acoustic wave (BAW)-based devices. Here, we extend the method to qualitatively assess device resonances and relative field strengths for surface acoustic wave (SAW)-based devices with straight channels and circular chambers driven at 6 MHz and 20 MHz. The fabrication and electrical characterization of hybrid BAW/SAW devices with glass channels are also discussed. Upon testing, the optimal device operating parameters are identified using impedance measurements, as well as visual identification of resonant frequencies using the swimming algae cells.

10:45

**1aPA8. Cell-like microparticles with tunable acoustic properties for calibrating devices.** Clara E. Tandar (Ctr. for Biomedical Eng., Brown Univ., 91 Waterman St., Ste. 211, Providence, RI 02912, clara_tandar@brown.edu), Ryan Dubay, Eric Darling (Ctr. for Biomedical Eng., Brown Univ., Providence, RI), and Jason Fiering (Draper Lab., Cambridge, MA)

Mechanophenotype of biological cells has demonstrated correlation with biomolecular states and cell function. Hence, new methods to measure mechanophenotype at high throughput are of growing interest. Acoustophoretic microdevices can characterize cell mechanical features; however, calibration particles with physiologically relevant properties are needed to quantify and optimize device performance. Currently, conventional polymer microspheres are rigid and do not replicate cell deformation and compressibility. To address this, we developed monodisperse, tunable, cell-like microparticles (MPs) from polyacrylamide hydrogel, fabricated with a microfluidic droplet generator. Size and compressibility are adjusted by fabrication parameters, and density is adjusted by incorporation of nanoparticles (NPs). Here, we present for the first time microparticles of reduced density and acoustic contrast (lower than unloaded MPs) achieved by loading MPs with nanoparticles of low molecular weight alkanes. We produced the NPs by sonication and photopolymerization before addition to the MP precursor. NP-loaded MPs were less dense than unloaded MPs at 1005.9 and 1013.6kg/m$^3$, respectively, and they exhibited negative acoustic contrast by acoustophoresis in aqueous medium while that of unloaded MPs was positive. These new particles extend the tunable range of acoustic contrast, mimicking and exceeding that of most biological cells and could also aid cell separation when conjugated to cells.

11:00

**1aPA9. Sensitivity of acoustic parameters on the reflected signal from a rigid porous medium in the low-frequency range of ultrasound.** Mustapha Sadouki (Acoust. and Civil Eng. Lab., Khems-Miliana Univ., none, Rte. Thenia el Had, Khemis-Miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz), Abdelmadjid Mahiou (Theor. Phys. and Radiation Matter Interaction Lab., Soumaa, Blida.Acoust. and Civil Eng. Lab., Khemis-Miliana Univ., none, Khemis-miliana, Algeria), and Nacera Souna (Acoust. and Civil Eng. Lab., Khems-Miliana Univ., none, Khemis-Miliana, Algeria)

Porous materials are two-phase media consisting of pores saturated with a fluid emerging from a solid skeleton. There are several parameters describing the porous medium. These parameters are classified into two categories: the high-frequency parameters and the low-frequency parameters. During the excitation by an acoustic wave, two cases can occur; the vibration of both phases simultaneously in the case where the structure is flexible, this case is well studied by Biot theory. For a rigid structure, it is the movement of the fluid that is taken into account; this last case is treated by the equivalent fluid theory which is a particular case of Biot theory. Previous work [Proc. Mtgs. Acoust. **45**, 045004 (2021)] has shown the influence of the parameters involved in the corrected Johnson-Allard model recently introduced by Sadouki [Phys. Fluids 33, (2021)] on the transmitted signal from a rigid-porous material in the low-frequency regime of ultrasound. The objective of this work is to show, once again, the influence of the same parameters on the reflected signal by providing a comparative study between transmitted and reflected modes.

11:15

**1aPA10. Acoustic erythrocytometer for mechanical cell probing.** Andreas Link (Biomedical Eng., Univ. of Glasgow, Glasgow City, United Kingdom) and Thomas Franke (Biomedical Eng., Univ. of Glasgow, Rankine Bldg., R522, Glasgow City G12 8LT, United Kingdom, thomas.franke@glasgow.ac.uk)

The mechanical properties of cells provide key insights into the type, differentiation and/or pathology of a cell. The mechanical analysis of a cell population, such as a blood sample, enables meaningful biological and medical interpretation for the diagnosis and monitoring of diseases. There are number of diseases, which alter the mechanical properties of human red blood cells (RBCs). Here, an acoustic method to probe both the viscous and elastic mechanics of single RBCs by SAW in a microfluidic device is presented. The device operates by exciting a surface acoustic wave in a microfluidic channel creating a stationary acoustic wave field of nodes and antinodes. RBCs are attracted to the nodes and are deformed. Using a stepwise increasing and periodically oscillating acoustic field, the static and dynamic deformation of individual red blood cells one by one was studied and the deformation by the Taylor deformation index D and relaxation times were quantified. The precision of the measurement allows to distinguish between individual cells in the suspension and provides a quantitative viscoelastic fingerprint of the blood sample with a resolution of a single cell. The method overcomes limitations of other techniques that provide averaged values and has the potential for high-throughput screening.

11:30

**1aPA11. Phase speed and attenuation determination of a monodisperse liquid at the single bubble resonance using a single and two microphone transfer function method.** Stanley A. Cheyne (Phys. & Astronomy, Hampden-Sydney College, Dept. of Phys., Hampden-Sydney, VA 23943, scheyne@hsc.edu), H O. Thurman, and R G. Holt (Phys. & Astronomy, Hampden-Sydney College, Hampden-Sydney, VA)

The phase speed and attenuation of a bubbly liquid were determined. The experimental configuration consisted of a vertical stainless steel tube partially filled with water and pressurized air flowing through a single hypodermic needle at the bottom. The sound source was located in the air-filled portion at the top of the tube. A SRS-785 spectrum analyzer was used to measure and calculate the transfer function. Once the impedance was found, the phase speed and attenuation was calculated from the complex wave number.

## Session 1aSA

## Structural Acoustics and Vibration: Tunable Metamaterials

Ganesh U. Patil, Cochair
*Mechanical Science and Engineering, University of Illinois Urbana-Champaign,
144 Sidney Lu Mechanical Engineering Building, MC-244, 1206 West Green Street, Urbana, IL 61801*

Benjamin M. Goldsberry, Cochair
*Applied Research Laboratories at The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758*

Elizabeth Smith, Cochair
*Mechanical Science and Engineering, University of Illinois at Urbana Champaign,
Mechanical Engineering Building, 1206 W Green St., Urbana, IL 61801*

**Chair's Introduction—9:00**

### *Invited Papers*

**9:05**

**1aSA1. Tunablility at the ultra-low-frequency via inerter-based elastic metamaterials.** Pai Wang (Mech. Eng., Univ. of Utah, 1495 E 100 S, MEK Bldg., Salt Lake City, UT 84112, pai.wang@utah.edu), Faisal Jamil (Intel, Portland, OR), Fei Chen, Robert G. Parker (Mech. Eng., Univ. of Utah, Salt Lake City, UT), and Bolei Deng (MIT, Salt Lake City, UT)

Inerter is a mechanical device capable of exhibiting an inertial effect that is orders-of-magnitude larger than its physical mass. By coupling linear motion with rotational motion of a built-in flywheel, the inerter generates a response force proportional to the relative acceleration between its two independent terminals. Here, we experimentally fabricate and characterize vibro-elastic metamaterial designs with embedded interters. Aided by computational simulations, our design aims to demonstrate a unique and fundamental advantage in forming a bandgap at extremely low frequencies. After fabrication, we perform wave-propagation testing on samples with both longitudinal (pressure) and transverse (shear) waves. The results show that our design can be made tunable by changing either the effective rotational inertia or the effective connection stiffness. This significantly enhances the metamaterial's applicability in mitigating real-world structural vibration with at ultra-low frequency and very long wavelength. Our data indicate that inerter-based design may outperform traditional locally resonant metamaterials and could be more suitable for broader application scenarios, such as seismic events.

**9:25**

**1aSA2. Reconfigurable metamaterial neuromorphic computing.** Mohammadreza Moghaddaszadeh, Mohamed Mousa (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), Buffalo, NY), Amjad Aref (Civil, Structural, and Environ. Eng., Buffalo, NY), and Mostafa Nouh (Mech. and Aerosp. Eng., Univ. at Buffalo (SUNY), 240 Bell Hall, Attn: Mostafa Nouh, Buffalo, NY 14260, mnouh@buffalo.edu)

Neuromorphic computing was originally introduced in electronic circuits to mimic neuro-biological architectures. In these systems, a physical agent (e.g., an electromagnetic or acoustic wave) propagates through multiple layers of metasurfaces which are trained to perform a computational task (e.g., classification). Despite their potential, current neuromorphic metasurfaces rely on passive designs which limits their computational power to a single task. Furthermore, attempts to realize these systems in the context of mechanical wave propagation have been very scarce. This work presents a neuromorphic metasurface which is designed to exploit elastic wave scattering to realize a physical computing environment. Owing to the reconfigurable design of the chosen unit cell, the neuromorphic metasurface can be tuned to conduct multiple distinct classification tasks without the need for remanufacturing. The designed subwavelength metasurface cell will be used to train a customized neural network with constant weights (representing the elastic wave propagation between different layers of metasurface) and trainable activation functions (representing the phase modulation at each layer of metasurface). To perform distinct classification tasks, the trainable parameter in the activation function will be tuned accordingly. This work opens up new avenues in high performance mechanical computing.

9:45

**1aSA3. Digitally tunable non-reciprocal wave propagation using piezoelectric metamaterials with synthetic impedance circuits.** Mustafa Alshaqaq (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30318, Atlanta, GA 30318, mshaqaq3@gatech.edu), Christopher Sugino (Mech. Eng., Stevens Inst. of Technol., Hoboken, NJ), and Alper Erturk (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

One common approach to enable non-reciprocal wave propagation in elastic media relies on introducing a directional bias by parameter modulation (e.g., stiffness modulation) in space and time. Researchers have successfully demonstrated non-reciprocal wave propagation by coupling elastic waveguides with magnetic electrical coil elements, piezoelectric elements shunted to capacitive analog circuits, or rotating mechanical resonators. The existing efforts are often limited by narrow operation frequency ranges and cumbersome implementations that lack smooth modulation (e.g., with analog circuit elements and switches). In this work, we demonstrate precise and smooth non-reciprocal wave propagation in a piezoelectric metamaterial by connecting each unit cell to a digitally controlled synthetic impedance circuit. The effective impedance of each unit cell can be externally controlled according to a desired space-time modulation scheme over a wide frequency range. We present numerical and experimental investigations on the dynamics of a piezoelectric metamaterial beam with 30 bimorph unit cells whose impedances can be varied smoothly. Specifically, we employ the spatiotemporal modulation on synthetic inductance circuits rather than capacitive. Experimental results are presented for various space-time modulation profiles, investigating the effects of the inductance value (target frequency), as well as the modulation frequency and amplitude on digitally tunable non-reciprocal wave propagation.

**10:05–10:20 Break**

10:20

**1aSA4. Dynamics of tunable multistable metastructures.** Myungwon Hwang, Yeongeun Ki (Mech. Eng., Purdue Univ., West Lafayette, IN), and Andres F. Arrieta (Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907, aarrieta@purdue.edu)

Connectivity yields unconventional properties. However, the attainable dynamics are strongly dependent on the unit cell size, restricting the effective behavior to narrow, high-frequencies bands. This is due to the nature of band gaps from local resonance or scattering, both of which are strongly related to unit's size (mass) and stiffness. We present multistable metastructures displaying strong nonlinear interactions between propagating transition waves and structural modes. We show how transition waves excite the same type of response in the metastructure's units regardless of the input excitation. This invariant response allows for efficient electromechanical energy transduction as the mechanical response can be tuned to electrical conversion circuits robustly. We also present a new dynamic phenomenon—solitonic resonance—leveraging soliton-structural mode interactions that enable multistable metastructures to exhibit extreme input-output energy exchange. By tuning the topology of our multistable metastructures we can transform energy input frequencies into output responses orders of magnitude apart. The presented metastructures break the dependence of the attainable unconventional dynamical properties on the unit cell's size. The dynamics of multistable metastructure provide a route to accelerating metamaterials adoption in engineering applications addressing the structural bandwidth.

## *Contributed Papers*

10:40

**1aSA5. Strongly tunable nonlinear MEMS resonators by electrothermoelastic buckling.** Ali Kanj (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 105 S Mathews Ave. Mech. Eng. Lab., Urbana, IL 61801, alimk2@illinois.edu), Paolo F Ferrari, Arend M. van der Zande, Alexander F. Vakakis, and Sameh Tawfick (Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

This talk presents the benefit of buckling in strongly tuning nonlinear MEMS resonators. In particular, the talk shows experimental buckling achieving more than 230% tuning in the natural frequency of drumhead MEMS resonators. Moreover, the experiments demonstrate that buckling switches the frequency response between purely stiffening, purely softening, and stiffening-to-softening nonlinearities. We tune the buckling state in these experiments by controlling the electric voltage and the temperature of the resonators. Therefore, these resonators undergo electrostatically-mediated thermoelastic buckling where specific combinations of temperature and voltage are required to access the distinct vibrational responses. We explain the observed linear and nonlinear responses by a reduced-order model (ROM) that lumps the resonator into a 1-degree-of-freedom mass connected to springs reflecting bending, stretching, electrostatic forces, thermal expansions, and residual stresses. The ROM mimics von Mises trusses to model the buckling in the membrane resonators without the need for exact geometry or structure to model the resonators. This developed ROM and the electro-thermoelastic buckling tunability present useful applications for on-chip acoustic devices in different fields such as signal manipulation, filtering, and MEMS waveguides.

10:55

**1aSA6. Frictional instability: A nonlinear mechanism to control shear wave responses in rough contact-based metamaterials.** Ganesh U. Patil (Mech. Sci. and Eng., Univ. of Illinois Urbana-Champaign, 144 Sidney Lu Mech. Eng. Bldg., MC-244, 1206 West Green St., Urbana, IL 61801, gupatil2@illinois.edu), Alfredo Fantetti (Dept. of Mech. Eng., Imperial College London, London, United Kingdom), and Kathryn Matlack (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Incorporating nonlinearity in periodic media not only enables enriched wave dynamics but also allows passive tunability of the wave responses. However, studies so far have been mostly focused on nonlinear mechanisms and responses of *longitudinal* wave propagation whereas *shear* propagation in the presence of strong nonlinearity is yet to be fully understood. In this talk, we study shear wave propagation through metamaterials with rough contacts including friction. The roughness of contacting surfaces results in structural instability causing the contacts to switch between different regimes—stick, partial slip, and gross slip—giving rise to strong nonlinearity. Moreover, due to the presence of friction, the contacts exhibit hysteretic nonlinearity, i.e., history-dependent response. In this study, we first experimentally evaluate the frictional properties of rough contacts by measuring high-frequency friction hysteresis loops. Then, we develop metamaterial with a periodic arrangement of these rough contacts and use the obtained frictional properties to numerically study nonlinear shear wave signatures. We evaluate higher harmonic generation and demonstrate how they can be tuned through excitation wave amplitude, external precompression, and surface roughness. These fundamental understandings can open new avenues

for designing tailored material with memory-dependent nonlinearity for controlling shear wave propagation.

**11:10**

**1aSA7. Exploration of tunable properties of a one-dimensional leaky-wave antenna.** Abigail D. Willson (Appl. Res. Lab, Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, adw5@psu.edu), Andrew S. Wixom, and Amanda Hanford (Appl. Res. Lab, Penn State Univ., State College, PA)

Leaky-wave antennas (LWA) are commonly used in beam-steering situations such as radars, imaging, and other such frequency scanning applications. LWA are particularly useful metamaterials due to their relatively simple structure of repeated cells in a linear array and their broad scanning range allowing for more use to come from a single frequency sweep. Among current methods to model and design acoustic LWA are notably high fidelity models such as finite element analysis and low fidelity models such as the acoustic transmission-line equations. Presented is a technique for a mid-fidelity model of a one-dimensional acoustic LWA using a lumped element system. This model is used to explore rapid evaluations of different tuning options in the elastic membranes separating the unit cells of the structure. Comparisons are made between time and frequency domains as well as modal and physical space simulations.



MONDAY MORNING, 5 DECEMBER 2022 — RAIL YARD, 9:00 A.M. TO 11:15 A.M.

## Session 1aSP

## Signal Processing in Acoustics: Machine Learning in Signal Processing

Trevor Jerome, Chair
*Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd., BLDG 3 #329, West Bethesda, MD 20817*

**Chair's Introduction—9:00**

### *Contributed Papers*

**9:05**

**1aSP1. Braid feature extraction and representation of active sonar data.** Bernice Kubicek (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52242, bernice-kubicek@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

As an active sonar system pings an object of interest, the moving platform collects data descriptive of both the target and background clutter. We hypothesize that the relevant features of the target echo tend to persist and smoothly morph over time in some domain as the sonar platform continues its route. This work leverages the preceding and presents initial findings of a feature extraction algorithm resulting in representations of active sonar data that persistently appear and are physically and statistically informed. This is done by isolating the targets' features from background clutter using a statistically meaningful threshold, extracting features onto a manifold representation, and mathematically describing and quantifying the extracted features. The background clutter is modeled using a Rayleigh probability density function, and the echo is isolated by keeping statistically significant responses. Feature extraction is performed on a ping-by-ping basis by minimizing the angle between the isolated echo samples and assigning a pseudo probability to sequential samples. These methods are used to extract features from ping sequences that are unique to targets and quantified using a correlation metric. Results reported are on simulated data and experimental field data. [Work supported by NREIP, NDSEG, and ONR, Grant No. N00014-21-1-2420.]

**9:20**

**1aSP2. Graph representation learning on braid manifolds.** Andrew J. Christensen (Elec. and Comput. Eng., Univ. of Iowa, 3100 Seamans Ctr., Iowa City, IA 52242, andrew-christensen@uiowa.edu), Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA), and Ivars Kirsteins (NUWCDIVNPT, Newport, NJ)

The accuracy of autonomous sonar target recognition systems is usually hindered by morphing target features, unknown target geometry, and uncertainty caused by waveguide distortions to signal. Common "black-box" neural networks are not effective in addressing these challenges since they do not produce physically interpretable features. This work seeks to use recent advancements in machine learning to extract braid features that can be interpreted by a domain expert. We utilize Graph Neural Networks (GNNs) to discover braid manifolds in sonar ping spectra data. This approach represents the sonar ping data as a sequence of timestamped, sparse, dynamic graphs. These dynamic graph sequences are used as input into a GNN to produce feature dictionaries. GNNs ability to learn on complex systems of interactions help make them resilient to environmental uncertainty. To learn the evolving braid-like features of the sonar ping spectra graphs, a modified variation of Temporal Graph Networks (TGNs) is used. TGNs can perform prediction and classification tasks on timestamped dynamic graphs. The modified TGN in this work models the evolution of the sonar ping spectra graph to eventually perform graph-based classification. [Work supported by ONR grant N00014-21-1-2420.]

9:35

**1aSP3. Modeling sonar performance using J-divergence.** Douglas Abraham (none, Ellicott City, MD, abrahad@ieee.org)

This presentation demonstrates the use of J-divergence as a performance measure for detection in a sonar system. The inherent inaccuracies between system-level performance and the cell-level (PD,PF) detector operating point used in traditional analysis open the door to using approximate performance measures such as J-divergence. The properties of J-divergence making it an appealing choice are covered: summing to accrue J-divergence across multiple independent measurements (e.g., from multiple source signals, waveforms, or arrays), a data-processing inequality dictating that processing cannot improve J-divergence, and an asymptotic relationship to the traditional (PD,PF) operating point. Simple forward models of J-divergence are presented for matched filters and energy detectors when applied to the standard signal models in Gaussian noise. A "design" J-divergence, which is chosen by the desired qualitative performance level, is used in simple equations to obtain the "design" SNR require to achieve it. This provides a direct replacement for the detection threshold (DT) term in the sonar equation that is easier to apply and evaluate. Example applications of J-divergence are presented illustrating its utility in the modeling and adaptation of current systems as well as the design and analysis of new ones.

9:50

**1aSP4. Imaging sonar performance estimation.** Brian O'Donnell (Sensors and Electromagnetic Applications Lab., Georgia Inst. of Technol., Atlanta, GA, brian.odonnell@gtri.gatech.edu), David J. Pate, and Daniel Cook (Sensors and Electromagnetic Applications Lab., Georgia Inst. of Technol., Smyrna, GA)

Imaging sonar systems including forward looking sonars, synthetic aperture sonars, and real aperture sonars produce large quantities of data and are often connected to automated target recognition (ATR) algorithms. Performance of these algorithms degrades, sometimes significantly, with changes in environmental reverberation, seafloor sediment characteristics, and target characteristics. However, most ATR algorithms have no capability to assess their own performance, flag data that is invalid, or convey when their results should not be trusted or utilized by humans or autonomous algorithms. Performance estimation algorithms can be used in conjunction with ATR to supplement these shortcomings. Estimates of signal-to-noise ratio, image contrast, and resolution can be used to assess deviation from nominal performance capabilities, while regressing ATR performance against imagery characteristics or other through-the-sensor metrics can be used to estimate false alarm rate.

10:05

**1aSP5. Recent results in shallow water acoustic channel estimation and equalization.** Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, 103 S Capitol St., Iowa City, IA 52242, ananya-sengupta@uiowa.edu), Nathan Kofron, and Eva Riherd (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA)

Shallow water acoustic channel estimation and equalization has been widely studied and explored across the last three decades, with several techniques being proposed that account for the rapidly time-varying nature of the channel. Despite advances in integrating robust channel equalization techniques with channel estimation, incorporating well-known models of acoustic propagation into the equalization setup has been hard. In this work, we will present some of our recent explorations in this regard involving a hybrid setting as well as a model-based simulation to test the performance of simple equalization schemes under a variety of channel conditions. The shallow water acoustic channel is first estimated from experimental field data, and the real-time channel estimates are used to simulate an end-to-end communication system with ambient noise at different signal-to-noise ratios (SNR). In tandem, we also simulate a time-varying shallow water acoustic channel under similar oceanic conditions using the Bellhop model, along with a communication system with similar signaling schemes and SNR variations. Equalization results from both the hybrid and model-based settings will be presented and implications of including machine learning to learn the channel parameters from both setups to inform the channel equalizers will be explored.

**10:20–10:30 Break**

10:30

**1aSP6. Classification between different vocal pathologies by using convolution neural network and continuous wavelet transforms.** Bhawna Rathi (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, 535 W. Michigan St., IT371, Indianapolis, IN 46202, brathi@iu.edu) and Timothy Hsu (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, IN)

Millions of Americans suffer from vocal damage that can negatively impact their daily lives and potentially incur large health care costs. Vocal damage is found frequently in at-risk populations, including singers, teachers, coaches, and telemarketers. The current standard method of diagnosis involves performing a laryngoscopy. Current research has shown that digital signal processing with acoustic machine learning methods can be used to distinguish healthy from unhealthy voices, but there has been limited prior work in classifying different pathology types from one another using only acoustic machine learning and signal processing methods. This study aims to design a convolution neural network (CNN) algorithm that can differentiate between different vocal pathologies by using an audio data file of the vowel sound /a/ of damaged voices as inputs. The audio dataset of different vocal pathologies has been obtained from the Indiana University Vocal Pathology Dataset. For the CNN algorithm, images are required, and continuous wavelet transforms (CWT) are calculated that show spectral energy across time in a visual format. The overall results will show the accuracy for two class classifications of different vocal pathologies and how accuracy changes when the different parameters of the CNN architecture are changed.

10:45

**1aSP7. Acoustic detection of drone range and type using nonuniform band energy features.** Kaliappan Gopalan (ECE, Purdue Univ. Northwest, ECE Dept., Hammond, IN 46323, kgopala@pnw.edu), Brett Y. Smolenski (North Point Defense, Rome, NY), and Darren Haddad (AFRL, Rome, NY)

With increased use of drones in a variety of situations, it is imperative that efficient means of detecting the type of unmanned aerial vehicle and/or its range are developed from security, privacy, and safety perspective. This paper describes the application of an acoustic feature set in a deep learning network for the estimation of the line-of-sight range of drones. The set of spectral energy values over nonuniform bands within the range of audio recordings of open space drone noise has been shown to predict the range to a reasonable degree of accuracy. The energy feature set, when augmented with low frequency spectral components, raised prediction accuracy to within 85 cm of mean error and a standard deviation of 12 m for test cases ranging from 10 m to 935 m. Additionally, the spectral band energy applied to classify the range quantized into 1 m intervals resulted in better than 87% accuracy with a fixed error of $\pm$50 cm over the entire range. Adding low frequency spectral components to the band energy set raised the correct range classification to 97%. For classification of seven tethered drones, the band energy feature set resulted in 99.9% accuracy.

11:00

**1aSP8. Detection and classification of drones using Fourier-Bessel series representation of acoustic emissions.** Kaliappan Gopalan (ECE, Purdue Univ. Northwest, ECE Dept., Hammond, IN 46323, kgopala@pnw.edu), Brett Y. Smolenski (North Point Defense, Rome, NY), and Darren Haddad (Information Exploitation Branch, Air Force Res. Lab., Rome, NY)

Detection, classification, and line-of-sight range estimation of drones are vital for security, safety, and privacy reasons. Representation of the audio emissions of drones in a Fourier-Bessel (FB) series expansion is proposed for the identification of a drone and/or the prediction of its range from an observation point. A deep learning network employing the FB series coefficients as the preprocessed input has been shown to classify accurately each of seven drones flying in a controlled environment in about 84 % of cases. For the case of any one of three drones flying outdoors, presence of the drone—as opposed to background noise—was detected correctly with few false positive and false negative results. Additionally, the range of the drone—from 2.5 m to 935.6 m—was estimated to be within $\pm$50 cm of actual line-of-sight distance in over 85 % of the available test cases.

# Session 1pAA

## Architectural Acoustics: Acoustical Challenges in Small Rooms II

Joseph Keefe, Chair
*Ostergaard Acoustical Associates, 1460 US Highway 9 North, STE 209, Woodbridge, NJ 07095*



### *Invited Papers*

**1:20**

**1pAA1. Recording studio sound isolation: A case study of challenges faced in measuring door transmission class (DTC) and noise isolation class (NIC) in small rooms.** Jessica S. Clements (Director of Acoust., Newcomb & Boyd, LLP, 303 Peachtree Ctr. Av NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com)

This case study examines the challenges faced in measuring the DTC and NIC between a recording studio control room and the studio recording space. Challenges included structural radiation around the tested partition, flanking paths, ADA ramp design, and improper use of the space during construction. The measurements were performed following the procedures of ASTM E2964-14 Standard Test Method for Measurement of the Normalized Insertion Loss of Doors and ASTM E336-19 Standard Test Method for Measurement of Airborne Sound Attenuation between Rooms in Budlings. The spaces were designed by an acoustical consultant and desired by the owner to have extremely high levels of sound isolation. Furthermore, the nature of the intended clientele necessitated high performance, a high finish level, and state of the art technology. The investigators were not part of the design team and were invited to measure the space to document and validate the final performance. This presentation will outline the findings, challenges in measurement, and the steps taken to increase the sound isolation performance.

**1:40**

**1pAA2. Case study of the acoustic design of on-air broadcast studios and a performance space in a historic building in Franklin, TN with other tenants.** Steven D. Pettyjohn (The Acoust. & Vib. Group, Inc., 5765 9th Ave., Sacramento, CA 95820-1816, spettyjohn@acousticsandvibration.com)

A historic building called 5th Square in Franklin Tennessee was renovated to provide space for a Museum of the Bible Radio facility. The renovation had begun and other tenants were preparing to move in such as law firms. This is a two story building with the on the 2nd floor. Two on-air radio studios, control room and a capture studio (performance area) were the main spaces. A new HVAC system was designed based on variable refrigerant. An analysis of this system with recommendations for designing the system to meet acoustic and vibration goals was included. The Capture Studio would be used for live performance of bands that would be streamed on-air. A design was required that would prevent the transmission to law offices while providing good acoustical results in the space. The contractor and owner of the historic building requested a field visit to show them how to install the sound rated walls, deal with existing concrete masonry walls, deal with repairing the walls and sealing around the walls to achieve the sound transmission loss goals. A review of the doors and windows was made, and the installation of the mechanical equipment was reviewed and recommendations made during the visit.

**2:00**

**1pAA3. The challenge and solutions to providing consistent masking sound levels in individual rooms.** Viken Koukounian (K.R. Moeller Assoc. Ltd., 3-1050 Pachino Court, Burlington, ON L7L 6B9, Canada, viken@logison.com)

The propagation of sound in the built environment—impacted by size and geometry of a space, finishings, furnishings, and fit-outs—is exceptionally complex. It is for this reason that experts in acoustics need to consider all features of all spaces, despite the availability of prescriptive tools—e.g., categorization, acceptable-level and reverberation time [design] schemes—which are insufficient towards assuring any acoustical outcome. The use of a sound masking system is one acoustical solution that can be used to meet acoustical objectives, such as acoustical comfort and speech privacy, by reducing the variability in temporal, spectral, and spatial properties of sound within a space. The following investigation utilizes commissioning data of sound masking systems to underline the need for dedicated control zones for individual rooms. More specifically, the overall and one-third octave band levels of masking sound in rooms within the same control zone are compared against the project's specifications (i.e., the target overall level and the reference masking spectrum). These results enforce opportunities to improve the performance of a sound masking system to realize more consistent perceptual (i.e., acoustical comfort and speech privacy) outcome.

**2:20**

**1pAA4. Evaluating the reverberation chamber as a small room: How room dimensions affect the generalization of testing results.** Evelyn Way (Res. & Development, Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, evelynway@gmail.com)

Testing acoustic properties is performed in specially designed reverberation chambers according to relevant ASTM, ISO, and other international standards. Performing this testing is complicated by the fact that practical considerations limit the size of the reverberation rooms. Where the dimensions of the room are similar in size to the wavelength of the frequency of interest, modal behavior becomes dominant, and the statistical analysis of room acoustics based on the diffuse sound field theory is not sufficient to characterize the sound field. However, standard methods implement formulas which assume a diffuse sound field to compare results from different labs. Using the Maxon Acoustics Lab, a purpose-built floor ceiling test lab comprised of two stacked, 300 m3 reverberation chambers, we will examine the various physical criteria for evaluating a theoretically diffuse sound field. Discussion will include historical debates concerning the validity of comparing measurements in different labs and an analysis of the Maxxon Lab through the lens of various methods for evaluating the acoustic environment in comparison to the statistical analysis for room acoustics. Methods will include analysis of modal density and distribution, sampling of the sound field per ASTM standards E90, E492, and C423, the Schroeder frequency, and other statistical analyses.

**2:40**

**1pAA5. Wave acoustic solutions in small rooms.** Peter D'Antonio (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com)

This presentation will discuss the growing application of wave based solutions due to the increased processing power of desktop and distributed cloud cluster computing. The causes and mitigation of four types of potential acoustical distortion in small rooms, below and above 200 Hz, will be presented. Below 200 Hz, approaches to minimize modal effects and the speaker boundary interference response will be presented. We will discuss limitations of traditional dimensional ratio determinations, more accurate wave-based predictions, importance of speaker and listener locations, and prediction and measurement of low frequency absorption devices. Above 200 Hz, comb filtering from strong specular reflections and poor diffusion are potential problems. We will describe the necessary ratio of surface width to incident wavelength for a specular reflection and its effect on comb filtering. Finally, we describe the importance of creating a mixing room design and the use of broad bandwidth diffusive surfaces, which can be both measured and simulated with a virtual goniometer boundary element program called VIRGO, from a 3D CAD file topology of any shape. This offers the acoustician the ability to evaluate surface topologies that are of interest in a room design quickly and inexpensively.

## *Contributed Papers*

**3:00**

**1pAA6. Effect of spatial separation of sources on speech intelligibility in an automotive environment.** Linda Liang (College of Civil Eng. and Architecture, Guangxi Univ., No.100 Univ. East Rd., Nanning, Guangxi 530004, China, 1015307923@qq.com) and Guangzheng Yu (Acoust. Lab., School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China)

Speech intelligibility has been shown to be enhanced when the target speech source is spatially separated from the interferer, which is related to the spatial unmasking phenomenon. However, this issue has not been studied in the small acoustical space of an automobile. This study, thus, examines the effect of spatial separation of sources on the speech reception threshold (SRT) in an automobile, and compared with the results in a weak-reflective listening room. The target was always presented at the front-passenger seat, and the interferer was presented at the front-passenger seat, right-back seat, mid-back seat, and left-back seat in sequence. The stimuli were synthesized using convolution with binaural room impulse responses measured on a dummy head in driver seat under different interferer locations. Sentence SRTs in Mandarin Chinese were measured via headphones virtually in an automobile and a listening room. Accordingly, the spatial release from masking (SRM) was obtained based on the SRT result. Results show that the SRM in automobile is always smaller than that in listening room, because the early reflections and seat-back occlusions cannot only obscure the binaural cues related to the source localization such as the interaural-time-difference and interaural-level-difference, but also influence the target-to-interferer ratio.

**3:15**

**1pAA7. Influence of sound source directivity on finite element simulation of small-room acoustics.** Zhichao Zhang (School of Phys. and Optoelectronics, South China Univ. of Technol., Bldg. 18, No. 381 Wushan Rd., Tianhe District, Guangzhou, Guangdong 510641, China, 202020130104@mail.scut.edu.cn) and Guangzheng Yu (School of Phys. and Optoelectronics, South China Univ. of Technol., Guangzhou, China)

Numerical simulation is a flexible and effective method for room acoustic design. Full-range simulation of room acoustics requires a combination of different numerical methods, in which wave acoustic methods (WAM) and geometric acoustic methods (GAM) are used for the low and high frequency region, respectively. In the general low-frequency WAM simulation, a sound source is often assumed to be a point source or loudspeakers are usually approximated by circular planar pistons. However, compared to a large room, the critical frequency between WAM and GAM should be higher in acoustic simulation of small rooms because of relative size between the wavelength and the room, and thus, the above simplifications of actual sound sources, typically loudspeakers, may lead to errors in terms of directivity in the frequency range that the WAM is applied for. Further errors in the desired room impulse responses or other room acoustic parameters caused by directivity errors needs quantitative analysis, and then evaluation can be made on whether it is necessary to consider a more accurate sound source directivity in the WAM simulation of small room acoustics.

## Session 1pAO

# Acoustical Oceanography, Underwater Acoustics, and Animal Bioacoustics: Shelfbreak Acoustics II

Ying-Tsong Lin, Cochair
*Woods Hole Oceanographic Institution, 266 Woods Hole Road, Woods Hole, MA 02543*

Martin Siderius, Cochair
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Andone C. Lavery, Cochair
*AOPE, Woods Hole Oceanographic Institution, 98 Water Street, Woods Hole, MA 02543*

### *Contributed Paper*

**1:00**

**1pAO1. Evidence of 3D effects for along shelf acoustic propagation during SBCEX21.** Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanogr. Inst., 86 Water St., Falmouth, MA 02543, bdecourcy@whoi.edu) and Julien Bonnel (Woods Hole Oceanogr. Inst., Woods Hole, MA)

The 2021 Seabed Characterization Experiment (SBCEX21) was performed on the New England Shelf Break (NESB) in May 2021 with the main objective of characterizing the NESB sediment geoacoustic properties. In this talk, we focus on low-frequency propagation (f<100 Hz) along the 200 m isobath line. Time-frequency dispersion of normal modes is estimated using a SUS charge signal recorded on a distant single hydrophone TOSSIT system. Geoacoustic inversion is performed, but two-dimensional (2D) acoustic propagation models fail to capture the observed behavior of higher order acoustic modes, and predict much shorter travel times than are seen in the data. By introducing a three-dimensional (3D) model environment, both parabolic equation and modal ray calculations confirm that the gradual shelf slope has a nontrivial influence on modal travel times. This comparison between 2D and 3D simulations of SBCEX21 acoustics and experimental data emphasizes the importance of 3D effects from sloped bathymetry on understanding propagation in a shelfbreak environment. These effects will need to be accounted for when doing geoacoustic inversion of SBCEX21 along-shelf low-frequency data.

### *Invited Paper*

**1:15**

**1pAO2. Acoustic propagation and ambient sound on the Chukchi shelf.** Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, meganb@arlut.utexas.edu) and Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This paper presents a yearlong record of acoustic propagation and ambient sound collected on the Chukchi Shelf collected as part of a set of experiments known as the Canada Basin acoustic propagation experiment (CANAPE) and shallow-water CANAPE which took place from 2016 to 2017. Over the course of the yearlong experiment, the surface conditions transitioned from completely open water to fully ice-covered. The propagation conditions in the deep basin were dominated by a subsurface duct; however, over the slope and shelf, the duct was seen to significantly weaken during the winter and spring. The combination of these surface and subsurface conditions led to changes in the received level of the deep-water tomography sources that exceeded 60 dB. The ambient sound data were analyzed using k-means clustering to quantify the occurrence of six spectral shapes over the yearlong experiment. Each cluster type was associated with a different sound generation process based on the correlations with environmental observations. The cluster observed most frequently was associated with wind-generated sound, and the cluster with the smallest number of observations was attributed to wind effects on frazil ice forming in open leads during the ice-covered season. [Work sponsored by ONR.]

## *Contributed Papers*

**1:35**

**1pAO3. Coupled effects of nonlinear internal gravity waves and seabed properties on underwater sound propagation.** Tzu-Ting Chen (Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS-9, Woods Hole, MA 02543, pingpingting326@gmail.com), Ying-Tsong Lin (Woods Hole Oceanogr. Inst., Woods Hole, MA), Linus Y.-S. Chiu (National Sun Yat-sen Univ., Kaohsiung, Taiwan), and William L. Siegmann (Math Sci., RPI, Troy, NY)

Underwater sound propagation can be affected by strong sound speed gradients induced by nonlinear internal waves in the ocean. Meanwhile, geoacoustic properties of the seabed control acoustic reflections. Experimental data collected at the shelfbreak on the northeast of the South China Sea were analyzed to investigate the joint effects. Both two-dimensional (2D) and three-dimensional (3D) numerical sound propagation models with realistic seafloor and oceanographic inputs have been established to study the nonlinear internal wave effects observed on hydrophone vertical line array moorings. Compared to the 3D model, the prediction errors and uncertainties of the 2D scheme neglecting the transversal/azimuth energy propagation are discussed. In addition, the impact of sediment properties on 3D propagation effects affecting underwater soundscapes in the area is investigated. [Work supported by the Office of Naval Research.]

**1:50**

**1pAO4. Changes to mixed layer acoustic energy from spice and isopycnal tilt.** Edward Richards (Ocean Sci., UC Santa Cruz, Ocean Sci., 1156 High St., Santa Cruz, CA 95064, edwardlrichards@gmail.com) and John A. Colosi (Oceanogr., Naval Postgrad. School, Monterey, CA)

Mixed layer acoustic propagation is predicted for fields that approximate two dynamically separate sources of sound speed variation: the tilting of isopycnals and "spice," density compensated temperature and salinity changes. These fields are decomposed from a transect measurement over approximately 1000 km of the northeast Pacific Ocean. At a frequency with only one mixed layer mode, both fields contain blocking features, defined as sound speed structures less than 5 km wide that create significant acoustic energy loss. The transect shows two distinct regions where blocking features are caused by tilt or spice, indicating the acoustic importance of these dynamic processes depends on location. Statistics of mixed layer acoustic energy are investigated at two frequencies, averaging one or three mixed layer modes, both with and without blocking features. The spice field was found to produce higher loss than the tilt field. Both fields create more loss at lower frequencies when locations with blocking features were included, and more loss at higher frequencies without blocking features. Mixed layer mode amplitudes demonstrate more mixed layer modes at the higher frequency increases mode coupling, both preventing large scale acoustic energy loss and creating a more consistent loss mechanism.

## *Invited Paper*

**2:05**

**1pAO5. Twenty-first century outer continental shelf and shelfbreak acoustics research: Methods, tools, and progress.** Timothy F. Duda (Woods Hole Oceanogr. Inst., M.S. 11, Woods Hole, MA 02543, tduda@whoi.edu), Arthur E. Newhall, Ying-Tsong Lin (Woods Hole Oceanogr. Inst., Woods Hole, MA), James Lynch (Woods Hole Oceanogr. Inst., Falmouth, MA), Glen Gawarkiewicz, Weifeng G. Zhang, Karl R. Helfrich (Woods Hole Oceanogr. Inst., Woods Hole, MA), Pierre F. Lermusiaux (MIT, Cambridge, MA), Keith von der Heydt, and John Kemp (Woods Hole Oceanogr. Inst., Falmouth, MA)

New findings in outer-shelf and shelfbreak acoustics have been enabled by experimental and computational advances over the last 25 years. The details of sound field variability caused by highly dynamic conditions often found in this regime have become measurable through advances in data collection technology. Furthermore, these details can now be computationally modeled more realistically. The coupling of more plentiful data and higher fidelity modeling has uncovered many new behaviors. It has also allowed us to quantify the effects on sound level and phase structure (coherence) of many outer-shelf physical features, as well as the temporal aspects of these variations. Key tools have been vessel dynamic positioning, underwater position finding, small mobile platforms, high-capacity multichannel receive arrays, data-assimilating regional ocean dynamical models, nonlinear wave modeling, and three-dimensional acoustic propagation modeling. Examples from the published work of the authors, and the work of others, of how these advances have fostered new knowledge of specific processes will be presented, as well as present-day challenges inspired by recent findings.

## *Contributed Papers*

**2:25**

**1pAO6. Influence of the vertical resolution when simulating ocean sound speed variability in mesoscale eddies.** Richard X. Touret (Ocean Sci. and Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, office 131, Atlanta, GA 30332, rtouret@gatech.edu), Guangpeng Liu (Oceanogr., Univ. of Hawaii, Honolulu, HI), Annalisa Bracco (Earth and Atmospheric Sci., Georgia Inst. of Technol., Atlanta, GA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Vertical resolution affects the representation of ocean sound speed according to a suite of regional simulations of the De Soto Canyon circulation in the Gulf of Mexico. Simulations have identical horizontal resolution of 0.5 km, partially resolving submesoscale dynamics, and increasing vertical resolution from 30 (i.e., comparable to what commonly used in mesoscale permitting or resolving hindcast and forecast products such as HYCOM) to 200 terrain-following layers. Simulations with 30- and 70-layers underestimate the ageostrophic contributions in and around the eddies below the mixed-layer and do not reproduce the sharp vorticity and density variations associated with the mesoscale circulations compared to the 140- and 200-layers runs. The ocean sound speed (based on the classical MacKenzie formula) was found to be far more variable when the submesoscale, ageostrophic circulations are captured also in their vertical structure and vertical contributions to the density field. Hence, the results of this study indicate that to better predict the influence of the submesocale oceanic circulation on ocean sound speed variability, model simulations should consider enhancing both horizontal and vertical resolution to resolve at least the first 3 baroclinic modes. To do so, more than 100 vertical layers were found to be needed in this study.

2:40

**1pAO7. Quantifying the influence of the vertical resolution in ocean circulation modeling on the acoustic propagation through mesoscale eddies.** Matthew McKinley (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30318, mmckinley31@gatech.edu), Richard X. Touret (Ocean Sci. and Eng., Georgia Inst. of Technol., Atlanta, GA), Guangpeng Liu (Dept. of Oceanogr., Univ. of Hawaii at Manoa, Honolulu, HI), Annalisa Bracco (School of Earth and Atmospheric Sci., Georgia Inst. of Technol., Atlanta, GA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Accurate numerical simulations of underwater acoustic propagation in a dynamic ocean—and its associated uncertainty—require using realistic environmental parameters as inputs and especially a high-fidelity representation of the expected spatio-temporal variability of the ocean sound speed in the volume of interest. In areas characterized by strong temperature and salinity variations (e.g., associated with long-living mesoscale eddies in the Gulf of Mexico), the approximate simulation of the 3D sound-speed field and its variability requires predictive oceanographic models capable of resolving such variations. This study investigates the impact of vertical resolution focusing on how it shapes the representation of the 3D sound speed variability through a suite of simulations of the northern Gulf of Mexico performed with a regional ocean model run at submesoscale permitting horizontal resolution (0.5 km) using increasing vertical resolution from 30 to 200 terrain-following layers over a one-month simulation interval (as described in the companion paper presented by Touret *et al.*). Geo-acoustic parameters were matched to the existing sediment database. In selected areas influenced by mesoscale eddies, ray tracing is used to determine the significance of increased resolution on acoustic propagation as a function of the sensor configurations and the expected sound speed variability.

2:55

**1pAO8. Effects of modelling assumptions of complex ocean floor topography on marine life.** James Crowcombe (Defence Sci. and Technol. Lab., Dstl Porton Down, Salisbury SP4 0JQ, United Kingdom, jcrowcombe@dstl.gov.uk), Tim Clarke (Defence Sci. and Technol. Lab., Salisbury, United Kingdom), Yvonne Mather (Defence Sci. and Technol. Lab., Fareham, United Kingdom), and Duncan Williams (Defence Sci. and Technol. Lab., Salisbury, United Kingdom)

Passive acoustic monitoring (PAM) is an important technique to assess the presence of marine mammals and, if necessary, mitigate the effects of anthropogenic noise sources upon them. The complexity of the ocean acoustic environment makes accurate localisation of marine mammal vocalisations difficult. It is, therefore, important to be able to predict the detection and localisation performance of PAM to ensure sensors are optimally placed and any resulting actions are suitably informed. A variety of acoustic models can be used to predict the performance of a sensor field ranging from simple 1D models up to full 3D models. Each type of model has advantages and disadvantages. Simple 1D models offer the most advantages in terms of solution time and minimal inputs but their predictions can have reduced accuracy. This is especially true in areas of complex ocean floor topography, such as on a shelf break, where complex 3D acoustic effects can occur. This study investigates the variation in the predicted acoustic field from 1D up to 3D models on a shelf break and how with increasing model fidelity the picture of the acoustic environment changes and its impact on acoustic monitoring advice. 

3:10

**1pAO9. Using a source of opportunity for self-localization of mobile underwater vector sensor platforms.** Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Transforming a network of few compact mobile underwater platforms, each equipped with a single acoustic sensor, into a distributed sensing array requires precise positioning of each mobile sensor. But conventional accurate underwater positioning tools rely on active acoustic sources (e.g., acoustic pingers), which imposes additional hardware and operational complexity. Hence self-localization (i.e., totally passive) methods using only acoustic sources of opportunity (such as surface vessels) for locating the mobile sensors of a distributed array appear as an attractive alternative. Existing underwater self-localization methods have mainly been developed for mobile platform equipped with time-synchronized hydrophones and rely only on the time-difference of arrivals between multiple pairwise combinations of the mobile hydrophones as inputs for a complex non-linear inversion procedure. Instead, we introduce a self-localization method, which uses a linear least square formulation, for two mobile time-synchronized vector sensor platforms based on their acoustic recordings of a distant surface vessel and their inertial navigation systems (INS) measurements. This method can be generalized to multiple vector sensor pairs to provide additional robustness towards input parameter errors (e.g., due to a faulty INS) as demonstrated experimentally using drifting buoys with inertial vector sensors deployed $\sim$100 m apart in shallow water.

3:25–3:40

**Panel Discussion**

## Session 1pBA

## Biomedical Acoustics: General Topics in Biomedical Acoustics I: Assessment of Tissue Material Properties

John M. Cormack, Cochair
*Center for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Institute, Department of Medicine, University of Pittsburgh Medical Center, Pittsburgh, PA 15261*

Matthew W. Urban, Cochair
*Department of Radiology, Mayo Clinic, Department of Radiology, Mayo Clinic, Rochester, MN 55905*

### *Contributed Papers*

**1:00**

**1pBA1. Surface excitation of focused shear wave beams in soft elastic media: Theory.** Branch T. Archer (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Yu-Hsuan Chao (Bioengineering Dept., Univ. of Pittsburgh, Pittsburgh, PA), John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Inst., Dept. of Medicine, Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA), Kang Kim (Bioengineering Dept., Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Shear wave propagation is employed in medical ultrasound imaging, because it reveals variation in the viscoelastic properties of tissue. Frequencies below 1 kHz are required for imaging with shear waves in soft tissue due to their high attenuation and low propagation speeds, compared to compressional waves with frequencies above 1 MHz used for ultrasound imaging. Shear waves exhibiting particle motion in the direction of propagation, referred to as longitudinally polarized shear waves, can be generated by applying longitudinal motion of a circular disk to the surface of a soft elastic medium. This approach is used in practice because it permits imaging of the longitudinal shear wave with a conventional ultrasound transducer that is coaxial with the source of the shear wave. Presented here are the theoretical framework and numerical simulations that illustrate effects of focusing on longitudinally polarized shear waves. Longitudinal, transverse, radial, and torsional source polarizations are considered. The present investigation was motivated initially by an experimental study in optics due to Dorn *et al.* [Phys. Rev. Lett. **91**, 233901 (2003)]. Our predictions for shear wave beams support their measurements of light beams revealing that the longitudinal electric field component produces a smaller focal spot than the transverse field component.

**1:15**

**1pBA2. Surface excitation of focused shear wave beams in soft elastic media: Experiment.** Yu-Hsuan Chao (Bioengineering Dept., Univ. of Pittsburgh, Pittsburgh, PA), Branch T. Archer (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), John M. Cormack (Ctr. for Ultrasound Molecular Imaging and Therapeutics, and Vascular Medicine Inst., Dept. of Medicine, Univ. of Pittsburgh Medical Ctr., Pittsburgh, PA 15261, jmc345@pitt.edu), Kang Kim (Bioengineering Dept., Univ. of Pittsburgh, Pittsburgh, PA), Kyle S. Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Shear waves are employed in medical imaging to reveal variations in the viscoelastic properties of soft tissues, which are useful biomarkers for pathologies such as breast lesions and liver disease. Shear wave excitation methods that employ acoustic radiation force or surface vibration with a small piston have limitations associated with imaging depth and shear wave amplitude. We introduce a new method for surface excitation of shear waves that employs longitudinal motion of a concave-shaped piston source to generate a focused shear wave beam, thereby increasing shear wave amplitude and penetration depth. Focused shear waves are excited in a gel phantom in the frequency range of 200 Hz to 400 Hz using a concave piston with 40 mm radius of curvature and 50 mm diameter. The wave field comprises both transverse and longitudinal displacement components, which are polarized perpendicular and parallel to the propagation direction, respectively. Transversely and longitudinally polarized wave fields are measured using ultrasound speckle tracking. The longitudinally polarized shear wave is of interest because it is measured in elastography applications. Preliminary comparisons exhibit good agreement between measured beam patterns and those predicted by an analytical theory for shear wave beam propagation due to surface excitation with a piston source.

**1:30**

**1pBA3. Nonlinear least-squares estimation of shear elasticity and shear viscosity in viscoelastic media.** Nicholas A. Bannon (Michigan State Univ., Dept. of Elec. and Comput. Enginiering, Michigan State Univ., East Lansing, MI 48824, bannonni@msu.edu), Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN), and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Rapid simulations of three-dimensional (3D) shear wave propagation in viscoelastic media with a Kelvin-Voigt model are enabled by Green's functions accelerated with graphics processing unit parallelization and high-performance computing resources. Cross-correlation analysis of the wave motion yields an estimate of shear elasticity, where the errors in the estimated values increase with rising shear viscosity. To reduce errors in the shear elasticity estimates, a nonlinear least-squares routine that accounts for the effects of propagation, attenuation, and dispersion is applied to 3D simulated shear wave data in viscoelastic media. The nonlinear least-squares approach also yields an estimate of the shear viscosity when applied to simulated particle velocity waveforms. Post-processing, cross-correlation analysis, and the nonlinear least-squares routine are evaluated on a desktop computer using MATLAB. Estimation of viscoelastic parameters is performed at the focal depth for several combinations of shear elasticity and shear viscosity. The errors in the estimated shear elasticities obtained from the nonlinear least-squares routine and cross-correlation are determined as a function of the input shear elasticity and shear viscosity values. The estimates provided by the nonlinear least-squares routine for shear elasticity and shear viscosity are shown to approach the simulation input values for each combination of viscoelastic parameters.

1:45

**1pBA4. Simulations of shear wave propagation in tissue based on kidney transplant histology.** Nicholas M. Sanger (College of Eng., North Dakota State Univ., 1315 Centennial Blvd., Fargo, ND 58105, nicholas.sanger@ndsu.edu), Luiz Vasconcelos, and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

End-stage kidney disease is the result of progressive declines in function and frequently requires a transplant. Maintaining kidney transplant health with regularly scheduled function tests or biopsies is necessary to adjust treatments. Ultrasound shear wave elastography (SWE) could provide an inexpensive, noninvasive modality to monitor allograft health. In this project, we built models of the kidney cortex from segmented kidney transplant histology images. We assigned viscoelastic properties, based on a Kelvin-Voigt model, to the major constituents, glomeruli, interstitia, tubules, and fluid and simulated the wave propagation through tissue with no disease, inflammation, interstitial fibrosis, and tubular atrophy. The resulting *in silico* wave motion in the time- and frequency-domains was characterized to compare the *in silico* and *in vivo* mechanical properties. We also attempted modeling the effects of perfusion by modifying the constituents' viscoelastic properties. We found the most differentiation for shear elasticity, shear viscosity, and phase velocity dispersion in the inflammation patients. When accounting for perfusion we observed increases in shear wave group velocity, shear viscosity, and dispersion, bringing the results from *in silico* models in closer agreement with *in vivo* results. These results are encouraging and demonstrate first steps of simulating the shear wave propagation based on histological models.

2:00–2:15 Break

2:15

**1pBA5. An ultrasonic bone assessment technique based on the spectral difference of signals from cortical and cancellous bone: an *in vivo* study.** Ann M. Viano (Phys., Rhodes College, 2000 N Parkway, Memphis, TN 38112, viano@rhodes.edu), Sarah I. Delahunt (Phys., Rhodes College, Austin, TX), and Brent K. Hoffmeister (Phys., Rhodes College, Memphis, TN)

Ultrasonic backscatter may be able to detect changes in bone caused by osteoporosis. This study assesses the relative *in vivo* performance of three ultrasonic backscatter parameters: cortical-cancellous mean (CCM), cortical-cancellous slope (CCS), and cortical-cancellous intercept (CCI). Measurements were performed at the left and right femoral necks of 88 healthy volunteers. CCM was determined by frequency averaging the spectral power difference in decibels between echoes from the outer cortical surface and backscatter from underlying cancellous bone. CCS and CCI were determined from the slope and intercept, respectively, of a line fitted to the spectral power difference. Linear regression analysis was used to compare measurements performed at the left and right femur. CCM demonstrated highly significant ($p < 0.0001$) correlations between left and right side measurements across nine choices of gate delay and width for analysis of both ten and thirty adjacent signals centered on the bone ($0.41 < R < 0.53$). CCS and CCI showed significant correlations only for five and three gate choices, respectively, for the ten signal analysis ($0.001 < p < 0.05$). These results indicate that CCM is sensitive to naturally occurring variations in bone tissue and may be sensitive to changes in bone tissue caused by osteoporosis.

2:30

**1pBA6. Ultrasonic properties of human scalp.** Cecille Labuda (Phys. and Astronomy, Univ. of MS, 108 Lewis Hall, University, MS 38677, cpembert@olemiss.edu), Blake C. Lawler, Shona C. Harbert, and Brent K. Hoffmeister (Phys., Rhodes College, Memphis, TN)

The goal of this study was to characterize the ultrasonic properties of human scalp. Thirty-two specimens were prepared from formalin-fixed scalp tissue from four human donors (age 35–65, 2 male, 2 female). Tissue specimens were mounted in acrylic frames with a $30 \times 30$-mm acoustic window. The specimens were scanned in a water tank with a broadband ultrasound transducer with center frequency 7.5 MHz using a motion-controlled system. The ultrasonic region of interest (ROI) was a $20 \times 20$-mm region with a step size of 410 $\mu$m. The signals acquired were analyzed to determine the speed of sound (SOS) and frequency slope of attenuation (FSA) at all scan locations (2500 locations per specimen). The mean $\pm$ standard deviation (SD) of the SOS and FSA over all the specimens respectively was $1536 \pm 9$ m/s and $1.88 \pm 0.51$ dBcm-1MHz-1. The SD within an ROI was 3–15 m/s for SOS and 0.15–0.85 dBcm-1MHz-1 for FSA, depending on specimen, indicating some heterogeneity in the ultrasonic properties of the tissue.

2:45

**1pBA7. Comparison of arterial mechanical properties measured with arterial dispersion ultrasound vibrometry and clinical arterial stiffness metrics.** Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN 55905, urban.matthew@mayo.edu), Hyungkyi Lee (Dept. of Radiology, Mayo Clinic, Rochester, MN), Tuhin Roy (Civil Eng., North Carolina State Univ., Raleigh, NC), Charles Capron (Mayo Clinic Graduate School of Biomedical Sci., Mayo Clinic, Rochester, MN), Francisco Lopez-Jimenez (Dept. of Cardiovascular Medicine, Mayo Clinic, Rochester, MN), Gina Hesley (Dept. of Radiology, Mayo Clinic, Rochester, MN), Wilkins Aquino (Duke Univ., Durham, NC), Murthy Guddati (Civil Eng., North Carolina State Univ., Raleigh, NC), and James Greenleaf (Dept. of Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Arterial stiffness is recognized as a highly clinically relevant and independent prognostic biomarker of cardiovascular disease. Arterial dispersion ultrasound vibrometry (ADUV) uses acoustic radiation force to generate propagating waves in the arterial wall and measures the wave motion to estimate arterial mechanical properties from fitting phase velocity dispersion curves to a waveguide model. ADUV measurements were made with a Verasonics V1 system with a L7-4 linear array transducer in 63 human subjects (27 male/36 female) from ages 50–88 years old. We made measurements at 10 time points within the cardiac cycle, termed cardiac stages. We estimated the phase velocity (median of $c_p$ from 400-600 Hz) and $G$ determined from matching the dispersion curves with those from a waveguide model incorporating the artery diameter and wall thickness. Within the same study visit, we measured the intima-media thickness and diameter changes through the cardiac cycle with a General Electric Logiq E9 or E10 scanner. We also tonometry to determine the carotid-femoral pulse wave velocity. We report here comparisons of median phase velocity, $c_p$, and inverted shear modulus, $G$, at different stages of the cardiac cycle with clinical metrics of arterial stiffness.

# Session 1pCA

## Computational Acoustics, Biomedical Acoustics, and Signal Processing in Acoustics: Physics-Informed Artificial Intelligence/Machine Learning for Acoustics (Hybrid Session)

Amanda Hanford, Cochair
*Penn State University, PO Box 30, State College, PA 16802*

Matthew G. Blevins, Cochair
*U.S. Army Engineer Research and Development Center, 2902 Newmark Drive, Champaign, IL 61822*

Brandon M. Lee, Cochair
*Mechanical Engineering, University of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109*

**Chair's Introduction—1:00**

### *Invited Papers*

**1:05**

**1pCA1. Predictive capability assessment for physics-guided learning of vortex-induced vibrations.** Gregory A. Banyay (Appl. Res. Lab., Penn State Univ., PO Box 30, State College, PA 16802, gab5631@psu.edu) and Andrew S. Wixom (Appl. Res. Lab., Penn State Univ., State College, PA)

We seek here a computationally parsimonious and credible means to simulate the complex phenomena of vortex-induced vibrations (ViV), as one tool to assist in mitigating risk associated with ViV-induced instabilities that can cause non-negligible structural acoustic response. To address current limitations in data-driven modeling, for which credibility assessment proves challenging, or physics-based simulation (i.e., constrained by governing partial differential equations (PDEs)), which often includes prohibitive computational expense, we explore recent state-of-the-art approaches to optimally combine these engineering disciplines via a physics-guided machine learning framework. One can expect that intersecting data-driven modeling with physics-guided simulation offers one means to both maximize the credibility of machine learning based approaches and minimize the computational expense of physics-based modeling approaches.

**1:25**

**1pCA2. A physics-guided model for predicting spectral and temporal variability of road traffic noise.** Mylan R. Cook (Dept. of Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com), Kent L. Gee, Mark K. Transtrum (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Shane V. Lympany, and Matthew F. Calton (Blue Ridge Res. and Consulting, Asheville, NC)

The National Transportation Noise Map (NTNM) provides daily averaged A-weighted equivalent sound levels across the continental United States (CONUS) due to road traffic. The NTNM maps the spatial variability of road traffic noise, but not the temporal or spectral variability. A physics-guided model was developed to predict the temporal and spectral variability of road traffic noise across CONUS. Empirical models were developed to predict hourly road traffic volume and vehicle class mix across CONUS based on publicly available traffic volume measurements and geospatial data. The Federal Highway Administration's Traffic Noise Model characterizes average spectral levels by vehicle class; by combining the empirical model with the Traffic Noise Model's characteristic vehicle class spectra, hourly traffic noise predictions across CONUS are made which include temporal and spectral variability. This physics-based modeling approach improves upon nation-wide mapping of road traffic noise. [Work supported by U.S. Army SBIR.]

**1:45**

**1pCA3. Mapping ambient sound levels using physics-informed machine learning.** Shane V. Lympany (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, shane.lympany@blueridgeresearch.com), Matthew F. Calton (Blue Ridge Res. and Consulting, Asheville, NC), Mylan R. Cook, Kent L. Gee, and Mark K. Transtrum (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Mapping the spatial and temporal distribution of ambient sound levels is critical for understanding the impacts of natural sounds and noise pollution on humans and the environment. Previously, ambient sound levels have been predicted using either machine learning or physics-based modeling. Machine learning models have been trained on acoustical measurements at geospatially diverse locations to predict ambient sound levels across the world based on geospatial features. However, machine learning requires a large number of

acoustical measurements to predict ambient sound levels at high spatial and temporal resolution. Physics-based models have been applied to predict transportation noise at high spatial and temporal resolution on regional scales, but these predictions do not include other anthropogenic, biological, or geophysical sound sources. In this work, physics-based predictions of transportation noise are combined with machine learning models to predict ambient sound levels at high spatial and temporal resolution across the conterminous United States. The physics-based predictions of transportation noise are incorporated into the machine learning models as a geospatial feature. The result is a physics-informed machine learning model that predicts ambient sound levels at high spatial and temporal resolution across the United States. [Work funded by an Army SBIR]

## *Contributed Papers*



**2:05**

**1pCA4. Training physics-informed neural networks to directly predict acoustic field values in simple environments.** Brandon M. Lee (Mech. Eng., Univ. of Michigan, 1231 Beal Ave. Ann Arbor, MI 48109, leebm@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

As acousticians turn to machine learning for solutions to old and new problems, neural networks have become a go-to tool due to their capacity for model representation and quick forward computations. However, these benefits come at the cost of obscurity; it is difficult to determine whether the proficiency of a trained neural network is limited by training effort, training dataset size or scope, or compatibility of the network's design with the data's underlying pattern of interest. For neural networks trained to provide solutions to the point-source Helmholtz-equation in axisymmetric single-path, two-path, and multi-path (ideal waveguide) environments with constant sound speed, the key limitations are the dataset composition and network design. This study examines the effects on performance and explainablity which result from providing physical information (governing equation and boundary conditions) to these neural networks, instead of only acoustic-field solutions generated from well-known analytic solutions. The outcome of using physics-informed neural networks (PINNs) for these simple environments informs their possible extension to more complex, realistic environments. This study emphasizes source frequencies in the 100's of Hz, depths up to 500 m, and ranges up to 10 km for sound speeds near 1500 m/s. [Work supported by the NDSEG fellowship program.]

**2:20–2:35 Break**

**2:35**

**1pCA5. Spectral-based cluster analysis of noise from collegiate sporting events.** Mitchell C. Cutler (Phys. and Astronomy, Brigham Young Univ., 1 Campus Dr., Provo, UT 84604, mitchellccutler@gmail.com), Mylan R. Cook, Mark K. Transtrum, and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

This paper presents on studies to characterize crowd response from recordings taken from 30 collegiate sporting events. Inferring crowd response from raw acoustic signals is challenging because they contain complex combinations of acoustic sources including crowd noise, music, individual voices, and PA system. First, the distributions of recorded half-second interval overall sound pressure levels from basketball and volleyball games, both men's and women's, are analyzed with regard to crowd size and venue. Using 24 one-third octave bands between 50 Hz and 10 kHz, half-second spectral levels from each type of game are then analyzed. Based on principal component analysis, 87% of the spectral variation in the signals can be represented with three principal components, regardless of sport, venue, or crowd composition. Using the resulting three-dimensional component coefficient representation, clustering analysis (using Gaussian mixture models) then finds nine different clusters. These clusters separate audibly distinct signals and represent various combinations of acoustic sources, including crowd noise, music, individual voices, and PA system.

**2:50**

**1pCA6. Challenges in siting a highway noise barrier with constraints of cost, zoning, and wetlands management.** Arthur W. van der Harten (Acoust., Acoust. Distinctions / Open Res. in Acoust. Sci. and Education, 400 Main St. Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com)

A noise barrier has been planned to reduce the impact of highway noise on a private property with restrictions on construction due to wetlands management, and stringent building code requirements, with respect to cost. Using the constraints of the site and local ordinances, an application of a genetic algorithm was designed in order to find the optimal barrier placement that complied with all site constraints. Fitness criteria were developed according to the area of the constructed barrier, as well as the amount of attenuation provided according to a Maekawa based screen attenuation calculation determined in a 3-D model of the terrain.

**3:05**

**1pCA7. Learning both a value and uncertainty label for seabed parameters from ship noise data.** Michael C. Mortenson (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX)

Because the seabed impacts sound propagation in the ocean, machine learning is being used for both seabed classification and to obtain estimates of individual seabed properties. This paper proposes a method to simultaneously estimate these properties and an associated uncertainty label. A residual neural network is trained and validated using synthetic ship noise spectrograms generated with a range-independent normal mode sound propagation model and a ship noise source spectrum. The data set includes 140 seabeds: In each, the top sediment layer has a random thickness and properties randomly chosen from five sets of bounds, which roughly correspond to clay, mud, sand, silt, and gravel. For each of the 140 sediments, a random selection of 405 combinations of ship speed, closest-point-of-approach range, and source depth are used resulting in 22k data samples. Each data sample is labeled with the true values of the sediment parameters as well as a label for the uncertainty level of each. The uncertainty levels are obtained from the Fisher information and qualify of the information content in the ship spectrogram about each parameter. Examples of how the residual neural network learns to perform regression for the parameter value and the uncertainty level will be presented.

**3:20**

**1pCA8. Pneumonia diagnosis algorithm based on room impulse responses using cough sounds.** Jin Yong Jeon (Medical and Digital Eng., Hanyang Univ., Dept. of Architectural Eng., Hanyang University, Seoul 04763, South Korea, jyjeon@hanyang.ac.kr), Na Young Kim (Medical and Digital Eng., Hanyang Univ., Seoul, South Korea), Sang Heon Kim (Internal Medicine, Hanyang Univ., Seoul, South Korea), Hong Jin Kim (Management Support, HicareNet, Seoul, South Korea), and Gwun Il Park (Business Innovation, HicareNet, Seoul, South Korea)

For data augmentation of the pneumonia diagnosis algorithm by deep learning, an image conversion method through a convolutional method utilizing spatial impulse and sound quality factors of cough sounds is proposed. First, reverberant spaces with different volumes were implemented,

and spatial impulse responses were generated for each space through computer simulation of spatial models according to sound source and receiver points. Sound quality analysis was performed to improve accuracy, and 2-D sound quality data of time series was converted into 3-D image data using the Gramian Angular Field (GAF) method for combination between heterogeneous data. As a result, 97.5% accuracy was obtained for the configured dataset. The result of this study is expected to be used to develop diagnostic algorithms for various respiratory diseases including pneumonia in the future.

**3:35**

**1pCA9. The feces thesis: Using machine learning to detect diarrhea.** Maia Gatlin (Georgia Inst. of Technol., 7220 Richardson Rd., Smyrna, GA 30080, maia.gatlin@gtri.gatech.edu), David S. Ancalle (Georgia Inst. of Technol., Atlanta, GA), Anthony Popa (Georgia Inst. of Technol., Smyrna, GA), Ashima Taneja, Cade Tyler (Georgia Inst. of Technol., Atlanta, GA), David Meyer (Univ. of Toronto, Toronto, ON, Canada), David L. Hu, and Alexis Noel (Georgia Inst. of Technol., Atlanta, GA)

Cholera is a bacterial disease which induces extremely liquid diarrhea. It affects millions of people, resulting in up to about 150,000 deaths per year. In this study, a sensor is developed which can non-invasively determine if an outbreak may occur in an area, acting as an early detection method so that resources can be employed to stop the rapid spread of the disease. The sensor uses a microphone to collect audio samples of various excretion events. The collected acoustic data are pre-processed to produce mel spectrograms which capture the distinct temporal frequency characteristics of each excretion event. These mel spectrograms are input into a pre-trained convolutional neural network to classify the event as either diarrheal or non-diarrheal with up to 98.1% accuracy. The algorithm is also capable of classifying other excretion events such as urination, flatulence, and defecation. The sensor developed here could be applied to identify other use cases such as tracking bowel movements for hospice patients or for those with inflammatory bowel diseases like Crohn's disease.

MONDAY AFTERNOON, 5 DECEMBER 2022 — LIONEL, 1:00 P.M. TO 3:30 P.M.

## Session 1pEA

## Engineering Acoustics: General Topics in Engineering Acoustics

Pratik Ambekar, Chair
*Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105*

### *Contributed Papers*

**1:00**

**1pEA1. Low-cost measurement-grade microphone powered by MEMS elements and preamplifier housed in 3D printed enclosure.** Peter J. Riccardi (Graduate Program in Acoust., The Pennsylvania State Univ., University Park, PA), Zane T. Rusk (Dept. of Architectural Eng., The Penn State Univ., 104 Eng. Unit A, University Park, PA 16802, ztr4@psu.edu), John A. Case, Heui Young Park (Graduate Program in Acoust., The Pennsylvania State Univ., University Park, PA), Eric Rokni (Graduate Program in Acoust., The Pennsylvania State Univ., State College, PA), and Stephen C. Thompson (Graduate Program in Acoust., The Pennsylvania State Univ., University Park, PA)

Acoustic measurement-grade microphones with flat frequency responses and adequate sensitivities are a necessary tool for many acousticians and vibroacoustic engineers. These microphones can often cost hundreds, if not thousands, of dollars. With the availability of microelectromechanical systems (MEMs) microphone elements and 3D printers, it is possible to construct drop-in replacements of these measurement grade microphones at the fraction of the cost. A MEMs system was designed with four elements in parallel to reduce uncorrelated noise. The system runs rail-to-rail on a 3.3VDC, Integrated Electronics Piezo-Electric (IEPE) powered preamplifier, producing a nominal sensitivity of 13 mV/Pa. The microphone body is 3D printed, and the final bill of material cost is less than 30 USD. The performance of the microphone was measured experimentally in a semi-anechoic chamber and verified by comparing the data to measurements of a commercial microphone: the PCB Piezotronics 378B02 ½" free-field microphone with a nominal sensitivity of 50 mV/Pa. This presentation will show that the MEMs microphone has a similar electrical noise floor, and lower acoustical noise floor below 1 kHz. The frequency response of the MEMs microphone is flat in the audio band through 10 kHz, making it a suitable replacement for acoustical measurements where its lower sensitivity can be tolerated.

**1:15**

**1pEA2. MEMS microphones as ultrasonic transducers.** Xiaoyu Niu (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX, xyniu@utexas.edu), Yuqi Meng, Zihuan Liu, Ehsan Vatankhah, and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

We demonstrate the transmission of ultrasound in air using a transducer that resembles a MEMS microphone in its construction. The device comprises a compliant 1 mm diameter diaphragm, a stiff perforated backplate electrode, and a back-volume. The diaphragm is driven using AC signals with peak values that exceed the pull-in voltage of the diaphragm. Relatively large diaphragm displacements are achieved as diaphragm oscillations

traverse the complete 2.30-micrometer diaphragm-backplate gap in response to excitation waveforms spanning from 40 kHz to 150 kHz. Large amplitude diaphragm vibration is advantageous for high SPL applications in air, as sound pressure is directly proportional to diaphragm displacement for a given operating frequency. Diaphragm vibration profiles are measured using a scanning laser Doppler vibrometer, and resultant acoustic pressure waveforms in air are measured using a broadband microphone. We demonstrate how nonlinear features of the electrostatic transducer can be exploited to generate loud, broadband signals. We also discuss interesting applications using an array of these transducers.

**1:30**

**1pEA3. Broadband multimode piezoelectric spherical shell transducers for underwater sound.** David A. Brown (ECE, Univ. Massachusetts Dartmouth, 151 Martine St., Ste. 123, Fall River, MA 027230000, dbrown@umassd.edu), Austin Souza, Corey Bachand, and Michael Bisbano (BTech Acoust., Fall River, MA)

Broadband multimode piezoelectric spherical shell transducers are investigated for underwater sound by exciting the zeroth and/or first order $(0+1)$ extensional modes of vibration with and without acoustical baffles of different types. Measurements of the frequency response (transmit pressure per volt or TVR), radiation patterns and electromechanical impedance reveal useful tradeoffs and design guidance for many practical applications. Measurements are in good agreement with theoretical closed form analytical results.

**1:45**

**1pEA4. A heat balance model to explain thresholds for thermoacoustic sound production in seawater using metal electrodes.** Michael S. McBeth (Res. and Appl. Sci., Naval Information Warfare Ctr. Atlantic, NASA Langley Res. Ctr., 100 West Taylor St., M.S. 060, Hampton, VA 23681, michael.s.mcbeth.civ@us.navy.mil)

Experiments conducted with solid metal wire electrodes in seawater generated second harmonic sound waves thought to be thermoacoustic in origin. Using sine wave bursts of three to twenty-five cycles of 10 kHz voltage across the electrodes, the applied voltage threshold for thermoacoustic sound production was observed to decrease almost linearly for bursts up to ten cycles. At the time of these experiments, around 2014, we were unable to explain these sound production threshold values. A mass density continuity equation with a negative source term is used to explore the observed threshold values. The negative source term represents the decrease in fluid density with each half cycle of applied voltage due to the ohmic heating. This approach builds on an earlier heat balance model that successfully explains an observed delay in thermoacoustic sound production. Model results are compared with the experimental observations.

**2:00**

**1pEA5. An improved high altitude ultrasonic anemometer.** Tim J. Cheng (Mech. Eng., Tufts Univ., 200 Boston Ave. 2600 Ste., Medford, MA 02155, timothy.cheng@tufts.edu), Tara Curran, Freidlay Steve (Mech. Eng., Tufts Univ., Medford, MA), Chris Yoder (Wallops Flight Facility, NASA, Wallops Island, VA), Don Banfield (Ames Res. Ctr., NASA, Mountain View, CA), and Robert D. White (Mech. Eng., Tufts Univ., Medford, MA)

This paper describes a sonic anemometer for low pressure (~10mbar) environments. The technology is being developed as a high accuracy 3-D wind measurement system for the surface of Mars and stratospheric balloons. Applications include climate research, balloon navigation, and detection of energetic events through infrasonic atmospheric gravity waves. The device was constructed from an ST 32 bit Nucleo microcontroller and a custom daughter board that interface with high performance application specific amplifiers and multiplexers. Total power draw is 1.5 W. The sensor head uses narrowband commercial ultrasound transducers operating near 40 kHz at voltages below 30 $V_{pp}$. The total mass of the head is 180 g with a volume 15 cm on a side. The resulting system can take speed of sound and 3-axis flow velocity data at 0.4 Hz with 10 cm/s RMS fluctuation. Bell jar testing demonstrated operation to $-5\,^{o}$C and 10 mbar. A balloon launch to 32 km will occur before the meeting and will result in the first wind measurements at this altitude with a digital sonic anemometer. We will report on results and compare to inertial and GPS. [Work supported by NASA-NNX16AJ24G and NASA-80NSSC20M0007. Thanks to the NASA Wallops Balloon Programs Office for technical support.]

**2:15**

**1pEA6. Experimental characterization of high intensity progressive ultrasound beams in air.** Yuqi Meng (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX 78712, yuqimeng@utexas.edu), Ehsan Vatankhah, Zihuan Liu, Xiaoyu Niu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Randall P. Williams (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Neal A. Hall (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

The majority of reported measurements on high intensity ultrasound beams in air are below 40 kHz and are performed on standing waves inside a guide. Here we present experimental characterization of high intensity progressive and divergent sound beams in air at 300 kHz. Measurements in this frequency range are challenging. Accurate characterization of high intensity sound beams requires a measurement bandwidth at least 10x the beam's primary frequency, as high intensity soundwaves steepen and form shocks, and therefore contain significant signal power at harmonic frequencies. A measurement bandwidth of at least 3 MHz is therefore required. Calibrated measurement microphones are generally not available in this frequency range. We have overcome this limitation by using a hydrophone with a calibrated response from 250 kHz—20 MHz. A narrowband piezoelectric transducer is used as the source in this study. The source is capable of generating tone burst waveforms centered at 300 kHz and with 160 dB SPL surface pressure. Cumulative wave steepening and shock formation are observed in on-axis measurements. The source's surface vibration profile is measured using a scanning LDV, and the vibration profile is imported into a numerical wide-angle KZK simulation for comparison against measured on-axis waveforms.

**2:30**

**1pEA7. Mechatronic acoustic research system for generating real large-scale dynamic datasets.** Austin Lu (Univ. of Illinois Urbana-Champaign, 503 E Clark St., Champaign, IL 61820, austinl8@illinois.edu), Ethaniel Moore, Arya Nallanthighall (Univ. of Illinois Urbana-Champaign, Champaign, IL), Mankeerat S. Sidhu, Kanad Sarkar, Manan Mittal, Ryan M. Corey, Paris Smaragdis, and Andrew C. Singer (Univ. of Illinois Urbana-Champaign, Urbana, IL)

To support spatial audio research, we aim to take recordings from complex acoustic environments with moving sources and microphones, however we observe a lack of research tools that can accomplish this. Past approaches recorded people engaging in various tasks, which produces rich data that unfortunately lacks repeatability. We propose using robots to recreate dynamic scenes without the inherent variability of human motion. To be useful, this Mechatronic Acoustic Research System must be remotely accessible, offer concise representations of dynamic scenes, support a variety of robot and audio devices, and synchronize robot motion. In this talk, we show how we solved these challenges. Remote experimentation is facilitated by our virtual interface, which uses a simple GUI to describe robot motion and audio playback/recording. A digital twin physical simulation is used for visualization and validation of motion paths. We propose using the Robot Operating System for multi-robot coordination so that networked robots can be incorporated with little overhead. We use MARS to run experiments where a cable-driven parallel robot moves a loudspeaker along a 3D path while being recorded from distributed Matrix Voice microphone arrays. We evaluate the measured audio to show repeatability of the system, justifying its use in research.

**2:45**

**1pEA8. Modeling nonlinear acoustic damping due to flow separation.**
Joseph Day (Univ. of Colorado Springs, 1420 Austin Bluffs Pkwy, Colorado Springs, CO 80918, jday@uccs.edu) and J. M. Quinlan (Univ. of Colorado Colorado Springs, Colorado Springs, CO)

Nonlinear acoustic damping has been observed in many high-amplitude acoustic systems as a result of flow separation and shear layer vortical motion, eventually transforming some of the acoustical energy into heat. The amount of nonlinear acoustic damping helps determine the nonlinear limit cycle amplitude, e.g., damping caused by baffle blades in a liquid rocket engine to reduce combustion instabilities. The damping mechanism is dependent on both the location and phase of flow separation. Identifying the flow separation is a function of both the boundary layer growth and the acoustically imposed pressure gradient. When the acoustic pressure gradient is adverse, the boundary layer is more prone to separation. Using this as a basis, a model can be created that is applicable to general geometry, which will then be used to approximate the nonlinear acoustic damping in various situations. The constructed model will be compared to established cases, such as an orifice in a duct, to validate the model.

**3:00**

**1pEA9. Security weaknesses in acoustic release system.**
Mak Gračić (Marine Technologies, Univ. of Haifa, 199 Aba Koushy Ave., Haifa 3498838, Israel, Mak.Gracic@fer.hr) and Roee Diamant (Marine Technologies, Univ. of Haifa, Haifa, Israel)

Underwater acoustic communication is increasingly perceived as a cost-effective ocean exploration and monitoring means. However, while carrying out these tasks, UWAC devices are left unattended over long periods of time and may become vulnerable to external attacks. Of particular interest are acoustic release systems that are standard equipment in anchored systems, and whose purpose is to allow safe release when needed. Here, the release command sent in cleartext, and an attacker may try to replicate it in order to disconnect key assets from a network by making them emerge or drift away from their anchor. In this work, we analyze the security weaknesses in acoustic release systems in terms of the number of secrete bits in the release command. We then provide an analysis for the average number of release attempts required for an attacker to mimic the release command. Demonstration is performed over a real acoustic release system using 5 different release commands. We show that the current acoustic release systems are extremely vulnerable for attacks. Our presentation will include the background for acoustic release communication commands, the security analysis and examples of the possible attacks.

**3:15**

**1pEA10. Numerical study on the aeroacoustics and interaction of two distributed-propulsion propellers in co- and counter-rotations.**
Sidharath Sharma (Univ. of Nottingham, Nottingham, United Kingdom), Guangyuan Huang (Univ. of Nottingham, University of Nottingham, Nottingham NG8 1BB, United Kingdom, guangyuan.huang@nottingham.ac.uk), Stephen Ambrose, and Richard Jefferson-Loveday (Univ. of Nottingham, Nottingham, United Kingdom)

Driven by increasing demands for a sustainable and eco-friendly future in aviation, distributed electric propulsion (DEP) systems have received much attention for their high aerodynamic efficiency. DEP systems of lower noise emissions are desired by customers and policymakers and therefore it is important to understand the aeroacoustics and interaction of distributed propeller systems. In this paper, the aeroacoustics of a simplified DEP system is numerically investigated. The system consists of two Mejzlik 2-blade-$9 \times 9$-inch propellers that are distributed side by side, with a tip-to-tip distance of 10 mm. Their rotating speed and freestream velocity are set as 6500 RPM and 12 m/s, respectively. The configurations of both co- and counter-rotation are considered. Compressible Large-eddy simulations are performed to obtain the flow solutions, and the Ffowcs Williams and Hawkings (FW-H) method is used to calculate the corresponding far-field acoustic solutions. The results present interaction effects for both configurations and compare against isolated propeller results. First, the thrust and the induced sound of each propeller are examined and secondly the impact of the interaction effects on the aerodynamic and acoustic performance is investigated. Finally, sound interference in the acoustic fields is assessed and compared for both co- and counter-rotation configurations.

## Session 1pID

## Interdisciplinary: Keynote Lecture

Subha Maruvada, Chair
*U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993*

**Chair's Introduction—4:00**

***Invited Paper***

**4:05**

**1pID1. Broadening participation in acoustics: Personal reflections and pathways forward.** Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

It has been over 30 years since the author embarked on pursuing acoustics and engineering as a career. At the first Acoustical Society of America's (ASA) Technical Committee in Architectural Acoustics meeting that the author attended in the early 1990s, there were two women in the filled room, of whom the author was one. Overall, these past three decades have provided a fulfilling and positive professional experience, but there have been moments where the author leaned on demonstrating resilience, sought to rise above feelings of not being good enough, and made career choices and conducted research differently than might have been expected. In this presentation, the author shares personal reflections on her experience, as well as thoughts on how the ASA and its members can build on recent momentum to continue to broaden participation in acoustics, particularly among underrepresented groups. The author was tasked to lead diversity, equity and inclusion efforts in the College of Engineering at the University of Nebraska – Lincoln as an Associate Dean of Faculty and Inclusion beginning in 2018. A summary of data, tactics, and strategies undertaken since then are shared as possible pathways forward, that could be replicated in others' research groups and communities.

# Session 1pMU

## Musical Acoustics: General Topics in Musical Acoustics I–Perception and Psychoacoustics

Andrew A. Piacsek, Chair
*Physics, Central Washington University, 400 E. University Way, Dept. of Physics, Ellensburg, WA 98926-7422*

### *Contributed Papers*

**1:00**

**1pMU1. Combination tones and multiphonics in a physics of music lab.** Randy Worland (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Laboratory exercises related to multiphonic tones in wind instruments have been developed for use in an undergraduate Physics of Music class. Although the concepts of nonlinear mixing and production of combination tones can be challenging to present at a non-technical level, a hand's-on (mouth's-on!) approach allows students to can gain a visceral and aural appreciation of this type of mixing and its musical use by wind players. The exercises make use of inexpensive free-reed pitch pipes. A pair of pipes can be played independently, producing a linear combination of the two sources, and as a pair of coupled nonlinear oscillators, resulting in combination tones. The differences between the two types of mixing are heard and viewed on a spectrogram in real time. Unlike multiphonics produced by advanced woodwind and brass players, every student can produce these combination tones with pitch pipes. Most can also sing a fixed pitch while playing a pipe to achieve another multiphonic effect. The exercises are readily extended to harmonicas, as available, where difference tones are often heard quite clearly, and the nonlinear mixing forms an essential part of the instrument's timbre. Additional multiphonic examples are also illustrated.

**1:15**

**1pMU2. Introducing acoustics as music in the interdisciplinary and inclusive classroom through ski-hill graph pedagogy to teach the meter fundamentals as sound (psychoacoustics).** Andrea Calilhanna (None, 2 Kayla Way, Cherrybrook, Sydney, New South Wales 2126, Australia, a.calilhanna@gmail.com)

Although music is sound (acoustics) and listeners experience the physics of acoustics as music, the meter of most music textbooks is the notated meter signature and group of beats notated in 'measures' on the page with vertical bar lines. Notation-based representation of the meter is antiquated music theory because new research in acoustics, cognitive neuroscience, and critically music theory (Cohn, 2020) informs otherwise. Cohn's modern meter theory, a distillation of contemporary meter theory from North America, is the first comprehensive meter theory that acknowledges the meter's psychoacoustic experience and augments notation-based understandings of the meter through the meter the fundamental mathematics of the Ski-hill graph. This paper is a music teacher's response to the availability of modern meter theory's Ski-hill graph to represent the meter's psychoacoustic (mind and body) experience. The paper illustrates how listeners articulate their quantification of low meter frequencies such as the duple meter in sets of pulses in ratio 2:1 initiated by 1.2 hz and 2.4 hz experienced concurrently. Also, the triple meter sets of pulses in ratios 3:1 0.8 hz and 2.4 hz (Nozaradan *et al.*, 2011).

**1:30**

**1pMU3. Perception of vibrato rate by professional singing voice teachers.** Joshua D. Glasner (Speech-Lang. Pathol., Delaware Valley Univ., 700 E Butler Ave. Doylestown, PA 18901, jdglasner@gmail.com) and John Nix (Music, Univ. of Texas: San Antonio, San Antonio, TX)

This study sought to investigate how voice clinicians perceive vibrato rate alterations when presented with controlled, synthesized singing voice samples which vary in vibrato rate and vibrato extent. Thirty-four professional voice teachers completed a twelve-item demographic survey and performed a visual sort and rate task (VSR). For the VSR task, each participant listened to twenty synthesized samples and sorted them from slowest vibrato rate to fastest vibrato rate. This task resulted in distance (i.e. individual perception of vibrato rate) and rank-difference measurements for each sample. Two generalized linear mixed effects models (GLMM) and one linear model (LM) were computed. Results for GLMM's found significant associations between vibrato extent and vibrato rate and both individual perception of vibrato rate and rank-difference. Results for the LM found no significant relationships between demographic information and absolute total ranking error. From the results of this study, it seems that both vibrato extent and vibrato rate influence the perception of vibrato rate for professional voice teachers. Neither age nor teaching experience seemed to relate to the ability to discern vibrato rate accurately.

**1:45**

**1pMU4. The impact of formal musical training on speech intelligibility performance—Implications for music pedagogy in high-consequence industries.** Alexandra L. Bruder, Akash K. Gururaja (Anesthesiology, Vanderbilt Univ. Medical Ctr., Nashville, TN), Clayton D. Rothwell (Dept. of Human Autonomy Systems, Infoscitex Corp., Dayton, OH), Suzanne A. Baillargeon (Anesthesiology, Vanderbilt Univ. Medical Ctr., Nashville, TN), Matthew S. Shotwell (Dept. of Biostatistics, Vanderbilt Univ., Nashville, TN), Judy R. Edworthy (School of Psych., Univ. of Plymouth, Plymouth PL4 8AA, United Kingdom), and Joseph J. Schlesinger (Anesthesiology, Vanderbilt Univ. Medical Ctr., 1211 21st Ave. South, Medical Arts Bldg., Ste. 422, Nashville, TN 37212, joseph.j.schlesinger@vumc.org)

In high-risk domains, accurate and timely communications while multitasking are necessary for safety and efficiency. Complex musical/acoustic environments cause hindered communication and awareness. This study used an audio-visual multi-tasking paradigm that examined speech intelligibility and if formal music training (FMT) can mitigate these challenges. Twenty-five students (16F/9M) from Vanderbilt University participated with varying levels of FMT: no FMT, 1–3 years, 3–5 years, and 5+ years of FMT. The study found that the degree of FMT significantly impacted non-response (p-value $<$ 0.001). Among participants with no FMT, the presence

of background music increased the odds of non-response by 1.5-fold (95% CI: 0.95, 2.37), conversely, participants with 5+ years of FMT had no decrease (OR: 0.97, 95% CI: 0.69, 1.36), showing that non-response in the presence of music worsens with each subgroup until 5+ years of FMT. The accuracy for all groups was similar ($p = 0.74$) and the background music decreased accuracy for all groups (OR: 0.67, 95% CI: 0.58, 0.76). Although levels of accuracy were similar for all, the 5+ FMT group responded less frequently, which may indicate increased working memory (i.e., phonological loop) and meta-cognition, a valuable skill in high-risk industry. Future research can explore the pedagogy of musical training.

**2:00**

**1pMU5. Uncovering the differences between the violin and erhu musical instruments by statistical analysis of multiple musical pieces.** Wenyi Song (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong NA, Hong Kong, wsongak@cse.ust.hk) and Andrew B. Horner (Dept. of Comput. Sci. and Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

The violin and erhu are two of Western and Chinese music's main bowed string instruments. Recent work has compared the different emotional characteristics between the violin and erhu on the *Butterfly Lovers Concerto*. In our study, we examine several hypotheses to investigate whether the previous studies' results hold generally. Four musical excerpts were extracted from four famous Chinese and Western classical pieces, and the excerpts were divided into four emotional categories: Happy, Sad, Agitated, and Calm. Based on the *Butterfly Lovers* results, we expected that: (1) the violin has a more Happy emotional characteristic than the erhu, while the erhu is comparatively more Sad, and (2) the violin is better at conveying high-Arousal excerpts. We used the Bradley–Terry–Luce (BTL) paired-comparison model to obtain the ranking scores and identify statistically significant differences between the two instruments. The erhu was consistently perceived as sadder than the violin for all Sad excerpts, while the violin was generally calmer and more agitated for those categories. Further study with more listeners and excerpts is needed to verify whether these results generally hold and at a statistically significant level.

**2:15**

**1pMU6. Emotion equalization app: A first study and results.** Man Hei Law (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong 000000, Hong Kong, mhlawaa@connect.ust.hk), Andrew B. Horner, and Hoi Ting Leung (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

In recent years, there is an increasing trend of mental health issues in society. It is important to identify mental first aid strategies that can be applied at an early age, whether emotional issues have been diagnosed or have yet to be found. Music has tremendous potential impact on changing emotional states since it can distract listeners from rumination on negative thoughts and engage them in a moment of musical enjoyment. This paper presents a new emotion equalization app that incorporates validated diagnosis tests (PHQ-9 and GAD-7) and an emotion measuring tool (SAM) for establishing a personalized therapy treatment using emotion rebalancing methods. By determining the emotional state of the user, songs are chosen and sequenced in a playlist using one of three proposed methods (consoling, relaxing, and uplifting) with a baseline method (random). With this systematic generation of playlists, the app can be used for personalized therapeutic treatment even for users without music background. In our experiment, the results showed positive changes in listeners' valence levels while there was no significant change in arousal. Furthermore, the relaxing and uplifting methods showed a significant effect on moving listeners from negative to more positive emotional states.

**2:30**

**1pMU7. The effects of vowel, pitch, and dynamics on the emotional characteristics of the Soprano, Alto, Tenor, and Bass Voices.** Bing Yen Chang (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Man Hei Law (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong 000000, Hong Kong, mhlawaa@connect.ust.hk), Hiu Ting Chan, and Andrew B. Horner (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Previous research on the soprano and tenor voices have shown that their emotional characteristics change with different vowel, pitch, and dynamics. This work continues the investigation with the alto and bass voices. Listening tests were conducted whereby listeners gave absolute judgements on the SATB voice tones over ten emotional categories with the data analyzed using logistic regression. High-arousal categories were stronger for loud tones, whereas low-arousal categories were stronger for soft tones. The categories Happy, Heroic, Romantic, and Shy had mostly upward trends across the pitch range, whereas Angry had an overall downward trend. Calm and Sad had an arch-shaped trend, while Scary had a U-shaped trend. Comic and Mysterious had different trends among the voices. The voices each exhibited different vowel trends, though vowel A was dominant for all the voice types in the categories Happy and Romantic. Dynamics had the strongest effect overall, followed closely by pitch, with both effects approximately twice as strong as the effect of vowel. Vowel U had the largest strength-of-expressiveness overall, with A second, O third, I fourth, and finally E last. These results give a quantified preliminary perspective on how vowel, pitch, and dynamics shape emotional expression in the SATB voices.

**2:45**

**1pMU8. A comparison of the emotional characteristics of western orchestral sustaining musical instrument families with different pitch and dynamics.** Hiu Ting Chan, Bing Yen Chang (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong), Andrew B. Horner (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong 00000, Hong Kong, horner@cse.ust.hk), and Man Hei Law (Comput. Sci. and Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Recent research has shown that the emotional characteristics of the bowed strings and brass change with different pitch and dynamics. This work compares how pitch and dynamics influence their characters in the western orchestral sustaining musical instruments. We conducted listening tests where listeners gave absolute judgements on the sounds over ten emotional categories and analyzed the data with logistic regression. For pitch, the categories Happy and Heroic had an arched shape that peaked at C6. Romantic, Calm, and Shy increased until C4 and decreased afterwards. Comic and Mysterious did not shown a clear common trend with pitch. Angry and Scary had a U-shape that was slightly stronger at the highest pitch, while Sad decreased with pitch. For dynamics, Happy, Heroic, Comic, and Angry were stronger for loud notes, while Romantic, Calm, Mysterious, Shy, and Sad were stronger for soft notes. For Scary, loud and soft notes were about the same. In the bowed strings and woodwinds, pitch and dynamics had about an equally important effect on the emotional characteristics, while pitch had a more important effect than dynamics in the brass. For the bowed strings, brass, and woodwinds, the particular instrument was more of a secondary factor though still important.

# Session 1pPA

## Physical Acoustics: General Topics Physical Acoustics I: Acoustic Manipulation and Atmospheric Propagation

Matthew J. Kamrath, Chair
*US Army Engineer Research and Development Center, 72 Lyme Road, Hanover, NH 03755-1290*

### *Contributed Papers*

**1:00**

**1pPA1. Analytical solution for a focused vortex beam radiated by a Gaussian source.** Chirag A. Gokani (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX 78713-8029, chiragokani@gmail.com), Yuqi Meng (Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX), Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Current interest in focused vortex beams is motivated by the ability to trap particles axially and laterally using the resulting radiation force. A simple closed-form solution is obtained in the Fresnel approximation for a sound beam radiated by a Gaussian source with time dependence $e^{-i\omega t}$, focal length $d$, amplitude distribution $\exp(-r^2/a^2)$, and azimuthal phase dependence $e^{in\theta}$, where $\theta$ is the angle in the plane perpendicular to the beam axis, and the integer $n$ is the topological charge, referred to here as the vorticity. The solution is in good agreement with the pressure field predicted in the paraxial region by numerical evaluation of the Rayleigh integral. Of interest in optics as well as acoustics is the distance from the minimum along the beam axis to the first local maximum, referred to as the vortex ring. The present solution yields $r_n = \eta_n d/ka$ for the vortex ring radius in the focal plane, where $k$ is the wavenumber, and $\eta_n = 1.69 + 0.965(n-1)$ for vorticities in the range $1 \leq n < O(20)$. Within this range the radius $r_n$ thus increases linearly with the vorticity. Results are also presented for dependence of the focusing gain on the vorticity. [CAG was supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

**1:15**

**1pPA2. Particle manipulation with acoustic waves based on Hertz-Mindlin mechanics.** Marina Terzi (Lab. d'Acoustique de l'Université du Mans, Av. Olivier Messiaen, Le Mans 72085, France, marina.e.terzi@gmail.com), Vincent Tournat (Lab. d'Acoustique de l'Université du Mans, Le Mans, France), Vladislav Aleshin (Institut d'Electronique, de Microélectronique et de Nanotechnologie, Villeneuve d'Ascq, France), Mathieu Chekroun, and Maxime Lanoy (Lab. d'Acoustique de l'Université du Mans, Le Mans, France)

Ultrasound noncontact particle manipulation (vibrotransportation) is of interest for a number of industrial applications: delivery of solid agents for defect healing, noninvasive excretion of undesirable particles, micromachine technology, and sorting of particles. It is known that a particle on a substrate can be moved by a combined action of acoustic waves and friction. A traditional theory describing the effect considers a particle as a material point capable of moving only against the wave, i.e., in a direction opposite to the wave propagation direction. Here, we use another approach based on Hertz-Mindlin mechanics in which a particle represents a deformable body whereas a zone of its contact with the substrate has a certain structure including evolving regions of stick and slip. The friction dynamics of the Hertz-Mindlin system with a mass comprises different regimes, including a consistent motion in a desired direction (not only the one opposite to the direction of wave propagation but also the one oriented like the wave), which allows a more elaborate manipulation or positioning of the particles. The presented numerical tool treats the problem of a particle on a substrate being moved by Rayleigh waves. We demonstrate various regimes in which steady movement is observed or not, depending on excitation parameters and particle inertia.

**1:30**

**1pPA3. Noncontact rotation of a small object using ultrasound standing wave and traveling wave.** Eimei Yamamoto (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe-shi, Kyoto-fu 610-0394, Japan, ctwh0374@mail4.doshisha.ac.jp), Natsumi Nakaoka, and Daisuke Koyama (Doshisha Univ., Kyotanabe, Kyoto, Japan)

In an acoustic standing wave generated in the air between a vibrator and a reflector, a small object is levitated near the nodal positions of the sound pressure where the acoustic radiation force and the gravity are balanced. By controlling the sound field spatially and temporally, the object can be transported without contact. In this report, we experimentally investigated the non-contact high-speed rotation of a small object using ultrasound, which can be applied to measurement techniques of the physical properties of liquids. The experimental system consists of a vibrating disc with four bolt-clamped Langevin-type transducers and a reflector, and an acoustic standing wave is generated between them. The effects of the tilt angle of the reflector with respect to the vibrator on the rotation speed of the object and the acoustic field were investigated. When the tilt angle was around 1.2°, the object was trapped in the air at 31.5 kHz by the standing-wave component in the vertical direction and rotated by the traveling-wave component propagating in the horizontal direction. The maximum sound pressure amplitude and the rotation speed of the object were changed with the tilt angle, and a larger sound pressure amplitude gave a larger rotation speed.

**1:45**

**1pPA4. Experimental study and modeling of the level-dependent acoustical behavior of granular particle stacks.** Guochenhao Song (Ray W. Herrick Lab., Purdue Univ., Ray W. Herrick Labs., 177 S, Russell St., West Lafayette, IN 47907, song520@purdue.edu), Zhuang Mo, and J. S. Bolton (Ray W. Herrick Lab., Purdue Univ., West Lafayette, IN)

Researchers have previously observed elastic modulus softening and increased damping when granular particle stacks are exposed to progressively increasing acoustical excitation levels. However, the level-dependent behavior of granular particle stacks is not well understood, and there are no comprehensive approaches to modeling those effects. Earlier, the authors measured the absorption coefficient of a stack of one type of granular activated carbon stack by using signals having different bandwidths and levels. In the present work, five more types of granular particle stacks were studied to validate and generalize the previous conclusions: i.e., both the modulus softening, and increased damping can be characterized by the total RMS fluid displacement at the sample surface. Therefore, a strain-dependent modulus and damping formula from the literature (based on cyclic loading tests on sand particles) was converted into a total RMS fluid displacement-dependent formula (based on acoustic measurements). In addition, a multi-layered model based on this displacement-dependent formula has been developed to iteratively update each layer's modulus, damping, and total RMS fluid displacement to solve for the particle stack's

acoustic properties. This approach allows modeling of the particle stack's acoustical behavior by using a single set of parameters, even for different level and bandwidth test signals.

**2:00**

**1pPA5. A finite difference approach to study the impact of boundary conditions on the acoustical behavior of particle stacks.** Zhuang Mo (Ray W. Herrick Lab., Purdue Univ., 177 S. Russell St., West Lafayette, IN 47907-2099, mo26@purdue.edu), Guochenhao Song (Ray W. Herrick Lab, Purdue Univ., West Lafayette, IN), Tongyang Shi (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and J. S. Bolton (Ray W. Herrick Lab., Purdue Univ., West Lafayette, IN)

The responses of particle stacks to incident sound waves show interesting features that are very different from those of a homogeneous continuum. Further studies of the acoustical performance of particle stacks can help both to discover potential noise control applications of these types of materials, and to help provide better insight into the internal status of the particle stacks. In the current study, a finite difference (FD) model for a particle stack was built based on the Biot poro-elastic theory. The intention in developing this model is to describe the acoustical behavior of particle stacks with consideration of not only the finite stiffness of the particles, but also the influence of gravity and friction between the particles and the inner wall of their enclosure: i.e., the cylindrical sample holder of a standing wave tube, in this work. The derivation of governing equations and boundary conditions is introduced, together with acoustic measurement results of particles stacks consisting of micron-scale glass bubbles, including absorption coefficients and surface impedances that are compared with the theoretical predictions. The possible application scenarios of such materials, and potential developments that will further improve the FD model will also be discussed.

**2:15–2:30 Break**

**2:30**

**1pPA6. Locating impulsive sound sources in microscale urban spaces.** Dorian Davis (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, dorian.davis1@udc.edu), Samba Gaye, Lirane Mandjoupa, Justin An, Wagdy Mahmoud, Lei Wang, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, techniques for localizing impulsive acoustic sources in an urban environment are presented. Of particular interest is the localization of sources in urban street canyons and enclosed urban areas. Sound propagation in an urban environment is strongly influenced by multiple reflections. In urban street canyons, multiple reflections tend to amplify with decreasing canyon width. A numerical investigation is performed to study the role multiple reflections on time difference of arrival (TDOA) and beamforming source localization techniques. Results of various urban street canyon and enclosed space geometries are investigated. The results and limitations of the TDOA and beamforming techniques for estimating source position in urban environments are discussed.

**2:45**

**1pPA7. Characterizing flow conditions of urban environments from wind-induced noise measurements: A semi-empirical wind noise model of the University of the District of Columbia campus.** Lirane Mandjoupa (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, DC 20008, max.denis2@udc.edu), Samba Gaye, Dorian Davis, Justin An, Wagdy Mahmoud, lei wang, and Max Denis (Univ. of the District of Columbia, Washington, DC)

In this work, the flow conditions within the University of the District of Columbia (UDC) urban campus are predicted from wind-induced noise. Wind-induced noise obtained from a collection of spatial distributed microphones are used to estimate the mean velocity airflow and wind noise distribution across the UDC campus. Wind speed and direction are estimated by fitting the second-order statistics of semi-empirical models of wind noise distribution from microphone measurements to analytical models in the least squares sense. The accuracy of the proposed is investigated for average microphone separation and time resolution. Comparisons of the wind speed and direction results to ultrasonic anemometer measurements are discussed.

**3:00**

**1pPA8. Scanning Doppler LIDAR wind profiles to inform near shore atmospheric acoustic propagation modeling.** Andrea Vecchiotti (Mech. Eng., The Catholic Univ. of America, Washington, DC), Hannah Blackburn, Kyle Kirian (Eng., East Carolina Univ., Greenville, NC), Joseph Vignola, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), Jeff Foeller (Eng., East Carolina Univ., Greenville, NC), and Teresa J. Ryan (Eng., East Carolina Univ., Dept. of Eng., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu)

This work presents a comprehensive experimental system to measure concurrent atmospheric acoustic transmission loss and meteorological conditions. A three-dimensional scanning Doppler lidar wind profiler captures real-time wind speed gradients at many locations along the acoustic propagation path of a simple pitch catch style study. A long-range acoustic device on an anchored pontoon sends known chirp sequences to a seven-channel receiver array at the water's edge at ranges up to approximately one kilometer. Additional synchronized meteorological observations include temperature, humidity, and wind measured with anemometers. The meteorological data stream is used to inform the sound speed gradient implemented in a parabolic equation based numerical model of atmospheric acoustic propagation. The model can account for sea surface roughness and accommodate a sound speed profile that changes along the propagation range. Model predictions are compared to measured transmission losses. An assessment of the value of the computational cost of incorporating the varying sound speed profiles in the model is presented.

**3:15**

**1pPA9. Evaluation of a gamma power, wrapped normal phase model for vertical and slanted atmospheric sound propagation.** Matthew J. Kamrath (US Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, matthew.j.kamrath@erdc.dren.mil)

As sound propagates through the atmosphere, turbulence causes fluctuations in the signal power and phase. These fluctuations complicate the signal processing required to detect, locate, and identify elevated sources. This presentation evaluates a model that assumes a gamma distribution describes the power and a wrapped normal distribution describes the phase. The gamma distribution approximates a log-normal distribution when the normalized power variance is small compared to one and an exponential distribution when the normalized power variance is one. The wrapped normal distribution approximates a normal distribution when the phase variance is small and a uniform distribution when the phase variance is large. In 2018, we measured vertical and slanted sound propagation at the National Wind Technology Center (NWTC). The ground-borne source emitted tones in the range 600–3500 Hz, and several microphones were mounted to a meteorological tower up to 130 m high. Overall, the gamma power, wrapped normal phase model accurately approximated the NWTC data across a wide range of scattering regimes. This presentation also discusses the largest discrepancies.

**3:30**

**1pPA10. Transfer function model-based Bayesian estimation of dissipation and sound speed in impedance tube measurements.** Ziqi Chen (Arch. Acoust., RPI, 110 8th St., Troy, NY 12180, chenz33@rpi.edu) and Ning Xiang (Arch. Acoust., RPI, Troy, NY)

The air dissipation, mainly due to the boundary effect at interior walls of impedance tubes, is noteworthy for normal incidence measurements of acoustic materials using the transfer function method. This work proposes a transfer function model-based Bayesian approach to estimate the dissipation factors. Other parameters, such as the sound speed and the microphone positions, are also critical, therefore, estimated as well, along with the dissipation. The measurement of the empty impedance tube with a rigid termination applies the two-microphone transfer function method. A hypothetical air layer assumed in front of the rigid backing enables the numerical comparison between the air layer's theoretical and experimental transfer function values. Experimental results incorporating the estimated dissipation coefficient validate the accuracy of the proposed approach. Furthermore, the estimated parameters are incorporated into further evaluations of material properties using the transfer function method, which effectively reduces the residual absorption error of the impedance tube measurement.

## Session 1pSC

## Speech Communication: Methods and Instrumentation in Speech (Poster Session)

Tian C. Zhao, Chair

*Department of Speech and Hearing Sciences/Institute for Learning and Brain Sciences, University of Washington, Box 367988, Seattle, WA 98195*

All posters will be on display from 1:00 p.m. to 4:00 p.m. Authors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

### *Contributed Papers*

**1pSC1. An investigation of interference between electromagnetic articulography and electroglottography.** Jessica Goel (Univ. of Florida, Florida PO Box 100174, 1225 Ctr. Dr., Rm. 2143, Gainesville, FL, jgoel@ufl.edu), Matthew Masapollo, Ratree Wayland, Rahul Sengupta, Allen Shamsi, and Karen W. Hegland (Univ. of Florida, Gainesville, FL)

The present study tested whether there is cross interference between state-of-the-art electromagnetic articulography (EMA) and electroglottography (EGG) during the acquisition of kinematic speech data. In Experiment 1, we calibrated EMA sensors with and without EGG electrodes present in the EMA field. In Experiment 2, EMA was used to record lip, tongue, and jaw movements for one male talker and one female talker, with and without simultaneous EGG recording. Collectively, the results provide no evidence of signal artifacts in either direction, suggesting that current EMA and EGG technology can be combined to reliably assess laryngeal and supralaryngeal motor coordination in speech.

**1pSC2. Concurrent visualization and analysis of acoustic, flesh-point motion, and electroglottography signals during speech production.** Jessica Goel (Univ. of Florida, Florida PO Box 100174, 1225 Ctr. Dr., Rm. 2143, Gainesville, FL, jgoel@ufl.edu), Matthew Masapollo (Univ. of Florida, Gainesville, FL), Yoonjeong Lee (Linguist., Univ. of California, Los Angeles, Los Angeles, CA), and Ratree Wayland (Univ. of Florida, Gainesville, FL)

Over the years, technological advances have enabled speech researchers to directly track the skilled, sound-producing movements of the vocal tract, both intraoral and laryngeal articulators normally hidden from view (the tongue, velum, and glottis) and orofacial articulators directly visible on talkers' faces (the lips and jaw). Despite these advances, however, no single instrument is capable of concurrently recording movements of all the articulators, which has impeded progress in characterizing inter-articulator control and coordination. To explore how inter-articulator coordination subserves linguistic structure, tools that co-register and temporally align different signals from different recording devices are necessary. This tutorial introduces optimal methods for studying the temporal coordination between laryngeal, intraoral, and orofacial articulators by combining various signals from electromagnetic articulography (EMA), electroglottography (EGG), and audio recordings, and displaying the time-aligned signals in the same analysis space. The multimodal data is processed using a set of MATLAB-based functions, which co-register and display positional and velocity trajectories of the lips, tongue, and jaw in tandem with the EGG waveform, *F0* trajectory, and acoustic waveform and spectrogram. The coordination of laryngeal and supralaryngeal speech movements can then be measured and analyzed. [Work supported by an Emerging Research Grant from the Hearing Health Foundation.]

**1pSC3. Examining a new paradigm for simultaneous evaluation of mismatch response (MMR) and complex auditory brainstem response (cABR) for speech in MEG.** Tian C. Zhao (Inst. for Learning and Brain Sci., Univ. of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu) and Samu Taulu (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

The mismatch response (MMR) is a common neural signature, both in M/EEG, to evaluate neural sensitivity to sound change. Furthermore, the complex auditory brainstem response (cABR) has gained wide research interest recently as it is argued to reflect early sensory encoding of complex sounds, such as speech, along the auditory pathway. While both measures are important in infants as babies undergo rapid speech learning, they also share the crucial drawback that it takes many trials and thus long recording times, prohibiting a wide usage in infant population. Here, we investigate a new and more efficient recording paradigm to simultaneously assess both MMR and cABR for speech in MEG. Adult participants are recorded under this new paradigm using simultaneous M/EEG with previously published speech stimuli. For MMR, we aim to replicate previously published results that MMR for a native speech contrast is more concentrated than for a non-native speech contrast. For cABR, we aim to extract a predominant spatio-temporal pattern from all MEG channels and examine its correlation with the EEG-recorded signal. Once the new paradigm can be validated in adults, it can be used in infant populations with much increased efficiency, opening the door for addressing new research questions.

**1pSC4. Extending the pixel difference metric from 2D to 3D/4D ultrasound.** Pertti Palo (Speech, Lang. and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, pertti.palo@taurlin.org) and Steven M. Lulich (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

It is frequently of interest to examine differences (e.g., the 1st or 2nd difference) in signals and images. Differencing can be employed for the detection of edges in images, or the detection of events in temporal sequences. One such event is the onset of articulatory movement in ultrasound image sequences of the tongue. It has been quantified using a form of Euclidean distance between successive frames of image scan line data, referred to as Pixel Difference. When applied to 3-D/4-D ultrasound data of the tongue (after scan conversion), Pixel Difference exhibited a surprisingly poor signal-to-noise ratio. In this presentation, we test the hypotheses that (1) scan conversion amplifies the effect of speckle noise near the top of the ultrasound image (i.e. above the tongue), and (2) the Pixel Difference signal-to-noise ratio in 3-D/4-D ultrasound differs from 2-D ultrasound by a power of 3/2 due to the added spatial dimension.

**1pSC5. Examining localization error in MEG for the complex auditory brainstem response (cABR) across early development.** Tian C. Zhao (Dept. of Speech and Hearing Sciences/Inst. for Learning and Brain Sci., Univ. of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu) and Samu Taulu (Dept. of Physics/Inst. for Learning & Brain Sci., Univ. of Washington, Seattle, WA)

The complex auditory brainstem response (cABR) has gained wide research interest over the last decade as it is argued to reflect early sensory encoding of complex sounds, such as speech, along the auditory pathway. Recent adoption of cABR recording in MEG, instead of 3 EEG electrodes, allows further examination of the underlying sources. In adults, while the onset component of cABR to transient events (e.g., consonants) can be reliably localized to the auditory brainstem, the frequency-following component to longer periodic events (e.g., vowels, lexical tones) may contain contribution from auditory cortex. This study thus expands to evaluate the localization accuracy of both onset response and FFR through simulation, both of which are crucial for speech. Simulated source activities are placed in the auditory brainstem versus superior temporal gyrus. Using the forward model, the activities are projected to MEG sensor space with realistic room noise and head movement. The simulated sensor space signal then undergoes inverse modeling to calculate source activation. Differences between the simulated source activity versus calculated source activity are calculated to reflect localization accuracy. This process is repeated using infant head models at 3, 6, 9, and 12 months and adult head model to examine changes across ages.

**1pSC6. Anatomical measures of the vocal tract in children ages 5 and 6.** Megan Diekhoff (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN) and Steven M. Lulich (Speech, Lang. and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu)

At the same time young children experience rapid physical growth, they are faced with the challenge of mastering the complexities of speech production. Capturing the dimensions of the vocal tract during early childhood is a crucial step in characterizing early speech acoustics and speech motor control. The present study partially replicates measures of the vocal tract first employed in previous studies. Magnetic resonance images (MRI) were obtained from two 5-year-old and six 6-year-old children. Anatomical measurements of the vertical vocal tract (VT-V), posterior cavity length (PCL), anterior cavity length (ACL) and overall vocal tract length (VTL) are reported for each child. Results are consistent with previous findings, and are foundational for a larger, stratified longitudinal study aiming to characterize the development of individuals' vocal tract anatomy between 5 and 9 years of age. [Work supported in part by NSF.]

**1pSC7. Evaluation of automatically generated tongue surfaces from three-dimensional/four-dimensional ultrasound recordings of sustained vowels.** Steven M. Lulich (Speech, Lang. and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu) and Rita R. Patel (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN)

Three-dimensional/four-dimensional (3-D/4-D) ultrasound is capable of imaging almost the entire tongue surface during the production of speech sounds, including sustained vowels. Manual tongue surface segmentation is slow and laborious, but typically yields good results. Automatic tongue surface segmentation is much faster and objective but can result in errors. In this presentation, we quantify the errors in a set of tongue surfaces that were automatically segmented using the previously published 3D-SLURP algorithm implemented in the WASL toolbox for MATLAB. The ultrasound recordings were of 20 children and 40 adults producing the sustained vowel [o:] with and without a narrow straw semi-occlusion in the mouth. Twenty of the 40 adults produced phonation at threshold phonation. Automatic and manual error detection and quantification methods will be compared to evaluate the accuracy of the automated algorithm when applied to the sustained vowel [o:]. [Work supported in part by NSF.]

**1pSC8. Suitability of online and offline remote recording for vowel reduction analysis.** Jenna T. Conklin (Carleton College, One North College St., Goodsell Observatory, Northfield, MN 55057, jconklin@carleton.edu) and Sophia Chuen (Carleton College, Northfield, MN)

Increasing demand for remote research techniques has led to several initial investigations of the suitability of remote recording for vowel production research. An initial consensus has emerged that remote recording may provide broadly accurate data for broad-strokes analysis, such as analyzing the relative positions of a speaker's vowels, although some minor distortions may occur; fine-grained analysis of sociolinguistic variation suffers greatly from remote data collection (Freeman and De Decker, 2021). This study investigates the quality of data obtainable through two remote recording methods (online recording through Gorilla and offline recording via smartphone) for a vocalic phenomenon requiring a medium level of sensitivity of analysis, namely vowel reduction. Ten subjects completed a shadowing task eliciting target words containing one of five English vowels in reduced or unreduced form, taking simultaneous online and offline remote recordings. To determine whether the variability introduced through the remote recording process was sufficient to alter the findings of the study, results were compared to data from a second group of subjects who completed the task in person in a traditional laboratory setting. This comparison to in-person data allows for an assessment of the suitability of online and offline remote recordings for vowel reduction research.

**1pSC9. Lenition measures: Neural networks' posterior probability versus acoustic cues.** Ratree Wayland (Linguist., Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, ratree@ufl.edu), Kevin Tang (English Lang. and Linguist., Heinrich-Heine-Universität Düsseldorf, Düsseldorf, Germany), Fenqi Wang (Linguist., Univ. of Florida, Gainesville, FL), Sophia Vellozzi (Linguist., Univ. of Florida, Gainesville, FL), Rahul Sengupta (Dept. pf Comput. and Information Sci. and Eng., Univ. of Florida, Gainesville, FL), and Lori Altman (Dept. of Speech, Lang., and Hearing Sci., Univ., Gainesville, FL)

This study compared a new approach of lenition measure to traditional acoustic-based methods. In this new approach, degrees of lenition are estimated from posterior probabilities generated by recurrent neural networks trained to recognize the sonorant and continuant phonological features. These two phonological features capture the range of surface manifestations, from a fricative to an approximant, of lenited voiced and voiceless stops in Spanish. Input to the networks is Mel-filtered log-energy computed from 25-ms windowed frames of each 0.5sec chunk of the input signals. When applied to lenition of intervocalic voiced and voiceless stops, /p, t, k, b, d, g/, in the corpus of Argentinian Spanish built by Google, the new approach yielded lenition patterns largely similar to those obtained using a quantitative acoustic method. Specifically, both approaches revealed that voiced stops were more lenited than voiceless stops, that lenition was more likely in unstressed syllables relative to stressed syllables and that degrees of lenition vary with place of articulation of the target phoneme and the height of surrounding vowels. However, a greater amount of variance was accounted for by the absolute and relative (to neighbouring segment) intensity measures of lenition in the acoustic method than the phonological posteriors.

**1pSC10. Decoding syntactic class from EEG during spoken word recognition.** McCall E. Sarrett (Psychol. and Brain Sci., Villanova Univ., Tolentine Hall 334, 800 E Lancaster Ave. Villanova, PA 19085, mccall.sarrett@villanova.edu), Alexa S. Gonzalez, Olivia Montañez (Psychol. and Brain Sci., Villanova Univ., Villanova, PA), and Joseph C. Toscano (Villanova Univ., Villanova, PA)

A fundamental issue in spoken language comprehension involves understanding the interaction of linguistic representations across different levels of organization (e.g., phonological, lexical, syntactic, and semantic). In particular, there is debate about when different levels are accessed during spoken word recognition. Under serial processing models, comprehension is sequential. In contrast, under parallel processing models, simultaneous activation of representations at multiple levels can occur. The current study investigates this issue by isolating neural responses to syntactic class distinctions from acoustic and phonological responses. EEG data were collected in an event-related potential (ERP) experiment in which participants ($N = 26$) listened to words varying in syntactic class (nouns versus adjectives) that were controlled for low-level acoustic differences via cross-splicing. Machine learning techniques were used to decode syntactic class from ERP responses over time. Results showed that syntactic class is decodable approximately 160–190 ms after the average syntactic point of disambiguation in the words, during which listeners are still processing acoustic information. This supports the prediction that different levels of representation have overlapping timecourses. Overall, these results are consistent with a parallel, interactive processing model of spoken word recognition, in which higher-level information—such as syntactic class—is accessed while acoustic analysis is still occurring.

**1pSC11. Comparison of coarse and fine sampling resolutions in vowel analysis.** Emily Grabowski (UC Berkeley, Dwinelle Hall, Berkeley, CA 94710, emily_grabowski@berkeley.edu)

There are many researcher degrees of freedom in phonetic analyses, especially in that of segments such as vowels. The focus of this study is one such degree of freedom, governing how frequently measurements are taken for an individual token in the corpus. Choices for this stage of analysis include averaging formants (single measurement), taking point measurements (multiple measurements), or sampling continuously across the token (many measurements). Less-frequent measurements are convenient techniques compatible with many common statistical methods, but risk oversimplifying the patterns in the data. In this project, I compare point and continuous measurements using Generalized Additive Mixture Models (GAMMs) on speech corpus data. GAMMs are a generalized model that can flexibly parameterize both curves and points based on a combination of random and fixed effects. To compare different measurement frequencies, I quantify the benefit of additional measurements in terms of information gain and variance explained. I also address the modeling of multidimensional measurements and identify the degree to which issues of multidimensionality such as collinearity can be accounted for in GAMMs.

**1pSC12. SpeakerPool: A remote speech data collection platform.** Tyler T. Schnoor (Linguist., Univ. of AB, 150 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, tschnoor@ualberta.ca) and Benjamin V. Tucker (Linguist., Univ. of AB, Edmonton, AB, Canada)

The collection of speech production data for academic use has traditionally been carried out by recording participants in-person. While certain types of research require traditional data collection methods, many factors, such as distance and world events, make remote collection a valuable alternative for other types of research. Despite the existence of capable technologies and the advantages of remote collection methods, to the authors' knowledge there is no freely accessible, reusable, and research-oriented platform for the remote collection of speech production data. We aim to address this with SpeakerPool: a web application that, in addition to providing an easy-to-use recording interface for participants, has built-in functionalities for the automation of many tasks. Users are able to access SpeakerPool regardless of platform using modern web browsers on both mobile and non-mobile devices. This greatly reduces compatibility limitations and bypasses the need to download software. In the present study, we discuss the general advantages and disadvantages of remote speech data collection and investigate the usefulness of SpeakerPool by collecting and analyzing a pilot data set of Malagasy speech.

**1pSC13. Developing a voice monitoring smartphone app: Acoustic acquisition and processing considerations.** Victoria S. McKenna (Commun. Sci. and Disord., Univ. of Cincinnati, 3225 Eden Ave., Health Sci. Bldg., Cincinnati, OH 45267, mckennvs@ucmail.uc.edu), Andres F. Llico (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Aaron Friedman (Otolaryngology-Head & Neck Surgery, Univ. of Cincinnati, Cincinnati, OH), Savannah N. Shanley (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Thomas Talavage (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Leigh M. Bamford (Elec. Eng. and Comput. Sci., Univ. of Cincinnati, Cincinnati, OH)

With the advent of smartphone technology, there has been an increase in at-home health monitoring. Yet, there are few applications (apps) available to track voice acoustics for those with voice disorders. Therefore, we completed two investigations into the acquisition and processing of the acoustic signal to help develop a voice monitoring app. Study 1: We investigated how microphone distance and phone tilt impact the accuracy of acoustic measures of voice. A total of 58 participants with ($n = 47$) and without ($n = 11$) voice disorders completed speech tasks of sustained vowels and the rainbow passage. Concurrent recordings were collected using the participant's phone, as well as a stationary headset microphone for comparison. We determined that the voice measures of fundamental frequency (Hz), voicing duration (seconds), and cepstral peak prominence (dB) were impervious to phone distance and tilt. Study 2: Focusing on the three acoustic measures from study 1, we investigated the correspondence between acoustic measures attained with clinically available software (e.g., Praat) and those from our own lab-developed algorithms specialized for on-app processing. Preliminary results show strong relationships ($r > .90$) between the different processing techniques. Further work is needed to understand how participant-specific factors (age, dysphonia severity) may improve algorithm accuracy.

**1pSC14. Modeling retroflex fricative variation in accented mandarin.** Fenqi Wang (Linguist., Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, fenqi@ufl.edu), Delin Deng, and Ratree Wayland (Linguist., Univ. of Florida, Gainesville, FL)

Since dental-retroflex fricative contrast is not consistently maintained in many southern dialects of Chinese, native speakers of these dialects may not accurately produce the Mandarin retroflex fricative /ʂ/. Consequently, /ʂa/ may be realized as [sa] (Duanmu, 2007). This study investigated the variation of the retroflex fricative /ʂ/ in a Chinese Mandarin speech corpus (DataTang, 2018). The corpus contains 200 hours of recordings of 600 speakers from different dialectal regions in China. Each recording was aligned at the phone level using Montreal Forced Aligner. The center of gravity of the acoustic energy (COG) of the target sounds was extracted using Christian DiCanio's Praat script. For statistical analysis, the generalized additive mixed-effects model (GAMM) was used. COG was the response variable. The following vowel's height, tone, and gender were factorial predictors. To evaluate the geographic effect, we used tensor product smooths by fricatives with the longitude and latitude of each speaker's birthplace (Chuang *et al.*, 2021). Our results suggested a more dental-like realization of the retroflex for speakers from southern China and significant effects of the following vowel's height and gender in the realization of Mandarin retroflex fricative.

**1pSC15. The acoustic profiles of vocal emotions in Japanese: A corpus study with generalized additive mixed modeling.** Fenqi Wang (Linguist., Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611-5454, fenqi@ufl.edu), Delin Deng, and Ratree Wayland (Linguist., Univ. of Florida, Gainesville, FL)

This study investigated the vocal emotions in Japanese by analyzing acoustic features from emotional utterances in the Online Gaming Voice Chat Corpus with Emotional Label (Arimoto and Kawatsu, 2013). The corpus contains the recorded sentences produced in 8 emotions by four native Japanese speakers who are professional actors. For acoustic feature extraction, Praat script ProsodyPro was used. Principle component analysis (PCA) was conducted to evaluate the contribution of each acoustic feature. In addition, a linear discriminant classifier (LDA) was trained with the extracted acoustic features to predict the emotion category and intensity. A

generalized additive mixed model (GAMM) was performed to examine the effect of gender, emotional category, and emotional intensity on the time-normalized f0 values. The GAMM's results suggested the effects of gender, emotion, and emotional intensity on the time-normalized f0 values of vocal emotions in Japanese. The recognition accuracy of the LDA classifier reached about 60%, suggesting that although pitch-related measures are important to differentiate vocal emotions, bio-informational features (e.g., jitter, shimmer, and harmonicity) are also informative. In addition, our correlation analysis suggested that vocal emotions could be conveyed by a set of features rather than some individual features alone.

**1pSC16. Fearless steps Apollo: Advancements in robust speech technologies and naturalistic corpus development from Earth to the Moon.** John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX), Aditya Joglekar (CRSS: Ctr. for Robust Speech Systems, The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, aditya.joglekar@utdallas.edu), and Meena Chandra Shekar (Elec., Univ. of Texas at Dallas, Dallas, TX)

Recent developments in deep learning strategies have revolutionized Speech and Language Technologies(SLT). Deep learning models often rely on massive naturalistic datasets to produce the necessary complexity required for generating superior performance. However, most massive SLT datasets are not publicly available, limiting the potential for academic research. Through this work, we showcase the CRSS-UTDallas led efforts to recover, digitize, and openly distribute over 50,000 hrs of speech data recorded during the 12 NASA Apollo manned missions, and outline our continuing efforts to digitize and create meta-data through diarization of the remaining 100,000hrs. We present novel deep learning-based speech processing solutions developed to extract high-level information from this massive dataset. Fearless-Steps APOLLO resource is a 50,000 hrs audio collection from 30-track analog tapes originally used to document Apollo missions 1,7,8,10,11,&13. A customized tape read-head developed to digitize all 30 channels simultaneously has been deployed to expedite digitization of remaining mission tapes. Diarized transcripts for these unlabeled audio communications have also been generated to facilitate open research from speech sciences, historical archives, education, and speech technology communities. Robust technologies developed to generate human-readable transcripts include: (i) speaker diarization, (ii) speaker tracking, and (iii) text output from speech recognition systems.

**1pSC17. Analyzing field audio of parent-child book reading activities across various audio capture devices.** Sathvik S. Datla (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, 10424 offshore Dr., Irving, TX 75063, ssd180007@utdallas.edu), Satwik Dutta (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX), Jacob Reyna (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Dallas, TX), Jay Buzhardt, Dwight Irvin (Juniper Gardens Children's Project, Univ. of Kansas, Kansas City, KS), and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

**L**anguage **EN**vironment **A**nalysis (LENA) is a light weight audio capture device commonly used to monitor language including frequency of parent-child interactions for research purposes. However, various factors including price and technical limitations often limit the use of LENA for low-income families. Using a smartphone can be a feasible alternative when LENA is not available to practitioners/families. Over the past three decades, significant advancements have resulted in smartphone platforms/microphones/technology, allowing for high quality recorded audio which are available in most households. In this study we compare audio quality and measure performance of several Automatic Speech recognition (ASR) engines on audio captured from iPhone and Android relative to LENA devices. Families who consented in this study recorded reading activities with their children at home using both personal smartphones and LENA. Some challenges we found include recording synchronization, unnecessary background noises, and uneven room acoustics. Audio quality comparison is measured using Speech-Signal-to-noise ratio (SSNR) metric. Both open-source and fine-tuned ASR models are explored, with results reported using overall Word Error Rate (WER), as well as separated by speaker–Child versus Adult. Results show that smartphone platforms can be used versus LENA for child-adult language assessment. [Work sponsored by NSF through Grant Nos. 1918032 and 1918012.]

**1pSC18. Towards developing speaker diarization for parent-child interactions.** Abhejay Murali (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, 800 W Campbell Rd., Richardson, TX 75080, abhejay.murali@utdallas.edu), Satwik Dutta, Meena Chandra Shekar (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Dallas, TX), Dwight Irvin, Jay Buzhardt (Juniper Gardens Children's Project, Univ. of Kansas, Kansas City, KS), and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

Daily interactions of children with their parents are crucial for spoken language skills and overall development. Capturing such interactions can help to provide meaningful feedback to parents as well as practitioners. Naturalistic audio capture and developing further speech processing pipeline for parent-child interactions is a challenging problem. One of the first important steps in the speech processing pipeline is Speaker Diarization—to identify who spoke when. Speaker Diarization is the method of separating a captured audio stream into analogous segments that are differentiated by the speaker's (child or parent's) identity. Following ongoing COVID-19 restrictions and human subjects research IRB protocols, an unsupervised data collection approach was formulated to collect parent-child interactions (of consented families) using LENA device—a light weight audio recorder. Different interaction scenarios were explored: book reading activity at home and spontaneous interactions in a science museum. To identify child's speech from a parent, we train the Diarization models on open-source adult speech data and children speech data acquired from LDC (Linguistic Data Consortium). Various speaker embeddings (e.g., x-vectors, i-vectors, resnets) will be explored. Results will be reported using Diarization Error Rate. [Work sponsored by NSF via Grant Nos. 1918032 and 1918012.]

# Session 1pSP

## Signal Processing in Acoustics: Signal Processing in Acoustics Poster Session

Trevor Jerome, Chair
*Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd., BLDG 3 #329, West Bethesda, MD 20817*

All posters will be on display from 1:30 p.m. to 3:30 p.m. Authors of odd-numbered papers will be at their posters from 1:30 p.m. to 2:30 p.m. and authors of evennumberedpapers will be at their posters from 2:30 p.m. to 3:30 p.m.

### *Contributed Papers*

**1pSP1. Real-time acoustic detection and identification of drones in operational conditions.** Matthew Tan (Stanford Univ., 1277 W. John Beers Rd., Stevensville, Mi 49237, MI 49127, Matthtan@stanford.edu), Brett Y. Smolenski (North Point Defense, Rome, NY), and Darren Haddad (AFRL, Rome, NY)

Unmanned Aerial Vehicles (UAV), commonly known as drones, can fly below the radar horizon, can have very small radar cross-sections, and can drop munitions, which makes them a serious potential threat to public facilities, such as airports and shopping centers. However, UAVs emanate unique acoustic signatures that can be used to identify them and track their locations using microphone arrays. This paper presents a deep-learning approach to performing noise-robust drone detection and identification in real-time. In particular, a Convolutional Neural Network (CNN) with densely connected layers at the output was employed. Detection was accomplished by including an additional output for when only background environmental noise is present. The inputs were log-magnitude FFTs extracted from 10 ms Hanning windowed frames with 50% overlap. As long as the training data include a higher or equal level of noise than the target data, accuracy at detecting and identifying seven drones was above 90% for environmental noise levels as low as 20 dB SNR. Current research is focused on using a Bayesian version of the network to enable the classifier to report confidence levels in its decisions as well as detect novel drones, i.e., when a detected craft is not represented in the training data.

**1pSP2. Using wavelets to compress underwater acoustic data from the Gulf of Mexico.** Avery C. Landeche (Dept. of Phys., Univ. of New Orleans, 644 Whitney Dr., Slidell, LA 70461, aclandec@uno.edu), Bradley J. Sciacca (Dept. of Phys., Univ. of New Orleans, Westwego, LA), Brandon Howe, Shaun Pies, Kendal Leftwich, and Juliette W. Ioup (Dept of Phys., Univ. of New Orleans, New Orleans, LA)

This research investigated characteristics of the ocean, specifically in the Gulf of Mexico, to find the best wavelet to use for compression of the recorded data. Recorded underwater acoustic data as well as other types of oceanic data were studied. The power spectral density before and after wavelet decomposition, compression, and recomposition (decompression) are compared to assess the quality of the compression. This research describes a method to choose the best wavelet to analyze ocean acoustic data, with future use in the study of sea surface temperatures and heights (tides). [This material is based upon work supported by the National Science Foundation REU program under DMR-2049188, and partially funded by the Naval Research Laboratory-Stennis Space Center.]

**1pSP3. TSUNet: Transformer-based single channel speech enhancement using SkipConv U-Net for cochlear implant (CI) users.** Nursadul Mamun (EE, Univ. of Texas, Dallas, 7650 McCallum Blvd. Apt-611, Dallas, TX 75252-6399, nursad49@gmail.com) and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

The presence of multi-speaker babble noise greatly degrades speech intelligibility for communications with cochlear implant (CI) users. Convolutional Neural Network (CNN) based speech enhancement has been popular for suppressing noise because it can localize spectro-temporal information within the feature set. However, suppressing noise without creating artifacts is challenging in low SNR environments, and even more so if the noise is speech-like such as Babble noise. Transformers have emerged as a useful architecture with innate global self-attention mechanisms to capture long-range dependencies and global context, which is a limitation for CNN. The Mel spectrogram has a strong correlation along both time and frequency axes, and thus captures contextual information beneficial for speech enhancement. In this study, we propose TSUNet, which leverages both Transformers and U-Net to capture sufficient low-level details of contextual information in the speech time-frequency domain. TSUNet employs Transformer layers at the bottleneck of CNN based on U-Net. We also incorporate a neural vocoder to synthesize speech from the time-frequency representation without using a contaminated phase. The performance of the proposed network is evaluated under simulated and real recordings of noisy speech including CI testing. The proposed network achieves very effective performance in both scenarios.

**1pSP4. Active target classification using a shallow neural network with dimension reduction.** Sung-Hoon Byun (Korea Res. Inst. of Ships & Ocean Eng., 32 Yuseong-daero 1312beon-gil, Yuseong-gu, Daejeon 34103, South Korea, byunsh@kriso.re.kr) and Youngmin Choo (Sejong Univ., Seoul, South Korea)

Techniques for automatically identifying sonar targets using machine learning algorithms are being actively developed, and modern algorithms based on deep learning are often considered first. However, deep learning based algorithms require a lot of data to train a neural network, and if there is not enough data, overfitting easily occurs and the performance on real data can be degraded. In fact, in the case of active target detection where there is not enough data, it has been reported that using a shallow neural network after extracting appropriately designed features from the raw data showed better detection performance than applying deep learning directly to

the raw data. With regard to this, we investigate the performance of a shallow neural network combined with dimensionality reduction techniques for active sonar target classification. In particular, several linear and nonlinear dimension reduction techniques are compared in terms of target classification performance, and the effects of the characteristics of background noise and the presence of reverberation on the target classification performance are discussed. [This work was supported by the research project PES4380 funded by KRISO.]

**1pSP5. Image enhancement by motion compensation of a moving array for underwater target detection.** Sea-Moon Kim (Ocean System Eng. Res. Div., Korea Res. Inst. of Ships and Ocean Eng., 32 Yuseong-daero 1312beon-gil, Yuseong-gu, Daejeon 34103, South Korea, smkim@kriso.re.kr), Pan-Mook Lee (Ocean System Eng. Res. Div., Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea), Yeon-Seong Choo (Dept. of Ships and Ocean Eng., Korea Univ. of Sci. and Technol., Daejeon, South Korea), and Sung-Hoon Byun (Ocean System Eng. Res. Div., Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea)

Recently synthetic aperture sonar is widely used for precise seafloor exploration or underwater target detection. In order to acquire clearer sonar images, accurate measurement of the platform motion is absolutely required. Various approach has been studied for the motion estimation and compensation for sonar image generation. A tank test for underwater target detection has been done using an array mounted on a vertical bar attached on a horizontally moving platform. The array was vibrating when moving and the motion was measured using asynchronous navigational sensors installed just beside the array. In this study images constructed by synthetic aperture processing before and after motion compensation are compared. Across track array movement as well as translational motion with tilt angle was applied during the experiment. Additional numerical analysis of target detection with various motion parameters are performed. The effect of the motion estimation error and the installation offset are also analyzed. [This work is financially supported by the research project PES4380 funded by KRISO.]



MONDAY EVENING, 5 DECEMBER 2022 NORTH COAST A, 7:00 P.M. TO 9:00 P.M.

## Session 1eID

## Interdisciplinary: Tutorial on Effective Media Interactions Training Workshop

Kerri Seger, Cochair
*Applied Ocean Sciences, Fairfax Station, VA 22039*

Wendy Beatty, Cochair
*American Inst. of Physics, Media Services, College Park, MD 20740*

**Chair's Introduction—7:00**

**7:05**

The Public Relations Committee and the AIP Media Services team present this hands-on workshop for meeting attendees who are interested in effectively communicating scientific work to the public on Monday evening, 5 December. This workshop counts as the required media training for anyone interested in serving as a media liaison for their Technical Committee. The workshop will consist of presentations by media professionals to provide a toolkit of techniques for engaging the media. Following that, small group activities will give participants an opportunity to discuss and apply those techniques in round table discussions and practice sessions with media representatives. Though not required, attendees may find it helpful to have prepared a one minute "elevator pitch" of their research and/or an email response to a media person who has just asked for an interview about a recent publication. These items can be honed during planned round table events.

# Exhibit

An instrument and equipment exhibition will be located in the Summit Foyer on the 4th floor.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Monday, 5 December, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including lite snacks and a complimentary beverage.

Tuesday, 6 December, 9:00 a.m. to 5:00 p.m.: Exhibit Open Hours including a.m. and p.m. breaks serving coffee and soft drinks.

Wednesday: 7 December, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. break serving coffee.

## Session 2aAAa

# Architectural Acoustics, Noise, and ASA Committee on Standards: Sound Transmission and Impact Noise in Buildings

Matthew Golden, Cochair
*Pliteq, 131 Royal Group Crecent, Woodbridge L4H 1X9, Canada*

Benjamin M. Shafer, Cochair
*PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406*



### *Invited Papers*

**8:00**

**2aAAa1. Variations in sound transmission loss through vertical mass-decoupled partitions installed with a comprehensive sample of resilient channels.** Benjamin M. Shafer (PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

Resilient channels are commonly employed as a sound isolation treatment in building partition assemblies. Originally designed and patented as a crack-mitigation component in gypsum wallboard partition assemblies, the development and application of resilient channels as a mass-decoupling treatment for sound isolation in building partitions has achieved widespread acceptance in the building industry. Several recent laboratory sound transmission loss (STL) studies illustrate statistically significant variability in STL for steel-framed partitions. This experimental analysis attempts to more comprehensively characterize the variability in STL through mass-decoupled vertical partitions that employ steel resilient channels as a sound isolation treatment. Ten different models of resilient channels produced by five different manufacturers were tested with both damped and undamped panel configurations in a same-laboratory, same-series research program. Over 120 total tests were completed in this experimental study. The results of this study and recommendations for sound isolation design and specification are discussed.

**8:20**

**2aAAa2. Real-time identification of airborne sound flanking paths using a 3-dimensional acoustic camera.** Andrew Schmidt (USG Corp., 700 N Highway 45, Libertyville, IL 60048, ASchmidt@usg.com), Adam O'Donovan (VisiSonics Corp., Riverdale, MD), and Austin Phillips (USG Corp., Libertyville, IL)

Sound transmission testing, whether conducted in the laboratory or under field conditions, is critical for quantifying sound transmission performance of building constructions and elements. When samples undergoing sound transmission loss testing are constructed in the laboratory, results can suffer from airborne sound flanking paths that may not be apparent through visual inspection of the test sample. In the field, noise reduction testing between spaces can be similarly influenced by airborne sound flanking paths that are not immediately obvious. One methodology is presented utilizing a portable calibrated device that generates accurate overlays of sound pressure level over visual images (the VisiSonics 5/64 Acoustic Camera), allowing for the successful identification of unwanted flanking paths in both laboratory and field environments. Data are presented from recent laboratory testing that show the efficacy of acoustic imaging to identify sound flanking paths in real-time, enabling users to optimize sound transmission performance of an assembly.

**8:40**

**2aAAa3. Lateral impact noise I: Theory and application.** Wayland Dong (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, California, CA 90404, wdong@veneklasen.com), John Lo Verde, and Samantha Rawlings (Veneklasen Associates, Santa Monica, CA)

Impact noise insulation has traditionally been thought of mostly in terms of vertical adjacencies. However, impact noise is transmitted in all directions though a building, and lateral and diagonally adjacent impact noise transmission can be significant. Lateral and diagonal impact transmission is a one-junction structural flanking path and is, therefore, amenable to statistical energy analysis-based calculation methods such as ISO 12354. In this paper, we review the relevant results from structural flanking path calculations and show how this provides a framework for understanding major features of lateral impact noise transmission such as the frequency dependence and effect of structure type. In turn, measurement of lateral impact noise provides a method of estimating the vibration reduction across junctions. These findings are illustrated with examples from field testing.

**8:55**

**2aAAa4. Lateral impact noise II: Practice and recommended procedures.** John Lo Verde (Veneklasen Assoc., 1711 Sixteenth St., Santa Monica, CA 90404, jloverde@veneklasen.com), Wayland Dong, and Samantha Rawlings (Veneklasen Assoc., Santa Monica, CA)

Impact noise insulation has traditionally been thought of mostly in terms of vertical adjacencies, and ASTM E1007 provides no guidance for performing such a measurement in a lateral, diagonal, or other non-vertical condition. Because lateral impact transmission can be significant, the authors have developed methods and practices for performing this measurement (see papers by the authors at ICSV 2017 and Inter-Noise 2019). ISO 16283-2 also describes procedures and methods for the measurements and analysis of lateral impact noise measurements. The various measurement methods are compared and evaluated. For some assembly types, there is an advantage to maintain a constant distance from tapping machine to the separating partition. Appropriate ratings for categorization of lateral impact isolation are discussed, and changes to E1007 to clarify the measurement method are proposed.

**9:10**

**2aAAa5. Adventures in lab testing: Optimizing a new construction floor-ceiling acoustics lab.** Evelyn Way (Research & Development, Maxxon Corp., 920 Hamel Rd., Hamel, MN 55340, evelynway@gmail.com)

The Maxxon Acoustics Lab, a purpose-built, floor-ceiling test lab was completed at the end of 2021, but the verification process was just beginning. Results coming out of the lab were verified using slabs tested at another lab and wood frame assemblies compared to historical data collected on comparable assemblies. The iterative process of modifying mounting conditions will be presented, and data will be analyzed based on test condition modifications and lab facility differences.

**9:30**

**2aAAa6. The use and application of pressure-based acoustical metrics adopted within the International Building Code.** Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com), John Lo Verde, and Wayland Dong (Veneklasen Assoc., Santa Monica, California, CA)

The 2021 International Building Code (IBC) includes pressure-based metrics as the field acoustical standards for conformance. These include normalized noise isolation class (NNIC) and normalized impact sound rating (NISR) (ASTM E336-20 and ASTM E1007-21, ASTM International; 2021 International Building Code, Section 1206, International Code Council). Pressure-based metrics were intentionally used in lieu of power based metrics. The authors here address why pressure-based metrics are more appropriate than power-based metrics for field verification, following from previous work [LoVerde and Dong, "Field impact insulation testing: Inadequacy of existing normalization methods and proposal for new ratings analogous to those for airborne noise reduction," JASA **118**, 638 (2005); LoVerde, Dong, and Rawlings, "Sound pressure-based ratings for evaluation of *in situ* sound isolation," Denver (2022)]. Furthermore, the authors shall present the application of these metrics as they pertain to Building Code requirements. This paper will include and present code interpretation as it relates to sample size and evaluation for compliance as it relates to Building Code and sampling of systems for acceptance.

**9:50–10:05 Break**

**10:05**

**2aAAa7. Noise from above: A summary of studies regarding the perceived annoyance due to impact sounds.** Markus Mueller-Trapet (National Res. Council Canada, 1200 Montreal Rd., Ottawa, Ontario K1A 0R6, Canada, Markus.Mueller-Trapet@nrc-cnrc.gc.ca), Sabrina Skoda (Hochschule Duesseldorf Univ. of Appl. Sci., Duesseldorf, Germany), Young-Ji Choi (Kangwon National Univ., Gangwon-do, South Korea), Iara Batista da Cunha, and Jeffrey Mahn (National Res. Council Canada, Ottawa, Ontario, Canada)

To support efforts of introducing an impact sound requirement into the National Building Code of Canada (NBCC), the National Research Council of Canada has initiated a long-term research project. In cooperation with the Hochschule Duesseldorf in Germany and Kangwon National University in Korea, several listening tests were performed to investigate the annoyance due to impact sound as it is perceived by building occupants, and how this annoyance relates to the results of standardized laboratory measurements. In this contribution, the different listening test setups will be described and compared. Tests were performed in the laboratory using loudspeakers, headphones, and also with a Virtual Reality headset. In addition, an online listening test was launched to expand the reach of such studies to the general population. The results will be summarized and discussed regarding appropriate rating methods. The goal of these studies is to gain a better understanding of the relevant factors that affect the perceived annoyance, such as the impact source type and the dominant frequency range, but also how the test environment influences the results.

**10:25**

**2aAAa8. Overview of technical certificate program from the Institute of Noise Control Engineering.** Matthew Golden (Pliteq, 131 Royal Group Crecent, Woodbridge, Ontario L4H 1X9, Canada, mgolden@pliteq.com), Ethan Bourdeau (Int. WELL Bldg. Inst. PBC, New York, NY), Jeffrey Fullerton (Acoustics, Bldg. Sci. Solutions, Intertek, Boston, MA), Herb Singleton (Cross Spectrum Acoust., East Longmeadow, MA), Karl Peterman (Vibro-Acoust., Swegon North America, Ajax, Ontario, Canada), and Andrew Barnard (Acoust., Penn State, Univ. Park, PA)

Standards are often misunderstood by even the most seasoned of acoustic professionals, let alone new technicians and engineers in the field, as they are often very complicated, written in piecemeal fashion by committees and are updated often. Some of the standards even conflict with one another. To help alleviate this problem, the Institute of Noise Control Engineering (INCE) is developing two technical certificate programs designed to educate and evaluate participants on the details of common standards used in the noise control industry. Each program will contain education, testing, and continuing education on ASTM International and Acoustics Society of America (ASA) standards. The first program will cover general acoustical measurements and environmental noise measurements. The

second program will cover field testing of building acoustics, including airborne and impact noise insulation as described in ASTM E336, ASTM E1007, and associated standards. The courses are currently being developed with a target launch date of early 2023. Comparisons to similar programs that have been developed in other countries will be included. A business case will be made on how these programs will save noise control engineering firms time and money in training new staff.

10:45

**2aAAa9. *In situ* testing methods for evaluating and predicting the impact isolation of high-performance floating floors - Part 1.**
Michael Raley (PAC Int., Canby, OR) and Peter Allen (ABD Eng. and Design, 321 Southwest 4th Ave. Suite 700, Portland, OR, pallen@abdengineering.com)

Floating floors using discrete isolators are commonly recommended to mitigate impact noise from fitness facilities; however, there has been little study of the performance of floating floors under the types of heavy/hard impacts that often cause noise and vibration complaints from fitness facilities. Furthermore, it can be difficult to create mock-ups of floating floor systems for *in situ* testing, making it difficult to adequately evaluate their performance. Here, the authors present the beginnings of a research program intended to develop effective methods for evaluating and predicting the performance of floating floors using *in situ* testing methods. Part 1 of the presentation covers the overall scope of the research program, a discussion of prior work by others that informed the research program, an overview of the measurement setup and methodology, and a discussion of the methods to capture, post-process, and analyze the data.



11:00

**2aAAa10. *In situ* testing methods for evaluating and predicting the impact isolation of high-performance floating floors - Part 2.**
Michael Raley (PAC Int., Canby, OR) and Peter Allen (ABD Eng. & Design, Inc., 321 Southwest 4th Ave. Suite 700, Portland, OR, pallen@abdengineering.com)

Floating floors using discrete isolators are commonly recommended to mitigate impact noise from fitness facilities; however, there has been little study of the performance of floating floors under the types of heavy/hard impacts that often cause noise and vibration complaints from fitness facilities. Furthermore, it can be difficult to create mock-ups of floating floor systems for in-situ testing, making it difficult to adequately evaluate their performance. Here, the authors present the beginnings of a research program intended to develop effective methods for evaluating and predicting the performance of floating floors using in-situ testing methods. Part 2 of the presentation covers the results of the measurements, including the effects of the following changes to the floating floor assembly: various air cavity depths, vented vs. un-vented air cavities, loaded vs. un-loaded floors, and concrete versus plywood floors. Finally, Part 2 presents the intended next steps in the research program.

## *Contributed Papers*

11:15

**2aAAa11. Evaluation of dry mass timber assemblies and comparison to cementitious toppings.** Aedan Callaghan (Pliteq, Inc., 4211 Yonge St., Suite 400, Suite 404, Toronto, Ontario M2P 2A9, Canada, acallaghan@pliteq.com) and Matthew Golden (Pliteq, Woodbridge, Ontario, Canada)

Mass timber construction is an increasingly common way of building taller and larger wood buildings utilizing renewable building materials with lower embodied carbon. Given the desire for the aesthetic design of exposed mass timber ceilings, it is common for these mass timber panels to require an additional subfloor decoupled by an acoustic interlayer to achieve IBC 1207 requirements. In the efforts to minimize added structural weight and overall embodied carbon, various dry linings were tested as an alternative to traditional cementitious toppings. A 175mm cross laminated timber (CLT) panel was constructed in the laboratory, and numerous ASTM E90 and ASTM E492 tests were performed. This analysis examines the role of dynamic stiffness of the acoustic interlayer and various mass layers in the resulting performance. Assembly transmission loss data from ASTM E90 testing are assessed to understand if it agrees with mass law.

11:30

**2aAAa12. Acoustic criteria and design approaches for fitness spaces in mixed-use buildings.** Robert Connick (Architectural Acoust. and Mech. Noise Control, Acentech, 33 Moulton St., Cambridge, MA 02138, rconnick@acentech.com)

Boutique fitness franchises have been popping up more often across the nation, and most high-rise residential towers include fitness centers located around and above residential spaces. When high impact exercise activities neighbor offices and residences, disturbances and complaints often follow suit. In this presentation, we look at the most common types of fitness spaces, from big name franchises to small scale fitness centers, in terms of their program of activities and severity of their impact, as well as what methods of noise and vibration mitigation are available and effective. Several case studies are presented.

TUESDAY MORNING, 6 DECEMBER 2022 SUMMIT E, 9:00 A.M. TO 11:00 A.M.

## Session 2aAAb

## Architectural Acoustics: Student Design Competition (Poster Session)

Robin Glosemeyer Petrone, Chair
*Threshold Acoustics, 141 W Jackson Blvd #2080, Chicago, IL 60604*

The Student Design Competition is sponsored by the ASA Technical Committee on Architectural Acoustics, with generous support from the Wenger Foundation, the Robert Bradford Newman Student Award Fund, and the National Council of Acoustical Consultants.

This competition is intended to encourage students in the disciplines of architecture, engineering, physics, and other curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics and noise control in the design of a facility in which acoustical considerations are of significant importance.

Design Scenario: The project involves designing a new 700-seat end-stage theatre to accommodate dramatic and musical performances, which includes a pit for up to 15 musicians for musicals. The building also includes a variety of rehearsal rooms.

All posters will be on display from 9:00 a.m. to 11:00 a.m.

TUESDAY MORNING, 6 DECEMBER 2022 GRAND HALL A, 8:30 A.M. TO 11:45 A.M.

## Session 2aAB

## Animal Bioacoustics: General Topics in Animal Bioacoustics I – Terrestrial

Rolf Müller, Chair
*Mechanical Eng., Virginia Tech, ICTAS II, 1075 Life Science Cir, (Mail Code 0917), Blacksburg, VA 24061*

### *Contributed Papers*

**8:30**

**2aAB1. Investigating the integration of biosonar sensing and flight control in bats on Borneo.** Rolf Müller (Mech. Eng., Virginia Tech, ICTAS II, 1075 Life Science Cir, (Mail Code 0917), Blacksburg, VA 24061, rolf.mueller@vt.edu), Benjamin C. Beiter (Mech. Eng., Virginia Tech, Blacksburg, VA), Trinity Blackman (Mech. and Aerosp. Eng., Univ. at Buffalo, Buffalo, NY), Yihao Hu, Michael Goldsworthy (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Jesse Granger (Biology, Duke Univ., Durham, NC), Grant Patterson (Eng. and Comput. Sci., Virginia State Univ., Petersburg, VA), Yohan Sequeira (Mech. Eng., Virginia Tech, Blacksburg, VA), Amaro Tuninetti (Cognit., Linguist., & Psychol. Sci., Brown Univ., Providence, RI), Adam Tyler (Mech. Eng., Virginia Tech, Blacksburg, VA), and Cara Webster (Biology, Texas A&M Univ., College Station, TX)

Many bat species rely on biosonar as their primary source of sensory information about their environments. Species that are able to navigate amid dense vegetation use this biosonar information to guide highly dexterous flight maneuvers. Understanding the connection between biosonar inputs and flight outputs poses a challenge, because the flight apparatus of bats has the most degrees of freedom of any flight system—whether natural or man-made. At the same time, the biosonar inputs consist of just two one-dimensional acoustic time signals (i.e., echoes received at the two ears). Due to the complexity and large variability in the echoes and the flight maneuvers of bats, understanding the input-output relationships requires the ability to collect large amounts of quantitative data on the acoustics and the flight kinematics. In addition, comparing different bat species could offer a window into the principles behind the evolutionary co-adaptation of biosonar and flight. To accomplish this, a cylindrical flight tunnel that integrates synchronized arrays of 50 high-speed video cameras and 32 ultrasonic microphones has been set up on the island of Borneo. This instrument is complemented by a set of custom deep-learning techniques that can handle the large amounts of data that are being produced.

8:45

**2aAB2. Field study of bats in Brunei Darussalam.** Cara Webster (Biology, Texas A&M Univ., College Station, TX, cfjwebster17@tamu.edu), Amaro Tuninetti (Cognit., Linguist., & Psychol. Sci., Brown Univ., Providence, RI), Jesse Granger (Biology, Duke Univ., Durham, NC), Syafi'ie Su'eif, Ibnurrafiq Ariffin, Ulmar Grafe (Faculty of Sci., Univ. Brunei Darussalam, Bandar Seri Begawan, Brunei Darussalam), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Studying model organisms in their natural environments is essential for advancing bioinspired science and technology. Echolocating bats can be models to inspire new technology for sonar sensing, mobility with flapping flight as well as the integration of these two capabilities. Bat species belonging to genera such as *Rhinolophus* and *Hipposideros* stand out for their ability to navigate and hunt prey in dense vegetation. To further the study of echolocation and flight strategies in these genera, field-based research methods have been used to observe, record, and capture bats in Brunei Darussalam. This work was conducted at Andulau Forest Reserve on the border of Belait and Tutong Districts, various other field sites in Tutong District, as well as at Mata Mata in Brunei-Muara District from June 2022 through August 2022. Harp traps were primarily used to capture bats, a less invasive capture technique compared to other types of netting and thought to be more effective against the sophisticated echolocation capabilities of these bats. Acoustic recordings were collected at trap sites and upon processing. These recordings can be utilized to advance passive acoustic monitoring and to improve software that can distinguish species based on sonar characteristics which can also support general biodiversity monitoring.

9:00

**2aAB3. Spatiotemporal dynamics of biosonar in navigating Bornean Rhinolophid and Hipposiderid bats.** Amaro Tuninetti (Cognit., Linguist., & Psychol. Sci., Brown Univ., Box 1821, Providence, RI 02912-9067, Amaro_Tuninetti@brown.edu), Yohan Sequeira (Virginia Polytechnic Inst., Blacksburg, VA), Jesse Granger (Duke Univ., Durham, NC), Cara Webster (Texas A&M Univ., College Station, TX), Benjamin C. Beiter (Virginia Polytechnic Inst., Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

To investigate the active sensing strategies used by echolocating bats of the genera Rhinolophus and Hipposideros, we have constructed a 9-meter long flight tunnel, which incorporates an array of 32 ultrasonic microphones distributed throughout the tunnel. Rhinolophus and Hipposideros are of special interest because of their highly flexible biosonar system; these bats emit pulses from their nasal cavities, using complex noseleaf structures to quickly and precisely alter the beam-form and direction of emissions. Additionally, each species utilizes a unique combination of constant-frequency (CF) and frequency-modulated (FM) ultrasonic signals with varying durations, repetition rates, and frequencies. We plan to trap several species of wild Bornean bats of these genera and fly individual bats through the tunnel; a time-of-arrival algorithm will be used to localize the position of each bat at the time of each biosonar pulse emission, and an amplitude-comparison method to measure the horizontal and vertical direction of each emission from the bat's noseleaf. We will also incorporate relatively simple foliage obstacles into the tunnel; this will create complex acoustic clutter and allow us to determine how bats of different species adjust their biosonar sampling strategies in order to navigate around novel obstacles in a cluttered environment.

9:15

**2aAB4. Machine learning methods for reconstructing the acoustic fields of bat biosonar.** Yohan Sequeira (Mech. Eng., Virginia Tech, Blacksburg, VA, yohans21@vt.edu), Amaro Tuninetti (Cognit., Linguist., & Psychol. Sci., Brown Univ., Providence, RI), Michael Goldsworthy (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Understanding the relationships between biosonar and flight in bats requires synchronized recordings of the echo inputs and the flight kinematics of the animals. Here, integrated arrays of 50 high-speed camera and 32 ultrasonic microphones have been set up to collect the data that is necessary for achieving this goal. The arrays are set up to record bats as they fly through an an obstacle course inside a cylindrical tunnel. Due to the complexity of the received signals in these scenarios, we have been developing custom strategies that rely on a combination of compressed sensing and deep learning to reconstruct the acoustic field inside the tunnel from the microphone-array measurements. By using a combination of the high-speed video and acoustic data, it can be attempted to determine the bat's location in the array from the image data and then reconstruct the properties of the sound fields that were emitted and received at this position using the acoustic array data. The goal of this research is to eventually determine the dynamic characteristics of the bat's biosonar emissions such as beampatterns and potentially time-variant characteristics as well as the biosonar inputs that the bats rely on for controlling their flight behaviors.

9:30

**2aAB5. Investigating of the impact of biommetic pinna dynamics on sonar tasks in natural environments.** Michael Goldsworthy (Elec. and Comput. Eng., Virginia Tech, 1075 Life Sci. Cir, ICTAS II (Mail Code 0917), Blacksburg, VA 24061, michaeljg@vt.edu), Salwani Osman (Comput. Sci., Univ. Of Brunei Darussalam, Bandar Seri Begawan, Brunei Darussalam), Grant Patterson (Eng. and Comput. Sci., Virginia State Univ., Petersburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Bats are known to perform incredible feats using their echolocation abilities, which include natural tasks, such as navigation through foliage and hunting prey, as well as tasks they have been trained to perform, such as classification of man-made objects. Much work has been done to engineer biomimetic systems that can perform these and similar tasks in a comparable way to bats. Biomimetic sonar devices have performed tasks such as identifying a geographic location, object classification, and foliage gap finding. Particular species of bats (notably those of the families Hipposideridae and Rhinolophidae) use rapid ear motion while making sonar calls, and the full effect and purpose of this motion is not fully understood. Our research goal is to perform sonar tasks with a biomimetic sonar head capable of both static and dynamic ear motion and investigate the effect of dynamic ear motion on captured echoes and performance on tasks. To this end, we have collected echo data in natural habitats with dense tropical vegetation in Brunei on the island of Borneo. Tasks currently under investigation include location identification and detection of man-made objects in foliage, and are performed by deep learning classifiers trained on echoes taken by the biomimetic sonar device.

9:45

**2aAB6. Autonomous localization and mapping based on biomimetic sonar in natural environments.** Grant Patterson (Eng. and Comput. Sci., Virginia State Univ., Petersburg, VA), Ju Wang (Eng. and Comp. Sci., Virginia State Univ., 1 Hayden Dr., Petersburg, VA 23806, jwang@vsu.edu), Michael Goldsworthy (Elec. and Comput. Eng., Virginia Tech, Blacksburg, VA), Salwani Osman (Comput. Sci., Univ. of Brunei, Bandar Seri Begawan, Brunei Darussalam), Adam Hinson, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Bats navigating in dense vegetation based on biosonar have to obtain the necessary sensory information from "clutter echoes," i.e., echoes that are superpositions of contributions of many reflecting facets (e.g., leaves). Since the locations and reflective properties of the individual facets are unknown, clutter echoes have to be treated as random signals that can neither be predicted nor—under most practical circumstance—be replicated. Nevertheless, prior research has shown that deep neural networks are capable of extracting fairly precise location information from clutter echoes. This raises the question whether clutter echoes could be used to provide navigational guidance in a conventional simultaneous localization and mapping (SLAM) framework, which is to a large extent dependent on precise and deterministic measurement to generate a localized map. Our hypothesis is that biomimetic sonar echoes indeed contain the information that is necessary support to local path planning, which can be utilized by a suitable deep learning architecture and training process. To investigate this issue, we have collected data in natural, heavily vegetated environments using a biomimetic sonar head that mimics the periphery of the biosonar system in horseshoe bats. The annotation of the data and the network training process is currently undergoing.

**10:00–10:15 Break**

**10:15**

**2aAB7. A bioinspired robot for investigating biosonar guidance of flapping flight in bats.** Adam Tyler (Mech. Eng., Virginia Polytechnic Instit. and State Univ., Blacksburg, VA, adamtyler365@vt.edu), Trinity Blackman (Aerosp. Eng., Univ. at Buffalo, Buffalo, NY), Yohan Sequeira, Benjamin C. Beiter (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Investigating how bats manage to control their complex flight apparatus using streams of biosonar echoes as inputs with an approach based on biomimetic robotics, we have designed a bioinspired flapping robot that replicates some of the features of the flight apparatus of bats: In particular, it incorporates a mechanism to translate one degree of freedom from a motor into flapping as well as folding of the wings. Throughout the flapping cycle, the wings are folding and unfolding continuously in a way that achieves maximum upward lift. Similarly to a bat, a wing membrane is stretched over the wings and pulled taut. The design is primarily 3-D printed to reduce weight and allow for rapid prototyping. Future iterations of the robot will continue to reduce the weight and add more degrees of freedom to enable greater maneuverability in the air. After flight has been achieved, acoustics in the form of ultrasonic microphones will be implemented on the robot for further navigational capabilities. It is important to optimize the robot for the maximum amount of lift while keeping the robot as light as possible to allow for a greater capacity for acoustics equipment while still maintaining the ability to fly.

**10:30**

**2aAB8. A comparative study of a bioinspired flapping bat robot and Bornean bats.** Trinity Blackman (Univ. at Buffalo, P.O. box 802, East Aurora, NY 14052, tb10933@gmail.com), Adam Tyler (Virginia Tech, Blacksburg, VA), Yohan Sequeira (Mech. Eng., Virginia Tech, Blacksburg, VA), Benjamin C. Beiter, and Rolf Müller (Virginia Tech, Blacksburg, VA)

Effectively characterizing the significance of bioinspired design for the advancement of robotics technology is a complex endeavor. Here, a direct comparative analysis was conducted to examine the differences between a bioinspired bat robot and bats. The unique maneuvering and biosonar capabilities of bats have long been recognized as a source of technical inspiration. However, the capabilities of bats in these areas still far exceed those of current robotics models. To evaluate these differences, we have been preparing to fly rhinolophid/hipposiderid bats and a bat-robot prototype through a flight tunnel instrumented with arrays of 50 high-speed cameras and 32 ultrasonic microphones. Using this approach, kinematic and acoustic data can be compiled to enable a thorough comparison of the bats and the robot. This data can support a quantitative analysis of key characteristics such as wing cycle, size ratios, lift and thrust capacity, as well as maximum carrying capacity. Based on insight from this data, strategic design iterations can be carried out to more accurately mimic bat flight with considerations for incorporation of biosonar-inpsired technology into the design. By continuing this process, an in-depth understanding of the bioinspired design approach, implementation, and impact to the design process can be achieved.

**10:45**

**2aAB9. Optimization approach to designing a bioinspired bat robot for flight and sonar integration.** Benjamin C. Beiter (Mech. Eng., Virginia Tech, 635 Prices Fork Rd., Blacksburg, VA 24061, bbeiter1@vt.edu), Yohan Sequeira (Mech. Eng., Virginia Tech, Blacksburg, VA), Adam Tyler (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), Trinity Blackman, and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

In addition to remarkable acoustic sensing and navigation abilities, bats are highly agile and capable fliers, achieving flight efficiency that exceeds that of not only the rest of the animal kingdom, but of all robots as well. We seek to design a robot inspired by the biological capabilites of bats to achieve artificial flapping flight integrated with an acoustic sensing ability. Coupled with the kinematic and dynamic data collection array, we define a process for optimizing the design of bat wings for efficient flight. We identify fitness functions such as flap speed and air subtended throughout a wing cycle and then optimize the size and shape of a wing to achieve desired setpoints of the fitness functions. An inverse kinematics design process can then be used to create a single degree of freedom cyclic mechanism that will achieve the desired wing flap and fold functions. By setting fitness function objectives for both engineering design requirements (weight, lift, aerodynamics) and biologically inspired goals (wing flexibility, similarity to bat flight, responsiveness to echolocation). With this we can procedurally design a bat robot based on updating understandings of bat flight and navigation, leading to a streamlined production process when combined with rapid manufacturing.

**11:00**

**2aAB10. Impact of environmental factors on the variability of the avian dawn chorus.** Kelsey B. Moore (Statistics, Brigham Young Univ., N283 ESC, Provo, UT 84602, kbmoore@byu.edu), Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Mylan R. Cook (Phys.ics and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall (California State Univ. Bakersfield, Bakersfield, CA), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The avian dawn chorus is especially prominent during the spring and summer when birds are breeding. The short-term and seasonal variation (timing, level, duration, and spectrum) of the avian dawn chorus is not well understood, and this study represents an initial effort to correlate changes in different chorus characteristics with different environmental variables. Acoustical data were gathered with a Larson Davis Sound Level Meter at the U.S. Fish and Wildlife Service Bear River Migratory Bird Refuge from March through August 2021 and 2022 and were analyzed for variation in start time and acoustical characteristics. Although it has been shown that the chorus start time is roughly correlated with civil twilight, we found that they ranged from 3 h before nautical twilight to half an hour after. Additionally, daily maximum SPL during a dawn chorus can vary during the peak breeding season by more than 40 dB. We analyzed the effect of several environmental variables on the dawn chorus, including nearby water levels, cloud coverage, wind, etc., as well as overall seasonal variation. Our findings highlight the importance of water and demonstrate the need for continued research and analysis of daily variation. Results will be discussed.

**11:15**

**2aAB11. Evolution of the dawn chorus throughout the breeding season at an inland migratory bird refuge.** Levi T. Moats (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmoats359@gmail.com), Mylan R. Cook, Kelsey B. Moore (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Lucas K. Hall (Biology, California State Univ. at Bakersfield, Bakersfield, CA), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The dawn chorus is a defining acoustic event for many environments with vocal bird populations. To study short and long-term changes in spectral characteristics of the dawn chorus at an inland migratory bird refuge, Larson Davis 831C sound level meters (SLM) were deployed at Utah's Bear River Migratory Bird Refuge from March to August in 2021 and 2022, overlapping with the breeding seasons of many resident and migratory species of birds. The SLMs collected one-second temporal resolution data in one third octave (OTO) frequency resolution both years, as well as audio recordings during 2022. The relative contributions of the bird-relevant OTO bands ($\sim$500 Hz–12.5 kHz) to the total energy are analyzed for each chorus. One result shows that the chorus's spectral composition varies markedly for different locations within the refuge throughout the breeding season. The results show the value of the relatively low-resolution data in studying overall chorus characteristics. Additionally, the audio recordings have been analyzed using BirdNET species identification software to correlate changes in chorus characteristics with changes in species and behavior over the breeding season.

11:30

**2aAB12. Parabolic completions in gibbon duets may signal appreciation of projectile motion.** David M. Schruth (Music, Durham Univ., NA, Durham NA, United Kingdom, dschruth@anthropoidea.org)

Most gibbon species produce salient duet calls at daybreak. Duets start with low frequency barks by males, followed by the female great call, and end with a short, and often complex, male-dominated coda. The female great call itself typically climaxes via a crescendoing increase in pitch, tempo, or both and characteristically features bilaterally symmetrical parabolic structures, which can manifest both in the distribution of vocal units over time as well as in frequency. Male codas appear to anticipate and even complete many of these female-initiated parabolas. Employing spectrograms of species-typical great calls from nearly all gibbon species ($n = 12$), I plotted coordinates of the upper-most frequency of each vocal unit. Using these $x = $time and $y = $frequency coordinates (plus x-differences), I tabulated the parabolas with the best possible second order polynomial fits for each species' great call. Measures of parabolic fit for each call were then compared to quantitative locomotor estimates for each species. All forms of parabolic assessment had positive correlations with leaping bout percentages across species. These results indicate that gibbon duets may function to signal a fundamental understanding of parabolic shapes—presumably useful in landing airborne locomotor (especially leaping) bouts spanning canopy elements—enabling concordant execution of arboreally projectile acrobatics.



TUESDAY MORNING, 6 DECEMBER 2022 NORTH COAST A, 8:30 A.M. TO 11:50 A.M.

## Session 2aAO

## Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Acoustical Remote Sensing, Navigation, and Passive Monitoring in the Polar Ocean I

Matthew Dzieciuch, Cochair
*Univ. of California, San Diego, Scripps Inst. of Oceanography, San Diego, CA 92122*

Hanne Sagen, Cochair
*Nansen Environ. and Remote Sensing Ctr., Jahnebakken 3, Bergen, 5007, Norway*

Peter F. Worcester, Cochair
*Scripps Inst. of Oceanography, Univ. of California, San Diego, 9500 Gilman Drive, 0225, San Diego, CA 92093-0225*

**Chair's Introduction—8:30**

### *Contributed Papers*

8:35

**2aAO1. Wind-driven ambient noise in seasonally ice-covered waters north of the Svalbard Archipelago.** Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Norwegian Defence Res. Establishment, Horten 3191, Norway, Dag.Tollefsen@ffi.no) and Helge Buen (Norwegian Defence Res. Est. (FFI), Horten, Norway)

This paper presents analysis of a one-year (2018–2019) recording of ambient noise (40–2000 Hz) at a seasonally ice-covered location on the continental slope between the Svalbard archipelago and the Nansen Basin, northeast Atlantic Arctic. Time series of ambient noise show highest correlations with ice concentration and wind speed. A log-wind speed regression model is fitted to spectral noise data for three categories of ice concentration. Wind-speed dependence decreases with increasing ice concentration and increases with frequency, except at high ice concentration. Periodicity in noise during the ice-covered season is related to the M2 and M4 tidal current constituents.

8:50

**2aAO2. Ambient sound observations from beamformed horizontal array data in the Pacific Arctic.** Colby W. Cushing (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, colby.cushing@utexas.edu), Jason D. Sagers, and Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Changes in the Arctic environment with regard to declining sea ice and changing oceanography are expected to alter the ambient sound field, affecting both the sound generating processes and the acoustic propagation. This talk presents acoustic recordings collected on the 150-m isobath on the Chukchi Shelf during the Canada Basin Acoustic Propagation Experiment (CANAPE), which took place over a yearlong period spanning October 2016 to October 2017. The data were recorded on a 52-channel center-tapered horizontal line array and adaptively beamformed to quantify the azimuthal directionality in long-term trends ambient sound under 1200 Hz as well as track specific sound events as they travel through space over time.

The acoustic data were analyzed in the context of wind speed and satellite imagery to identify the dominant sound generation mechanisms. Automated identification system (AIS) data were also incorporated to determine sources of ship generated sound and seismic profiler activity observed in the acoustic recordings. This talk will provide an overview of the long-term trends and describe a subset of results from the beamforming. [Work Supported by ONR.]

**9:05**

**2aAO3. Statistical and spatial characteristics of ocean ambient noise up to 1900 Hz on the Chukchi Shelf in the Arctic affected by climate change.** Kathryn Fung (MIT/WHOI, Woods Hole, MA) and Julien Bonnel (Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu)

This paper analyzes a year of underwater ambient noise data recorded in the Arctic on the Chukchi Shelf as part of the 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE). A broadband (50–1900 Hz) statistical study is performed to analyze noise variability and its relationship to environmental drivers, notably the local presence of ice and the presence/absence of the Beaufort duct in the experimental area. Both environmental factors are found to significantly affect the noise levels. Local ice coverage tends to decrease ambient noise at all frequencies, while the presence of the Beaufort duct tends to increase ambient noise for frequencies below 1 kHz. The lowest ambient noise levels are, thus, found when the sea is ice covered, but the duct is absent. Furthermore, the study explores the link between noise level and distant ice drift magnitude. The ambient noise levels are shown to be highly correlated with distant (up to 1400 km) ice drift for frequencies between 300 and 1500 Hz. [Work supported by the Office of Naval Research.]

**9:20**

**2aAO4. Shipping noise modelling in the thawing Arctic Ocean.** Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Kerri D. Seger, Christopher Verlinden, and Andrew Heaney (Applied Ocean Sciences, Fairfax Station, VA)

One of the earliest and clearest signals of the impact of anthropogenic induced climate change has been the severe reduction in sea ice extent in the Arctic Ocean, particularly in the summer. Predictions of an ice free summer range from 2030 to 2050. The reduction of ice extent and thickness opens the possibility for greater shipping traffic through polar regions for international shipping (cutting the transit time from Europe to Asia), for mine and indegenous nations logistics support and for high north fishing. The increased number and presence of ships is a potential threat to the sensitive marine species that live in the arctic and have evolved in a region devoid of anthropogenic sounds. Effective management of the region requires an understanding of the potential increases in shipping noise levels and in their relative levels to natural backround sound, such as wind and ice noise. In this paper, we present a modelling study of 2013–2019 shipping noise and look to build a model to predict the shipping noise in 2030 based upon the combination of national forecast economic plans and ice modeling. Mapping the projected levels of shipping noise into excess noise (level above wind and/or background ice levels) provides a methodology for developing management decisions consistent with other noise measurement and mitigation projects. Comparison of models with passive observations of arctic soundscapes will be presented.

**9:35**

**2aAO5. Towards the determination of appropriate sampling schemes within tidewater glacierized fjords.** Matthew C. Zeh (Dept. of Chemistry and Phys., Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, matthew.zeh@belmont.edu), Ginny Catania (Dept. of Geo. Sci. and Inst. for Geophys., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Marcy Davis, and Dan Duncan (Inst. for Geophys., The Univ. of Texas at Austin, Austin, TX)

Using an optimal duty cycle (DC) can extend the length of recording deployments without greatly reducing the information collected and is critical for measurements in dynamic, remote environments like glacierized fjords. The DC selection is often based on the recording system (battery life/memory) rather than the information in the data. We propose an improved, data-informed method to determine the optimal DC. Fifty-two hours of ambient acoustic data were collected from two underwater vertical line arrays moored near Hubbard and Turner Glaciers in Southeast Alaska in June 2021, prior to a year-long deployment. Two-second-averaged power density spectra were calculated for each of the four channels of both moorings. Spectra were clustered for each recording channel using the k-means clustering algorithm. The process was repeated for a range of DCs from 100% to 1%. Comparisons between clustered prototypes and cluster observations were made between the full and reduced duty cycle (RDC) recordings using 2, 3, 4, 5, and 6 clusters. For DCs greater than 22% (when using four clusters), the information content within the RDC recordings was not significantly enhanced. While this result is specific to this glacierized fjord, we propose this method to optimize DC selection for other passive monitoring applications.

**9:50**

**2aAO6. Determining the speed dependent source level of a snowmobile traveling on sea-ice.** Emmanuelle Cook (Oceanogr., Dalhousie Univ., 1355 Oxford St., Halifax, Nova Scotia B3H 3Z1, Canada, emmanuellecook@dal.ca), John Winters, Katrina Anthony (Oceanogr., Dalhousie Univ., Postville, Newfoundland, Canada), David R. Barclay, and Eric Oliver (Oceanogr., Dalhousie Univ., Halifax, Nova Scotia, Canada)

As part of the Sustainable Nunatsiavut Futures project, a field experiment to determine the acoustic properties and underwater radiated sound level of a snowmobile was designed and executed. The fieldwork consists of lowering acoustic recorders under the sea ice and driving a snowmobile with a known position and velocity to evaluate its speed-dependent source level. This experiment is the first step toward collaborative Dalhousie and community research on underwater sound as it relates to the marine habitat, human use of the ocean, and sea-ice in Nunatsiavut. Due to COVID-19, the planning stages were coordinated virtually, and the fieldwork in Nunatsiavut was conducted by local Inuit Research Coordinators (IRCs), while a twin Dalhousie-led experiment was conducted in Caraquet, New Brunswick. A single hydrophone sensor was used in Nunatsiavut, and a vertical array of hydrophones was used in Caraquet to obtain underwater sound data from a moving snowmobile. Skidoo specifications for each site were recorded as well as sea-ice thickness, temperature, salinity, and sound-speed data were collected. Spectrograms of skidoos traveling at different speeds were computed. Comparisons between received levels at different velocities, sites, and ranges are shown, and the impact of sea ice and snowmobile specifications on received levels are discussed.

**10:05–10:20 Break**

**10:20**

**2aAO7. The coordinated arctic acoustic thermometry experiment—CAATEX.** Matthew A. Dzieciuch (SIO/UCSD, 2728 Arnoldson Ave., San Diego, CA 92122, mad@ucsd.edu), Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Espen Storheim (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), John A. Colosi (Naval Postgrad. School, Monterey, CA), Richard A. Krishfield (Woods Hole Oceanogr. Inst., Woods Hole, MA), Stein Sandven, and Florian Geyer (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway)

The coordinated arctic acoustic thermometry experiment (CAATEX) was a joint U.S.-Norwegian trans-Arctic acoustic propagation experiment with a design comparable to the 1994 TransArctic Propagation (TAP) experiment. The goal was to measure the changes in low-frequency sound propagation due to changes in ocean heat content and salinity, and ice conditions. Two 35 Hz acoustic transceiver moorings, one in the Nansen Basin and one in the Beaufort Sea, were deployed along with four other receiving moorings. All six moorings were equipped with vertical hydrophone arrays, recording transmissions every 36 h from fall 2019 to fall 2020. Each mooring also recorded temperature and salinity time series along the vertical extent of the hydrophone arrays, ice thickness using an upward-looking

sonar, and ocean bottom pressure. The acoustic travel-times allow comparison of present-day heat content to the 1994 measurement, but there are several other points of comparison that are sensitive to environmental changes. These include transmission loss, acoustic scattering, and the acoustic arrival structure. These measurements of ocean and ice processes, and the acoustic propagation and ambient sound will improve our ability to monitor, communicate, and navigate in the Arctic Ocean.

**10:35**

**2aAO8. Modelling of sound propagation across the Arctic Ocean using oceanographic fields and oceanographic data.** Espen Storheim (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway, espen.storheim@nersc.no), Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Matthew A. Dzieciuch (SIO/UCSD, San Diego, CA), and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

During the coordinated arctic acoustic thermometry experiment (CAATEX), two acoustic sources transmitted 35 Hz binary m-sequences in a yearlong experiment. The signals were transmitted across the Arctic Ocean and detected at distances up to 2700 km. The aim was to make measurements similar to those carried out in 1994 and 1998, to investigate the changes in sound propagation due to changes in the ocean temperature, ocean stratification, and thinning of the sea ice. Changes in propagation conditions over the last decades were investigated using time-series of data from ocean reanalysis, as well as available oceanographic data along the transect. Range-dependent fields were constructed and used as environmental input to acoustic models to produce time series of arrival times and vertical arrival structure. This approach allows for an examination of the sensitivity of low-frequency sound propagation to both vertical and large-scale horizontal changes in the ice-ocean environment. The predicted time series of acoustic travel times and arrival structure will be presented and compared with the observations.

**10:50**

**2aAO9. Acoustic networks in high Arctic Ocean observing systems.** Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway, Hanne.Sagen@nersc.no), Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Espen Storheim, Astrid Stallemo, Helene R. Langehaug (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Stein Sandven (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Richard A. Krishfield (Woods Hole Oceanogr. Inst., Woods Hole, MA), and John A. Colosi (Oceanogr., Naval Postgrad. School, Monterey, CA)

Sustained *in situ* ice-ocean observations are sorely lacking in the Arctic, limiting research on climate, weather, ice-ocean processes, and geophysical hazards. A sustained network of advanced multipurpose underwater moorings and drifting buoys in the Nansen and Amundsen Basins that included acoustic and other instrumentation would make a substantial contribution to a high Arctic Ocean observing system. Such a network would provide point measurements of ocean parameters, large-scale temperature measurements using acoustic thermometry, acoustic geo-positioning of underwater floats and gliders, and passive acoustic measurements for detection of marine mammals, geohazards, and human generated noise. Optimal design of such a network of fixed moorings and drifting platforms requires accurate knowledge of the ice-ocean environment to determine the acoustic properties. Such a sustained network would build on the successful basin-wide coordinated arctic acoustic thermometry experiment (CAATEX). In this presentation, the focus will be on the oceanographic and acoustic characteristics of the Nansen and Amundsen Basins using observations made during CAATEX. The ability of several climate models and reanalysis products to describe the oceanographic characteristics as well as their usefulness in predicting low-frequency acoustic propagation will be evaluated.

**11:05**

**2aAO10. New developments in submarine cable technology can facilitate acoustics in Polar regions and on the global scale.** Bruce M. Howe (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, bhowe@hawaii.edu) and Hanne Sagen (NERSC, Nansen Environ. and Remote Sensing Ctr., Bergen, Norway)

The subsea telecommunications cable industry is expanding their present single purpose infrastructure to include ocean observing capability. Science Monitoring And Reliable Telecommunications (SMART) Subsea Cables is working to integrate temperature, pressure, and seismic acceleration sensors into commercial cables (~70 km spacing) to support climate and ocean observation, sea level monitoring, and tsunami and earthquake early warning on the global scale. Furthermore, telecom rated branching cables with power feed units supporting multipurpose "nodes" are becoming a reality. Acoustic capability can be integral to both. Major uses of these nodes include supporting low frequency transceivers enabling basin scale tomography and geo-positioning of mobile assets and docking for autonomous undersea vehicles (AUVs). Enabled by cabled power, these would be part of the fixed/mobile acoustic tomography system measuring ocean heat content at the speed of sound and more generally for transporting energy, data, and acquiring multidisciplinary data throughout a large volume of the ocean. Both variants and hybrids between can support the necessary acoustics contribution to ocean observing. Two proposed systems can support polar applications: Far North Fiber Express connecting Norway/Finland/Ireland with Japan via the Canadian Northwest Passage, and the NSF proposed SMART cable connecting New Zealand with McMurdo Base, Antarctica.

**11:20**

**2aAO11. Simulation of acoustic reflection from Arctic ice.** Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, chotiros@utexas.edu)

The acoustic reflection from an ice sheet in the arctic, as observed by an upward looking sonar, is simulated. The reflection from a smooth ice sheet is modeled using OASES, using material properties that may be found in the published literature. Depending on the bandwidth of the signal, specular reflections from the top and bottom of the ice, as well as one or more resonances that are identified as leaky Lamb waves should be detectable. If the speed of the Lamb wave is faster than the speed of sound in water, it will leak energy back into the water. The leaky Lamb wave has a characteristic resonant frequency that is a function of the ice thickness and shear speed. The reflection and backscatter from a rough ice sheet is modeled using SPECFEM2D. Scattering from a rough interface may obscure the expected signals, depending on the severity of the roughness. [Work supported by ONR, Ocean Acoustics Program.]

**11:35**

**2aAO12. Midfrequency sound propagation and reverberation in a deep ice-covered ocean.** Anatoliy N. Ivakin (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aniv@uw.edu), Kevin L. Williams (Univ. of Washington, Seattle, WA), John E. Joseph, and D. Benjamin Reeder (Naval Postgrad. School, Monterey, CA)

Two experiments on midfrequency acoustic transmission under ice in the Beaufort Sea are discussed. The APL-ICEX14 measurements were made using 3500 Hz 5s-long CW pulses for fixed geometry of source and receiver at ~30 m depths and ~720 m range. The NPS-ICEX16 experiment was performed using CW pulses and LFM sweeps from two mobile 950–3000 Hz transmitters and five spatially separated hydrophones at different combinations of source-receiver ranges (0.3–10km) and depths (45–183m).

2a TUE. AM

For analysis of recorded timeseries, a modeling approach is suggested that considers several types of arrivals contributing to the received signal at different time intervals. The direct arrivals corresponding to nearly horizontal propagation are described using a hybrid PE-&ray-based approach, which employs a fast PE code for propagation in a stratified ocean with flat ice-free surface to account for effects of structure and dynamics of the arctic surface duct and then adds a field of an image source whose strength is defined by a complex reflection loss factor of the ice cover at small grazing angles. The direct signal is followed by reverberation coda that is modeled by bottom- and ice-bounced arrivals with corresponding reflectivity and scattering strengths at steep angles. Potential applications to remote sensing in ice-covered environments are discussed. [Work supported by ONR.]

TUESDAY MORNING, 6 DECEMBER 2022 LOOKOUT, 8:30 A.M. TO 11:10 A.M.

## Session 2aBAa

## Biomedical Acoustics, Computational Acoustics, and Signal Processing in Acoustics: Deep Learning in Ultrasound Imaging and Tissue Characterization I

Aiguo Han, Cochair
*Univ. of Illinois at Urbana-Champaign, 306 North Wright St., Urbana, IL 61801*

Xiaoming Zhang, Cochair
*Mayo Clinic, 200 1st ST SW, Rochester, MN 55905*

### *Invited Papers*

**8:30**

**2aBAa1. Narrowing the distribution of ultrasound image quality using machine learning and deep learning.** Christopher Khan (Vanderbilt Univ., Nashville, TN), Ying-Chun Pan (Biomedical Eng., Vanderbilt Univ., Nashville, TN), and Brett Byram (Vanderbilt Univ., 2301 Vanderbilt Pl, Nashville, TN 37235, brett.c.byram@vanderbilt.edu)

Ultrasound image quality varies substantially across different subjects. In some cases, this means ultrasound images are non-diagnostic. Overcoming these non-diagnostic exams is a common goal for advanced ultrasound beamforming algorithms. Recently, new beamforming approaches using machine learning and deep learning have been proposed by a number of groups to overcome ultrasound's image quality issues. Our group has proposed several methods relying on both machine learning and deep learning approaches. We will also show how physics-based machine learning methods can lead directly to deep learning methods, and we can use the development and performance of these methods to generate insight into the underlying structure of ultrasound data. We will also show that rather than leading to artificial gains, deep learning methods can be used to actually increase the available information in the form of improved dynamic range compared to delay and sum beamforming. The improvement is 15–20 dB, and we can achieve this improvement in both clean and highly cluttered data. Finally, we will show that ultrasound beamformers can be trained with unlabeled *in vivo* data in order to learn the underlying distribution of clutter in particular *in vivo* scenarios (e.g. echocardiography). This leads to improvements in imaging performance and can be used to generate insight into the interaction of different sources of image degradation *in vivo*.

**8:55**

**2aBAa2. Deep estimation of viscoelastic and backscatter quantitative ultrasound.** Ali Tehrani (Concordia Univ., Montreal, Quebec, Canada), Ivan Rosado-Mendez (Univ. of Wisconsin-Madison, Mexico City, Mexico), and Hassan Rivaz (Concordia Univ., 1455 Maisonneuve ouest, EV005.154, Montreal, Dept. of Electr. and Comput. Eng., Concordia, Montreal, Quebec H3G 1M8, Canada, hrivaz@gmail.com)

Deep learning is an ideal tool to solve inverse problems, which are often ill-posed and require incorporation of *a priori* information. We focus on solving two well-known inverse problems that entail the estimation of (1) viscoelastic (VE) and (2) backscatter quantitative ultrasound (QUS). These properties are of critical clinical value but are not currently available from B-mode images. On the first front, we propose a novel technique called PICTURE (Physically Inspired ConsTraint for Unsupervised Regularized Elastography) [Tehrani, Rivaz, MICCAI, (2022)], where we impose additional physics-based constraints on the deformation vector field within our loss function and show that it substantially improves the quality of lateral displacement estimation. We develop semi- and unsupervised methods to tackle the problem of lack of ground truth training datasets in real experiments. On the second front, we propose a novel method for segmenting regions of ultrasound images without any patching based on scatterer number densities [Tehrani *et al*. TUFFC (2022)]. Our segmentation maps can divide the image into irregular regions of fully developed speckle (FDS) or underdeveloped speckle. When moving from simulation to real datasets, we exploit domain adaptation methods using concepts similar to the popular reference phantom method in QUS.

## *Contributed Paper*

**9:20**

**2aBAa3. Endoscopic photoacoustic with deep learning for evaluating early detection of the gastrointestinal.** Ruei-Wen Ou, Wei-Cheng Lin (Dept. of Elect. Eng., National Chin-Yi Univ. of Technol., Taichung, Taiwan), Hsiao-Chuan Liu (Dept. of Ophthalmol., Univ. of Southern California, Los Angeles, CA), and JIAN-XING WU (Dept. of Elect. Eng., National Chin-Yi Univ. of Technol., No. 57, Sec. 2, Zhongshan Rd., Taiping Dist., Taichung 411030, Taiwan, jian0218@gmail.com)

Colorectal cancer is increasing rapidly every year. At present, traditional medical endoscopic probes are large in size, slow in imaging, and insufficient resolution. Here, we propose a system combining a customized probe-based photoacoustic system with deep learning to improve photoacoustic hardware to achieve a smaller size of the probe and a higher imaging resolution. On the other hand, deep learning was imbedded into the proposed photoacoustic system to achieve highly accurate results of the classifications for potentially helping physicians to identify lesions as the second opinion. In this study, a customized probe with a diameter of 9 mm was used to replace the original probe with a diameter of 1 cm in the hospital. The laser provides 15 W average power with an approximate 300 $\mu$J pulse energy. The laser beam is formed by the lens after focusing, and the fiber couple can convert the light into parallel light. After the probe rotates 360°, a slip ring will trigger the ultrasound to receive particle displacements caused by the thermal propagations. A GRIN lens with a diameter of 1 mm was used to focus the scattered light. The proposed system can generate an 800 $\mu$m resolution. The collected images are classified by deep learning algorithms, including AlexNet, GoogLeNet, and ResNet, to differentiate polyps from tumors. After comparing these four image classification methods, ResNet_18 is finally used for image classification, which helps the attending doctor reduce fatigue and quickly identify a disease.



**9:35–10:00 Break**

## *Invited Paper*

**10:00**

**2aBAa4. Ultrasound strain elastography-based characterization of *in vivo* breast lesions using deep learning.** Jingfeng Jiang (Biomedical Eng., Michigan Technolog. Univ., 1400 Townsend Drive, M&M 309, Houghton, MI 49931, jjiang1@mtu.edu) and Zhengfu Xu (Mathematical Sci., Michigan Technolog. Univ., Houghton, MI)

Ultrasound strain elastography showed great promise for breast lesion characterization. With the readily available computational power in the clinical workflow, machine learning-based is gaining acceptance in cancer imaging applications. In this study, we combine the latest developments in motion tracking that enables us to obtain high-quality axial strain and full shear strain images to test the deep-learning-based prediction of breast lesion malignancy. To our knowledge, total shear strain images have not been widely used for breast lesion differentiation. More specifically, a PDE-based regularization method [Guo *et al*., Ultrasonic Imaging (2015)] developed by our group has been improved to get high-quality two-dimensional displacement data. From those high-quality displacement data, total shear strain images can be constructed. We adopted three variants of convolution neural network (CNN) models with attention modules to predict breast lesion malignancy by combining B-mode, axial strain, and total shear strain images and their radiomic features. From our internal database, 150 cases of pathologically-confirmed breast ultrasound data [Hall *et al*., UMB (2003)] with data augmentation are used to evaluate our machine learning models. Our initial testing results are encouraging with the accuracy and area under the curve being around 0.75.

## *Contributed Papers*

**10:25**

**2aBAa5. Assessment of transfer learning ultrasound elastography: A breast cancer phantom study.** Justin An (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008, max.denis@udc.edu), Tasneem Abdus-Shakur, and Max Denis (Univ. of the District of Columbia, Washington, D.C.)

In this work, a phantom study is performed to investigate the feasibility of quantitative tissue stiffness assessment of breast cancer masses using transfer learning ultrasound elastography. A transfer learning ultrasound elastography model is developed to classify the breast masses into quantifiable Young's modulus (kilopascals, kP) values. The transfer learning model combines features of B-mode images and elastograms from Google's deep learning model AlexNet. The B-mode images and elastograms from a calibrated phantom with elastic inclusions are used to train and validate the model. Thereafter, the model is used to quantify Young's modulus of inclusions from an uncalibrated breast phantom. The accuracy of the transfer learning results with and without the inclusion of the B-mode is discussed.

**10:40**

**2aBAa6. Ultrasound elastography evaluation of age-related eye lens nucleus stiffness: A porcine eye study.** Tasneem Abdus-Shakur (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008, max.denis@gmail.com), Justin An, and Max Denis (Univ. of the District of Columbia, Washington, D.C.)

In this work, ultrasound elastography is employed to evaluate age-related changes of eye tissues. Of particular interest is the eye lens nucleus stiffness. Age-related changes in the eye lens nucleus stiffness are one of the most important causes of cataract. Ultrasound elastogram studies are performed by mechanical scanning porcine eyes using a handheld ultrasound system. Employing deep learning techniques, tissue stiffness assessments are made to differentiate porcine eyes with and without cataracts. Results and limitations of the elastogram assessment will be presented and discussed.

10:55

**2aBAa7. Particle image velocimetry for estimating shear wave elasticity imaging (SWEI) in the esophagus dysplasia.** Wei-Cheng Lin, Ruei-Wen Ou (Dept. of Electr. Eng., National Chin-Yi Univ. of Technol., Taichung, Taiwan), Hsiao-Chuan Liu (Dept. of Ophthalmol., Univ. of Southern California, Los Angeles, CA), and Jian-Xing Wu (Dept. of Electr. Eng., National Chin-Yi Univ. of Technol., No. 57, Sec. 2, Zhongshan Rd., Taiping Dist., Taichung 411030, Taiwan, jian0218@gmail.com)

In this study, a 128-channel high-frequency pulse generator/receiver system was used to emit multi-angle wave propagation, and the generator plane excitation was used to generate a focused beam, which induced the phenomenon of particle displacement in the esophagus phantom recorded. Shear wave electrography is generated by using a micro-ultrasound probe (ARFI), which excites the shear waves with the output acoustic energy. In addition, we perform the method of PIV proper orthogonal decomposition to better qualify and quantify the differences between the PIV and Doppler, especially for analysis block into PIV for particle tracking, in which the external environment, with a Doppler similarity index of 82%, and PIV mainly uses the Doppler algorithm for analysis and obtains 97 frames, and fixed 0.33 ms time difference to output pictures. Each pass interrogation area is set to 128 in sequence, the distance is defined in calibration, in which the low filter contrast of image-based validation is set to 0.004, and the bright filter objects are set to 0.5. Through the analysis results of PIV, each particle's speed and movement are critical for the hospital to think. The relative position of the lesion and the effect of ARFI on the lesion are to be understood by understanding the direction of movement of the contrast agent. Overall, the shear wave and Doppler output amplitudes provided the best image quality and PIV results for esophagus dysplasia.

TUESDAY MORNING, 6 DECEMBER 2022 — MILL YARD A, 8:30 A.M. TO 11:45 A.M.

## Session 2aBAb

## Biomedical Acoustics: Look How Big You've Gotten! A Story of Droplets and Ultrasound I

Kevin J. Haworth, Cochair
*Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3939, Cincinnati, OH 45267-0586*

Mario L. Fabiilli, Cochair
*Univ. of Michigan, 1301 Catherine St., 3226A Med Sci I, Ann Arbor, MI 48109*

### *Invited Papers*

8:30

**2aBAb1. Probing the mechanisms of acoustic droplet vaporization.** Mitra Aliabouzar (Radiology, Univ. of Michigan, 1301 Catherine St., 3226A Med. Sci. Bldg. I, Ann Arbor, MI 48109, aliabouz@umich.edu)

Phase-shift emulsions (PSEs) undergo a phase-transition to a gaseous state when subjected to sufficient pressures delivered by an ultrasound source. This phenomenon, termed acoustic droplet vaporization (ADV), has broadened the scope of ultrasound-based applications, resulting in novel diagnostic and therapeutic techniques. Since the introduction of ADV, design and fabrication methods of PSEs have progressed substantially. Development of PSEs with different thermophysical properties (e.g., perfluorocarbon core and size) has enabled modulation of ADV dynamics for specific applications. A variety of microscopy techniques have been utilized to characterize the response of different PSEs under varying acoustic conditions at relevant timescales for diagnostic and therapeutic applications. Ultra-high-speed microscopy enabled study of ADV dynamics from the inception of the vapor nucleus at nanosecond time scales to the growth of the generated bubbles seconds post ADV. The kinetics of ADV-induced drug release have been captured at multiple timescales via high-speed fluorescence microscopy. Confocal and atomic force microscopy techniques have provided insight into ADV-induced changes to the environment surrounding a PSE/bubble. Additionally, great progress has been made in the theoretical framework to predict ADV dynamics under different acoustic conditions. This talk will review physical mechanisms underpinning ADV and highlight new emerging applications.

**8:55**

**2aBAb2. Activatable perfluorocarbon nanoemulsions stabilized by oligo(ethylene glycol) bisphosphonate dendrons: Methods of preparation and characterization.** Hagop Abadian, Salima El-Yahklifi (CNRS, Univ. of Strasbourg, Strasbourg, France), Benjamin Ayela, Delphine Felder-Flesch (Superbranche, Strasbourg, France), Marc Schmutz (CNRS, Univ. of Strasbourg, Strasbourg, France), and Marie Pierre Krafft (CNRS, Univ. of Strasbourg, Inst. Charles Sadron, 23 rue du Loess, Strasbourg 67034, France, krafft@unistra.fr)

Perfluorocarbons (PFCs) combine unique capacities (biological inertness, gas solubilization, low intermolecular interactions, extreme hydrophobicity, etc.), which qualify them as critical actors in emerging fields [M. P. Krafft and J. G. Riess, Adv. Colloid Interface Sci. *294*, 102407 (2021) ]. Microbubbles (MBs) focus sustained interest for biomedical imaging and therapy. Nanometric PFC droplets (NDs) can accumulate in tumors and be vaporized on demand at the target site using ultrasound or other stimuli, generating MBs. We identified a lack of well-defined, biocompatible, and versatile amphiphiles specifically tailored for PFC-ND stabilization and acoustic droplet vaporization control. So, we developed a series of oligo(ethylene glycol) (OEG) bisphosphonate dendrons that, in combination with phospholipids, provide closer control over the size and stability of perfluorohexane-NDs [Felder-Flesch *et al.*, Eur. Pat. 22305549.2 (2022)]. These NDs were easily converted into MBs with a half-life five time longer than for reference MBs. We will discuss the methods of preparation and characterization of the new PFC nanoemulsions. Temperature-dependent cryo-TEM studies identified key intermediates in the phase-shift of the NDs into MBs. The bisphosphonate dendrons also stabilize iron oxide nanoparticles for incorporation in the NDs' phospholipid shell.



## *Contributed Papers*

**9:20**

**2aBAb3. Assessing the effect of poly (ethylene oxide-b-lactide) block copolymer on the characteristics of perfluoropentane phase-shift nanoemulsions.** Nour Al Rifai (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, Cardiovascular Ctr. 3950, Cincinnati, OH 45267-0586, alrifang@ucmail.uc.edu), Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), and Shameel Abid (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, Cincinnati, OH)

Acoustic droplet vaporization is the ultrasound-mediated phase shift of liquid perfluorocarbon emulsions into echogenic microbubbles. To reduce the emulsion's interfacial surface tension and minimize coalescence, the perfluorocarbon is coated with a surfactant. Copolymer shelled phase-shift nanoemulsions have emerged as potential diagnostic and therapeutic agents. The goal of this study is to understand how adding Poly (ethylene oxide-b-lactide) (PEO-PLLA) to a Pluronic F-68 shell formulation impacts the perfluoropentane (PFP) nanoemulsion size and polydispersity index (PDI). PFP nanoemulsions coated with Pluronic F-68 or a Pluronic F-68: PEO-PLLA (1:0.05 (v/v)) blend was prepared using high shear pressure homogenization (LV1, Microfluidics International, Corp.). Nanoemulsion size distributions and concentrations were measured (n=5) using a Beckman Coulter Multisizer 4. At 10,000 psi pressure and 1 passage through the homogenizer, the copolymer blend and Pluronic F-68 nanoemulsions had modal diameters of $0.73 \pm 0.09$ $\mu$m and $1.25 \pm 0.09$ $\mu$m, respectively. Additionally, the concentration was larger ($6.6 \pm 0.1\times10^{-2}$ vs $5.56 \pm 0.2\times10^{-2}$ ml/ml) and the PDI smaller ($0.070 \pm 0.004$ vs $0.129 \pm 0.004$) for the copolymer blend. All differences were statistically significantly different ($p < .05$). The inclusion of PEO-PLLA with Pluronic F-68 reduced the PFP nanoemulsion size and polydispersity, which may reduce experimental variability in acoustic droplet vaporization experiments.

**9:35**

**2aBAb4. Towards a phase diagram for the process of acoustic droplet vaporization.** Francois Coulouvrat (CNRS, Sorbonne Universite - 4 place Jussieu, Inst. Jean Le Rond d'Alembert, Paris 75005, France, francois.coulouvrat@upmc.fr), Thomas Lacour, and Tony Valier-Brasier (Inst. Jean Le Rond d'Alembert, Sorbonne Univ., Paris, France)

The phase-change of a liquid droplet exposed to an oscillating acoustic field is known as "acoustic droplet vaporization." It potentially represents a versatile tool for medical applications. In an attempt to understand the complex mechanisms that drive the vaporization process, a theoretical and numerical model is developed to describe the time evolution of a three-phase phase model, made of an initial (nucleus) bubble of vapor perfluorocarbon, at the center of a liquid droplet of the same perfluorocarbon immersed in water. The effect of an encapsulating layer can also be taken into account. The model is solved numerically to compute the vapor bubble and liqud droplet evolution with time. The dynamics are sorted into six different regimes depending on their characteristics and on the system ultimate fate. Those regimes can be organized within a phase diagram that synthesizes all the possible dynamics, predicting whether the complete vaporization occurs or not depending on the two control parameters, the amplitude and the frequency of the driving acoustical field. In particular, the dependance of the vaporization threshold with frequency is discussed with two different behaviors at low or high frequency.

**9:50**

**2aBAb5. The effects of varying ultrasound parameters on oxygen scavenging via acoustic droplet vaporization.** Rachel P. Benton (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3950, Cincinnati, OH 45267-0586, bentonrp@ucmail.uc.edu), Kateryna Stone (Internal Medicine/Cardiol., Univ. of Cincinnati, Cincinnati, OH), Nour Al Rifai (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Abigail R. Clark (Boonshoft School of Medicine, Wright State Univ., Cincinnati, OH), and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Acoustic droplet vaporization (ADV) is a process that phase transitions liquid droplets into gas microbubbles via ultrasound. ADV results in oxygen scavenging from the surrounding fluid into the microbubbles. The objective of this study was to determine the effect of ultrasound parameters on oxygen scavenging. A microfluidic device was used to produce perfluoropentane droplets with a modal diameter of $1.14 \pm 0.04\,\mu$m and a polydispersity index of $0.08 \pm 0.02$. Droplets were diluted to a concentration of $4.7 \times 10^{-4} \pm 0.4 \times 10^{-5}$ ml/ml in 95% oxygenated water. The oxygen partial pressure ($pO_2$) of the water was measured before and during ADV. An EkoSonic ultrasound catheter (2.35 MHz, 1.5 MPa peak negative pressure, 47 W pulse average power) nucleated ADV while either varying burst period or pulse duration (n = 5). Pre-ADV $pO_2$ was $558 \pm 5$ mm Hg for all experiments. Peri-ADV $pO_2$ dropped to $294 \pm 6$, $316 \pm 10$, and $356 \pm 12$ mmHg for a pulse duration of 17 $\mu$s and burst periods of 0.450 ms, 0.725 ms, and 1.000 ms, respectively. The peri-ADV $pO_2$ dropped to $331 \pm 14$, $342 \pm 10$, $313 \pm 10$ mmHg for a burst period of 1.000 ms and pulse durations of 17.0, 23.4, and 37.9 $\mu$s, respectively. A significant difference was seen between the amount of oxygen scavenging for the lowest and highest burst periods (p = 0.0012) and pulse durations (p = 0.027).

**10:05–10:20 Break**

## *Invited Paper*

**10:20**

**2aBAb6. Volatile nanodroplets for neurological applications.** Harriet Lea-Banks (Phys. Sci., Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, Ontario M4N 3M5, Canada, harriet.lea-banks@sri.utoronto.ca), Sheng-Kai Wu (Phys. Sci., Sunnybrook Res. Inst., Toronto, Ontario, Canada), Ying Meng (Harquail Ctr. for Neuromodulation, Sunnybrook Res. Inst., Toronto, Ontario, Canada), Flavia Venetucci Gouveia (Neurosci. and Mental Health, The Hospital for Sick Children Res. Inst., Toronto, Ontario, Canada), Clement Hamani (Harquail Ctr. for Neuromodulation, Sunnybrook Res. Inst., Toronto, Ontario, Canada), and Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Toronto, Ontario, Canada)

While current neuromodulation techniques are invasive, inaccurate, and/or unable to penetrate deeply, focused ultrasound (FUS) offers a way to study and treat the nervous system that is non-invasive, precise, and able to reach targets deep within the brain. Using ultrasound to trigger the delivery of pharmaceuticals from nanodroplets combines the spatial precision of FUS with the temporal control of psychoactive drugs. Here, I will present our work developing ultrasound-responsive nanodroplets to deliver anesthetics in the brain for neuromodulation. This work has shown how nanodroplets, loaded with pentobarbital, can be vaporized with FUS and inhibit activity in specific brain regions. This platform is now being investigated for the treatment of several neurological and psychiatric disorders, including agitation and aggression in a model of Alzheimer's disease.

## *Contributed Papers*

**10:45**

**2aBAb7. Acoustic cluster therapy (ACT®) for improved treatment of cancer and brain diseases.** Melina Mühlenpfordt (EXACT Therapeutics AS, Høgskoleringen 1, Trondheim 7034, Norway, melina.muhlenpfordt@ntnu.no), Andrew Healey, Svein Kvåle (EXACT Therapeutics AS, Oslo, Norway), Marieke Olsman, Annemieke van Wamel, Sofie Snipstad, and Catharina de Lange Davies (Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

Systemic injections of chemotherapeutics often deliver only minor fractions of the administered dose to the targeted pathology, resulting in unwanted toxicity towards healthy tissue. In brain tissue, drug delivery is impaired by the highly selective nature of the blood brain barrier impeding the influx of most substances. Acoustic Cluster Therapy (ACT®) is a platform for targeted therapeutic enhancement, facilitating increased local drug transfer. The platform is comprised of ACT® clusters, a mix of microbubbles and microdroplets and low intensity (diagnostic) ultrasound insonation. Intravenously injected ACT® clusters circulate freely through the body before activation by localised insonation at the targeted pathology. In the ultrasound field, microbubbles transfer energy to microdroplets, which undergo a liquid-to-gas phase transition, forming larger ACT® bubbles that transiently deposit in the targeted microvasculature. Further exposure to ultrasound results in controlled volume oscillation of the ACT® bubbles, inducing a range of biomechanical effects. The effects lead to enhanced extravasation across the endothelial barrier, facilitating the transport of co-administered chemotherapeutics to the targeted tissue. Proof of concept studies showed an increased therapeutic efficacy of standard of care drugs, when combined with ACT® treatment. Furthermore, ACT® treatment induced a controlled and temporal opening of the blood brain barrier observed by the extravasation of fluorescent macromolecules into brain tissue.

**11:00**

**2aBAb8. Nanodroplets for methicillin-resistant *Staphylococcus aureus* (MRSA) eradication in murine diabetic wounds.** Virginie Papadopoulou (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, 116 Manning Dr., 9004 Mary Ellen Jones Bldg., CB 7575, Chapel Hill, NC 27599-7575, papadopoulou@unc.edu), Ashelyn Sidders, Kuan-Yi Lu, Amanda Velez (Microbiology and Immunology, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Phillip Durham (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Duyen Bui (Microbiology and Immunology, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Paul A. Dayton (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Brian Conlon, and Sarah Rowe (Microbiology and Immunology, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Bacterial biofilms are the leading cause of delayed healing in chronic wounds. Aminoglycoside antibiotics, such as gentamicin (Gent), are ineffective against biofilm cells as they maintain proton motive force below the threshold for drug uptake. We employ a novel aminoglycoside adjuvant, palmitoleic acid (PA), to facilitate drug uptake. Here, we propose a dual strategy to eradicate a chronic wound infection; utilizing ultrasound-stimulated nanodroplets (US+ND) to improve the penetration of the novel drug combination Gent/PA. A chronic wound infection model was established in SKH-1 hairless mice; 6–8 week old mice were treated with streptozocin to induce diabetes. A circular wound was created on the back of the mice and infected with methicillin-resistant *Staphylococcus aureus* (MRSA). Wounds were treated topically with Gent and/or PA, twice daily for four days. One daily antibiotic treatment was combined with US+ND. On day 5, mice were euthanized and the wound area was excised and plated to enumerate bacterial survivors. Neither Gent nor Gent/PA reduced bacterial burden but both treatments were significantly improved by applying US+ND to improve

drug penetration. Importantly, bacteria were eradiated from 3 out of the 8 wounds in the Gent/PA group that were treated with US+ND. These data show that improving the penetration of a novel anti-biofilm drug combination is a viable strategy to eradicate biofilms in chronic wounds. [Some authors are inventors on patents related to this work.]

**11:15**

**2aBAb9. *In vivo* ultrasound imaging of macrophages using acoustic vaporization of internalized superheated nanodroplets.** Lalit Chudal (Radiology, UT Southwestern Medical Ctr., Dallas, TX), Caroline de Gracia Lux (Radiology, UT Southwestern Medical Ctr., 5323 Harry Hines, Dallas, TX 75390-8514, Caroline.Lux@UTSouthwestern.edu), Jacques Lux, and Robert Mattrey (Radiology, UT Southwestern Medical Ctr., Dallas, TX)

Ultrasound (US) not only detects but also interacts with its contrast agents, potentially allowing the use of perfluorocarbon-filled nanodroplets (NDs) or microbubbles (MBs) for both the detection and treatment of deep-seated tumors. Cells labeled using MBs to allow their detection *in vivo* with clinical ultrasound scanner presents several advantages over PET and MRI: Ultrasound is 2–3 orders of magnitude more sensitive and can detect a single acoustically labeled cell *in vivo*. Furthermore, with high ultrasound intensities, drugs or genes that are loaded on NDs or MBs can be delivered with spatiotemporal control. Although the acoustic labeling of stem cells for *in vivo* imaging using MBs has been reported, the ability to load and image internalized low boiling point NDs has not been explored yet. We used perfluorobutane NDs to acoustically label macrophages, which were viable following post internalization and ultrasound-mediated phase change. We documented that internalized NDs are stable for at least 8 h, which is needed for macrophages to accumulate to diseased sites. Labeled macrophage accumulated in the liver of healthy rats as stable intracellular NDs were vaporized into MBs using a clinical scanner (Siemens ACUSON Sequoia, 10L4 transducer at 4 MHz, 1.4 MI) and remained visible after vaporization.

**11:30**

**2aBAb10. Multiplex molecular ultrasound imaging with perfluorocarbon nanodroplets.** Austin Van Namen, Sidhartha Jandhyala, Catalina-Paula Spatarelu (Thayer School of Eng., Dartmouth College, Hanover, NH), and Geoffrey P. Luke (Thayer School of Eng., Dartmouth College, 15 Thayer Dr., Hanover, NH 03755, geoffrey.p.luke@dartmouth.edu)

Because of their small size and better stability, perfluorocarbon nanodroplets (PFNDs) are emerging as an appealing alternative to microbubbles as ultrasound imaging agents. We have developed a method to simultaneously distinguish between two populations of PFNDs in the same imaging volume. The method relies on synthesizing the PFNDs to contain cores with two different boiling points. One population of PFNDs contains a perfluoropentane core (boiling point = 28 °C), while the other contains a perfluorohexane core (boiling point = 56 °C). The low-boiling-point PFNDs undergo a single liquid-to-gas phase transition in response to an acoustic vaporizing trigger, while the high-boiling-point PFNDs recondense after each acoustic trigger, enabling repeated vaporization. We have optimized a custom imaging sequence including multiple acoustic triggers that can be used to accurately differentiate between the two populations of PFNDs. The estimated relative concentration between the two nanodroplets was highly linear ($R^2$ and gt; 0.99) in phantom studies. The high-boiling-point PFNDs were molecularly targeted to the epidermal growth factor receptor. Cell-culture results demonstrate that the non-targeted PFNDs can act as a delivery control, enabling more precise molecular imaging. Overall, these results show that multiplex ultrasound imaging could be an effective method for a variety of molecular imaging applications.

## Session 2aCA

# Computational Acoustics and Biomedical Acoustics: Numerical Approaches for Complex Media and Geometries I

Zhongquan Charlie Zheng, Cochair
*Mech. and Aerosp. Eng., Utah State Univ., 4130 Old Main Hill, Mech. and Aerosp. Eng., Logan, UT 84322*

D. Keith Wilson, Cochair
*Cold Regions Res. and Eng. Lab.y, U.S. Army Engineer Res. and Development Ctr., U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290*

### *Invited Papers*

**9:00**

**2aCA1. Representation of random media in acoustic scattering calculations.** D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil) and Vladimir E. Ostashev (U.S. Army Eng. Res. and Development Ctr., Hanover, NH)

Turbulence, internal waves, surface roughness, and other environmental variations in the atmosphere and ocean randomly scatter sound. Realistic representation of these variations is important for numerical wave propagation calculations. In principle, there are two primary approaches to create these representations: (1) physics-based, dynamical models for the atmosphere or ocean and (2) kinematic synthesis of random fields with prescribed statistical properties that do not necessarily capture the medium dynamics. The most common kinematic approach involves synthesizing fields from randomly phased Fourier modes. For statistically inhomogeneous media, the Fourier modes generalize to empirical orthogonal functions. An alternative kinematic approach, called *filtered Poisson processes*, distribute spatially localized functions with randomized positions and orientations. Quasi-wavelets (QWs) are a filtered Poisson process intended for turbulence and other self-similar media. While both the Fourier and QW approaches can be formulated to reproduce specified second moments of the random field, if the representations are constructed too sparsely, higher-order moments such as the kurtosis will be unrealistic. The kinematic approaches also underlie phase screen methods, which can be quite useful when the Markov approximation is valid.

**9:20**

**2aCA2. Split-step simulations of sonic boom propagation beyond the lateral cutoff in a turbulent atmosphere.** Alexander N. Carr (Aeroacoustics Branch, NASA Langley Res. Ctr., MS 461, Hampton, VA 23681, alexander.carr@nasa.gov), Joel B. Lonzaga (Structural Acoust. Branch, National Aeronautics and Space Administration, Hampton, VA), and Steven A. Miller (Mech. and Aerospace Eng., Univ. of Florida, Gainesville, FL)

Recent flight tests during the Quiet Supersonic Flights 2018 (QSF18) study reported sonic booms heard outside of the primary carpet region. In the absence of turbulence, the lateral cutoff region separates the primary sonic boom carpet from the shadow zone, where the sonic boom signal experiences significant attenuation. However, when turbulence is present in the atmospheric boundary layer (ABL), additional scattering of the sonic boom to the shadow zone region occurs. A method is presented for simulating sonic boom propagation in a turbulent atmospheric boundary layer beyond the lateral cutoff region into the shadow zone. A split-step method is used to integrate a partially one-way equation for the acoustic pressure. Inhomogeneous turbulence, representative of the ABL, is generated in the computational domain with a Fourier synthesis approach. Distributions of several loudness metrics in the shadow zone region for a sonic boom N-wave and a shaped boom are examined. Increasing both turbulence root-mean-square velocity and integral length scale are found to increase the average loudness of booms in the shadow zone. (This research is supported by the Commercial Supersonic Technology Project of the National Aeronautics and Space Administration under Grant No. 80NSSC19K1685.)

**9:40**

**2aCA3. Sound field reconstruction in urban environments: Application to enhance mapping of urban microspaces.** Max Denis (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008, max_f_denis@hotmail.com), Samba Gaye, Lirane Mandjoupa, Dorian Davis, Justin An, Wagdy Mahmoud, and lei wang (Univ. of the District of Columbia, Washington, D.C.)

In this work, reconstruction techniques for the spatial interpolation and extrapolation of sound fields in urban environments are presented. Gaussian processes are generally used for sound field reconstruction from limitedly observations of isotropic acoustic fields. However, this model is often not applicable for the anisotropic urban environments including urban street canyons and enclosed spaces, when the complexity of the sound field is high in the mid-frequency regime, unless diffusely reflecting boundaries are assumed. Two different techniques are compared for reconstructing the sound field: the least-squared method and the Kirchhoff-Helmholtz integral equation method. Of particular interest is the reconstruction of the sound field with a minimal number of irregularly and arbitrarily

distributed microphone measurements. Therefore, the techniques will not require knowledge of the microphone positions. A successive series approximation approach is presented to enhance the microscale prediction of the Kirchhoff-Helmholtz integral equation method. The sound field reconstruction results from limited urban environment observations for both methods are presented and discussed.

**10:00–10:15 Break**

## *Contributed Papers*

**10:15**

**2aCA4. A numerical and experimental study of micrometeorological effects on urban sound propagation.** Samba Gaye (Univ. of the District of Columbia, 4200 Connecticut Ave. NW, Washington, D.C. 20008, samba.gaye1@udc.edu), Lirane Mandjoupa, Dorian Davis, Justin An, Wagdy Mahmoud, lei wang, and Max Denis (Univ. of the District of Columbia, Washington, D.C.)

In this work, the influence of micrometeorological effects on sound propagation in an urban street canyon is investigated numerically and experimentally. Numerical simulations of acoustic propagation are based on sound particle propagation simulation method. Numerical data are generated for urban street canyons of various widths and height ratios. Experimental data are obtained from longitudinal measurements of urban street canyons in the United States. Temperature and wind profiles are obtained from ultrasonic anemometers and thermocouples. Measurements within and outside the street canyon are of particular interest. The experimental data are useful in integrating micrometeorological effects into the acoustic propagation model. Preliminary numerical results and measurements are presented and discussed.

**10:30**

**2aCA5. Parabolic equation with arbitrary variations in the sound speed and Mach numbers of the medium velocity.** Vladimir E. Ostashev (U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu), D. Keith Wilson (U.S. Army Eng. Res. and Development Ctr., Hanover, NH), Didier Dragna, and Jules Colas (LMFA, Ecole Centrale de Lyon, Ecully, France)

Among computational techniques in atmospheric and ocean acoustics and other fields such as seismic wave propagation, the parabolic equation (PE) approach is one of most popular now. The PE is well suited to small computers, large domains, and high frequencies. It can handle many complicated phenomena such as atmospheric and ocean stratification and refraction, scattering by turbulence, internal waves, and other inhomogeneities, ground impedance and ocean bottom interactions, and propagation over slowly varying terrain and ocean bathymetry. PEs are usually formulated for small variations in the sound speed and small Mach numbers of the medium velocity. However, these assumptions might result in significant phase errors of propagating sound waves. In this presentation, a PE is formulated for arbitrary variations in the sound speed and arbitrary Mach numbers. This new PE is surprisingly simple and can be used in various fields of acoustics and physics. It can be implemented numerically with minimal modifications to existing Crank-Nicholson PE solvers described in Section 11.2 of Ostashev and Wilson, *Acoustics in Moving Inhomogeneous Media, Second Edition* (2015).

**10:45**

**2aCA6. Application of the finite element method to atmospheric sound propagation over impedance discontinuities.** Ray Kirby (Ctr. for Audio, Acoust. and Vib., Univ. of Technol., Sydney, Univ. of Technol., Sydney, Broadway, Ultimo, New South Wales 2007, Australia, ray.kirby@uts.edu.au)

Outdoor noise propagation typically involves the propagation of sound over mixed ground conditions, including variations in the surface impedance. Diffraction occurs at the interface between different ground conditions, and this can significantly affect local sound attenuation. Popular approaches to modelling impedance discontinuities include the boundary integral method, parabolic equations, and semi-empirical methods. This paper presents an alternative approach that takes advantage of the generality of the finite element method. It was shown recently that the semi analytic finite element method can be used to solve the exact governing wave equation for atmospheric sound propagation in a complex vertically stratified medium. This allows for the inclusion of arbitrary temperature and wind profiles provided the problem is range independent. It is shown here that it is relatively straightforward to extend this approach to include multiple surface impedance discontinuities, provided each discrete surface is uniform in the axial direction. This is achieved using point collocation to enforce the axial boundary conditions over each discontinuity. Predictions are generated for single and multiple discontinuities, and comparisons are made against predictions obtained using other computational methods.

**11:00**

**2aCA7. Learning coordinate systems for fast and accurate acoustic modeling.** Aaron Charous (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, acharous@mit.edu) and Pierre F. Lermusiaux (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

Many popular numerical methods, such as finite difference and spectral methods, rely on simple domain geometry and environmental smoothness. Unfortunately, these features are rarely found in real-world simulations. We propose learning coordinate transformations with deep neural networks to facilitate acoustic modeling in complex media. Using automatic differentiation, we obtain new coordinate systems by solving various optimization problems that map the computational domain to a rectangular grid. Different choices of the objective function are utilized to attain different goals, including (i) mapping complicated boundaries such as the seafloor to straight lines and (ii) reducing the rank of two-dimensional slices of the refractive index. The first choice allows for domain decomposition to accurately model sharp discontinuities in density and sound speed. The second drastically accelerates a non-intrusive reduced-order modeling technique referred to as the dynamical low-rank approximation. Using realistic ocean test cases, we compare the performance of our learned coordinate systems with domain decomposition to the classic approach of smoothing sharp discontinuities, and we compare full-rank, low-rank, and coordinate-system-accelerated low-rank solutions of the three-dimensional parabolic wave equation.

# Session 2aMU

## Musical Acoustics: General Topics in Musical Acoustics II—Sound Production and Radiation

Andrew C. Morrison, Chair
*Natural Sci., Joliet Junior College, 1215 Houbolt Dr, Joliet, IL 60431*

### *Contributed Papers*

**9:00**

**2aMU1. Low-frequency directional characteristics of a gamelan gong.** Samuel D. Bellows (Dept. of Phys. and Astron., Brigham Young Univ., Brigham Young Univ. Dept. of Phys., Provo, UT 84602, samuel.bellows11@gmail.com), Dallin T. Harwood, Kent L. Gee, and Timothy W. Leishman (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT)

The structural modes of gamelan gongs often have clear impacts on their far-field directivity patterns with the number of directional lobes corresponding to the associated structural mode shapes. Many of the lowest modes produce dipole-like radiation with the dipole moment determined by the positions of the nodal and antinodal regions. Spherical harmonic and multipole expansions facilitate further understanding of the gongs' low-frequency directional characteristics. The expansions also yield practical simplifications to model their radiation.

**9:15**

**2aMU2. A comparative study of the directional characteristics of two gamelan gongs.** Dallin T. Harwood (Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, dth37@byu.edu), Samuel D. Bellows, Joseph E. Avila (Phys. and Astron., Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT)

While previous research on the structural modes of gamelan gongs has revealed important insight into its unique acoustics, little attention has been dedicated to study the directionality of their acoustic radiation. This study compares high-angular resolution spherical directivity measurements of two Balinese gamelan gongs of different sizes. The directional characteristics are usually closely connected with measured structural modes at both low and high frequencies. However, the directivities of the large and small gong for the same structural mode shape are not always consistent, particularly at higher frequencies. Thus, changes in modal frequencies due to the scaling of the gongs do not always indicate similar acoustical radiation patterns.

**9:30**

**2aMU3. Directivity of the muted trumpet.** Joseph E. Avila (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602, joeavila@byu.edu), Samuel D. Bellows, Timothy W. Leishman, and Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT)

The directivity function of a played musical instrument describes the angular dependence of its acoustic radiation and diffraction about the instrument, musician, and musician's chair. In this study, high angular resolution directivity data were acquired in an anechoic chamber of a muted trumpet being played by a seated musician. The chair height and horizontal displacement ensured that the geometric center of the instrument's radiating region fell at the circular center of a computer-controlled semi-circular array of 36 microphones positioned at $\Delta\theta = 5°$ polar-angle increments. Azimuthal rotations progressed in $\Delta\varphi = 5°$ increments, such that the measurements involved 2,521 unique positions over a sphere. Additional measurements at a position within the rotating reference frame facilitated post-processing. The musician played chromatic scales at each rotation position, and this process was repeated for straight, cup, and wow wow mutes in order to draw comparisons in the directivity patterns of each mute to the unmuted trumpet. Radiation behind the musician increased as a result of the mute, and mute-dependent changes to the directivity patterns primarily occurred above 1 kHz.

**9:45**

**2aMU4. Objective identification of note-to-note transitory segments in clarinet playing.** Makayle Kellison (Rollins College, Dept. of Phys., Rollins College - Box 2743, Winter Park, FL 32789, mkellison@rollins.edu), Whitney L. Coyle (Rollins College, Winter Park, FL), and Montserrat Pàmies-Vilà (Dept. of Music Acoust. - Wiener Klangstil (IWK), Univ. of Music and Performing Arts Vienna, Vienna, Österreich, Austria)

Transients are generally identified according to a user dependent set of standards—for example, one could choose a threshold that is defined between 10% and 90% of the max mouthpiece pressure or 20%–80%. However, this choice is always at the researcher's discretion. Presented here will be an objective option for identifying the transitory region between notes when studying clarinet playing tests by using the second derivative of the envelope of the mouthpiece pressure during clarinet playing tests. This work will offer the comparison of two methods of transition classification (1) ΔT, the interval between two extrema of the second derivative mentioned above, and (2) the interval of transition found with a threshold-based method, in order to then validate that the ΔT method relates to the tongue-reed-contact duration Tc (3) the interval between the maximum and minimum slope of the first derivative of a reed vibration signal surrounding a transition. We will then show that ΔT is a reliable and repeatable replacement for the arbitrary identification options that have been available thus far.

**10:00–10:15 Break**

**10:15**

**2aMU5. Using the lattice Boltzmann method to study the open-pipe end correction.** James A. Temple (Rollins College, 1000 Holt Ave., Winter Park, FL 32789, jtemple@rollins.edu), Whitney L. Coyle, and Adrien David-Sivelle (Rollins College, Winter Park, FL)

The lattice Boltzmann method (LBM) is a well-known and often used computational technique to simulate air-flow in musical instruments. Most LBM simulations in musical acoustics published in the past have used unrealistic values for air viscosity and have focused their study on other aspects of the instrument than the open-end. Due to recent experimental discoveries, it is now interesting to focus more on the behavior at the open end of the a musical instrument, such as an organ pipe, while also including realistic playing parameters and fluid characteristics. This paper will discuss the model improvements necessary to investigate the end correction of open-ended musical instruments with LBM. Comparison of results will be made with a Comsol Multiphysics model and experimental work using transmission electronic speckle pattern interferometry (TESPI).

10:30

**2aMU6. Broadening the scope of measurement and analysis of vibrations of an organ pipe employing intensity probe, simulations, and high-speed camera.** Paolo Bordoni (School Of Industrial and Information Eng., Polytechnic Univ. of Milan, Milan, Italy), Jozef Kotus, Piotr Odya (Multimedia Systems, Gdansk Univ. of Technol., Gdansk, Poland), Fabio Antonacci (School of Industrial and Information Eng., Polytechnic Univ. of Milan, Milan, Italy), and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

This paper shows an integrated approach to measure, analyze, and model phenomena occurring in an organ pipe driven by pressurized air. The aim of this paper is two-fold, i.e., to measure the pressure signal and the intensity field around the mouth by means of an intensity probe and to visualize and observe the motion of the air jet, which represents the excitation mechanism of the system. This is realized through two techniques, i.e., measurements conducted on a Bourdon organ pipe and numerical simulations of the air flow in a pipe of the same geometry. Measurements were carried out in an anechoic chamber using the Cartesian robot. Simulations were performed on a server equipped with graphical cards, and the results were visualized employing ParaVIEW software. Moreover, an analysis was conducted by observing phenomena in both audio and video signals. A high-speed camera was employed to make the jet getting out from the windway visible. This was realized using the steam that produced the reaction of dry ice and hot water. This choice has been adopted to ensure safety conditions in the measurement environment. Finally, a comparison between the sound spectrum measured and the spectrum resulted from simulations was made.

10:45

**2aMU7. Imaging standing waves in the resonator of a flue organ pipe.** Lucia Baquerizo (Dept. of Phys., Rollins College, Box 2743, Winter Park, FL 32789, lbaquerizo@rollins.edu), Quinn Fuse, Makayle Kellison, and Thomas R. Moore (Dept. of Phys., Rollins College, Winter Park, FL)

The acoustic standing wave inside a transparent flue organ pipe has been imaged using transmission electronic speckle pattern interferometry (TESPI). Optically imaging acoustic standing waves in a pipe requires sensing the pressure-induced change in index of refraction of the air, and imaging this change in index has been accomplished in the past using a variety of methods. However, the presence of the flowing air that is responsible for creating the sound of an organ pipe masks the optical effects attributable to the standing wave. We demonstrate that the optical effects of the standing wave can be separated from those of the flow by spectrally filtering high-speed TESPI images at acoustic frequencies. The resulting images show both expected and unexpected details of the standing wave inside an organ pipe. [Work supported by NSF through Grant No. PHY–2109932.]

2a TUE. AM

## Session 2aNS

## Noise and Psychological and Physiological Acoustics: Methods for Community Noise Testing and Analysis I

Alexandra Loubeau, Cochair

William Doebler, Cochair
*NASA Langley Res. Ctr., MS 463, Hampton, VA 23681*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**2aNS1. Assessing community noise annoyance: Two decades of the international technical specification ISO/TS 15666:2003.** Charlotte Clark (Population Health Res. Inst., St. George's, Univ. of London, Cranmer Terrace, London SW17 0RE, United Kingdom, chclark@sgul.ac.uk), Truls Gjestland (SINTEF Digital, Trondheim, Norway), Lisa Lavia (Noise Abatement Society, Brighton, United Kingdom), Hilary Notley (UK Dept. for the Environ., Food and Rural Affairs (Defra), London, United Kingdom), David Michaud (Health Canada, Canadian Federal Government, Ottawa, Ontario, Canada), and Makoto Morinaga (Kanagawa Univ., Kanagawa, Yokohama, Japan)

The robust assessment of noise annoyance is of key importance given that it is the most prevalent community response in populations exposed to environmental noise. In 1993, the International Commission on Biological Effects of Noise Community Response to Noise team began formalizing a standardized methodology for assessing noise annoyance, which resulted in reporting guidelines and recommendations later published as a Technical Specification (TS) by the International Standards Organization in 2003 (ISO/TS 15666: 2003). This TS, intended to inform the international community on the quantification of the exposure-response relationship between noise exposure and annoyance, has been in circulation for nearly two decades and has been updated in 2021 (ISO/TS 15666: 2021) by an international working group (ISO TC43/SC1/WG62). This paper reviews use of the 2003 TS, identifies common adaptations in use, and summarizes the revisions. Methodological issues arising from the use of the 5-point verbal and the 11-point numeric scale questions and the scoring of "highly annoyed" are discussed. The revisions are designed to encourage further standardization in noise annoyance research. This paper highlights research needs that if addressed would strengthen the methodology underlying the assessment of noise annoyance, including multidimensional assessments of annoyance.

**8:30**

**2aNS2. Preliminary analysis of urban sound data and the correlation to public health using a mobile phone application.** Kimberly A. Riegel (Phys., Farmingdale State College, 652 Timpson St., Pelham, NY 10803, kriegel@qcc.cuny.edu) and Jody Resko (Social Sci., Queensborough Community College, Bayside, NY)

Obtaining accurate community sound data that allows for the real time annoyance response can be challenging. A mobile phone application called Auditive was developed to collect health history, annoyance, and sound level data from the respondents on a community scale. A pilot study was conducted in the spring of 2022 where 60 respondents uploaded approximately 900 sound and annoyance measurements. To improve the functionality of the application, the preliminary study data were examined for trends and correlations as well as obvious technological and data issues. The type of respondent as well as the kinds of sound recorded were examined. Some basic correlations between factors will be presented. After a review of the data, some modifications of the application were made to improve the quality of the collected data. The second version of the app was launched in Fall of 2022, and data were recorded with a new set of respondents. Comparisons of the data quality and the impact of the improvements to the application were analyzed.

**8:50**

**2aNS3. Urban air mobility community noise test planning.** Stephen A. Rizzi (NASA Langley Res. Ctr., MS 461, 2 N. Dryden St., Hampton, VA 23681, s.a.rizzi@nasa.gov) and Donald S. Scata, Jr. (Office of Environ. and Energy, Federal Aviation Administration, Washington, D.C.)

The term "advanced air mobility" has been adopted by NASA to describe safe, sustainable, affordable, and accessible aviation for transformational local and intraregional missions. By this definition, advanced air mobility includes both "rural" and "urban" applications including cargo and passenger transport missions, and other aerial missions (e.g., infrastructure inspection). There will be a range of aircraft types performing such missions, including small and medium unmanned aircraft systems (UAS), electric conventional takeoff and landing (eCTOL) aircraft, and electric vertical takeoff and landing (eVTOL) aircraft. Urban air mobility (UAM) is a challenging use case for transporting cargo and passengers in an urban environment and is a new opportunity for aviation that could revolutionize the

transportation system. The National Aeronautics and Space Administration and the Noise Division of the Federal Aviation Administration Office of Environment and Energy have initiated discussions for planning UAM community noise test(s) at the end of this decade. This presentation discusses the test goals, candidate test objectives, and some of the activities needed in preparation for the test(s). It also draws distinctions between the type of study envisioned (observational versus staged) and between it and recent and planned studies on large fixed-wing transports and commercial supersonic transports.

**9:10**

**2aNS4. The effects of night-time aviation noise exposure on sleep disturbance and annoyance.** Charlotte Clark (Population Health Res. Inst., St. George's, Univ. of London, Cranmer Terrace, London SW17 0RE, United Kingdom, chclark@sgul.ac.uk), Mari Toomse-Smith (National Ctr. for Social Res., London, United Kingdom), Mathias Basner (Unit for Experimental Psychiatry, Div. of Sleep and Chronobiology, Dept. of Psychiatry, Univ. of Pennsylvania, Perelman School of Medicine, Philadelphia, PA), James Trow (Noise Consultants, Ltd., Warrington, United Kingdom), Franziska Marcheselli, Dhriti Mandalia, Rebecca Steinbach (National Ctr. for Social Res., London, United Kingdom), Joan Morris (Population Health Res. Inst., St. George's Univ. of London, London, United Kingdom), George Gibbs (Noise Consultants, Ltd., Warrington, United Kingdom), and Elena Marcus (Population Health Res. Inst., St George's Univ. of London, London, United Kingdom)

This paper will report on the methodology for a new UK study of aviation night-noise exposure on health. Funded by the United Kingdom Department of Transport (DfT), the study is examining the effects of aviation night-noise exposure for a range of night-time periods on sleep disturbance and annoyance. The study involves a cross-sectional survey of 4000 participants living near eight UK airports to assess associations of aircraft noise exposure at night and subjective sleep disturbance and annoyance, as well as an objective sleep disturbance study of 170 participants, where physiological assessments of sleep disturbance will be linked to aircraft noise exposure at the participant's home. The study will deliver exposure-response functions showing how time-averaged metrics such as $L_{Aeq,8h}$, $L_{Aeq,1h}$, N60 relate to subjective and objective sleep disturbance, and annoyance, which could be used to inform updates to the DfT's Transport Analysis Guidance (TAG) and subsequently aviation night-noise policy in the UK. It will also explore if a relationship for objective sleep disturbance can be estimated for event-related metrics such as $L_{Amax}$ and sound exposure level. Effect modification will be examined; quantifying whether some population groups may be more vulnerable to the effects of aviation night-noise on sleep disturbance and annoyance.

**9:30–9:50 Break**

**9:50**

**2aNS5. NASA quesst mission—Community response testing plans.** Peter Coen, Alexandra Loubeau, Jonathan Rathsam (NASA Langley Res. Ctr., Hampton, VA), and Gautam H. Shah (NASA Langley Res. Ctr., Mail Stop 254, 100 Nasa Rd., Hampton, VA 23681, gautam.h.shah@nasa.gov)

The National Aeronautics and Space Administration (NASA) has made a commitment to deliver data to the International Civil Aviation Organization's Committee on Aviation Environmental Protection (ICAO CAEP) defining community response to sounds from supersonic aircraft designed such that their sonic boom is replaced with a soft "thump" sound. The dataset will be a correlation of public perceptions of these sounds to the corresponding acoustic levels and will support efforts to develop international standards for permissible noise from supersonic overflight. The NASA Quesst mission is developing the X-59, a unique research aircraft capable of quiet supersonic flight, and will use the aircraft in a series of community overflight tests to measure acoustic levels of its "sonic thump" and related public response. This presentation will provide an overview of the Quesst mission, including X-59 development, acoustic validation, and community testing with a focus on the plans and technical goals for those community response tests. In addition, results of a NASA-sponsored international workshop on strategies and considerations for key aspects of the community test phase of the Quesst mission, including estimating noise exposure levels and conducting surveys to support the dataset development, will be briefed.

**10:10**

**2aNS6. NASA quesst mission—Site selection process for community testing.** Gautam H. Shah (NASA Langley Res. Ctr., Hampton, VA), Joseph J. Czech (Harris Miller Miller & Hanson, Inc., 700 District Ave., Suite 800, Burlington, MA 01803, jczech@hmmh.com), and David M. Richwine (NASA Langley Res. Ctr., Hampton, VA)

The National Aeronautics and Space Administration (NASA) has committed to deliver data to the International Civil Aviation Organization's Committee on Aviation Environmental Protection (ICAO CAEP) defining community response to sounds from supersonic aircraft designed such that their sonic boom is replaced with a soft "thump" sound. NASA is developing the X-59, a unique research aircraft capable of quiet supersonic flight, and will use the aircraft in a series of four to six community overflight tests (2024–2026) to measure acoustic levels of its "sonic thump" and related public response in support of the agency's commitment to ICAO. Unlike most NASA flight research, which is conducted primarily at the agency's facilities, the community tests will occur at various locations across the contiguous United States and involve flight over the general population in order to obtain a nationally-representative dataset. The process for identifying both the airfields and community test locations involves multiple operational and technical considerations relative to both aircraft and data collection requirements, including flight operations, geographic, and climate diversity as well as population demographics. This presentation provides an overview of the ongoing selection process being applied to address those considerations while ultimately ensuring an representative community response dataset.

**10:30**

**2aNS7. Updated noise dose range of NASA's X-59 aircraft estimated from propagation simulations.** William Doebler (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov) and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA)

A follow-on study to Doebler and Loubeau ["The noise dose range of the X-59 estimated from propagation simulations," JASA **150**(4), A208 (2021)] is described. In the previous work, propagation simulations of NASA's X-59 sonic thump were conducted using two near-field pressure solutions from NASA's Cart3D computational fluid dynamics (CFD) code. The near-fields were propagated through realistic atmospheric profiles from the Climate Forecast System Version 2 database across the USA, and perceived level statistics of the thumps were presented. The current work repeats the previous analyses, instead using near-field CFD solutions from NASA's Fully Unstructured Navier-Stokes (FUN3D) code for the same aircraft conditions (Mach 1.4 at 53 200 feet and Mach 1.3 at 43 000 feet), which achieve the minimum and maximum cruise loudness. NASA and contractors in the X-59 development and community noise testing project mutually agreed upon using FUN3D for generating near-field pressures. Sonic thump loudness statistics using the FUN3D near-fields are similar to those of Cart3D. The min loudness condition is about 1 dB (PL) greater when using the FUN3D solution, and the max loudness condition results are nearly identical. Understanding the X-59's dose range for various flight and atmospheric conditions is important for planning X-59 community noise surveys.

**10:50**

**2aNS8. Performance evaluation of a shaped sonic boom detector and classifier.** Blaine M. Harker (Blue Ridge Res. and Consulting, 29 N Market St., Suite 700, Asheville, NC 28801, blaine.harker@blueridgeresearch.com), Shane V. Lympany (Blue Ridge Res. and Consulting, Asheville, NC), and Juliet A. Page (Blue Ridge Res. and Consulting, Cambridge, MA)

NASA will soon fly the X-59 aircraft over selected communities to evaluate community responses to shaped sonic booms. Community tests will include dozens of deployed acoustic sensors capable of measuring and detecting shaped sonic boom waveforms *in situ* for rapid onboard analysis. To this end, we present a sonic boom detector and classifier. The sonic boom detector identifies a shaped sonic boom within a measured acoustical waveform by calculating the cross-correlation with a template shaped sonic boom waveform. The sonic boom classifier determines whether the identified event is indeed a shaped sonic boom based on the correlation coefficient and the calculated noise exposure level. We evaluate these algorithms using simulations of on- and off-design X-59 sonic boom waveforms injected into previously measured 30 s ambient noise recordings. Results of this case study indicate that the detector identifies a sonic boom with an accuracy of $\pm$100 ms in 99.98% of the cases. Furthermore, for a given 30 s measurement, the classifier shows true-positive rates of approximately 0.9999 when the false-positive rate is $10^{-3}$. The case study demonstrates that the recommended onboard sonic boom detector and classifier should be highly capable of identifying shaped sonic booms.

**11:10**

**2aNS9. Estimating sonic boom metrics across a community using a Kalman filter.** Shane V. Lympany (Blue Ridge Res. and Consulting, 29 N Market St., Suite 700, Asheville, NC 28801, shane.lympany@blueridgeresearch.com) and Juliet A. Page (Blue Ridge Res. and Consulting, Cambridge, MA)

As part of the Quesst mission, NASA will fly the X-59 aircraft over selected communities to evaluate community responses to low-intensity sonic booms. The purpose of these community tests is to determine the dose-response relationship between the noise exposure metrics and the community response. The independent variables for the dose-response relationship are the noise exposure metrics experienced by survey respondents within the community. Two sources of noise exposure metrics are available in each community: measurements at sparse locations throughout the community, and calculations from propagation models across the community. Both the measurements and calculations are subject to uncertainty. A Kalman filter is proposed to combine the measured and calculated noise exposure metrics to obtain the best estimate of the true noise exposure metrics across the community. The noise exposure metrics estimated by the Kalman filter have lower uncertainty than either the measured or calculated noise exposure metrics alone. Simulations demonstrate that the Kalman filter produces a more accurate estimate of the true noise exposure metrics than other noise estimation methods.

## Session 2aPAa

# Physical Acoustics, Structural Acoustics and Vibration, and Biomedical Acoustics: Effective Medium Theories in Acoustics

Michael B. Muhlestein, Chair
*Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755*



### *Contributed Papers*

**8:00**

**2aPAa1. Derivation of a second-order acoustic wave equation in an air-saturated porous medium from a volume-averaged transport model.** Gregory W. Lyons (U.S. Army Eng. Res. and Development Ctr., Information Technol. Lab., 3909 Halls Ferry Rd., Vicksburg, MS 39180, gregory.w.lyons@erdc.dren.mil) and Carl R. Hart (U.S. Army Eng. Res. and Development Ctr., Cold Regions Res. and Eng. Lab., Hanover, NH)

For propagation of high-amplitude acoustic waves in a rigid porous frame saturated by air, significant departures from a constant flow resistivity model have been observed. This departure can be modeled as a quadratic modification to the linear Darcy's drag law, known as Forchheimer's correction. To study the physical cause and relative significance of this nonlinear correction with respect to convective nonlinearity, a second-order acoustic wave equation is derived from volume-averaged equations of mass, momentum, and entropy conservation. Due to the scale separation between the wavelength and the porous structure, the particle velocity is approximated as the sum of irrotational macroscopic and incompressible microscopic fields. The porosity, tortuosity, and permeability are defined and incorporated into the macroscopic conservation equations. Porous drag terms are also derived in the macroscopic momentum equation for Darcy's law and Forchheimer's correction. Conservation equations for fluid parameter perturbations are obtained, and a second-order wave equation is derived. Dimensional analysis, in terms of the acoustic Reynolds number and Darcy number, describes the criteria for significant Forchheimer and convective nonlinearity. A case is obtained for a Westervelt-like equation for approximately progressive plane waves. These results inform modeling of nonlinearities for sound propagation in porous media under practical conditions.

**8:15**

**2aPAa2. Numerical analysis of the limits of effective medium theory for nonlinearly propagating dispersive waves.** Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil) and Kyle G. Dunn (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., Hanover, NH)

Acoustic metamterials are an increasingly popular approach to control both linear and nonlinear sound propagation. In this approach, heterogeneous structures are engineered to behave as predetermined continuous media. A key requirement for a system to be accurately approximated as a homogeneous material for wave motion is that the smallest length scale of the waveform be much larger than any length scale associated with the microscale structure of the system. However, the fact that the characteristic length scales of a finite-amplitude waveform can change due to nonlinear effects means that care must be taken when analyzing such signals in an acoustic metamaterial. This talk presents recent work toward understanding the breakdown of effective medium theory as finite-amplitude waves propagate in a heterogeneous system and weak shocks or solitons begin to form. A finite-difference time-domain code was used to model the propagation of a pulse of sound in a one-dimensional propagation domain using both a full description of the system and an effective-medium description accounting for effective mass density, compressibility, coefficient of nonlinearity, and dispersion. The difference between the solutions of the two models provides a method by which one can estimate the error associated with using effective properties.

**8:30**

**2aPAa3. Acoustic performance of a lattice of closely spaced hard inclusions in a soft medium.** Gyani Shankar Sharma (School of Mech. and Manufacturing Eng., UNSW Sydney, J17, Ainsworth Bldg., Sydney, New South Wales 2052, Australia, gyani2511@gmail.com)

Marine vessels are covered with an acoustic coating for underwater noise control. Acoustic coatings comprise a soft rubber-like medium embedded with periodic inclusions. Common inclusion types include cavities or hard metallic scatterers. A coating design comprising hard inclusions can exhibit consistent performance under hydrostatic pressure. In this work, analytical and numerical models are developed to predict the acoustic performance of a lattice of closely spaced hard inclusions in a soft medium. The analytical model is based on homogenization which allows the modelling of a complex medium as a homogenous medium with effective material and geometric properties. The effective properties account for the resonance of the inclusions and multiple scattering of waves between them. The numerical model is based on the finite element method and is developed using COMSOL Multiphysics. Results show that the dipole resonance of hard inclusions in a soft medium leads to high sound absorption. Reducing the spacing between the inclusions enhances multiple scattering of waves, resulting in higher sound absorption.

## Session 2aPAb

# Physical Acoustics and Structural Acoustics and Vibration: Frontiers of Resonant Ultrasound Spectroscopy and Its Applications I

Christopher M. Kube, Cochair
*Eng. Sci. and Mech., The Pennsylvania State Univ., 212 Earth and Engineering Sci. Bldg., Univ. Park, PA 16802*

Matthew Cherry, Cochair
*Air Force Res. Lab., 2230 10th St., Fairborn, OH 45433*

Rasheed Adebisi, Cochair
*UDRI, Univ. of Dayton, 141 Firwood Dr., Shroyer Park Ctr., Dayton, OH 45419*

**Chair's Introduction—9:15**

## *Invited Papers*

**9:20**

**2aPAb1. Resonant ultrasound spectroscopy and some extensions.** Julian D. Maynard (Phys., Penn State Univ., 104 Davey Lab, Box 231, Univ. Park, PA 16802, maynard@phys.psu.edu)

Resonant ultrasound spectroscopy (RUS) is a method in which knowledge of the shape, mass, and some natural frequencies of a sample of solid material may be used to determine the elastic properties of the material. Customarily, the natural frequencies are measured by driving the sample with a sine wave in one transducer and monitoring the sample's response with a second transducer; when the frequency of the drive is swept through a natural frequency, a tuning curve is measured, from which the natural frequency and a quality factor may be determined. Usually, a Rayleigh-Ritz method involving rectangular parallelepiped or cylindrical sample shapes and polynomial basis functions is used to analyze the data to determine the elastic properties. This talk will discuss the customary method and some extensions, including: 1) overcoming a limitation that for some problematic materials (e.g., elastomers), the method can measure only one elastic constant accurately; 2) modifying the method to include piezoelectric materials; 3) overcoming the inaccuracy of the method for samples having sharp interfaces between different materials.

**9:45**

**2aPAb2. Resonant ultrasound spectroscopy: How much information lies in higher frequencies.** Farhad Farzbod (Mech. Eng., Univ. of Mississippi, 1764 Univ. Circle, Room 203, University, MS 38677, farzbod@olemiss.edu)

Resonance ultrasound spectroscopy (RUS) is a well-established experimental technique for measuring all the elastic properties and a few anelastic properties of a material. A sample with known geometry is excited to vibrate in a wide range of frequencies, and its resonant frequencies are detected. The resonant frequencies depend on elastic constants; as such, a complete set of elastic constants can be extracted from the resonant frequency values. In this work, we investigate first the sensitivity of resonant frequencies to some of the elastic constants, and second, how much data are in the resonant frequencies as they go higher. It has been speculated that the higher modes have more info about the off-diagonal elastic constants and shear modes of vibrations. We will investigate this hypothesis and study resonant frequencies' limiting behavior.

## *Contributed Papers*

**10:10**

**2aPAb3. Generalized inverse problems in resonant ultrasound spectroscopy.** Juraj Olejnak (Inst. of Thermomechanics, Czech Acad. of Sci., Dolejškova 1402/5, Prague 18200, Czechia, olejnak@it.cas.cz), Petr Sedlak, Hanus Seiner, Kristyna Zoubkova, Pavla Stoklasova, and Tomas Grabec (Inst. of Thermomechanics, Czech Acad. of Sci., Prague, Czechia)

Determination of the elastic constants by RUS is an inverse problem because experimentally obtained resonant frequencies cannot be directly recalculated into the elastic constants. Instead, an approximate spectrum is calculated from the dimensions and crystallographic orientation of the sample, its mass, and a set of 'guessed' elastic constants, and the difference between this approximate spectrum and the experiment is iteratively minimized. RUS has been used for the determination of either the elastic constants, or crystallographic orientations of the material in the past, but the recent advancements in RUS methodology, in particular, the employment of the scanning laser vibrometry for identification of the vibrational modes, enable inverse determination of most of the input parameters simultaneously. We propose an extension of the classical RUS inversion procedure that allows us to precisely identify the crystallographic orientation and

dimensions of the sample in addition to the elastic coefficients. The proposed algorithm was applied to generally oriented iron single crystals. After the shape and orientation optimization, we achieved an unprecedented match between calculated and measured spectrum, including a very high number of utilized resonant modes ($>$300). We show that the highest modes are extremely sensitive to the crystallographic orientation.

**10:25–10:40 Break**

**10:40**

**2aPAb4. Principal component analysis for complex geometry modal identification.** Luke Beardslee (Los Alamos National Lab., Bikini Atoll Rd., D446, Los Alamos, NM 87545, lbeardslee@lanl.gov), Timothy J. Ulrich (Los Alamos National Lab., Los Alamos, NM), and Parisa Shokouhi (Eng. Sci. and Mech., The Pennsylvania State Univ., Univ. Park, PA)

Recent advances to resonant ultrasound spectroscopy (RUS) to include finite element methods (FERUS) have expanded its application to more complex geometries. When measuring a sample of complex geometry, it is often challenging to retrieve all modal information from a single point, this is due to the complexity of the mode shapes and likeliness of landing on a node location. To capture all the modal information, many measurement points are used to superimpose into a final resonant spectrum, and these measurements points are often based on intuition or experience. The work presented in this abstract proposes a simple method of finding crucial measurement locations on a complex sample using singular value decomposition (SVD) and principal component analysis (PCA). Applying SVD to a group of forward calculated eigenvectors, a set of measurement locations can then be extracted from the high amplitude locations identified from the subset of eigen images. This method of PCA for mode identification has been shown to accurately identify essential measurement locations.

**10:55**

**2aPAb5. Laser-based resonant ultrasound spectroscopy in samples with compliant boundary conditions.** Matthew Cherry (Air Force Res. Lab., Wright-Patterson Air Force Base, OH, matthew.cherry.2@us.af.mil), Rasheed Adebisi (UDRI, Univ. of Dayton, Dayton, OH), Juan Pastrana (Southwest Ohio Council for Higher Education, WPAFB, OH), Syndey Giannuzzi (Southwest Ohio Council for Higher Education, Orlando, FL), Tyler Lesthaeghe (UDRI, Univ. of Dayton, Dayton, OH), and David Torres Reyes (Air Force Res. Lab., WPAFB, OH)

Laser-based resonant ultrasound spectroscopy (LRUS) is a promising technique for measuring elastic properties in samples of exceedingly miniature sizes where traditional piezoelectric transducers present a limitation. However, the performance of an LRUS system is highly dependent on the geometry and boundary conditions of the specimen. More specifically, samples manufactured from a substrate, such as micro-pillars, have a compliant boundary at the base which needs to be accounted for. In this work, sub-millimeter, free-standing rectangular parallelepipeds were manufactured from tungsten material. These samples provided excellent measurements of resonances as well as mode shapes. A similar pillar was manufactured and left attached to the tungsten block to study the outcome due to alterations on boundary conditions. The results show a significant attenuation on all signal content in modes with higher frequency than the fundamental bending resonance. However, excellent images of mode shapes were obtained once the sample was detached from the substrate. Furthermore, silicon cantilevers were manufactured from wafers, and similar effects were observed. The theoretical justification for poor performance in higher-order modes resulted in sample redesigns that provided superior fidelity resonance measurements.

**11:10**

**2aPAb6. Probing bonded interfaces with finite-element resonant ultrasound spectroscopy.** Paul Geimer (Los Alamos National Lab., P.O. Box 1663, Mail stop D446, Los Alamos, NM 87545, pgeimer@lanl.gov), Timothy J. Ulrich, and Amber Whelsky (Los Alamos National Lab., Los Alamos, NM)

Resonant ultrasound spectroscopy (RUS) has long been applied for non-destructive quantification of elasticity, geometry, and density in material samples. Improvements in mode imaging and finite element analysis now provide validation and predictive models for a growing range of sample types. We build on recent work which applied finite-element RUS to characterize layered samples. Our study takes a closer look at the utility of RUS to evaluate the bonded interface between layers, with extension to thin films. Forward modeling of bond properties provided insights into mode-dependent frequency shifts for various types of homogeneous bond changes such as thickness and stiffness. Experimental work consisted of RUS testing on Al-Ti samples diffusion-bonded at different pressures. Despite identical base materials, samples bonded at 2 MPa and 15 MPa exhibit spectral differences not explained by geometry variance alone, indicating sensitivity to the bond interface. Though highly geometry and material dependent, the sensitivity of RUS to interface properties appears to be roughly an order of magnitude smaller than to the constitutive material properties, emphasizing careful experimentation.

**11:25**

**2aPAb7. Resonant ultrasound spectroscopy applications: Beyond canonical geometry.** Rasheed Adebisi (UDRI, Univ. of Dayton, 141 Firwood Dr., Shroyer Park Ctr., Dayton, OH 45419, rasheed.adebisi.1.ctr@us.af.mil), Tyler Lesthaeghe (UDRI, Univ. of Dayton, Dayton, OH), Matthew Cherry (Air Force Res. Lab., Wright-Patterson Air Force Base, OH), and Shamachary Sathish (UDRI, Univ. of Dayton, Dayton, OH)

Resonant ultrasound spectroscopy (RUS) is a technique that uses a combination of experimentally measured resonant frequencies and physics-based computation of these frequencies to determine the entire set of single crystal elastic constants of a material. Computation of the resonances is most often accomplished using the Rayleigh–Ritz energy minimization method but this requires knowledge of mass density and analytical representation of the shape of the material. As a result, the sample will be manufactured into a canonical geometry such as cylinder or a rectangular parallelepiped such that its displacement field can be represented by a set of basis function. In this approach, deviation from a perfect canonical geometry can have a significant impact on the estimate elastic constants during the inversion. To overcome this limitation, this project describes a finite element method combined with x-ray computed tomography to model a sample with complex shape for computation of resonance frequencies and in the inversion of the elastic constants.

**11:40**

**2aPAb8. Underwater resonant mode excitation using cavitation.** Ankush Gupta (Mech. Eng., Boston Univ., 110 Cummington Mall, Dept. of Mech. Eng., Boston Univ., Boston, MA 02215, ag1@bu.edu), Mark J. Cops (Triton Systems, Inc., Chelmsford, MA), and R. Glynn Holt (Phys. & Astron. Dept., Hampden-Sydney College, Hampden-Sydney, VA)

Characterizing underwater objects is often accomplished with optical imaging, but this task is rendered difficult or impossible if the water is turbid or the object is encrusted or buried. Often imaging is capable of identifying approximate dimensions, but lacks the ability to differentiate between naturally occurring and manmade objects. However, every resonant target possesses its own (nearly) unique ID: its vibrational modal spectrum, if it can be excited. This study reports on the use of strongly collapsing cavitation bubbles, generated via dielectric breakdown, to impulsively excite resonant modes of an underwater object. [Work funded by Department of the Navy.]

# Session 2aPP

## Psychological and Physiological Acoustics: Cochlear Implants and Computation

Terrin N. Tamati, Chair
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**Chair's Introduction—9:00**

### *Contributed Papers*

**9:05**

**2aPP1. Processing of linguistic and indexical information in adult cochlear implant users.** Terrin N. Tamati (Dept. of Otolaryngology, Ohio State Univ., 1608 Aschinger Blvd., Columbus, OH 43212, terrintamati@gmail.com) and Aaron C. Moberly (Dept. of Otolaryngology, Ohio State Univ., Columbus, OH)

Normal-hearing (NH) listeners simultaneously make use of both linguistic (speech content) and indexical (talker-specific) information in speech perception. For adult cochlear implant (CI) users, limitations in the talker-specific details conveyed by their devices may impact processing dependences between linguistic and indexical information. The current study examined the extent to which linguistic and indexical information are processed interdependently in adult CI users, using speeded phoneme and talker gender classification tasks. Higher-performing adult CI users (open-set word recognition >65% correct) reported the target vowel (phoneme classification) or target talker gender (gender classification). The non-target dimension varied by the number of talkers (phoneme classification) and the number of words (gender classification). Largely consistent with NH findings, adult CI users displayed asymmetric interference between linguistic and indexical dimensions. Phoneme classification was less accurate and slower with high talker variability. In contrast, gender classification was not affected by word variability. Processing dependencies also varied across individual CI users and related to auditory and cognitive-linguistic abilities. These preliminary findings suggest that adult CI users may make use of degraded linguistic and indexical information during speech processing. However, the extent to which they do so may depend on individual auditory and cognitive-linguistic factors.

**9:20**

**2aPP2. Music appreciation on cochlear implant simulation with contour and temporal cues.** Berliana N. Sari (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Dept. of Eng. Phys., Inst. Teknologi Sepuluh Nopember, Surabaya 60111, Indonesia, berlnindah@gmail.com) and Dhany Arifianto (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Surabaya, Indonesia)

The musical appreciation of the cochlear implant simulation user on the contour and temporal cues and the effect of the coding strategy are determined. Cochlear implant simulations were processed using continuous interleaved sampling (CIS) and spectral peak (SPEAK) coding strategies with varying numbers of channels. The experiments consist of contour identification, temporal cue discrimination, and familiar melody identification. The results show cochlear implant simulation users are poor at recognizing contour cues compared with normal hearing, but good at discriminating temporal cues from tempo and rhythm varied with contour, achieving scores close to normal hearing. A poor score was observed in familiar melody identification with pitch contour, compared to normal hearing scores, which correlated ($r = 0.99$) with the recognition of contour cues. Temporal cue information from rhythm helped cochlear implant simulation users recognize familiar melodies with an almost 40% increase in score compared to melodies without rhythm (p 0.05). Music appreciation was poor on the variation of coding strategy with the number of channels 2 to 6 CIS and good on the SPEAK coding strategy (compared with 22 channels condition). The studies emphasize the need for advances in coding strategies to deliver pitch contour cues to cochlear implant processing.

**9:35**

**2aPP3. Do cochlear implant users experience the same amplitude modulation rates in each ear?** Simin Soleimanifar (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, 901 South Sixth St., Champaign, IL 61820, simins2@illinois.edu), Prajna BK (Speech & Hearing Sci., UIUC, Champaign, IL), and Justin M. Aronoff (Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Cochlear implant (CI) users often produce different vocal fundamental frequencies (F0s) when using the left versus the right CI device. Since pitch perception and production are linked, differences in produced F0s likely indicate differences in perceived vocal pitches for the same F0 when using different ears. One possible explanation for different pitch perceptions with each ear is that, while F0s are typically encoded in the amplitude modulations (AM) of electrical stimulation, the resulting temporal pitch may differ for each ear. This study aimed to investigate if CI users perceive the same AM rates as different pitches with each ear. Five CI users participated in an AM pitch matching study using place-pitch-matched electrodes in the two ears. They were given two sounds with AM and asked to turn a dial, changing the AM rate in one ear to make the pitch of the second sound similar to the pitch of the first sound. The preliminary results showed that some participants perceived different pitches in the two ears for the same AM rate, and some did not, suggesting that the same AM-encoded F0s may be perceived as a different vocal pitch, but only for some CI users.

**9:50**

**2aPP4. Spatial cues on normal hearing and cochlear implant simulation with different coding strategies.** Ria R. Amelia (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Dept. of Eng. Phys., Inst. Teknologi Sepuluh Nopember, Surabaya 60111, Indonesia, riarizelia@gmail.com) and Dhany Arifianto (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Surabaya, Indonesia)

Cochlear implant users are known to have limited access to spatial cues. This study investigated the perception of spatial cues in normal-hearing listeners and cochlear implant simulation users. Perception of spatial cues is assessed for performance in determining the direction of the sound and understanding the speech. The results show that cochlear implant simulation users still have access to spatial cues, just like normal- hearing listeners. Normal-hearing listeners and cochlear implant simulation users can perceive spatial cues in ILD and ITD. Both can accurately identify the direction of the sound ($\text{slope} \approx 1.00$ and of $\text{set} \approx 0.00°$). Cochlear implant simulation

users can understand sentences as well as normal-hearing listeners (PCW = 113.64 rau) by using the coding strategy SPEAK in all channels or CIS with channel above 8. Perception of spatial cues in normal-hearing listeners and cochlear implant users can be improved by listening with two ears and spatially separating the target-masker position. The largest improvement in spatial cue perception was obtained from the head shadow effect (normal-hearing (NH) = 12.96, cochlear implant simulation users (CI) = 59.02), followed by binaural summation (NH = 5.72, CI = 19, 86) and binaural squelch (NH = 3.76, CI = 7.66).

**10:05–10:20 Break**

**10:20**

**2aPP5. The effects of interaural correlation and interaural place of stimulation asymmetry on binaural fusion.** Justin M. Aronoff (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, 901 South Sixth St., Champaign, IL 61820, jaronoff@illinois.edu), Simin Soleimanifar, Prajna BK, Mona Jawad (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, Champaign, IL), and Leslie R. Bernstein (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., Farmington, CT)

The combination of information at the two ears resulting in the perception of a singular auditory percept is referred to as binaural fusion. The stimulation received by bilateral cochlear implant (CI) users sometimes fails to foster binaural fusion. The goal of this study was to assess, for CI users, the extents to and manners in which interaural correlation of stimulation and the degree of interaural place of stimulation asymmetry affect binaural fusion. Bilateral CI users were presented with 1000 Hz pulse trains modulated by envelopes for which the interaural correlation was manipulated. Additionally, one electrode from one ear was paired with different electrodes from the other ear in order to manipulate interaural asymmetry. Listeners indicated the spatial diffuseness of the sound they perceived and whether they perceived a unitary auditory "image" by rotating a dial to manipulate a visual representation of their perception superimposed on a picture of a human head. Additionally, listeners could move the lateral position of the visual representations left or right, indicating perceived laterality. The results suggest that, independent of the degree of interaural asymmetry, the interaural correlation of the envelope was related directly to the degree of perceived binaural fusion.

**10:35**

**2aPP6. Spatial relationships between interaural differences in a room.** William Hartmann (Michigan State Univ., 749 Beech St., East Lansing, MI 48823, wmh@msu.edu)

The interaural ratio, as it occurs for a human head in a room, can be represented by a filter with a complex transfer function of the spatial location of the head. The log of this transfer function has real and imaginary parts, respectively, the interaural level difference (ILD) and the interaural phase difference (IPD). Because the filter is causal, the ILD and IPD spatial functions are Hilbert transforms of each other if the interaural filter is a minimum phase function of the head location. Interaural differences for a spherical "head" with tiny probe microphones, separated by a head diameter of 17.5 cm, were measured by Mr. Zane Crawford in an empty room having a volume of 63 $m^3$ and a reverberation time of 0.4 s. The head was moved in increments of 2.54 cm over a two dimensional grid 102 cm by 23 cm. Measurements at 250 Hz showed that the ILD and IPD were almost perfectly related by Hilbert transforms. Measurements at 500 and 1000 Hz showed increasing deviations. Because a function is orthogonal to its Hilbert transform, the ideal ILD is orthogonal to the IPD. Therefore, at locations where the ILD is the most stable function of space, the IPD is the most unstable and vice versa.

**10:50**

**2aPP7. Suppressing reverberation in cochlear implant stimulus patterns using time-frequency masks based on phoneme groups.** Kevin Chu, Leslie Collins (Elec. and Comput. Eng., Duke Univ., Durham, NC), and Boyla Mainsah (Elec. and Comput. Eng., Duke Univ., 101 Science Dr., Durham, NC, boyla.mainsah@duke.edu)

Cochlear implant (CI) users experience considerable difficulty in understanding speech in reverberant listening environments. This issue is commonly addressed with time-frequency masking, where a time-frequency decomposed reverberant signal is multiplied by a matrix of gain values to suppress reverberation. However, mask estimation is challenging in reverberant environments due to the large spectro-temporal variations in the speech signal. To overcome this variability, we previously developed a phoneme-based algorithm that selects a different mask estimation model based on the underlying phoneme. In the ideal case where knowledge of the phoneme was assumed, the phoneme-based approach provided larger benefits than a phoneme-independent approach when tested in normal-hearing listeners using an acoustic model of CI processing. The current work investigates the phoneme-based mask estimation algorithm in the real-time feasible case where the prediction from a phoneme classifier is used to select the phoneme-specific mask. To further ensure real-time feasibility, both the phoneme classifier and mask estimation algorithm use causal features extracted from within the CI processing framework. We conducted experiments in normal-hearing listeners using an acoustic model of CI processing, and the results showed that the phoneme-specific algorithm benefitted the majority of subjects. [Work supported by NIH through Grant No. R01DC014290-05.]

**11:05**

**2aPP8. Objective discrimination of bimodal speech using the frequency following response: A machine learning approach.** Erica Eng, Can Xu, Sarah Medina, Fan-Yin Cheng (Speech, Language, and Hearing Sci., Univ. of Texas at Austin, Austin, TX), René Gifford (Hearing and Speech Sci., Vanderbilt Univ. Med. Ctr., Nashville, TN), and Spencer Smith (Speech, Language, and Hearing Sci., Univ. of Texas at Austin, 10410 Skyflower Dr., Austin, TX 78759, spencer.smith@austin.utexas.edu)

Bimodal hearing, which combines a cochlear implant (CI) with a contralateral hearing aid, provides significant speech recognition benefits relative to a monaural CI. Factors predicting bimodal benefit remain poorly understood but may involve extracting fundamental frequency and/or formant information from the non-implanted ear. This study investigated whether neural responses (frequency following responses, FFRs) to simulated bimodal signals can be (1) accurately classified using machine learning and (2) used to predict perceptual bimodal benefit. We hypothesized that FFR classification accuracy would improve with increasing acoustic bandwidth due to greater fundamental and formant frequency access. Three vowels (/e/, /i/, and /ʊ/) with identical fundamental frequencies were manipulated to create five bimodal simulations (vocoder in right ear, lowpass filtered in left ear): Vocoder-only, Vocoder +125 Hz, Vocoder +250 Hz, Vocoder +500 Hz, and Vocoder +750 Hz. Perceptual performance on the BKB-SIN test was also measured using the same five configurations. FFR classification accuracy improved with increasing bimodal acoustic bandwidth. Furthermore, FFR bimodal benefit predicted behavioral bimodal benefit. These results indicate that the FFR may be useful in objectively verifying and tuning bimodal configurations.

# Session 2aSA

## Structural Acoustics and Vibration, Engineering Acoustics, and Musical Acoustics: Additive Manufacturing and Acoustics

Christina Naify, Cochair
*Univ. of Texas at Austin, Austin, TX 78758*

Michael R. Haberman, Cochair
*Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

John Granzow, Cochair
*Performing Arts Technol., Univ. of Michigan, 1100 Baits Dr., Ann Arbor, MI 48109*

James P. Cottingham, Cochair
*Physics, Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402*

**Chair's Introduction—8:30**

### *Invited Papers*

**8:35**

**2aSA1. Directional passive acoustic structures inspired by the ear of *Achroia grisella*.** Lara Díaz-García (Ctr. for Ultrasonic Eng., Univ. of Strathclyde, 99 George St., Glasgow G1 1RD, United Kingdom, lara.diaz-garcia@strath.ac.uk), Andrew Reid, Joseph Jackson-Camargo, and James Windmill (Ctr. for Ultrasonic Eng., Univ. of Strathclyde, Glasgow, United Kingdom)

The need for small directional microphones is patent in the current market. From smartphones to hearing aids, a small microphone capable of rejecting ambient noise is highly desirable. Most MEMS microphones are omnidirectional and have to resort to arrays to achieve directionality, effectively counteracting the reduced size that they offer in the first place. For this reason, we use bio-inspiration and turn to nature to find examples of solutions to this problem. The female specimens of the moth *Achroia grisella* are capable of monoaural directional hearing, which they use to track the males' mating calls. It is believed that they achieve directionality solely due to the morphology of their tympana. To test it, we first produce a multiphysics simulation of the structure that serves as a starting point. For experimental measurements, additive manufacturing is chosen for its ease and cost-efficiency. 3D-printed samples of the same model are examined through micro-CT scanning and then measured using laser-Doppler vibrometry to determine their frequency and directivity responses. The results of both approaches are compared, and it is found that the structure does indeed show directionality with the second eigenfrequency showing a hypercardioid-like pattern towards the front of the moth.

**8:55**

**2aSA2. Binder jet printing lead-free piezoelectric ceramics with sintering aids.** David Schipf (U.S. Naval Res. Lab., Washington, D.C.), Celeste A. Brown (U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, D.C. 20375, celeste.brown@nrl.navy.mil), Gregory Yesner, and Matthew D. Guild (U.S. Naval Res. Lab., Washington, D.C.)

In this study, we explore binder jet printing and sintering piezoelectric ceramic samples consisting of barium titanate (BaTiO3) and small amounts of sintering aids. Binder jet printing of piezoelectric ceramics shows promise for being a preferred additive manufacturing method for piezoelectric ceramics. Thus far binder jet printed piezoelectric ceramics have suffered from lower piezoelectric and dielectric properties than conventionally manufactured piezoelectric ceramics. This is mainly due to the high porosity of sintered binder jet printed samples. This investigation uses sintering aids that enable printed samples to sinter at lower temperatures and compact into higher densities than printed samples consisting of pure BaTiO3. In this presentation, we will discuss our selection of sintering aids, our investigation of sintering temperature profiles, and the properties of sintered samples. We will show a comparison of properties between our printed and sintered samples with BaTiO3 samples manufactured using uniaxial pressing. We will finish by discussing future steps in printing and sintering fully dense piezoelectric ceramics given the current state of this technology. [Work funded by the Office of Naval Research.]

9:15

**2aSA3. Ultrasonic nondestructive characterization and its role in the development and implementation of additive manufacturing processes.** Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ.,212 Earth-Engr Sci. Bldg., Univ. Park, PA 16802, aza821@psu.edu), Olivia Cook (Eng. Sci. and Mech., Penn State Univ., Univ. Park, PA), Nancy Huang (Mater. Sci. and Eng., Penn State Univ., Univ. Park, PA), Christopher M. Kube (Eng. Sci. and Mech., Penn State Univ., Univ. Park, PA), and Allison Beese (Mater. Sci. and Eng., Penn State Univ., Univ. Park, PA)

Advancements in manufacturing processes, such as metal 3D printing, are deeply reliant on our understanding of the resulting internal features and microstructures that dictate material behavior. Microstructure characterization is often relegated to techniques that require extensive sample sectioning and surface preparation, which are inherently limited to a small portion of the bulk material. In this presentation, I will show how elastic wave propagation methods (namely, ultrasonic testing) can be combined with physics-based models to extract microstructural parameters in fit-for-service parts. Example results are given for binder jet printed metals (namely, stainless steel 316 and SS316 infiltrated with bronze) where microstructure is characterized over large volumes nondestructively. These methods are correlated to both destructive metrics of microscale features and mechanical properties, which are linked to processing conditions and sample geometry. Finally, I will provide a broader outlook for the impact these techniques may have on the development and implementation of quality assurance protocols for additively manufactured parts.

9:35

**2aSA4. Old meets new: 3-D printing and the art of violin-making.** Mary-Elizabeth Brown (Montreal, Quebec H2W2H5, Canada, maryelizabethbrown@gmail.com)

Since violin-making emerged in the early 16th century, luthiers have incorporated new technology and scientific principles into their craft in an effort to create different sounds, increase resonance, amplify volume, and create more ease of playing. In late 2018, a Canadian interdisciplinary team spearheaded by the Ottawa Symphony Orchestra asked the question: What happens when you transpose those same principles to the 21st-century? In this session, participants will hear the story of the 3-D printed instruments that made international news from the perspective of the lead musician on the team, discover the manufacturing iterations and materials that led to the final versions used in the premiere of the first-ever concerto written for 3-D printed instruments and symphony orchestra, and explore the sound concepts behind these instruments as the centuries-old tradition of violinmaking meets 21st century technology. Join us as we delve into the juncture of old and new, examine the results of such interdisciplinary work and look to the future of 3-D printing in the realm of classical music and beyond.

**10:05–10:20 Break**

## *Contributed Papers*

10:20

**2aSA5. Dynamic characterization of fused deposition modeling-printed polymers with variable infills using homogenization techniques.** Christina J. Naify (Appl. Res. Labs. at UT Austin, Austin, TX 78758, christina.naify@gmail.com) and Colby W. Cushing (Appl. Res. Labs. and the Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Fused deposition modeling (FDM) printing is a common type of additive manufacturing, which builds up a 3-D volume by laying down filament layer-by-layer. To save material and reduce weight, many commercial software packages utilize infill, which is a repeating lattice pattern with a lower density than the solid material. Users can specify the density of the lattice, as well as the geometry of lattice, with as many as a dozen pre-designed options available to choose from in some commercial software packages. The lattice designs have impact on mechanical and dynamic properties, such as speed of sound, of the finished product. Despite the strong effect on final acoustic performance of infill, until now there has not been a methodical approach to predict the relationship between infill as a design parameter and dynamic properties. In this study, we use finite element analysis to implement homogenization techniques commonly used to study metamaterials in order to predict directional sound speeds for infills over a range of densities. The base material used in the study is polylactic acid (PLA). Additionally, samples with various infills are printed and sound speeds are measured. The experimental results are compared to predicted values.

10:35

**2aSA6. Modular speaker design enabled by multi-material additive manufacturing.** A. J. Lawrence (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, ajlawrence@utexas.edu), Jared Allison, and Christina J. Naify (Appl. Res. Labs., UT Austin, Austin, TX 78758)

One of the most promising features of additive manufacturing is ability to reduce part count and assembly by printing complex shapes. This feature is further highlighted by utilizing multi-material printing in which stiff and compliant materials can be deposited by the same system. Air-acoustic speakers are traditionally comprised of several assembled components with different material properties including a stiff enclosure, stiff diaphragm, and compliant suspension. In this presentation, we will describe a modular acoustic speaker design. The design consists of fully printed structural components (which are threaded together, and assembled with a voice coil and magnet). The basic design as well as printing process will be described and acoustic measurements reported. A range of printed designs, which can be interchanged among each-other will be presented, and future design ideas which take advantage of the printing process explored.

**10:50**

**2aSA7. Ultrasonic methods for the characterization of additively manufactured 316L stainless steel.** Colin L. Williams (Eng. Sci. & Mech., The Pennsylvania State Univ., 212 Earth-Eng. Sci. Bldg., Univ. Park, PA 16802, clw5756@psu.edu), Parisa Shokouhi (Eng. Sci. & Mech., The Pennsylvania State Univ., State College, PA), Matthew H. Lear (The Appl. Res. Lab., The Pennsylvania State Univ., State College, PA), Carly Donahue, and Colt J. Montgomery (Los Alamos National Lab., Los Alamos, NM)

This research utilizes linear and nonlinear ultrasonic techniques to establish a linkage between microstructure and macroscale mechanical properties of additively manufactured (AM) stainless steel 316L samples. The specimens are manufactured using two methods: laser-powder bed fusion and traditional wrought manufacturing. Using the nonlinear ultrasonic method of second harmonic generation, the acoustic nonlinearity parameter is estimated in samples with different heat treatment levels intended to alter microstructural and mechanical properties. Linear ultrasonic parameters including wave speed and resonant frequency are additionally measured. Mechanical properties are obtained through tensile testing of coupons corresponding to the test samples. Microstructural information for the samples is obtained using electron backscatter diffraction to help elucidate the relationships between microstructure, mechanical properties, and ultrasonic response. Results indicate correlations between the nonlinearity parameter and both ultimate tensile strength and yield strength, where nonlinearity generally decreases as sample strength increases, particularly in the AM samples. We hypothesize that microstructural evolution of grain characteristics across different heat treatments influences trends in measured nonlinearity, as well as substructures at smaller scales such as dislocations. These results show promising evidence for the feasibility of AM parts qualification using nondestructive nonlinear ultrasonic testing.

**11:05**

**2aSA8. In-process volumetric sensing of defects in multiple parts during powder bed fusion using acoustics.** Nathan Kizer (Eng. Sci. and Mech., The Pennsylvania State Univ., 212 Earth and Eng. Sci. Bldg., Univ. Park, PA 16802, njk19@psu.edu), Edward Reutzel, Corey Dickman (Appl. Res. Lab., The Pennsylvania State Univ., Univ. Park, PA), and Christopher M. Kube (Eng. Sci. and Mech., The Pennsylvania State Univ., Univ. Park, PA)

Additively manufactured (AM) metals have been gaining popularity due to their advantages over traditionally manufactured metals. AM processes can produce complex geometries unachievable in other methods. However, AM metals remain susceptible to traditional defects such as delaminations during manufacturing. Such defects can cause to build failure or loss of part integrity making them unsuitable for many structural applications. These defects often result from incorrect selections of process parameters for particular parts. In-situ monitoring systems using ultrasonic inspection have been proposed to sense and mitigate defects. This presentation describes experimental efforts to integrate a plurality of ultrasonic transducers into powder bed fusion AM. Nine 9Cr-1Mo stainless steel parts with identical dimensions were produced using powder bed fusion. Each part was printed with various laser power and speeds to produce a range of build qualities. Nine ultrasonic transducers were integrated into the build substrate to simultaneously monitor the nine AM parts during the build. Results of these measurements will be highlighted. This demonstration indicates the potential for using ultrasound to help determine optimal process parameters for reducing defects in AM parts. Additionally, this work supports progress toward closed-loop feedback for on-the-fly corrections.

**11:20**

**2aSA9. Integrating ultrasound and x-ray computed tomography to evaluate the microstructure of binder jet stainless steel 316L components.** Olivia Cook (Eng. Sci. and Mech., Penn State Univ., Univ. Park, PA, ojc3@psu.edu), Nancy Huang (Mater. Sci. and Eng., Penn State Univ., Univ. Park, PA), Robert L. Smithson (3M Co., St. Paul, MN), Christopher M. Kube (Eng. Sci. and Mech., Penn State Univ., Univ. Park, PA), Allison Beese (Mater. Sci. and Eng., Penn State Univ., Univ. Park, PA), and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., Univ. Park, PA)

Binder jet printing (BJP) is a promising additive manufacturing method with benefits in sustainability, material selection, and geometric design freedom. However, issues related to part quality persist, necessitating reliable inspection and characterization strategies. Traditional protocols involving sectioning and extensive sample preparation may miss crucial information about a components' microstructure due to volumetric variations. This study explores ultrasonic inspection of binder jet SS316L tensile specimens containing spatially varying grain size and porosity by measurement of longitudinal wave speed and attenuation. The ability of ultrasound to detect porosity is evaluated by cross referencing wave speed and attenuation data with porosity data gathered from x-ray computed tomography (XCT). Three-dimensional pore volumes were collapsed into two-dimensional maps such that ultrasound and XCT could be compared in a point-by-point fashion. After tensile testing, the location of failure was compared against wave speed and attenuation extremes. The results show the potential of ultrasound as well as important considerations related to the inspection of additively manufactured parts with complex microstructures.

## Session 2aSC

## Speech Communication, Architectural Acoustics, Psychological and Physiological Acoustics, and Noise: Acoustics and Communication in Healthcare Settings (Hybrid Session)

Tessa Bent, Cochair
*Speech, Language, and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408*

Erica E. Ryherd, Cochair
*Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816*

Melissa M. Baese-Berk, Cochair
*Univ. of Oregon, 1290 Univ. of Oregon, Eugene, OR 97403*



**Chair's Introduction—9:00**

### *Invited Papers*

**9:05**

**2aSC1. Hospital acoustics characterization and context.** Ilene J. Busch-Vishniac (Sonavi Labs., 1100 Wicomico Rd., Suite 600, Baltimore, MD 21230, ilene@sonavilabs.com)

In the last 20 years, attention has been paid to characterize and improve the acoustic environment of hospitals, but much remains to be done. For instance, hospital operations often prevent patients from sleeping well. When the HCAHPS survey of hospital patients included a question about noise preventing sleep, it was routinely the lowest score patients awarded to hospitals. A few preliminary studies also have shown hospitals to be poor to fair spaces for speech intelligibility. The delicate balance between maintaining privacy and establishing good communication is particularly challenging for hospital patients because they often have hearing impairments or are medicated and potentially less able to focus. Noise in hospitals has also prompted the shift to written orders for lab work and pharmaceuticals to avoid errors. A relatively new area of research interest is the stress impact of hospital soundscapes. Hospital patients are a vulnerable population, suffering from anxiety about their medical condition, and the sounds to which they are exposed (such as moans) can exacerbate the problem. Hospital staff as well show the impacts of noise-related stress. This talk summarizes the current state of hospital acoustics.

**9:30**

**2aSC2. Healthcare communication in acoustical consulting practice.** Ben Davenny (Acentech, 33 Moulton St., Cambridge, MA 02138, bdavenny@acentech.com) and Alex Odom (Acentech, Cambridge, MA)

With respect to speech communication in healthcare facilities, acoustical consultants are often concerned with overheard speech in the context of patient privacy. Low background sound is often a culprit with speech privacy problems, and examples will be given in exam rooms and common areas. Case studies of both poor speech privacy and poor speech communication will be given along with proposed solutions. Finally, popular science communications from the authors' corporate blog during the COVID pandemic on speech communication and personal protective equipment will be discussed.

**9:55**

**2aSC3. Ambient noise characteristics in long-term care facilities.** Brian J. Puckett (Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S 67th St., Architectural Eng., Omaha, NE 68182, puckett.brian@huskers.unl.edu), Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, Omaha, NE), Tessa Bent (Speech, Language, and Hearing Sci., Indiana Univ., Bloomington, IN), Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR), and Natalie Manley (Univ. of Nebraska Medical Ctr., Omaha, NE)

The built environment in senior living communities can enhance or detract from a resident's quality of life. This work focused on one specific physical characteristic of the environment that has not been well documented in recent literature: noise. Noise can negatively impact patient-caregiver speech communication—a paramount aspect of healthcare delivery. A communication breakdown can be particularly problematic among older adults, a population that can have a higher incidence of sensory impairments or cognitive decline. In this study, ambient noise data were collected at several senior living communities. Sound level meters were deployed in various locations across the facilities, including resident rooms, common lounge areas, nurse stations, therapy rooms, staff break rooms, and corridors. Noise data were collected over 72-hour periods during both weekdays and weekends to understand typical acoustical conditions of senior living communities during different levels of activity, occupancies, and times of day. The findings aid in understanding what

acoustical conditions exist in senior living communities, such as overall sound levels, fluctuations over time, and spectral content, and how these conditions can impact healthcare speech communication for aging populations.

**10:20–10:35 Break**

**10:35**

**2aSC4. Assessing speech communication in the operating room.** Navin Viswanathan (Communication Sci. & Disorders, The Pennsylvania State Univ., 1000 Sunnyside Ave., Lawrence, KS 66045, navin@ku.edu), Paul Rudy, Tara Krishnan, Margaret Brommelsiek, and Gary Sutkin (Univ. of Missouri, Kansas City, MO)

Smooth communication in the operating room (OR) is essential. However, its acoustic environment is typically composed of machine and human caused sounds such as patient temperature monitoring devices, systems, waste management systems, and irrelevant conversations that impede speech communication. These challenges are exacerbated by the need for combinations of facial protective equipment (FPPE) that both attenuate critical speech information as well as obviate relying on visual information for understanding speech. In order to faithfully capture the challenges to speech perception in this environment, we first recorded various interfering sounds based on a qualitative study of over 59 surgeries across different specializations. We analyzed their acoustic characteristics and obtained subjective measures of perceived interference from 60 surgical team members who all worked in the OR. In a follow up study, we assessed the acoustic consequences of different facial ppe combinations on produced speech. In both studies, we identified critical impediments to smooth OR communication. These include high decibel levels of interfering sounds, attention engaging irrelevant speech, and substantial attenuation of speech frequencies when using typical FPPE combinations. These studies constitute our preliminary steps to identifying and alleviating speech perception challenges in the OR.

**11:00**

**2aSC5. Vocal health for mask-wearing healthcare providers.** Victoria S. McKenna (Communication Sci. and Disorders, Univ. of Cincinnati, 3225 Eden Ave.; Health Sciences Bldg., Cincinnati, OH 45267, mckennvs@ucmail.uc.edu) and Renee L. Gustin (Otolaryngology-Head & Neck Surgery, Univ. of Cincinnati, Cincinnati, OH)

With the onset of the COVID-19 pandemic, healthcare providers are now required to wear face masks throughout the day. However, masks impact communication by creating a barrier to sound. Our previous studies show that masks attenuate mid-to-high-range frequency content and reduce vowel articulatory space. Mask-wearing healthcare workers also report significant increases in vocal effort compared to unmasked speech, as well as increased vocal effort at the end of their workday. Increased vocal effort co-occurred with a reduction in relative fundamental frequency (RFF) offset cycle 10—an acoustic indicator of vocal effort and laryngeal tension. With this in mind, we developed recommendations to help overcome some of the challenges of masked communication, with a specific focus on vocal health maintenance. A small pilot study in mask-wearing healthcare providers found that providers were able to learn and recall vocal health information with >90% accuracy over one week and implement 4 out of 6 vocal recommendations into their workday. RFF offset cycle 10 improved (Cohen's $d = 0.25$) following strategy implementation. More research in a larger group of providers is needed to understand the long-term effects of mask-wearing on vocal health, and the behavioral strategies healthcare workers can use to prevent voice problems.

**11:25**

**2aSC6. Barriers to successful communication between healthcare providers and patients.** Tessa Bent (Speech, Language, and Hearing Sci., Indiana Univ., 2631 East Discovery Parkway, Bloomington, IN 47408, tbent@iu.edu), Melissa M. Baese-Berk (Univ. of Oregon, Eugene, OR), Erica E. Ryherd (Architectural Eng., Univ. of Nebraska - Lincoln, Omaha, NE), and Natalie Manley (Div. of Geriatrics, Gerontology and Palliative Medicine, Univ. of Nebraska Med. Ctr., Omaha, NE)

Effective communication between healthcare providers and patients is crucial for optimal health outcomes yet barriers exist that may decrease successful information transfer. In this talk, we provide an overview of our research program that brings together two central challenges for provider-patient communication: the acoustic environment and the linguistic characteristics of the speech. We describe a corpus of medically-related sentences, varying in their frequency and familiarity characteristics. We evaluated the intelligibility of these sentences in quiet, speech-shaped noise, and hospital noise for both young and older adults. Results demonstrated that combining low familiarity words with any noise source causes steep word recognition declines for both listener groups, and that older, but not young, adults had more word recognition difficulties in hospital noise compared to speech-shaped noise. In the future work, we are extending these studies to investigate comprehension and recall of orally presented health information for different populations (e.g., nonnative speakers) under a range of conditions (e.g., audio-visual, speakers wearing masks). These investigations integrate research on hospital soundscapes with foundational concepts from the speech perception literature to address a crucial real-world context. [Supported by IU Institute for Advanced Study and James S. McDonnell Foundation.]

# Session 2aSP

## Signal Processing in Acoustics and Structural Acoustics and Vibration: Active Control of Sound and Vibration (Hybrid Session)

Yangfan Liu, Cochair
*Purdue Univ., Ray W. Herrick Labs., Purdue Univ., 177, South Russell St., West Lafayette, IN 47907-2099*

Scott D. Sommerfeldt, Cochair
*N249 ESC, Brigham Young Univ., Provo, UT 84602*



**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**2aSP1. Active structural acoustic control using the weighted sum of spatial gradients method.** Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, N249 ESC, Brigham Young Univ., Provo, UT 84602, scott_sommerfeldt@byu.edu), Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT), Pegah Aslani (Consumer and Clinical Radiation Protection Bureau, Health Canada, Provo, UT), and Yin Cao (Dept. of Intelligent Sci., Xi-an Jiaotong-Liverpool Univ., Provo, UT)

Active structural acoustic control focuses on implementing vibration-based control on a structure with the objective of achieving attenuation of the radiated acoustic field. It has been shown that utilizing control metrics that are integrally associated with the desired control result will generally lead to greatly improved control results. The weighted sum of spatial gradients (WSSG) method has been developed to provide global attenuation of the acoustic field radiated from a structure. It utilizes multiple spatial gradients that form the control metric, and which are correlated with various radiation mechanisms associated with the structure. The WSSG method will be briefly overviewed and the implementation of the method will be discussed. It will also be shown that the method is related to the concept of acoustic radiation modes, which have been used to describe radiation from structures. The WSSG method has been applied to several types of structures, and both computational and experimental results will be shown for flat plates and cylindrical shells to demonstrate the effectiveness of the control method.

**8:25**

**2aSP2. Manifold learning for global active noise control: Sensor interpolation.** Bo-Ru Lai (National Tsing Hua Univ., Hsinchu, Taiwan), You-Siang Chen (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Road, Hsinchu, Taiwan, Hsinchu 30013, Taiwan, s108033851@m108.nthu.edu.tw), and MingSian Bai (National Tsing Hua Univ., Hsinchu, Taiwan)

To achieve global control, unmeasured frequency responses (FRs) of a control source and preselected control points are required. In this paper, sensor interpolation (SI) approaches based on plane wave decomposition (PWD), kernel ridge regression, deep neural network (DNN) are presented. A broadened control region encompassing a large number of measured and fictitious control points can be obtained using the SI techniques. PWD is a two-stage procedure that begins with trimming down plane wave components, followed by an extraction of the associated complex amplitudes. In kernel ridge regression, a reproducing kernel in the Hilbert space (RKHS) is utilized to interpolate continuous relative transfer functions. The DNN-based SI approach is inspired by the kernel method. Convolutional neural networks and the like are exploited to interpolate the FRs at the fictitious points. Simulations are conducted for a thirty-microphone uniform linear array. From the results, an excellent fit between the FRs at the fictitious points and the ground truth generated by the image source method can be obtained below the spatial aliasing frequency. The efficacy of the interpolated FRs in relation to global control is demonstrated via an example of feedforward active noise control using a linear loudspeaker array.

## *Contributed Paper*

**8:45**

**2aSP3. A constrained adaptive active noise control filter design method via online convex optimization.** Yongjie Zhuang (Mech. Eng., Purdue Univ., Ray W. Herrick Labs., 177 S, Russell St., West Lafayette, IN 47907, zhuang32@purdue.edu) and Yangfan Liu (Mech. Eng., Purdue Univ., West Lafayette, IN)

In practical active noise control (ANC) applications, various types of constraints may need to be satisfied, e.g., robust stability, disturbance enhancement, and filter output power constraint. Some adaptive filters such as leaky LMS have been developed to apply required constraints indirectly. However, when multiple constraints are required simultaneously, satisfactory noise performance is difficult to achieve by tuning only one leaky factor. Another filter design approach that may achieve better noise control performance is to solve a constrained optimization problem. But the computational complexity of solving such a constrained optimization problem for ANC applications is usually too high even for offline design. Recently, a convex optimization reformulation is proposed which significantly reduces the required computational effort in solving constrained optimization problems for active noise control applications. In the current work, a constrained adaptive ANC filter design method is proposed. The previously proposed convex formulation is improved so that it can be implemented in real-time. The optimal filter coefficients are then redesigned continuously using online convex optimization when the environment is time-varying.

## *Invited Papers*

**9:00**

**2aSP4. An online identification method for virtual sensing in ANC system.** Shota Toyooka (Faculty of Eng. Sci., Kansai Univ., 3-3-35,Yamatecho, Suita-shi, Osaka-fu 564-8680, Japan, k927766@kansai-u.ac.jp) and Yoshinobu Kajikawa (Faculty of Eng. Sci., Kansai Univ., Suite-shi, Osaka-fu, Japan)

Active noise control (ANC) system generates anti-noise with the same amplitude and opposite phase to reduce unwanted acoustic noise based on acoustic wave superposition. In general, ANC can generate a zone of quiet (ZoQ) around the error microphone position. However, if the microphone cannot be placed at the desired position, the reduction effect may not be sufficient in the desired position. In such a case, we need to use the virtual sensing method to achieve noise reduction at the desired position without placing the physically microphone. In virtual sensing method, secondary path variation can significantly degrade performance, and secondary path modeling errors can cause the system to diverge. In this presentation, we introduce a new online identification method for auxiliary filter-based virtual sensing in ANC system. Virtual sensing with online identification requires re-estimation of not only the secondary path but also the auxiliary filters. The proposed online identification method has the capability of online identification of auxiliary filters in addition to the conventional online identification of secondary paths. We show that through computer simulations the noise reduction can be maintained by applying the proposed online identification method even when the secondary path varies.

**9:20–9:30 Break**

## *Contributed Paper*

**9:30**

**2aSP5. Automatic speech recognition for unmanned aerial vehicles.** Gamal Bohouta (Elect. & Comput. Eng. Dept., Florida Inst. of Technol.y, 220 E Univ. Blvd, apt 701, Melbourne, FL 32901, gidris2015@my.fit.edu)

Unmanned aerial vehicles (UAVs), also known as unmanned aerial systems (UASs), are quickly becoming a ubiquitous technology, poised to enter some key large-scale markets in the very near future. Fleets of such vehicles will be required in these large-scale deployments for commercial, industrial, and emergency response, along with the ability to efficiently control these fleets. Voice control and communication between human operators and these fleets will become imperative. This paper explores the framework for building an automatic speech recognition (ASR) use to the control of unmanned aerial vehicles (UAVs). The ARS system will be used by Aeronyde Corporation to fully-autonomous fleets with minimal human intervention. Aeronyde Corporation is working to shift and advance the current industry thinking of unmanned platforms from Remotely Piloted Aircraft (RPA) to fully-autonomous fleets with minimal human intervention. The Aeronyde Avionics package enables a single operator to control and monitor missions of many drones in real time anywhere in the world. The “1 Pilot – Many Drones” approach to aerial data collection is revolutionary for Big Data aggregation and analytics of the 4th Industrial Revolution. Multi-UAV autonomous aerial systems will transform data acquisition for many commercial applications, including: agriculture and forestry, railroad inspections, pipeline inspections, powerline inspections, windmill inspections, terrain mapping, search and rescue, firefighting and police work, and border patrol.

## *Invited Papers*

**9:45**

**2aSP6. A directional active noise control algorithm based on all-pass/minimum phase decomposition.** Yiming Wang (Purdue Univ., 2120 McCormick Rd., apt 711, West Lafayette, IN 47906, wymchihiro@gmail.com) and Yangfan Liu (Purdue Univ., West Lafayette, IN)

In this paper, we propose the design of a directional active noise control (ANC) system, which can suppress the noise coming from one specific direction without affecting the noise coming from other directions. The proposed directional ANC system integrates the sound extraction feature of a beamforming system with the active noise canceling feature of an ANC system. Instead of using the signals collected by the reference microphones as the reference signals directly, the directional ANC system uses the signal filtered by carefully designed minimum phase beamforming filters as the new reference signal, in which the noise signal coming from the look-direction of the beamformer is enhanced and at the same time noise signals coming from other directions are suppressed. Usually, the filtered signal of a beamforming system cannot be used as the reference signal of an ANC system because the delays introduced by the beamforming process degrade the performance of the ANC system. To resolve the problem, a spectral factorization technique is introduced and used to extract the minimum phase component of the beamforming filters which decreases the required delays of the beamforming system.



**10:05**

**2aSP7. An active noise control filter design method based on machine learning algorithms.** Dan Ding (Ancsonic Co., Ltd., Kexing Sci. Park D3-801, Shenzhen, Guangdong 518000, China, dingdan@ancsonic.com), Peisheng Wang (Beihang Univ., Shenzhen, China), and Yiming Wang (me, Purdue Univ., West Lafayette, IN)

Filter design is the essential problem in active noise control. In the case of FIR filter design, the whole problem can be formulated as a convex optimization. In practice, limited by computation capability, IIR filters are preferred but more challenging to design. On the one hand, the existence of zeros could cause instablity; on the other hand, the problem can not be formulated as a convex optimization problem. This work introduces a new IIR filter design method, which stablizes the system by modeling the ANC controller as casade parametric biquadratic IIRs and optimizes their cofficients by using machine learning algorithms. A neural network with zero input but learnable bias and network weights is used to generate the filter coefficients; and the loss function consists of a reward term for reducing noise and penalty terms for breaking the constraints. Training the neural network with gradient based machine learning algorithm directly leads to a group of coefficients that achieves good ANC performance and is subject to given constraints. It turned out that no big and deep neural network is required for this method to perform well and therefore it is computationally efficient. Experiments showed that the proposed method was able to consistently yield good ANC filters for a variety of given acoustical environments.

## *Contributed Paper*

**10:25**

**2aSP8. A multi-channel hear-through filter design approach and its applications.** Juhyung Kim (Mech. Eng., Purdue Univ., 325 S Chauncey Ave., West Lafayette, IN 47906, kim2714@purdue.edu), Yongjie Zhuang (Mech. Eng., Purdue Univ., West Lafayette, IN), and Yangfan Liu (Purdue Univ., West Lafayette, IN)

Sound characteristics change when propagating through materials and structures, such as vehicle cabins or space partitions. These structures usually block the sound propagation path and cause significant distortion to the sound heard by the other side of the structures while, in many applications, clear speech communication is needed. Especially during the Pandemic of Covid-19, it has become common to have space partitions at face-to-face service locations, for example, places providing food or banking services, for the prevention of virus spread. However, the degraded conversation sound quality caused by these partitions can negatively affects the service quality. A multi-channel active hear-through system is developed in the current work to provide a solution to provide more realistic sound environments for conversation through partitions which can preserve both the spectrum and the spatial distribution of the original sound field. This would also be useful for the improvement of augmented reality experiences and sound transmission quality in divided spaces. Compared with a direct inverse filter design, the coupling effect across channels and the superposition effect of original and reproduced sound are considered in the proposed method.

# Session 2aUW

## Underwater Acoustics and Acoustical Oceanography: Mud Acoustics I

Charles W. Holland, Cochair
*Elec. and Comput. Eng., Portland State Univ., Portland, OR 97207*

Stan Dosso, Cochair
*School of Earth and Ocean Sci., Univ. of Victoria, Victoria V8W 2Y2, Canada*

Allan D. Pierce, Cochair
*Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**2aUW1. "Mud acoustics" and the ocean acoustics program.** Kyle M. Becker (Office of Naval Res., 875 N Randolph St., Arlington, VA 22203, kyle.becker1@navy.mil), Robert H. Headrick, and Thomas C. Weber (Office of Naval Res., Arlington, VA)

Shallow water acoustics is one of the major thrusts of the Office of Naval Research Ocean Acoustics Program and has been an important area of interest to Navy programs since at least the work of Pekeris, Ewing, and Worzel in the 1940's. In the 1950's and early 1960's, there was a flurry of activity both in the US and UK focused on propagation within and reflection from the seabed and the acoustical characteristics marine sediments. This work formed the basis of a research program carried out by the applied ocean acoustics branch of the acoustics division at the Naval Research Laboratory in the 1970's. In April 1991, with the Cold War coming to an end, and a shift in interest from blue water to the littorals, ONR sponsored a workshop on shallow water acoustics at the Woods Hole Oceanographic Institution. This meeting, including underwater acousticians, geologists, geophysicists, and physical oceanographers, largely set the stage for the next 30 years of shallow water acoustics research supported by ONR. Prominent efforts include the ONR Geoclutter Program and the Shallow Water 2006 experiments, both focused on regions dominated by sandy bottoms. This talk describes more recent efforts in regions characterized by muddy bottoms.

**8:30**

**2aUW2. Overview of the seabed characterization experiment.** Preston S. Wilson (Walker Dept. Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu) and David P. Knobles (Phys., Knobles Sci. and Anal., Austin, TX)

The seabed characterization experiment (SBCEX) is a multi-disciplinary, multi-institutional international research project devoted to the following scientific goals: (1) to understand the physical mechanisms that control acoustic propagation in fine-grained sediments, (2) to quantify uncertainties in the estimation of seabed parameters, (3) to correlate the observed horizontal variations in the acoustical field with the measured horizontal variability of the seabed, and (4) to assess the performance of the resulting geoacoustic models, inversions, and statistical inference methods. To facilitate these scientific goals, research cruises were undertaken in 2015, 2016, 2017, 2021, and 2022 primarily at a site known as the New England Mud Patch. The work of the project through 2019 was documented in a special issue of the **IEEE Journal of Oceanic Engineering**, Vol. 45, No. 1 that appeared in January 2020, as well as in peer-reviewed papers in other journals. Here a brief overview of the results through the 2020 special issue will be presented, as well as an overview of the new field work competed in 2021 and 2022. [Work supported by ONR.]

**8:55**

**2aUW3. Geologic processes in marine "muddy" sediments.** Jason Chaytor (U.S. Geological Survey, Woods Hole Coastal and Marine Sci. Ctr., Woods Hole, MA, jchaytor@usgs.gov)

Marine muds, or "muddy" sediments, represent a broad range of sediments containing a mix of clay and silt size components, often with some smaller amount of sand sized and larger material. Sediments characterized by these broad definitions comprise most of the modern global seafloor, except for areas proximal to terrigenous inputs, such as continental shelves and glacial margins, which can receive and retain a higher proportion of coarser sediment. Beyond the coastal zone, the distribution of marine sediments is driven by a combination of geological, biological, and physical oceanographic processes that themselves are a response to variations in environmental conditions (e.g., global climate, nutrient availability, tectonics, terrestrial erosion). At the local scale, sediment source variation, seafloor bathymetry, primary transport and near-bed currents, resuspension and reworking, and bioturbation contribute to the short- and

long-term geologic nature of marine muds. The structure, quantitative compositional descriptions, and physical properties of marine muds from several environments will be compared and the techniques used to analyze them will be presented. New methods that may facilitate an improved understanding of the role and importance of muddy seafloor and shallow sub-seafloor sediments in influencing the acoustic environment will be highlighted.

**9:20**

**2aUW4. The role of biological processes in muddy sediments.** Kelly M. Dorgan (Dauphin Island Sea Lab., 101 Bienville Blvd., Dauphin Island, AL 36528, kdorgan@disl.org) and W. Cyrus Clemo (Dauphin Island Sea Lab., Dauphin Island, AL)

Muddy sediments are inhabited by diverse communities of animals that burrow, construct and irrigate tubes, ingest and egest sediments, and produce hard structures such as shells and exoskeletons. These infaunal animals mix sediments through bioturbation and modify the physical properties of sediments. Bulk density can be increased through compaction along burrow walls and production of compact fecal pellets or decreased by excavation of burrows and mixing of compacted sediments. Burrows and other biogenic structures can increase heterogeneity and consequently sound attenuation. Characterization of infauna using functional groups based on how their activities and body types affect their sediment environments can simplify the diversity of these animals. Some species, such as those that build tubes from shell fragments, have a disproportionately large impact on acoustics. Both infaunal and microbial communities feed on and modify the organic matter in muds, which may affect grain-grain interactions. Understanding the effects of biological processes on acoustics is important in interpreting acoustic data both in the upper layer of sediments and in deeper layers in which paleo-biological activities are preserved as well as in using acoustics as a tool for studying infaunal communities and their activities.



**9:45**

**2aUW5. Quantifying data information content to resolve seabed structure in geoacoustic inversion.** Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, British Columbia V8W 2Y2, Canada, sdosso@uvic.ca), Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR), Julien Bonnel (Woods Hole Oceanogr. Inst., Woods Hole, MA), Dag Tollefsen (FFI, Horten, Norway), Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, British Columbia, Canada), and Jan Dettmer (Dept. of Geosci., Univ. of Calgary, Calgary, Alberta, Canada)

This paper considers the importance of quantitative model selection and general parameterizations in estimating and interpreting seabed profiles in geoacoustic inversion, with application to data collected on the New England Mud Patch. In particular, the seabed structure that can be resolved depends on the information content of the acoustic data set under consideration, which varies with a number of factors, including the physics of the seabed acoustic interaction, frequency content of the data, and measurement and theory errors. Quantitative model selection applied to general parameterizations ensures the inclusion of seabed structure that is reliably sensed by the data while avoiding spurious structure. Data sets considered here include ship noise, modal dispersion, and wide-angle reflection coefficients. In each case, seabed models consistent with the data information content are estimated as part of the inversion using trans-dimensional and/or Bernstein-polynomial parameterizations. Results for all data sets indicate a low sound-speed mud layer over higher-speed sand; however, the ability to resolve structure within the mud layer, such as a transition to higher speeds near the mud base and possibly a weak positive gradient in the upper mud, depends on the information content of the various data sets. [Supported by the Office of Naval Research.]

**10:10–10:25 Break**

## *Contributed Papers*

**10:25**

**2aUW6. *In situ* acoustic coring systems for investigating the acoustic and physical properties of mud.** Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423, klee@arlut.utexas.edu), Megan Ballard, Andrew R. McNeese, Geno Gargas (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Dante D. Garcia, and Preston S. Wilson (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

*In situ* measurements of compressional and shear wave speed and attenuation provide direct characterization of marine sediment acoustic properties at near ambient conditions, as opposed to measurements that are conducted on samples removed from the seabed. Probe-based acoustic measurement techniques utilize piezoelectric sources and receivers that penetrate the seabed to obtain spatially localized depth-dependent records of compressional and shear waves in the sediment. When such measurements are coupled with conventional coring techniques, sediment samples can be collected from the propagation path between probes, allowing for comparison between the *in situ* acoustic records and quantities obtained from *ex situ* core analyses, such as porosity, grain size, or organic matter. Two acoustic coring measurement systems have been developed to investigate the acoustic, physical, and biological properties of soft sediment—the acoustic coring system, a gravity-corer-based platform capable of penetrating the seabed up to several meters, and the acoustic multi-corer, a seabed lander that samples the upper tens of centimeters of sediment near the water-seabed interface. The design and operation of these measurement platforms will be described, along with their capabilities and limitations. Some representative results from field deployments of these systems will be presented. [Work sponsored by ONR.]

**10:40**

**2aUW7. Direct measurements of acoustic and physical properties in mud and comparison to sediment acoustics models.** Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758-4423, klee@arlut.utexas.edu), Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), Kelly M. Dorgan (Dauphin Island Sea Lab., Dauphin Island, AL), and Jason Chaytor (United States Geological Survey, Woods Hole, MA)

Sediment acoustics models relate seabed physical properties to acoustic wave speed and attenuation, and direct measurements of sediment acoustic and physical properties provide a means to test such models. We examine direct measurements from two field experiment sites with muddy sediments that contain different ratios of clay, silt, sand, and organic components, comparing the acoustic data with sediment acoustic model predictions using

parameters obtained from cores as model inputs. In the New England Mud Patch, sediment sound speed profiles up to several meters below the seabed surface were measured using *in situ* acoustic probes mounted to a gravity corer. At a second site near Mobile Bay, *in situ* probe measurements of sound speed, attenuation, and shear speed were obtained in the upper 20 cm of sediment, and diver cores were collected and scanned with an acoustic core logger. Sediments from both sites were analyzed for porosity and grain size distribution. The applicability of various models to the mud found at each site will be discussed, as well as effects of infauna and organic matter, which were also measured, but are not explicitly included in the models. [Work sponsored by ONR.]

**10:55**

**2aUW8. Use of the rupture induced, underwater sound source in the New England mud patch.** Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), Preston S. Wilson, Dante D. Garcia (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Julien Bonnel (Woods Hole Oceanogr. Inst., Woods Hole, MA), James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Charles W. Holland (Electr. and Comput. Eng., Portland State Univ., Portland, OR)

The rupture induced, underwater sound source (RIUSS) is a device utilized in underwater acoustics experiments and surveys to create high amplitude, broadband acoustic pulses. The device functions by placing a rupture disk over an evacuated chamber and mechanically breaking the disk (either by striking on demand or via hydrostatic pressure) at a specified depth to initiate a volume collapse that produces an impulsive acoustic waveform. A new configuration of the device has been developed such that multiple chambers can simultaneously be deployed along with acoustic and oceanographic recorders. This device was deployed in the New England Mud Patch in support of the 2022 seabed characterization experiment along with a variety of acoustical recorders to investigate the frequency dependent, geoacoustic properties of the muddy seabed. Discussion will focus on the newly developed source and the various methods for which the source was configured and deployed in support of the experimental requirements. Results from a number of measurements will be discussed to demonstrate the ability of the source to generate long-range propagation signals, create interface waves in the seabed, and facilitate novel methods of measuring the angle of intromission within the seabed. [Work Supported by ONR.]

**11:10**

**2aUW9. Broadband core and resonance logger measurements of sound speed and attenuation in the New England mud patch.** Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, 8 College Rd., Durham, NH 03824, g.venegas@unh.edu), Kevin M. Lee, Megan Ballard, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Extensive measurements of sound speed and attenuation have been performed on sandy sediments over the last few decades, which have motivated a variety of physical sediment acoustics models. Recently, the seabed characterization experiment (SBCEX) was performed in 2015–2017, in part, to quantify the geoacoustic properties of a clayey-silt accumulation, known as the New England Mud Patch (NEMP). Due to the variability and frequency range reported by the various direct measurements and geoacoustic inversions, the sound speed measured in the NEMP lacked any noticeable frequency dependence. In 2022, an additional coring cruise was completed in the NEMP area and included shipboard core and resonance logger (CARL) measurements, expanding upper limit of the band to 1 MHz. CARL measured sound speed and attenuation as a function of depth from 100 kHz to 1 MHz using a time-of-flight technique, and sound speed along the entire core length from 12 kHz to 16 kHz using a resonance technique. Positive dispersion and depth dependance was observed in the 200 kHz to 1 MHz band. The frequency and depth dependence of the CARL measurements will be presented and compared with previously published model fits to the 2015–2017 SBCEX dataset. [Work sponsored by ONR.]

**11:25**

**2aUW10. Acoustic seabed characterization with autonomous underwater vehicles.** Ying-Tsong Lin (Woods Hole Oceanogr. Inst., 266 Woods Hole Road, Woods Hole, MA 02543, ytlin@whoi.edu), Jason Chaytor (U.S. Geological Survey, Woods Hole, MA), Gregory Packard, Tzu-Ting Chen, and Natalie Kukshtel (Woods Hole Oceanogr. Inst., Woods Hole, MA)

Autonomous underwater vehicles (AUVs) have been proven to be useful platforms to for acoustic seabed surveys with a variety of sonar systems. In this talk, results from two recent field experiments will be presented. One of the experiments was conducted in the New England shelf and upper slope areas using an AUV equipped with a towed hydrophone array and a 3.5 kHz sound source. The objective was to study the fine grained sediment (mud) layer on the shelf and the mutli-layer sub-bottom stratigraphy on the upper slope. The highlight of this first experiment is on the internal reflections from the mud layer at lower grazing angles, as well as the sub-bottom layering structure near a submarine landslide site. The other experiment took place at the Atlantis II Seamounts, and a deep water AUV was utilized with payloads including a multibeam echosounder, a sidescan sonar, a sub-bottom profiler, and a still image camera. The objective was to investigate the seafloor condition on the seamount summit plateau, of which the area is about 15 km $\times$ 5 km. Survey results showing sediment distributions and benthic marine life habitats will be presented. [Work supported by the Office of Naval Research.]

**11:40**

**2aUW11. Impacts of infauna on sediment heterogeneity and acoustic propagation.** Kelly M. Dorgan (Dauphin Island Sea Lab., 101 Bienville Blvd., Dauphin Island, AL 36528, kdorgan@disl.org), Madeline R. Frey, W. Cyrus Clemo (Dauphin Island Sea Lab., Dauphin Island, AL), Gabriel R. Venegas (Ctr. for Acoust. Res. and Education, Univ. of New Hampshire, Durham, NH), Kevin M. Lee, Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

The sea floor is inhabited by diverse communities of animals that mix sediments through bioturbation and modify the physical properties of sediments. Infaunal activities, such as burrowing, construction of burrows and tubes, and ingestion and compaction of sediments into fecal pellets, can compact and dilate sediments and change grain size distribution on small scales. These impacts increase heterogeneity of sediment structure, potentially leading to increased acoustic attenuation, especially at higher frequencies, and increased variability in sound speed. We examined the impacts of infaunal communities from Mobile Bay, AL, and the New England Mud Patch on acoustic properties of sediments through measurements on sediment cores collected at each site. We measured sound speed and attenuation at six different orientations within a core and characterized the infaunal community within the core to determine how infauna affect variability in acoustic propagation. We hypothesize that infauna increase variability both on very small scales, among orientations within a core, and on local scales, among cores collected from the same site. These data are compared with acoustic measurements over a broader frequency range to determine how infauna affect variability in acoustic propagation both over different spatial scales and across a range of frequencies.

## Session 2pAA

## Architectural Acoustics, Musical Acoustics, and Signal Processing in Acoustics: Recording Studios (Hybrid Session)

Bennett M. Brooks, Cochair
*Brooks Acoust. Corp., 49 N. Federal Highway, #121, Pompano Beach, FL 33062*

K. Anthony Hoover, Cochair
*McKay Conant Hoover, 5655 Lindero Canyon Rd., Suite 325, Westlake Village, CA 91362*

**Chair's Introduction—1:00**



### *Invited Papers*

**1:05**

**2pAA1. Evaluation of a rehearsal/recording room project for symphonic orchestra based on ISO23591.** Alexandre Maiorino (School of Music, Univ. Federal do Rio Grande do Norte, R. Cel. João Medeiros, Lagoa Nova, Natal, Rio Grande do Norte 59077-080, Brazil, alexandre.maiorino@ufrn.br)

The School of Music of the Federal University of Rio Grande do Norte is one of the fewer Federal Institutions in Brazil that is equipped with a recording studio attending a technical course in recording production. In 2021, the School of Music started a plan for its expansion with the project of a 7-storey building that will house a recording studio connected with several rehearsal rooms. One of the rooms was specifically planned to hold rehearsals and recordings of the UFRN Philharmonic Orchestra and the Big Band Jerimum Jazz. The objective of this research is to show the procedures used to evaluate, during the planning processes, the acoustics of the Philharmonic's rehearsal room using as reference the new standard ISO 23591. Results showed that it was possible to identify initial acoustical problems and indicate solutions for the best modification of the space, even in the initial stages of the project. The room, with approximately 235 $m^2$ and 6 m in height, was readjusted to a height of 9 m so that the space could comply with the recommendations of the standard. Next challenge will be to properly plan its isolation, since the new school library is above this room.

**1:25**

**2pAA2. Satisfying curriculum and rewarding curiosity—A case study in recording studios for education, experimentation, and verification.** Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Suite 3, Lowell, MA 01854, alex@fermata.biz)

While the acoustic performance goals of a recording studio in an academic setting have much in common with those of a commercial studio, the school diverges quickly from the business. The owner, users, and activities unique to the academy shift and expand priorities, motivating additionaldesign features. Students of audio learn to practice critical thinking informed by critical listening. It is not enough, however, that recording studios for education offer spaces withbest-in-class acoustics for listening and sound recording. Present and future generations of creative recordists must master problem solving that includes acoustic sleuthing, exploring a deliberately wide vocabulary of sonic spaces on campus to prepare them for success working in the wide range of architectural forms that await them in the course of future professional work. The snapshot of acoustic features presented here is the current solution to the ever-evolving curricular demands of one successful sound recording program.

**1:45**

**2pAA3. Bigger than they appear: Electronic architecture in recording studios and what that means to the musician and engineer.** Russ Berger (RBDG, 2717 North Surrey Dr., Carrollton, TX 75006, russberger2@gmail.com) and Mark Hornsby (RBDG, Plano, TX)

Technological advancements in electronic architecture (EA), and specific adaptations of that technology to the challenges of deploying it in small spaces, have made it an affordable application for recording studios. When combined with appropriate natural acoustics, this approach can achieve acoustical results not possible by treatments alone. The apparent volume, surface treatments, and resulting reverberant energy can be modified in real time beyond the constraints of the physical space, both for what the performer experiences within the space, and for what the engineer can capture in a recording. As a follow up to a paper presented in Seattle 2021, captured audio and video demonstrations from actual recording sessions of a variety of instrumentation and ensembles will illustrate EA's practical use in the studio and what it means to the musician and the engineer.

2:05

**2pAA4. Virtual acoustics in a modern recording studio.** Wieslaw Woszczyk (Music Res., McGill Univ., 527 Sherbrooke St., Room A-636, Montreal, Quebec H3K 3G9, Canada, Wieslaw.Woszczyk@mcgill.ca), Richard King, Ying-Ying Zhang, and Aybar Aydin (Music Res., McGill Univ., Montreal, Quebec, Canada)

Acoustical designs of recording studios have evolved over several decades to accommodate changes in the recording process/practice and in the available footprint of real estate. In many cases, special rooms were added to optimize acoustics for specific purpose: drum rooms, piano rooms, vocal rooms, reverb chambers, serving to isolate sounds and thus achieve better control of balance, auditory perspective, and sound color in recording. Today, workstations, plugins, and virtual acoustics setup allow for a replacement of the traditional design of a recording studio with one incorporating virtual acoustics that puts lesser demands on real estate. The talk will present a modern approach to 3D recording based on virtual assets that permits both simultaneous and sequential processes in recording of music while maximizing creative flexibility and acoustic experiences of artists.

2:25

**2pAA5. A modern metric for acoustic clarity in critical listening environments.** Andy Swerdlow (Criterion Acoust., 705 Central Ave., Unit #4, New Providence, NJ 07974, aswerdlow@criterionacoustics.com)

Characteristics of acoustics in small rooms have become less mysterious, more predictable and measurements more common than in past decades. Modern metrics for characterizing the acoustical performance of critical listening environments have not followed the recent predictive improvements. An Acoustic Clarity metric with a variable time window (Rear Reflection Clarity, $C_{RR}$) is proposed for small (under approximately 200m$^3$) critical listening environments such as recording studio control, mastering, film post, screening, audiophile listening, and similar rooms. The variable time window is defined by the arrival time difference between direct sound and the reflection from the rear wall, which is typically the strongest reflection and is opposite the on-axis response of the loudspeakers. Measured impulse response data collected *in situ* is presented and analyzed as case studies to characterize $C_{RR}$. For simplicity, this presentation will focus on only the stereo components of surround-sound environments.

**2:45–3:00 Break**

3:00

**2pAA6. Comparison of recording studio control room operational response measurements for single, stereo, and immersive audio monitor configurations.** Tiago F. Mattos (Brooks Acoust. Corp., 49 N. Federal Highway, # 121, Pompano Beach, FL 33062, tiagomattos@brooksacoustics.com) and Bennett M. Brooks (Brooks Acoust. Corp., Pompano Beach, FL)

Audio monitoring in recording studio control rooms has evolved continuously since the beginning of multitrack systems. In recent years, control rooms have adapted to using multiple audio monitors (loudspeakers) needed for the immersive audio experience. The primary technical recommendations for determining the acoustical quality of a control room are given in EBU-TECH-3276 and ITU BS.1116. The results of measuring the Operational Room Response Curve (ORRC) can differ significantly for only one audio monitor operating compared to the two monitors required for stereo. For multichannel immersive audio configurations, there can also be significant differences in the measured ORRC. A recent immersive system, known as Dolby Atmos, follows the Dolby Laboratories recommendations which generally comply with the EBU/ITU specifications. The goal of this research is to analyze and compare the measurements of the ORRC per the EBU/ITU/Dolby standards for various audio monitor configurations in a control room. These include individual monitors operating alone, two monitors in a stereo configuration operating at the same time, and combinations of monitors in the multichannel immersive audio system. The impact of the coupled system of room acoustics and multiple loudspeakers on the decision quality for the studio user will be addressed.

3:20

**2pAA7. Measurement of recording studio control room operational response for various audio monitor configurations using a binaural artificial head.** Tiago F. Mattos (Brooks Acoust. Corp., 49 N. Federal Highway, # 121, Pompano Beach, FL 33062, tiagomattos@brooksacoustics.com) and Bennett M. Brooks (Brooks Acoust. Corp., Pompano Beach, FL)

The primary technical recommendations for measuring the Operational Room Response Curve (ORRC) in a recording studio control room, given in EBU-TECH-3276 and ITU BS.1116, imply the use of a single microphone located at the "reference listener position" as well as other "worst case" locations. It is known that the results of measuring the ORRC in this way can differ significantly for only one audio monitor (loudspeaker) operating compared to the two monitors required for stereo, and for multichannel immersive audio configurations, such as Dolby Atmos. These differences can alter the way that program material will be perceived by the studio users (mix and master engineers) who employ two ears to make production decisions, not one! It is proposed to use a binaural artificial head system for ORRC measurements, to better simulate real-world listening conditions. This research analyzes and compares the measurements of the ORRC made using the standard single microphone locations to measurements made using a binaural artificial head, for various monitor configurations. The impact of these findings on the decision quality for the studio user will be addressed.

3:40

**2pAA8. Sound isolation of recording studios.** K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Suite 325, Westlake Village, CA 91362, thoover@mchinc.com)

Sound isolation to and from quality recording studios is critical to their success. Proper design based on informed understanding of both airborne and structureborne transmission is essential, because retrofitting isolation can be challenging or even prohibitive, and because one-size-fits-all recommendations may be unsatisfactory. This presentation will review some issues and evaluation methods that help to guide successful, cost-effective designs. Also discussed will be examples including a world-class studio with remarkable history that was encroached by an expanded loud facility, floated constructions that were not actually floated, and an approach that has helped to convince clients of the level of required isolation.

**4:00**

**2pAA9. Non-cuboid iterative room optimizer.** Peter D'Antonio (Res., Redi Acoust., LLC, 262 Martin Ave., Highland, NY 12528, peter.dantonio@rediacoustics.com) and Rinaldi P. Petrolli (Res., Redi Acoust., LLC, Florianópolis, Santa Catarina, Brazil)

The growing processing power of desktop and distributed cloud cluster computing is playing a larger role providing computational architectural acoustic solutions. This presentation will provide a status update on the Non-cuboid Iterative Room Optimization software called NIRO. It offers an iterative approach to full bandwidth optimization of critical listening rooms. The program uses a finite element method (FEM), image source model (ISM), and non-dominating sorting genetic algorithm (NSGA) to simultaneously optimize the room geometry of any shaped room, including boundary admittances, for any number of listeners and loudspeakers. It minimizes the weighted sum of the modal response and speaker boundary interference response, the spatial uniformity around the mix position and reflections in the reflection free zone (RFZ). With an optimal room setup, acoustical diffusors and absorbers with a specified resonant frequency, bandwidth, and efficiency are added to the model to optimize the acoustic properties. A combination of FEM, ISM, and an additional geometrical acoustics model is used to generate full bandwidth impulse responses. The RFZ, envelopment, reverberation time, low-frequency response, and temporal decay are optimized and evaluated. Following the addition of HRTFs, stereo auralization can be generated. Proof of performance and application examples will be presented.

**4:20**

**2pAA10. The evolution of Blackbird Studio C.** Peter D'Antonio (RPG Acoust. Syst. LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com) and George Maassenburg (Schulich School of Music, McGill Univ, Montreal, Quebec, Canada)

Starting in 2002 John and Martina McBride set out to create several small rooms, each one unique in design and technology. George Massenburg (GM) was asked to coordinate the design of one of the rooms and given the mandate to make it the most advanced room imaginable. *GM* specified that the room should offer an accurate monitoring environment *for more than one listener*, a reasonable presentation of how materials will sound in multiple circumstances outside of the space, a flexible environment that doesn't compromise musical and artistic contexts, an accurate representation of virtual sources (sources spread across one or more loudspeakers) to more than one listener, a room with linear, *supportive* ambience, which as much as possible, has near-equal decay rates across as much of the frequency spectrum as possible. In June 2004, GM contacted Peter D'Antonio with a proposal to collaborate on a new surround music monitoring/control/recording room, which became Studio C. The resulting diffusor design covering all four walls and ceiling, with corner bass absorption, will be presented. Since opening the room has been used successfully and has recently installed a Dolby Atmos system. The presentation will also present user perceptions from 2005 to the present.

**4:40–5:10**
**Panel Discussion**

# Session 2pAB

## Animal Bioacoustics: General Topics in Animal Bioacoustics II—Marine

Alyssa W. Accomando, Chair
*Biologic and Bioacoust. Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106*

**Chair's Introduction—1:00**

### *Contributed Papers*

**1:05**

**2pAB1. Assessing vocal activity patterns of leopard seals (*Hydrurga leptonyx*) In the Bransfield Strait, Antarctica using machine learning.** Andrea Bonilla-Garzón (K. Lisa Yang Ctr. for Conservation Bioacoust., Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, nab242@cornell.edu), Shyam Madhusudhana (K. Lisa Yang Ctr. for Conservation Bioacoust., Cornell Univ., Ithaca, NY), Robert P. Dziak (Pacific Marine Environ. Lab., NOAA, Newport, OR), and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoust., Cornell Univ., Ithaca, NY)

Leopard seals (*Hydrurga leptonyx*) are widely distributed pack-ice seals in Antarctic and sub-Antarctic waters. As apex predators, these animals play a crucial role in the Southern Ocean food web. Data on population-level changes in their abundance and distribution may be a useful indicator of ecosystem-level changes in this unique and fragile environment. Over the past few decades, many studies have conclusively shown that passive acoustic monitoring (PAM) is an effective tool to monitor the abundance and distribution of vocally active marine mammals in remote and inaccessible areas for extended periods. However, handling and analyzing the vast amount of PAM data being collected remains challenging. Within the scope of this effort, we explored the use of a machine learning algorithm (convolutional neural network; CNN) to automatically detect the 'low double trill'; one of the most common leopard seal vocalizations, in three years of continuous acoustic data recorded in the Bransfield Strait, Antarctica between 2005 and 2008. After optimizing the algorithm, we evaluated its detection performance on various temporal scales (weeks, days, hours) to assess if CNNs are useful for monitoring leopard seal populations at ecologically relevant scales in the Southern Ocean.

**1:20**

**2pAB2. Automatic detection and classification system based on convolutional neural networks for bearded seal vocalizations in Alaska.** Christian D. Escobar-Amado (Electr. and Comput. Eng., Univ. of Delaware, 139 The Green, Newark, DE 19716, escobarc@udel.edu) and Mohsen Badiey (Electr. and comput. Eng., Univ. of Delaware, Newark, DE)

Bearded seal vocalizations were recorded by four spatially separated receivers on the Chukchi Continental Slope in Alaska in 2016–2017. Bearded seals vocalizations are often analyzed manually or by using automatic detections that are manually validated. An automatic detection and classification system (DCS) based on convolutional neural networks (CNNs) is proposed. The DCS is divided into two sections. First, regions of interest (ROI) containing potential bearded seal calls are detected through a 2D normalized cross-correlation of the measured spectrogram and a representative template of the two calls of interest considered in this work. Second, CNNs are used to classify the ROIs to determine if they are noise or a specified vocalization. The CNNs are trained on 80% of the ROIs manually labeled from one of the recorders and validated on the remaining 20% with a classification accuracy above 95.5%. When assessing the generalization performance of the DCS, we tested on the remaining three recorders located at different positions, obtaining a precision above 89.2% for the main class of the two types of calls. This study provides evidence that CNNs are suitable for classifying bearded seal vocalizations from ROIs found by classic detection techniques. [Work supported by ONR via Grant No. N00014-18-1-2140.]

**1:35**

**2pAB3. Machine learning models can predict bottlenose dolphin health status from whistle recordings.** Brittany L. Jones (National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brittany.jones@nmmf.org), Jeremy Karnowski (Univ. of California San Diego, San Diego, CA), Jessica Sportelli, and Sam Ridgway (National Marine Mammal Foundation, San Diego, CA)

Recently, there have been incredible strides in applying machine and deep learning methods to predict the presence of disease states in humans from recordings of voice. Pattern analyses of dolphin whistles have historically depended on human operators visually and audibly differentiating between sound types. The training and implementation of machine and deep learning strategies for audio analyses reduce human bias and may improve feature detection. Our team at the Navy Marine Mammal Program in San Diego, CA has developed a substantial vocal catalog of whistle recordings for a group of focal dolphins. Additionally, an extensive health history database for each animal is maintained as part of a preventive medicine program. Together, these result in a unique, labeled dataset comprised of tens of thousands of whistles (input) emitted during differing health states (output) that have been leveraged for training machine and deep learning models to classify health status from whistles. We present preliminary results that suggest that health information may be encoded across dolphin whistle characteristics, similar to changes in the human voice. Further, we describe the applied goals for testing and implementing these innovative tools for early predicting changes in dolphin health status from non-invasive recordings.

1:50

**2pAB4. Determining the spatial and temporal variability of false killer whales (*Pseudorca crassidens*) and short-finned pilot whales (*Globicephala macrorhynchus*) in the Hawaiian Islands from passive acoustic monitoring.** Brijonnay Madrigal (Marine Mammal Res. Program, Univ. of Hawai'i at Mānoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744, bcm2@hawaii.edu), Jennifer L. McCullough (NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), Marc Lammers (Hawaiian Islands Humpback Whale National Marine Sanctuary, NOAA, Kihei, HI), Erin Oleson (NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), Lars Bejder (Marine Mammal Res. Program, Univ. of Hawai'i at Mānoa, Kaneohe, HI), and Aude Pacini (Marine Mammal Res. Program, Univ. of Hawai'i at Mānoa, Kaneohe, HI)

False killer whales (*Pseudorca crassidens*) and short-finned pilot whales (*Globicephala macrorhynchus*) are resident, toothed whale species to the Hawaiian archipelago. False killer whales are considered of high concern in Hawai'i with the insular population listed as endangered. Passive acoustic monitoring (PAM) is an effective technique for studying these highly social odontocete species on large temporal and geographical scales. Moored hydrophones were deployed inside and outside Marine Protected Areas (MPAs) through the SanctSound project, allowing leveraging of long-term datasets to assess the acoustic behavior of these species within and beyond the Hawaiian Islands Humpback Whale National Marine Sanctuary and Papahānaumokuākea Marine National Monument. High Frequency Acoustic Recording Packages (HARPs) (200 kHz SR) and SoundTraps (48 kHz SR), recording on various duty cycles, were deployed between 2018–2022 within MPAs (monument: n = 51 months; sanctuary: n = 76 months) and outside MPAs (n = 60 months). PAMGuard detectors were used to detect whistles, clicks, and burst pulses. Detection features were extracted using the PAMpal package and events classified to species using BANTER. Acoustic monitoring provides key information about the value of MPAs and a better understanding of habitat use and behavior of these species.

2:05

**2pAB5. Acoustic density estimation of North Atlantic right whales (*Eubalaena glacialis*) in Cape Cod Bay, MA, USA.** Marissa Garcia (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, mg2377@cornell.edu), Irina Tolkova (School of Eng. and Appl. Sci., Harvard Univ., Cambridge, MA), Shyam Madhusudhana (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), Kaitlin Palmer (SMRU Consulting, Vacouver, British Columbia, Canada), Ashakur Rahaman (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY), Scott Baker, Debbie Steel (Marine Mammal Inst., Oregon State Univ., Newport, OR), Charles Mayo, Christy Hudak (Ctr. for Coastal Studies, Provincetown, MA), and Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Univ., Ithaca, NY)

With fewer than 400 individuals remaining, critically endangered North Atlantic right whales (*Eubalaena glacialis*; NARWs) embody New England's foremost marine conservation challenge. In spring, a large portion of the NARW population visits Cape Cod Bay (CCB), MA, USA, a critical foraging area. Traditionally, aerial surveys have documented the abundance and distribution of NARWs in CCB. In this work, we employ passive acoustic monitoring to accomplish this task. This method offers a cost-effective and largely automated approach toward improved temporal resolution of observations. Regularly and universally produced by NARWs of all ages and sexes, the upcall vocalization serves as a proxy for NARW presence. This contact call's highly stereotyped signal structure renders it optimal for detection through machine learning. From February to May 2019, five marine autonomous recording units (MARUs) continuously monitored the underwater soundscape in CCB. Manual upcall annotations across 20 days were used to evaluate a NARW automatic detector that ran over the full recording duration. Multiple arrivals across the MARUs were then matched through time-difference-of-arrival association. After estimating population-level calling rates, NARW density in CCB was calculated across the season. Results will be compared to visual survey observations conducted by the Center for Coastal Studies.

**2:20–2:35 Break**

2:35

**2pAB6. Directional detection of simulated whale calls and ambient noise in a busy harbour.** Carolyn Binder (Defence Res. and Development Canada, 9 Grove St., Dartmouth, Nova Scotia B2Y3Z7, Canada, carolyn.binder@forces.gc.ca), Sean Pecknold (DRDC Atlantic Res. Ctr., Halifax, Nova Scotia, Canada), Calder L. Robinson (Dartmouth, Nova Scotia, Canada), and S. B. Martin (Halifax, JASCO Appl. Sci., Dartmouth, Nova Scotia, Canada)

Timely information on marine mammal presence in, or near, sonar training areas can be used to minimize the risk of harming these animals. Defence Research and Development Canada (DRDC) has recently been investigating a variety of acoustic technologies to facilitate real-time marine mammal monitoring in high noise (including ship noise and sonar signals) environments. DRDC assessed the directional detection and localization capability of a network of sub-surface volumetric and vertical line arrays against the noisy backdrop of the busy harbour during an experiment conducted in Bedford Basin, NS. Playbacks of Southern Resident killer whale calls and North Atlantic right whale upcalls, as well as simulated calls, were transmitted. To assess variability in bearing estimates for whale calls in different frequency bands, an acoustic source was fixed at a single location for half of the experiment and dipped over the side of a boat at seven locations around the recorders during the other half. During this talk, ambient noise and environmental measurements will be presented along with acoustic localization results.

2:50

**2pAB7. A yearlong record of acoustic propagation and ambient sound in a seagrass meadow.** Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78665, meganb@arlut.utexas.edu), Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kyle Capistrant-Fossa (Marine Sci. Inst., Univ. of Texas at Austin, Port Aransas, TX), Andrew R. McNeese, Prithika Sen, Thomas S. Jerome (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Kenneth H. Dunton (Marine Sci. Inst., Univ. of Texas at Austin, Port Aransas, TX)

Seagrasses are sentinel species whose structures are sensitive to variations in environmental conditions and thus reliable indicators of long-term changes in sea level and regional climate. The biological and physical characteristics of seagrasses are known to affect acoustic propagation. Gas bodies contained within the seagrass tissue as well as photosynthetic-driven bubble production result in free gas bubbles attached to the plants and in the water. The detachment of gas bubbles from the plants is also a source of ambient sound. In this work, we deployed a measurement system in a shallow (1.3 m deep) seagrass meadow that included an acoustic projector, a set of receiving hydrophones, an instrumentation pressure vessel that housed the electronics controlling the acoustic data acquisition and data storage, and a suite of environmental sensor loggers. Acoustic propagation and ambient sound were collected every ten minutes for a period of one year. Coincident measurements of water temperature, salinity, dissolved oxygen, and photosynthetically active radiation were also collected and used to interpret the acoustic data. We present preliminary results from the yearlong deployment of the acoustic system in a moderately dense seagrass bed dominated by Thalassia testudinum (turtle grass) in Corpus Christi Bay, Texas. [Work supported by NSF.]

3:05

**2pAB8. Anthropogenic sound sources in expanded southern resident killer whale critical habitat.** Emily N. Drappeau (Fisheries, NOAA, 50 Lower College Rd., Kingston, RI 02881, emily_drappeau@uri.edu)

During the last decade, Southern Resident Killer Whale (SRKW) sightings in the Salish Sea have declined significantly. In 2021 NOAA expanded the critical habitat for SRKWs to include waters off of coastal Washington Oregon and California. As little is known about this territory, it is crucial to identify and understand potential threats. Underwater explosive sounds have been recorded and analyzed using passive acoustic monitors on the Washington Coast. These recordings were reviewed visually and aurally for key characteristics to determine their source. According to previously published

descriptions, most of the approximately 16,500 sounds of high quality that were analyzed matched the characteristics of seal bombs, which are typically used in fishing operations to deter marine mammals. Using spatial and temporal sounds recorded in 2018 as a baseline, occurrences in 2019 and 2020 were further examined. Multiple lines of evidence have led us to believe that these explosions are associated with fishing, not military activity and are a potential threat to SRKW. It is this project's objective to continue to use published signatures to study the further changes. Passive acoustic monitoring is a vital mechanism in studying the emerging threats to SRKW outside the Salish sea.

**3:20**

**2pAB9. A simple approach for marine mammal noise-induced threshold shift prediction from down-the-hole piling noise exposure.** Shane Guan (Div. of Environ. Sci., BOEM, 45600 Woodland Rd., Sterling, VA 20166, guan@cua.edu)

In assessing auditory effects on marine mammals from noise exposure, a dual-criterion is currently used by regulators to estimate the noise-induced threshold shift (NITS). Such criterion classifies anthropogenic noise into two mutually exclusive categories: Impulsive and non-impulsive. However, in real world situations, marine mammals are often exposed to complex noise field that contains both impulsive and non-impulsive components, thus makes it difficult or even impossible to accurately assess their NITS from noise exposure. One example is down-the-hole (DTH) pile installation, which generates both impulsive noise from percussive drilling/striking and non-impulsive noise from debris removal simultaneously. In this study, a relatively simple approach is proposed that employs kurtosis values to quantify the impulsiveness of two DTH piling noises datasets for six different marine mammal functional hearing groups. A kurtosis adjustment approach that has been suggested for predict human hearing loss from noise exposure was then used to build correction factors for estimation of NITS of marine mammals exposed to DTH piling noises. Further research on marine mammal NITS from complex noise exposure is needed to validate and improve this model.

**3:35**

**2pAB10. Temporary threshold shift from continuous 20–40 kHz hyperbolic upsweeps in bottlenose dolphins (*Tursiops truncatus*).** Jason Mulsow (Biologic and Bioacoustic Res., National Marine Mammal Foundation, 2240 Shelter Island Dr. Suite 200, San Diego, CA 92106, jason.mulsow@nmmf.org), Carolyn E. Schlundt (Peraton Corp., San Diego, CA), Alyssa W. Accomando (Biologic and Bioacoustic Res., National Marine Mammal Foundation, San Diego, CA), and James J. Finneran (US Navy Marine Mammal Prog., NIWC Pacific Code 56710, San Diego, CA)

Naval continuous active sonar (CAS) can operate at high duty cycles and result in sound exposure levels (SELs, in dB re 1 $\mu Pa^2 s$) that may induce temporary threshold shift (TTS) in marine mammals. Estimating the impacts of CAS on marine mammal hearing is difficult however, given the scarcity of TTS data for long-duration tonal exposures. For this ongoing study, bottlenose dolphin TTS was measured following exposure to continuous playback of 19-s hyperbolic upsweeps from 20–40 kHz, at multiple sound pressure levels and exposure durations from 2 to 60 min (100% duty cycle). The upsweep was based on characteristics of naval CAS, but frequency-shifted into the dolphin's range of best hearing to predict how baleen whales might be affected by CAS exposures below 10 kHz. Hearing was measured using both psychophysical and auditory brainstem response measurements. To date, the greatest cumulative exposures of 180 dB SEL (normal-hearing dolphin) and 183 dB SEL (dolphin with elevated hearing thresholds) resulted in less than 6 dB of behavioral TTS. Auditory brainstem response amplitudes for these exposures did not indicate noise-induced fatigue. Future testing will evaluate TTS following exposure to 28-kHz pure tones with comparable SELs. [Work funded by US Navy Living Marine Resources.]

## Session 2pAO

# Acoustical Oceanography, Underwater Acoustics, and Signal Processing in Acoustics: Acoustical Remote Sensing, Navigation, and Passive Monitoring in the Polar Ocean II

Matthew Dzieciuch, Cochair
*Univ. of California, San Diego, Scripps Inst. of Oceanogr., San Diego, CA 92122*

Hanne Sagen, Cochair
*Nansen Environ. and Remote Sensing Ctr., Jahnebakken 3, Bergen, 5007, Norway*

Peter F. Worcester, Cochair
*Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, San Diego, CA 92093-0225*



### *Contributed Papers*

**1:00**

**2pAO1. Multi-frequency systems for arctic acoustic navigation and communications.** Lee Freitag (AOPE, Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, lfreitag@whoi.edu), Craig Lee, Jason Gobat (APL/UW, Seattle, WA), Sandipa Singh, Dennis Giaya, Eric Gallimore, and Keenan Ball (AOPE, Woods Hole Oceanogr. Inst., Woods Hole, MA)

Autonomous systems, including gliders, floats, and propeller-driven AUVs, require navigation under Arctic ice to be able to geolocate during their missions. In addition, acoustic communications provides for unidirectional or bidirectional data and command flow with the autonomous systems. Over the past ten years, steady progress has been made on implementing and demonstrating these acoustic systems in the Beaufort Sea north of Alaska. The eventual goal is a multi-frequency capability that uses 35 Hz for very long range navigation, 900 Hz for long-range navigation and communications, plus 10 kHz for close-range use. The different frequencies offer different advantages, and each is used accordingly. At 35 Hz, navigation transmissions span the water-column and survive reflection from the underside of the ice, allowing pan-Arctic ranges. The 900 Hz navigation and communications signals persist in the shallow (150 m) duct in the Beaufort at ranges up to several hundred kilometers, which is useful for under-ice survey transits. Finally, at 10 kHz, compressed data can be off-loaded and new mission files transferred via buoys on the ice. Recent work in 2021 and 2022 has continued to develop these capabilities, and this talk summarizes results and describes future directions.

**1:15**

**2pAO2. Acoustic float tracking with the Kalman smoother.** Paul Chamberlain (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093, pchamber@ucsd.edu), Bruce Cornuelle, Lynne Talley (Scripps Inst. of Oceanogr., La Jolla, CA), Kevin Speer, Cathrine Hancock (Florida State Univ., Tallahassee, FL), and Stephen Riser (Univ. of Washington, Seattle, WA)

Acoustically tracked subsurface floats provide insights into ocean complexity and were first deployed over 60 years ago. A standard tracking method uses a Least-Squares algorithm to estimate float trajectories based on acoustic ranging from moored sound sources. However, infrequent or imperfect data challenge such estimates. Acoustic tracking is currently the only feasible strategy for recovering float positions in the sea ice region, a focus of this study. Acoustic records recovered from under-ice floats frequently lack continuous sound source coverage. This is because environmental factors such as surface sound channels and sea ice attenuate acoustic signals, while operational considerations make polar sound sources difficult to deploy. Here we present a Kalman Smoother approach that, by including some estimates of float behavior, extends tracking to situations with more challenging data sets. The Kalman Smoother constructs dynamically constrained, error-minimized float tracks using all possible position data. The Kalman Smoother is applied to previously-tracked floats from the southeast Pacific (DIMES experiment), and the results are compared with existing trajectories constructed using the Least-Squares algorithm. The Kalman Smoother is also used to reconstruct the trajectories of a set of previously untracked, acoustically-enabled Argo floats in the Weddell Sea.

**1:30**

**2pAO3. Under-ice 4-8 kHz acoustic transmission and communication in ducted environments in the Arctic Ocean.** Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Horten 3191, Norway, Dag.Tollefsen@ffi.no), Paul van Walree, Trond Jenserud, and Vidar Forsmo (Norwegian Defence Res. Est. (FFI), Horten, Norway)

Underwater communications and navigation for small platforms can typically involve acoustic systems operating at kHz frequencies. For under-ice applications, this motivates studies of acoustic transmission in present-day Arctic conditions. This paper presents data from recent mid-frequency (4–8 kHz) under-ice acoustic transmission and communication experiments to ~10 km range in the Nansen Basin and in the Beaufort Sea of the Arctic Ocean. For shallow sources and receivers, delay profiles show early arrivals due to ducted paths followed by a group of strong arrivals due to bottom-interacting paths. The relative amplitudes of these groups depend on duct and under-ice properties. We demonstrate that despite shallow (~25 m) source/receiver depths, good communication performance can be achieved in these single- and double-ducted environments by exploiting the bottom-interacting paths. Simulations with the Bellhop ray model support the findings.

**1:45**

**2pAO4. Environmentally adaptive acoustic communication and navigation for underice autonomous operation.** Henrik Schmidt (Mech. Eng., MIT, 77 Massachusetts Ave., Cambridge, MA 02139, henrik@mit.edu)

The Arctic acoustic environment has changed dramatically, with multi-year ice being less abundant and the influx of warmer Pacific water, the so-called "Beaufort Lens," creating a local maximum in the sound speed at a depth of 50–80 m. In contrast to the classic Arctic sound speed profile which had a monotonic increase in sound speed yielding a surface duct with strong ice interaction, the warm water lens creates a lower duct supporting

extremely efficient propagation to long ranges. On the other hand, this double duct environment is decremental to short range communication and navigation of underwater vehicles. Thus, for a transmitter in one channel, the lens creates distinct shadow zones in the other channel at ranges between 1 and 4 km, which are the typical operational ranges of small AUVs, severely affecting navigation and communication performance. In addition, the temporal variability of is significant due to internal waves. During ICEX20 MIT and WHOI demonstrated an integrated communication and navigation concept where a network of ice-moored modem buoys with GPS tracking were used to track an under-ice AUV using regular CTD updates of optimal ranging, which in combination with a novel dynamic model constrained navigation fusion engine demonstrating GPS-grade navigation accuracy over several hours of operation. Using an EOF framework for updating the onboard environmental awareness on the AUV, the concept supports autonomous depth selection for optimal communication connectivity. [Work supported by ONR and UWDC/ASL.]

**2:00**

**2pAO5. Fast indicators for robust acoustic communication and navigation in a double-ducted polar environment.** EeShan Bhatt (Woods Hole Oceanogr. Inst., 77 Massachusetts Ave., 5-223, Cambridge, MA 02139, ebhatt@whoi.edu), Bradli Howard (MIT-WHOI Joint Program, Cambridge, MA), and Henrik Schmidt (Massachusetts Inst. of Technol., Cambridge, MA)

A floating ice-buoy system equipped with GPS and acoustic modems provided communication and navigation to a Bluefin 21 Autonomous Underwater Vehicle (AUV) for under-ice operations in the Beaufort Sea, in March 2020. Acoustic modems were placed above and below the "Beaufort Lens," at 30 and 90 m, where a data- and model-based autonomy selected which depth to use for transmitting messages to the AUV. Similarly, the AUV ran a depth adaptive path planning behavior to maintain acoustic communications with the ice buoy modems. The untethered mission showed a non-diverging navigation solution with the average error roughly 0.1% of the total 11 km traveled. This talk will summarize key metrics introduced for automated, on-the-fly processing of acoustic packets to navigate within a reliable acoustic channel and minimize navigation error. [Supported by Office of Naval Research.]

**2:15**

**2pAO6. Acoustic receptions at close ranges measured on autonomous underwater vehicles in the Beaufort Sea.** Luis O. Pomales Velázquez (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI, luispomales@uri.edu), Isaac B. Salazar, Cristian E. Graupe, Jessica Desrochers (Ocean Eng., The Univ. of Rhode Island, Newport, RI), Sarah E. Webster (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Rapid warming of the Pacific Summer Water layer strengthens a subsurface duct in the Beaufort Sea, allowing for long-range propagation at low frequencies. An array of tomography sources was deployed within the duct as part of the Canada Basin Acoustic Propagation Experiment (CANAPE) to study acoustic propagation in this environment. The moored transceivers provide measurements of acoustic propagation at several ranges from 176 to 285 km. Additionally, two Seaglider vehicles equipped with hydrophone receivers navigated in and around the CANAPE array and recorded the transmissions from the moored sources at ranges as far as 530 km and as close as 2 km. A spatially variable sound speed environment was generated from in-situ data measured by the Seagliders and CTD casts from research vessels. Acoustic arrivals measured on the vehicles were matched to range-dependent acoustic predictions made with a broadband Parabolic Equation model to estimate source-receiver range. Acoustic receptions from multiple moored sources were used to localize the Seagliders. Here, we examine the close range (2–25 km) receptions and their impacts on acoustic localization.

**2:30–2:45 Break**

**2:45**

**2pAO7. Travel-time variability during the 2016–2017 deep-water Canada Basin Acoustic Propagation Experiment.** Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, San Diego, CA 92093-0225, pworcester@ucsd.edu), Matthew Dzieciuch, Heriberto J. Vazquez, and Bruce Cornuelle (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The Arctic Ocean is undergoing dramatic changes in response to increasing atmospheric concentrations of greenhouse gases. The decreases in ice extent, the near disappearance of multiyear ice, and changes in the stratification of the ocean all have important implications for underwater acoustic propagation. During the 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE), a long vertical receiving array was embedded within an ocean acoustic tomography array of six acoustic transceivers with a radius of 150 km. The impulse response of the ocean was measured every four hours using broadband signals centered at about 250 Hz. The observed travel-time variability was extraordinarily low, reflecting both the low internal-wave energy level and sparseness of mesoscale eddies in the Canada Basin. The peak-to-peak travel time variability of the early, resolved ray arrivals was only a few tens of milliseconds, and the standard deviations over the entire year were only a few milliseconds. The travel-time spectra show increasing energy at lower frequencies and enhanced semidiurnal variability, presumably due to some combination of the semidiurnal tides and inertial variability. The travel-time fluctuations are roughly an order of magnitude smaller than is typical in midlatitudes at similar ranges.

**3:00**

**2pAO8. Ocean acoustic tomography in the Beaufort Gyre.** Heriberto J. Vazquez (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA, hvazquezperalta@ucsd.edu), Bruce D. Cornuelle, Peter F. Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

An ocean acoustic tomography array with a radius of 150 km was installed in the central Beaufort Gyre during 2016–2017 for the Canada Basin Acoustic Propagation Experiment (CANAPE). Five transceivers were deployed in a pentagon shape with a sixth transceiver at the center and a long vertical receiving array northwest of the central mooring. At least 12 refracted-surface-reflected (RSR) ray arrivals with lower turning points at depths between 500 and 3500 m were resolved in the acoustic receptions at all receivers. Travel-time anomalies were computed relative to a range-dependent sound-speed reference made by objectively interpolating annual-average sound-speed profiles constructed from the temperature data at each mooring. The travel time anomalies were inverted to estimate the 3-D sound-speed anomaly, including corrections to the positions of sources and receivers consistent with the uncertainty from long-baseline acoustic navigation systems at each mooring. Although the deep water in the Canada Basin is nearly homogeneous in temperature and salinity and highly stable (slowly warming in response to geothermal heating), it proved necessary to allow for a sound-speed change in the deep ocean to obtain consistent inversions, suggesting that the sound-speed equation at high pressure and low temperature is in error by about 0.1–0.2 $ms^{-1}$.

**3:15**

**2pAO9. Toward predicting Arctic Ocean acoustic travel times using an Earth system model.** Siobhan Niklasson (New Mexico Tech, Los Alamos National Lab., PO Box 1663, Los Alamos, NM 87545, sniklasson@lanl.gov), Milena Veneziani, Charlotte Rowe (Los Alamos National Lab., Los Alamos, NM), Susan Bilek (New Mexico Tech, Socorro, NM), Stephen Price, Andrew Roberts (Los Alamos National Lab., Los Alamos, NM), Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), and Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The hydroacoustic environment of a rapidly warming Arctic Ocean will be impacted by changing thermohaline structure, increased marine traffic, changes in sea ice coverage, and likely increases in microseism/storm noise. This will lead to obsolescence for today's Arctic Ocean acoustic models just as the need for understanding and monitoring the Arctic acoustic environment becomes more critical. To make sophisticated predictions for coming

conditions, we need fully-coupled Earth system models. Los Alamos National Laboratory is contributing to developing the Department of Energy's Energy Exascale Earth System Model (E3SM), which can potentially predict the ocean and sea ice conditions necessary to drive an acoustics model in a rapidly-evolving Arctic Ocean. We present a preliminary analysis of a comparison of acoustic travel times in the Canada Basin calculated from E3SM simulations with measured travel times from the 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE) and travel times computed from ice-tethered-profiler measurements of acoustic properties in the water column. The goals of this effort are to provide boundary ocean conditions to an acoustic model, to quantify the ocean acoustic implications of climate change, as well as to create a climate-aware atlas of global acoustic noise that could be applied, for example, in signal detection.

**3:30**

**2pAO10. Ice-tethered acoustic buoys for real-time active and passive monitoring in the Western Arctic.** Altan Turgut (Naval Research Lab., Acoust. Div., Washington, D.C. 20375, altan.turgut@nrl.navy.mil) and Jeffrey A. Schindall (Naval Res. Lab., Washington, D.C.)

Real-time monitoring of channel impulse response functions in the Beaufort Lens was performed by five Ice-Tethered Acoustic Buoys (ITABs) during March-September 2021. Each ITAB had an eight-element vertical line array and a 900-1000 Hz acoustic source, providing a reciprocal acoustic transmission of up to a 300 km range. Acoustic navigation of one ITAB was successfully tested using the acoustic travel-time from the other ITABs. In addition, a strong single arrival through the Beaufort Lens was observed on each ITAB. Simulations using the measured Q-functions [T. C. Yang, IEEE J. Ocean. Eng. **30**(4), 865–880 (2005)] showed a reliable long-range acoustic communication through the Beaufort lens within the 900–1000 Hz frequency band. Finally, a passive ITAB was deployed in March 2022 to measure third-octave under/in-ice ambient noise in the Western Arctic. Both experiments showed that ITABs can be used for under-ice acoustic navigation and communication as well as real-time ambient noise measurements in the Western Arctic. [Work supported by the ONR.]

**3:45**

**2pAO11. Prospects for acoustic remote sensing and acoustic system performance in the Beaufort Gyre region.** Timothy F. Duda (Woods Hole Oceanogr. Inst., Mail Stop 11, Woods Hole, MA 02543, tduda@whoi.edu)

In the Beaufort Gyre region north of Alaska, the vertical interleaving of the near-surface temperature maximum, the warm Pacific Summer Water (PSW), the cool Pacific Winter Water, and the Atlantic layer make for unusual acoustic conditions. The dynamics of these upper ocean layers cause typical complex and turbulent-like oceanic flow that causes geographically variable heat content and acoustics. A sound duct in the PWW, below the PSW and above the Atlantic Layer, filling $\sim$75 to 225 m depth, is prominent but not universal. Erosion of the PSW warm layer by either vertical or horizontal mixing processes often breaks the duct, with an additional seasonal cycle of ducting effectiveness, and allows sound to escape to interact with the scattering and attenuating surface. The duct enables communication and navigation signaling, but with that interruption caveat. It also allows passive sensing, best for sound that enters the duct and moves through it. A challenge is to utilize sound that exists outside the duct. This includes sounds emitted outside the duct by mammals, fracturing ice, or anything else. This sound will either cycle through the duct at steep angle, or stay above the duct and repeatedly reflect from the top boundary. The combination of unducted sound being subject to strong attenuation by ice or wave interaction (the typical high-latitude situation) and high variability of the duct effectiveness, imposes limitations on the usefulness of the otherwise convenient duct.

**4:00**

**2pAO12. Double duct propagation in the Arctic.** Arthur B. Baggeroer (Mech. and Electr. Eng., Massachusetts Inst. of Technol., Room 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

Recent experiments in the Beaufort Sea encountered the "Beaufort Lens," a hydrographic feature named by Russian acousticians, where warmer Pacific water enters the Bering Straits and rests above Atlantic water. This creates a "double duct" subsurface duct with this "lens" being near ubiquitous in space and time throughout the Beaufort. As part of the ONR Task Force Ocean program we found subsurface ducts in the Irminger Sea over the Reykjanes Ridge not reliably predicted by HYCOM. Propagation in these "double ducts" has some remarkable features and is well understood using normal modes and analogies to potential wells in quantum mechanics. These include the (i) filling of individual wells preferentially according to the modal phase speeds; (ii) tunneling phenomena when the phase speeds approaches the height of the potential barrier or maximum speed separating the ducts; (iii) issues of degeneracy when the modal phase speeds appear to cross; (iv) appearance of "mini convergence zones" for the transmission losses in each duct; (v) differences in modal group speeds with very small differences in phase speeds; and (vi) very low transmission loss when both source and receiver are in the subsurface duct separated from the water and/or ice surfaces.

## Session 2pBAa

# Biomedical Acoustics, Computational Acoustics, and Signal Processing in Acoustics: Deep Learning in Ultrasound Imaging and Tissue Characterization II

Aiguo Han, Cochair
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Xiaoming Zhang, Cochair
*Mayo Clinic, 200 1st St. SW, Rochester, MN 55905*

### *Invited Paper*

**1:00**

**2pBAa1. Deep learning-based super-resolution ultrasound microvascular imaging.** Pengfei Song (Electr. and Comput. Eng., Univ. of Illinois Urbana-Champaign, 405 N. Mathews Ave., Beckman Inst. 4041, Urbana, IL 61801, songp@illinois.edu)

Super-resolution ultrasound microvascular imaging (SR-UMI) is a fast-growing field that leverages contrast microbubbles (MB) to achieve super-resolution and track blood flow. Conventional SR-UMI is primarily based on MB localization and tracking, which is a challenging task *in vivo* with complex MB responses subject to various tissue characteristics and imaging settings. Conventional MB localization and tracking approaches based on experimentally calibrated or engineered MB templates typically fall short of capturing the widespread distribution of MB signals *in vivo*, leading to suboptimal SR-UMI performance. In contrast, deep learning (DL) provides many powerful tools to better represent the complex landscape of MB and blood flow signals that lead to enhanced SR-UMI. In this talk, I will first introduce several DL-based MB localization techniques with discussion on challenges and solutions involving DL training within the context of SR-UMI. I will then introduce several other novel DL-based SR-UMI techniques that do not rely on MB localization or tracking to achieve super-resolution and infer blood velocity. Some of the techniques are contrast-free. *In vivo* imaging examples from chorioallantoic membrane as well as mouse brain will be presented as validations of the DL-based SR-UMI techniques. *In vivo* applications in cancer and Alzheimer's disease will also be presented and discussed in this talk.

### *Contributed Papers*

**1:25**

**2pBAa2. Deep learning-based fast and dense microbubble localization for ultrasound localization microscopy.** YiRang Shin (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois at Urbana-Champaign, 405 N Mathews Ave., Urbana, IL 61801, yirangs2@illinois.edu), Matthew R. Lowerison, Zhijie Dong, Xi Chen, Qi You, Nathiya V. ChandraSekaran, Daniel A. Llano, Mark Anastasio, and Pengfei Song (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Ultrasound localization microscopy (ULM) has gained substantial attention owing to its ability to super-resolve minute blood vessels at clinically relevant imaging depth. However, accurate localization of individual microbubbles (MBs) in areas with high MB concentration and overlapping point spread functions (PSFs) remains a challenge. Furthermore, existing localization methods based on pre-determined MB PSFs cannot reflect the highly non-stationary PSFs that vary spatially with MB concentration and nonlinear responses, ultrasound diffraction, and imaging settings. To address these limitations, we implemented the DECODE (DEep COntext Dependent) neural network that was recently developed for optical imaging on ULM. DECODE constructs a Gaussian mixture model with a mixed MB count and localization loss to output the probability, uncertainty, and sub-pixel location corresponding to each MB detection, achieving accurate identification and localization of MBs. Notably, DECODE was trained with realistic simulation data that incorporates MB brightness, movement, ultrasound system noise, and PSFs produced by a generative adversarial network that encodes the internal distribution of MB PSFs obtained from *in vivo* ultrasound imaging. In high MB-density regime, simulation studies demonstrated that DECODE improved MB detection rate from 41% to 95%, and reduced localization error from 109.4 $\mu$m to 32.5 $\mu$m (20 MHz) when comparing to conventional MB localization techniques. DECODE also demonstrated improved *in vivo* ULM imaging in mouse brain.

**1:40**

**2pBAa3. On the effect of training simulation point spread function in deep-learning based super-resolution ultrasound imaging.** Xi Chen (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois Urbana-Champaign, 405 N Mathews Ave., Urbana, IL 61801, xichen30@illinois.edu), Qi You, Zhijie Dong, Matthew R. Lowerison, and Pengfei Song (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Deep learning (DL)-based super-resolution ultrasound microvascular imaging (SR-UMI) has gained interest in recent years owing to the success of DL in medical imaging applications. Mainstream DL techniques require an abundance of labeled training data. Since high-quality *in vivo* training data are challenging to obtain for medical imaging, most of the current studies have relied on simulations to generate training data of microbubbles (MBs) flowing in vasculature. This study investigates the effect of different MB point spread function (PSF) generation methods on the performance of DL models for SR-UMI. We used Gaussian shaped PSF (Gaussian-PSF), numerical simulation software (Field II-PSF), and experimentally collected

PSFs (Exp-PSF) to generate DL training sets. The training sets were used to train DL models that performed both MB localization- and non-localization-based SR-UMI. The quality of the models trained with different datasets were evaluated based on their *in vivo* performance (mouse brain, chicken embryo), with optical imaging as ground truth for some applications. We discovered that Exp-PSF achieved the best performance overall, due to its high resemblance to real experimental data. Finally, we present a pipeline for obtaining Exp-PSF and integrating them with ultrasound MB simulation, with a publicly available Exp-PSF dataset provided via Github.

**1:55–2:10 Break**

## *Invited Paper*

**2:10**

**2pBAa4. Applications of ultrasound image reconstruction using deep learning.** Dongwoon Hyun (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, dongwoon.hyun@stanford.edu), Leandra Brickson (Electr. Eng., Stanford Univ., Palo Alto, CA), Louise Zhuang (Electr. Eng., Stanford Univ., Stanford, CA), Walter A. Simson (Radiology, Stanford Univ., Palo Alto, CA), Gianmarco Pinton (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., Chapel Hill, NC), and Jeremy Dahl (Radiology, Stanford Univ., Palo Alto, CA)

Deep learning has gained tremendous popularity as a tool for ultrasound beamforming and image reconstruction. In previous work, we trained deep neural networks (DNNs) to estimate the echogenicity of a medium, to improve acoustical and electronic signal-to-noise ratio (SNR) in channel data, and to detect targeted microbubbles nondestructively for real-time ultrasound molecular imaging. Here, we present several advancements to each application. First, we compare the speckle- and noise-reducing performance of DNNs trained with simple linear Field II simulations of photographic images versus that of DNNs trained with full wave finite-difference time domain numerical simulations containing realistic abdominal walls and the resulting image degradation artifacts. We further extend our nondestructive molecular imaging DNN to incorporate spatiotemporal information using an extended simulation study to increase specificity for stationary bound microbubbles and further improve nondestructive molecular imaging.



## *Contributed Papers*

**2:35**

**2pBAa5. Transcranial ultrasound imaging using pulse-echo ultrasound and deep learning: A numerical study.** Zixuan Tian (Univ. of Illinois at Urbana-Champaign, 307 N Wright, Urbana, IL 61820, zixuant5@illinois.edu), Yun Jing (Acoust., Penn State Univ., State College, PA), and Aiguo Han (Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Phase aberration caused by skulls is a main challenge in transcranial ultrasound imaging for adults. Aberration could be corrected if the skull profile (i.e., thickness distribution) and speed of sound (SOS) are known. We previously designed a deep learning (DL) model to estimate the skull profile and SOS using pulse-echo ultrasound signals. This study's objective is to develop strategies to improve the estimation and evaluate the effectiveness of aberration correction in transcranial ultrasound imaging. Acoustic simulations were performed using k-Wave in this numerical study. The following strategies were used to improve estimation: (1) A phased array was used instead of a single-element transducer; (2) Channel radiofrequency data were used instead of beamformed data as the DL model input; (3) A DL model was developed to incorporate physics into architecture design and model training. Compared with previously reported results, these strategies improved the correlation coefficient between the estimated and ground-truth values from 0.82 to 0.94 for SOS, and from 0.98 to 0.99 for thickness. Simulated transcranial images of point targets with phase correction using the estimated SOS and thickness values showed significantly reduced artifacts than those without correction. The results demonstrate feasibility of the proposed approach for transcranial ultrasound imaging.

**2:50**

**2pBAa6. Raw channel data-based phase aberration correction for ultrasound localization microscopy using conditional generative adversarial networks.** Zhijie Dong (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois Urbana-Champaign, 4031 Beckman Inst., 405 N Mathews Ave., Urbana, IL 61801, zhijied3@illinois.edu), YiRang Shin, Xi Chen, Qi You, Matthew R. Lowerison, Mark Anastasio, and Pengfei Song (Beckman Inst. for Adv. Sci. and Technol., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Phase aberration (PA) significantly deteriorates imaging quality for transcranial imaging because of large speed of sound discrepancy and complex geometry of the skull. Recently, ultrasound localization microscopy (ULM) has demonstrated promising transcranial imaging results of deep brain microvasculature through intact skull. Robust PA correction is essential for high-fidelity transcranial ULM imaging because microbubble localization is highly susceptible to PA. Most existing PA correction methods rely on "guide stars," which may not be practical for ULM because the microbubble concentration may be too high to serve as guide stars. Furthermore, most conventional PA correction techniques assume globally or locally homogeneous delay law, thus resulting in suboptimal correction performance. To address these issues, we propose a deep learning-based PA correction approach that directly operates on the pre-beamformed, raw channel data. By taking advantage of the structured microbubble signals and using conditional generative adversarial networks to utilize specific aberrated channel data, our proposed method achieved pixel-wise PA correction. *In silico* studies using micro-CT scanned mouse skulls demonstrate that the lateral beam width at $-20$ dB is improved from $0.82 \pm 0.33$ mm to $0.48 \pm 0.12$ mm at 20 MHz applying proposed method, and the mean localization error is reduced from $67.99 \pm 21.50$ $\mu$m to $21.45 \pm 14.84$ $\mu$m.

**3:05**

**2pBAa7. Deep-learning based *insitu* ultrasound thermometry for thermal ablation monitoring.** Ajay Anand (Goergen Inst. for Data Sci., Univ. of Rochester, 250 Hutchinson Rd., Rochester, NY 14534, ajay.anand@rochester.edu), Ashwin Ramesh (Dept. of Comput. Sci., Univ. of Rochester, Rochester, NY), Seungju Yeo (Dept. of Mech. Eng., Univ. of Rochester, Rochester, NY), Narges Mohammadi, Mujdat Cetin (Dept. of Electr. and Comput. Eng., Univ. of Rochester, Rochester, NY), and Diane Dalecki (Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY)

Noninvasive *in situ* temperature measurement (thermometry) is an attractive means of mapping the region of thermal damage during thermal ablation treatments. Existing methods of ultrasound thermometry are ineffective beyond 50 °C due to multiple physical limitations, including a non-monotonic relationship between temperature and sound speed that reaches a plateau around 60 °C, tissue phase transitions and deformation. An ultrasound-based technology that can monitor treatment over the entire therapeutic temperature range is desirable clinically. This paper describes a deep learning-based approach that uses thermometry data from the periphery of the heating zone (where temperatures are less than 50 °C) to infer temperature throughout the treatment zone. Spatiotemporal 2D temperature maps from 3–12 s HIFU heating exposures (in 0.5 s increments) were generated (using COMSOL) with a subset used for training the network and the rest for testing. Peripheral temperature values (excluding the first 5 mm closest to the axial focus), scalar time values, and a binary flag indicating heating/cooling were inputs to the network, and the temperature profile axially through the HIFU focus was predicted. The temperature prediction accuracy was better than 0.5 °C during heating and cooling. This paper will also address robustness to noise in the input temperature measurements and discuss future directions including experiments with ultrasound backscatter data and strategies to explicitly incorporate heat diffusion physics into the learning paradigm.

**3:20**

**2pBAa8. Contrast free super-resolution microvessel imaging based on deep learning.** Qi You (Beckman Inst., Univ. of Illinois Urbana-Champaign, 405 N. Mathews Ave. M/C 251, Urbana, IL 61801, Urbana, IL 61801, qiyou3@illinois.edu), Matthew R. Lowerison, YiRang Shin, Nathiya V. ChandraSekaran, Xi Chen, Zhijie Dong, Daniel A. Llano, Mark Anastasio, and Pengfei Song (Beckman Inst., Univ. of Illinois Urbana-Champaign, Urbana, IL)

Super-resolution microvascular imaging (SRMI) based on ultrasound localization microscopy (ULM) breaks the diffraction limit of conventional ultrasound by exploiting intravascular microbubbles. However, microbubble injection makes ULM not entirely noninvasive and less practical. On the other hand, power Doppler (PD) is contrast-free and widely available in clinic, but PD's spatial resolution falls short of revealing tissue microvasculature. In this study, we propose a deep learning-based, contrast-free SRMI method to improve the spatial resolution of PD. We designed a modified U-Net structure with 4 layers in depth with 64 to 512 channels. Ultrafast ultrasound data (1000 Hz frame rate) acquired from 16 mice were made into the 4080 data blocks used for training. Reconstructed ULM images using 32 s acquisition served as the labelled ground truth. Contrast-free ultrasound data using 0.4 s acquisition were used as the network input. The results have shown that the proposed DL-based method can significantly improve the spatial resolution of contrast-free power Doppler without creating fake vessels. The full width half-maximum (FWHM) of the cross-sectional profile of a microvessel was improved from 66 to 24 $\mu$m. Robust generalizability of the proposed DL method was also demonstrated on various tissues and organs that were absent in training.

## Session 2pBAb

## Biomedical Acoustics: Look How Big You've Gotten! A Story of Droplets and Ultrasound II

Kevin J. Haworth, Cochair
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Mario L. Fabiilli, Cochair
*Univ. of Michigan, 1301 Catherine St., 3226A Med. Sci. I, Ann Arbor, MI 48109*

### *Invited Papers*



**1:15**

**2pBAb1. Ultrasound-activated perfluorocarbon nanodroplets in ophthalmology.** Ronald H. Silverman (Ophthalmology, Columbia Univ. Irving Medical Ctr., 635 W 165th St., Room 711, New York, NY 10032, rs3072@cumc.columbia.edu), Raksha Urs, Gulgun Tezel (Ophthalmology, Columbia Univ. Irving Medical Ctr., New York, NY), Mark Burgess (Riverside Res., New York, NY), and Jeffrey Ketterling (Radilolgy, Weill Cornell Medical Ctr., New York, NY)

Glaucoma, a leading cause of blindness, can result from increased intraocular pressure resulting from elevated outflow resistance. In this preliminary study, we consider the potential of ultrasound-activated perfluorocarbon nanodroplets (NDs) to increase the permeability of the outflow pathway. 100nm diameter NDs with a perfluoropentane core and lipid shell were introduced into the anterior chamber (AC) of *ex vivo* pig and *in vivo* rat eyes. Imaging (18 and 28 MHz) was performed with a weakly focused (F = 2.9) beam. Activation was induced by increasing the array sub-aperture to tighten the focus to F = 0.62 centrally. Images showed NDs to be distributed widely within the AC, from which they could enter the outflow pathway by virtue of their small dimension. At 28 MHz, NDs were activated at a peak negative pressure of 5 MPa, corresponding to a mechanical index of 1.8.Two days after treatment, rat eyes showed no sign of inflammation, but a few small gas bubbles were present in the AC. In this application, NDs are advantageous as cavitation nuclei with respect to conventional microbubble contrast agents because of their higher density and smaller diameter ($<1/10^{th}$ that of microbubble agents), which facilitate entry into the outflow channels. Future preclinical *in vivo* studies will assess the effect of treatment on IOP and outflow facility.

**1:40**

**2pBAb2. Application of low-boiling point phase-change nanoemulsions for sonothrombolysis.** Jinwook Kim, Kathlyne Bautista, Ryan Deruiter (Biomedical Eng., UNC Chapel Hill, Chapel Hill, NC), Bohua Zhang, Huaiyu Wu, Leela Goel (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Xiaoning JIang (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), and Paul A. Dayton (Biomedical Eng., UNC Chapel Hill, CB 7575, Chapel Hill, NC 27599, padayton@email.unc.edu)

Intravascular thrombus remains a major challenge in human health, resulting in substantial morbidity and mortality for our aging population. Current treatment approaches are focused on antithrombotic treatment or mechanical thrombectomy, both of which have challenges with efficacy and side effects. Ultrasound-enhanced sonothrombolysis, including that enhanced by microbubbles, has been studied for decades as potential alternative therapy. However, clinical efficacy of ultrasound driven thrombolysis has remained elusive. One hypothesis has been that for contracted venous clots, the dense clot matrix resists permeation of microbubble cavitation agents, and superficial cavitation alone is slow and weakly effective. We hypothesized that sub-micron phase change nanoemulsions could penetrate a clot and begin lysis from the interior. Using video microscopy during clot treatment with nanoemulsions and ultrasound, we observed cavitation mediated clot erosion initiating internally to the clot. Cavitation detection imaging shows this same trend of internal cavitation with nanoemulsions, whereas traditional microbubbles cavitate only near the clot surface. Lysis rate studies demonstrate statistically faster lysis of contracted clots with phase change nanoemulsions, compared to 5x larger microbubbles of the same formulation. We conclude that phase change nanoemulsions may play a substantial role in reinvigorating sonothrombolysis potential.

**2:05**

**2pBAb3. Fibrin-targeted phase shift microbubbles outperform fibrin-targeted microbubbles for the treatment of microvascular obstruction.** John J. Pacella (Medicine, Univ. of Pittsburgh, 3500 Terrace St., Pittsburgh, PA 15213, pacellajj@upmc.edu)

Introduction: Mortality from AMI has decreased in recent years; however, microvascular obstruction (MVO) occurs frequently, ultimately limits myocardial salvage, is associated with rising post-AMI CHF, and has no definitive therapy. To address this urgent unmet need, we have developed an image-guided acute therapy, termed "sonoreperfusion" (SRP), that resolves MVO via ultrasound-targeted microbubble cavitation (UTMC). We previously used standard MBs with fibrin-targeting and demonstrated enhanced reperfusion compared to standard MBs with non-targeting. However, phase shift microbubbles are much smaller than standard MBs. Thus, we compared the SRP efficacies of fibrin-targeted phase shift microbubbles (FTPSMB) to standard size fibrin-targeted microbubbles (FTMBs).

Methods: MVO of the rat hindlimb was created by injecting microthrombi into the left femoral artery under contrast-enhanced ultrasound (CEUS) guidance. DEFINITY® MBs were infused (2 ml/h) through the right external jugular vein for CEUS. Following 10 min of stable MVO, a transducer was positioned vertically above the hindlimb to deliver therapeutic US pulses during concomitant administration of FTMBs/FTPSMBs (3 ml/h). Results and conclusion: FTPSMB treatment resulted in a greater increase in the blood volume (dB) and flow rate (dB/sec) than FTMB after each 10-minute treatment session owing to their small size and more effective thrombus penetration. Studies to explore the underlying molecular mechanisms associated with SRP treatments are underway

## *Contributed Papers*

**2:30**

**2pBAb4. Low cost and low energy 3D volumetric histotripsy using nanodroplet vaporization.** Bar Glickstein (Bio Medical Eng., Tel Aviv Univ., Ramat Aviv, Tel Aviv 6997801, Israel, barg1@mail.tau.ac.il) and Tali Ilovitsh (Bio Medical Eng., Tel Aviv Univ., Tel Aviv, Israel)

Low pressure histotripsy may facilitate current treatments that require extremely high pressures. Here, a new technology platform for low-cost and low-energy 3D volumetric ultrasound histotripsy using nanodroplets was developed. The two-step approach involves the vaporization of the nanodroplets into gaseous microbubbles via a 1D rotating imaging probe (center frequency of 3.5 MHz). This transducer is situated within a therapeutic transducer that operates at a center frequency of 105 kHz. The therapeutic transducer is used to implode the vaporized nanodroplets and trigger potent mechanical effects in the surrounding tissues. Rotating the imaging transducer creates a circular vaporized ND shape, that upon their implosion generates a round shape lesion. In comparison, an elongated lesion shape is formed when using a standard 1D imaging transducer for ND activation. Initial optimization experiments were performed in tissue-mimicking phantoms, and *ex-vivo* chicken liver samples. Next, nanodroplet-mediated histotripsy was tested in a breast cancer tumor model in mice. The results confirm the generation of significant lesions and tumor tissue debulking compared to all control groups. Our approach facilitate the creation of large volume mechanical damage in tissues and solid tumors.

**2:45**

**2pBAb5. Nanoparticle-mediated histotripsy using dual-frequency histotripsy pulsing: Comparison of bubble-cloud characteristics and ablation efficiency.** Connor W. Edsall (Biomedical Eng. and Mech., Virginia Polytechnic Inst. and State Univ., 325 Stanger St., 340 Kelly Hall, Blacksburg, VA 24061, edsallcw@gmail.com), Laura Huynh (Mater. Sci. and Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), Yasemin Yuksel Durmaz, Waleed Mustafa (Biomed. Eng., Istanbul Medipol Univ., Istanbul, Turkey), and Eli Vlaisavljevich (Biomedical Eng. and Mech., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

Nanoparticle-mediated histotripsy (NMH) is a targeted ablation method using perfluorocarbon-filled nanoparticles to generate bubble-clouds at pressure levels (9–12 MPa) significantly below the histotripsy intrinsic threshold (>25 MPa). Prior studies have also shown a significant reduction in ablation efficiency compared to conventional histotripsy, likely from reduced bubble expansion and bubble-cloud density. Here, we investigate the bubble-cloud characteristics and ablation efficiency for NMH using dual-frequency pulsing. We hypothesize this method will increase ablation efficiency by increasing the bubble-cloud density and individual bubble expansion. High-speed optical imaging was used to characterize the cavitation threshold, cloud dimensions, and bubble-density of bubble clouds generated in agarose tissue phantoms, with and without perfluorohexane-filled nanocones, exposed to single-cycle dual-frequency pulses using a 500 kHz–3 MHz array transducer (1:1 pressure ratio). Ablation efficiency was investigated using red blood cell phantoms. Results showed dual-frequency NMH predictably produced smaller, denser, and more well-confined bubble-clouds and increased ablation efficiency compared to previous single-frequency studies, with complete ablation of the focal volume observed within 2000 pulses. This study demonstrates the potential of enhancing NMH ablation efficiency with dual-frequency pulsing and highlights the need for further studies to optimize NMH pulsing parameters for future clinical therapy development.

**3:00**

**2pBAb6. Assessing the effects of droplet diameter on oxygen scavenging in the presence of hemoglobin via acoustic droplet vaporization.** Kateryna Stone (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3950, Cincinnati, OH 45267, matiaska@mail.uc.edu), Demetria Fischesser (Personal Health Care R&D, Procter and Gamble, Cincinnati, OH), Nour Al Rifai, Rachel P. Benton (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Anwesa Basa (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, Cincinnati, OH), and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Reducing bioavailable oxygen ($O_2$) during reperfusion following an ischemic event can reduce cell death. $O_2$ in whole blood (WB) is either bound by hemoglobin (measured as $O_2$ saturation, $sO_2$) or dissolved in plasma (measured as partial pressure of $O_2$, $pO_2$). 98% of $O_2$ is bound by hemoglobin *in-vivo*. Acoustic droplet vaporization (ADV) scavenges dissolved $O_2$, but its ability to scavenge hemoglobin-bound $O_2$ is unknown. The goal of this study was to assess how varying liquid droplet diameter affects $O_2$ scavenging from bovine WB. Anticoagulated WB was prepared for *in-vitro* flow phantom experiments to mimic physiological arterial blood. 0.82 $\mu$m diameter droplets were manufactured using high-shear pressure homogenization. 1.34 $\mu$m and 7.02 $\mu$m diameter droplets were manufactured using microfluidics. Droplets ($5\times10^{-4}$ ml/ml) were infused through an EkoSonic catheter and exposed to ultrasound. $sO_2$ and $pO_2$ were measured with a blood gas analyzer. A post-ADV reduction in $pO_2$ of $5.8 \pm 2.6$ mmHg, $16.4 \pm 4.9$ mmHg, and $30.7 \pm 4.5$ mmHg was observed with 0.82 $\mu$m, 1.34 $\mu$m, and 7.02 $\mu$m droplets, respectively. No change in $sO_2$ was observed with 0.82 $\mu$m and 1.34 $\mu$m droplets, but a $sO_2$ decrease of $5.8 \pm 1.5\%$ was seen with 7.02 $\mu$m droplets.

## Session 2pBAc

## Biomedical Acoustics: Tribute to Fellows and Award Winners of the BATC: 2022

Kenneth B. Bader, Chair
*Univ. of Chicago, 5835 South Cottage Grove Ave., Dept. of Radiology, MC 2026, Q301B, Chicago, IL 60637*

### *Invited Papers*

**3:40**

**2pBAc1. Backscatter techniques for ultrasonic bone assessment.** Brent K. Hoffmeister (Phys., Rhodes College, 2000 N Parkway, Memphis, TN 38112, hoffmeister@rhodes.edu)

Osteoporosis is a bone disease that affects hundreds of millions of people worldwide. The name osteoporosis literally means "porous bone" reflecting the changes in bone density and microstructure that lead to increased risk of fracture. Ultrasonic backscatter techniques have been proposed as a way to detect these changes. Backscatter measurements are performed by propagating ultrasonic pulses into bone and receiving signals returned from the porous interior of the tissue. Numerous backscatter techniques have been developed over the last two decades for ultrasonic bone assessment. This presentation reviews many of those techniques and compares their relative performance in a recent *in vivo* study using measurements at the femoral neck, a common location for osteoporotic fracture.

**4:00**

**2pBAc2. My contributions to ultrasound metrology.** Subha Maruvada (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov)

My work in ultrasound metrology has centered around developing pre-clinical testing methods for therapeutic ultrasound devices. These test methods include radiation force balance systems and methods specifically for high-intensity therapeutic ultrasound, time-delay spectrometry methods for measuring hydrophone sensitivity, acoustic characterization of tissue-mimicking material that has been approved as a medical device development tool and a comparison protocol of computational and experimental acoustic characterization for therapeutic ultrasound devices. The purpose of this work at the FDA is to develop bench testing methods to ensure proper review of medical ultrasound devices to evaluate device safety and performance. The test instruments and methods developed have helped promote the development and recognition of voluntary consensus standards, as well as FDA industry guidance, that can provide the basis for least-burdensome data collection requirements by manufacturers and allow the FDA to assess device safety and effectiveness in a more consistent, methodical and scientifically rigorous manner.

**4:20**

**2pBAc3. A mechanistic journey in shock wave lithotripsy research.** Pei Zhong (Thomas Lord Dept. of Mech. Eng. and Materials Sci., Duke Univ., Durham, NC, pzhong@duke.edu)

As a remarkable engineering innovation, shock wave lithotripsy (SWL) has revolutionized the treatment of kidney stone disease since 1980s with worldwide clinical acceptance. Over the past three decades, the long journey to understand the transient interaction of focused shock waves with kidney stones and soft tissues has propelled fundamental research at multiple fronts, including (1) the generation, propagation, and interaction of various stress waves inside the bulk stone materials, and surface acoustic waves associated with Mach stem formation at the stone boundary; (2) dynamic fracture and comminution processes in relation to lithotripter field characteristics; and (3) cavitation-induced stone surface erosion and vascular injury. In this talk, I will present illustrative examples to highlight chronologically the progressive understanding regarding the mechanisms of stone comminution and tissue injury, and how the acquired knowledge has influenced the technology evolution and clinical practice in SWL. I will also discuss how the fundamental concepts and techniques of SWL have been widely utilized in other therapeutic ultrasound and lithotripsy applications; and future perspectives, especially related to 3D cavitation detection and monitoring, as well as multi-spark shock wave generator development.

**4:40**

**2pBAc4. Advancing ultrasound and microbubble-mediated drug delivery in the brain and spinal cord.** Meaghan O'Reilly (Phys. Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Rm C736a, Toronto, Ontario M4N3M5, Canada, moreilly@sri.utoronto.ca)

When combined with intravenously administered microbubbles, ultrasound can transiently open the so-called 'blood-brain barrier' (BBB) and 'blood-spinal cord barrier' (BSCB), resulting in increased vascular permeability to enable delivery of therapeutics to the central nervous system (CNS). Thus this technique holds transformative potential for patients suffering from CNS disorders. Achieving successful opening of these barriers while mitigating lasting tissue changes requires control over the transmitted sound, as well as knowledge of the interaction of the transmitted sound field with the microbubbles. In this talk I will review contributions made to monitoring and controlling BBB opening, as well as efforts by my group to translate this intervention to the spinal cord. Recent preclinical *in vivo* findings and *ex vivo* results in human vertebrae will be reported.

# Session 2pCA

## Computational Acoustics and Biomedical Acoustics: Numerical Approaches for Complex Media Geometries II

Zhongquan Charlie Zheng, Cochair
*Mech. and Aerosp. Eng., Utah State Univ., 4130 Old Main Hill, Logan, UT 84322*

D. Keith Wilson, Cochair
*Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., U.S Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290*

### *Invited Papers*

**1:00**

**2pCA1. A numerical analysis of the importance of local second-order nonlinear phenomena on acoustic wave propagation in complicated media.** Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil) and Kyle G. Dunn (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., Hanover, NH)

Efforts to reduce the full second-order nonlinear equations of motion down to a single nonlinear wave equation are frustrated by the presence of the Lagrangian density, a combination of terms that involve both the particle velocity and the acoustic pressure. The Lagrangian density is usually discarded, leading to the much simpler Westervelt equation, with the argument that it is associated with local nonlinear phenomena and that cumulative nonlinear phenomena will dominate its contribution over distance. However, this argument may not be valid for waves propagating through heterogeneous and complicated environments, such as a metamaterial. In order to understand whether the terms associated with local nonlinear phenomena are important in these types of environments, we have implemented a finite-difference time-domain method for two sets of equations: the full second-order equations and a set of equations having neglected the Lagrangian density. In this paper we will present propagation predictions for heterogeneous environments using both sets of equations in order to quantify the error obtained by neglecting the local nonlinear phenomena.

**1:20**

**2pCA2. A time domain boundary element method for acoustic scattering by lined surfaces in a subsonic uniform mean flow.** Fang Hu (Mathematics and Statistics, Old Dominion Univ., 5115 Hampton Blvd., Dept. of Math & Stat, Norfolk, VA 23529, fhu@odu.edu) and Douglas M. Nark (NASA Langley Res. Ctr., Hamptom, VA)

A time domain boundary integral equation with Burton-Miller reformulation is presented for acoustic scattering by surfaces with liners in a uniform mean flow. The Ingard-Myers impedance boundary condition is implemented using a broadband multipole impedance model which is in turn converted into time domain differential equations to augment the boundary integral equation. The coupled integral-differential equations are solved numerically by a March-On-in-Time (MOT) scheme. While the Ingard-Myers condition is known to support Kelvin-Helmholtz instability due to its use of a vortex sheet interface between the flow and the lined surface, it is found that by neglecting a second-order derivative term in the current time domain impedance boundary condition formulation, the instability can be effectively avoided in computation. Proposed formulation and implementation are validated with numerical examples. Moreover, a minimization procedure for finding the poles and coefficients of the broadband multiple impedance expansion is formulated by which, unlike the commonly used vector-fitting method, passivity of the model is ensured. Numerical tests show the proposed minimization approach is effective for modeling liners that are commonly used in aeroacoustic applications.

**1:40**

**2pCA3. Immersed-boundary time-domain simulation of acoustic pulse scattering from a single or multiple gas bubble(s) of various shapes.** Jiacheng Hou (Mech. and Aerosp. Eng., Utah State Univ., Logan, UT), Zhongquan Charlie Zheng (Mech. and Aerosp. Eng., Utah State Univ., 4130 Old Main Hill, Logan, UT 84322, zzheng@usu.edu), and John S. Allen (Mech. Eng., Univ. of Hawaii Manoa, Honolulu, HI)

Acoustic scattering and resonances resulting from a point pulse on a single or multiple gas bubbles are simulated using a time-domain simulation. The time histories of scattering pressure and velocity, both outside and inside the bubbles, are obtained simultaneously with an immersed-boundary method implementation. The acoustic resonances of the bubbles are investigated for various bubble numbers, sizes, shapes and interior gas parameters. For several cases, the scattering and resonance behaviors are compared with the existing theoretical and experimental results.

**2:00**

**2pCA4. Numerical modeling of low-frequency scattering from targets buried under rough seafloors.** Roberto Sabatini (Embry-Riddle Aeronautical Univ., 1 Aerospace Blvd., Daytona Beach, FL 32114, sabatinr@erau.edu), Yan Pailhas (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy), Paul Cristini (Lab. de Mecanique et d'Acoustique, Marseille, France), and Angeliki Xenaki (Ctr. for Maritime Res. and Experimentation, La Spezia, Italy)

The acoustic backscattering of elastic bodies lying across the seafloor is a problem of great interest in many underwater applications, such as detecting and classifying buried objects. In this work, the backscattered wavefield of meter-sized targets is investigated numerically in the 1–50 kHz frequency band. The simulator solves the three-dimensional linearized equations of continuum mechanics. The computations are performed through a multi-GPU solver based on a multi-grid staggered finite-difference time-domain method (cf. Sabatini et al., 181st ASA Meeting, Seattle, Washington, 2021). The effects of the seafloor roughness and the target burial depth on the spectrum of the scattered field are more specifically analyzed. This study demonstrates the usefulness of direct numerical simulations of acoustic scattering from realistic elastic objects in complex underwater environments to improve the capabilities of sonar systems in detecting and identifying buried objects.

**2:15**

**2pCA5. An open-source GPU-accelerated application of the elastodynamic finite integration technique (EFIT) to three-dimensional wave propagation and scattering in anisotropic materials of arbitrary geometries.** Seth Golembeski (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, sgole1@unh.newhaven.edu) and Eric A. Dieckman (Mech. Eng., Univ. of New Haven, West Haven, CT)

As ultrasonic testing (UT) becomes increasingly widespread, accurate numerical simulations of the complex wave propagation behavior in solids become increasingly useful. This is especially true for complex geometries and material properties created by new manufacturing processes such as those use to create Additively Manufactured Metals (AMMs). Here, we present an open-source fully-anisotropic Elastodynamic Finite Integration (EFIT) implementation written in the Julia language which can be deployed to high-performance computers for large-scale parallel simulations. Results of benchmarks against existing isotropic implementations, published data, and collected measurements will be presented.

**2:30**

**2pCA6. Numerical exploration of acoustic dipole helical wave generation.** Grant Eastland (Test and Evaluation, Naval Undersea Warfare Ctr. Div., Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil) and Peter J. Curry (Naval Information Warfare Ctr. Atlantic, North Charleston, SC)

The focus of this talk is on acoustic helical waves, which are a type of wave that transmits phase information across a section of the plane linearly about the azimuth and can transfer angular momentum via the axial phase profile and have recently been applied to navigation and communications systems. A phased array of four crossed circular baffled transducers each producing a pressure field 90° out of phase adjacent to one another are used to create the pressure field. The far field pressure is determined utilizing a paraxial ray approximation for ideal sources but applied to the more complex conceptualization. The construction is effectively two perpendicular dipoles 90° out of phase and from superposition of the arrangement produce the helical wave. The mathematical model will be summarized and simulated. An interesting approximation of the total field is also given, relating the helical wave to a Bessel beam. Numerical examination of the arrangement is examined with visualizations of the pressure field, and discussion is given regarding transients created when the direction of helix changes.

**2:45**

**2pCA7. Proximity detection by a mobile phone using inaudible signal.** Yegor D. Sinelnikov (Acoust., Zebra Technol., 126 Liberty Ave., Port Jefferson, NY 11777, yegorasha@yahoo.com)

Mobile devices find widespread use in the factories, warehouses, hospitals. Audio features simplify people's life and improve their productivity. We demonstrate that inaudible audio produced and recorded by a mobile device has a potential for a facial feature recognition. An Android device was mounted on a moving stage near the head and torso simulator inside the listening room. The device produced an inaudible signal with frequency content above 10 kHz. A set of acoustic responses was recorded in several location relative to the head and torso. For each location the 2D images were recorded and were post processed into 3D reconstruction. Good correlation between the spectral representation of acoustic responses and 3D reconstruction was observed. The results demonstrate feasibility to compliment mobile phone security with acoustic biometric authentication.



**3:00**

**2pCA8. A fast solver framework for acoustic hybrid integral equations.** Meydan Kaplan (Ben-Gurion Univ. of the Negev, David Ben Gurion Blvd. 1, Be'er Sheva 8410501, Israel, meydank@post.bgu.ac.il) and Yaniv Brick (Ben-Gurion Univ. of the Negev, Be'er Sheva, Israel)

Reliable modeling of the scattering by acoustically large and geometrically complex objects can be achieved by means of subdomain-dependent problem formulation and a numerically rigorous solution. While the objects' inhomogeneity has driven the development of differential equation formulations and solvers, integral equation formulations, where the object's background is modeled via a Green's function, are advantageous for unbounded domains. In the hybrid integral equations approach (Usner *et al.*, 2006), the interaction of separate subdomains with external fields is described by pertinent integral equations. Their Galerkin discretization leads to a dense blocked stiffness matrix. The development of compressed representations of the matrix, which are necessary for the treatment of large systems, becomes non-trivial due to the multitude of integral equation kernels and the different geometrical and physical characteristic of the subdomains. As part of the development of a fast hybrid integral equation solver framework, we consider the case of objects composed of large inhomogeneous volumes, modeled as incompressible fluids, and of simplified solids, modeled via surface integral equations. A hybrid integral equation formulation is derived and solved numerically. The iterative solution is accelerated by employing the butterfly-compressed hierarchical representation of the stiffness matrix, recently used for acoustic volume integral equations (Kaplan, 2022).

# Session 2pEA

## Engineering Acoustics, Computational Acoustics, and Structural Acoustics and Vibration: Automotive Acoustics (Hybrid Session)

Xiaoshi Su, Cochair
*Toyota Res. Inst. of North America, 1555 Woodridge Ave., Ann Arbor, MI 48105*

Michael R. Haberman, Cochair
*Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

### *Invited Papers*

**1:00**

**2pEA1. Automotive cabin acoustics and audio.** Roger Shively (JJR Acoust., LLC, 8313 61st St. Court West, University Place, WA 98467, rshively@jjracoustics.com)

We spend most of our quality listening time either at home or in a car. Unlike a concert hall – where we spend the least amount of time listening, and on which there has been more than a century of research done – there has very little research done for the home, and even less for the car. And this is why we put so much effort into the science of sound in homes and even more so into the science of sound in cars. Mr. Shively will explore the evolving path to becoming an audio engineer or an NVH engineer: the acoustic path. And, he will address the question of why sound quality does really matter, and the overlapping engineering disciplines which pursue sound quality. He will discuss how the audio systems are expanding to integrate Active Noise Control (ANC), Advanced Driver-Assistance Systems (ADAS), and Acoustic Vehicle Alert Systems (AVAS) for quiet car sound design and autonomous driver alerts in the pursuit of sound quality and how that is achieved. He will review the classical challenges of acoustics in a car and the new challenges for OEM audio system design and how they are being met. And, finally, he will touch on the evolution of expectations in automotive audio in existing markets and, more critically, emerging markets.

**1:20**

**2pEA2. Performance evaluation of automotive Hands-free microphones using subjective and objective metrics and their correlation.** Yu Du (Harman Int., 30001 N. Cabot Dr., Novi, MI 48377, yu.du@harman.com)

Hands-free, or voice, microphone is a standard device used in cars to enable hands-free telephony. The speech intelligibility and quality (SI&SQ) associated with such a microphone represents the ultimate performance considered by users. However, the SI&SQ cannot be derived from microphone datasheets. Furthermore, the complex and constantly changing automotive acoustical environment under different driving conditions makes it very challenging to choose the proper hands-free microphone design(s) for certain vehicle model(s). To establish the relationship between microphone characteristics commonly described by directivities and frequency-responses with the SI&SQ perceived by drivers and passengers in automotive applications, a study is conducted using three common types of automotive hands-free microphones. Their SI&SQ are evaluated using both subjective and objective metrics described in ANSI S3.2-2009, S3.5-1997 and ITU-T Recommendation P.862.2. The correlation of results obtained from different evaluation metrics is analyzed. It is shown that the speech intelligibility index (SII) calculated objectively using ANSI S3.5-1997 correlates approximately linearly with the SI data obtained from the subjective S3.2-2009 method. With some mathematical mapping, a modified SII can be derived that also correlates well with the mean-opinion-score predicted by P.862.2. The newly proposed modified SII method can be an effective tool to guide automotive hands-free microphone designs.

**1:40**

**2pEA3. Automotive audio system evaluation over headphones based on the biaural vehicle impulse responses of different listening positions: A case study of a specific audio system.** Yukun Pei (Dept. of Music and Performing Arts Professions, New York Univ., 35 West 4th St., New York, NY 10012, pyken@foxmail.com), You Li (Dept. of Music and Performing Arts Professions, New York Univ., New York, NY), and Pablo Ripollés (Depart. of Psychol., New York Univ., New York, NY)

In recent years, car manufacturers have consistently upgraded the audio systems of their vehicles, with audio aficionados adding further modifications to them. The acoustics of the vehicle cabin and the sound effects of the audio systems have become one of the most important topics of the Research and Development Departments of manufacturers. For example, the selected vehicle in this experiment has implemented different spatial audio algorithms in its models to achieve a better listening experience. By capitalizing on impulse responses, binaural audio technology provides the opportunity and the flexibility to virtually generate the sound effects of a particular space without the requirement of physically being in that space. We used binaural technology and the vehicle to develop a standardized procedure for the evaluation of car audio systems. A perceptual listening test was integrated into this study to verify the procedure and to further evaluate this specific audio system.

**2:00**

**2pEA4. Electric vehicle warning sounds: On road and immersive audio detection results for 20 subjects.** Michael Roan (ME, Penn State, 1430 Linn St., State College, PA 16803, mjr110@psu.edu), Luke Neurauter (VTTI, Blacksburg, VA), Michael Beard (ME, Virginia Tech, Blacksburg, VA), and Marty Miller (VTTI, Blacksburg, VA)

The number of electric vehicles on the road is rapidly increasing. Due to the decreased sound produced by these vehicles at low speeds there is significant concern that pedestrians and bicyclists will be at increased risk of vehicle collisions. Because of this potential for collisions, govenrnemts have institutes regulations governing additive vehicle warning sounds for electric vehicles. This research presents results on the detectability of six electric vehicle acoustic warning sounds using two different hardware systems. Detectability was initially by on-road participant tests and replicated in an immersive reality lab. Results were analyzed through both mean detection distances and probability of detection. This research aims to verify the lab environment as it will allow for a broader range of potential test scenario's, more repeatable tests, and faster test sessions. Along with pedestrian drive by tests, experiments were conducted to evaluate stationary vehicle acoustics, 10 and 20 km/h drive by acoustics, and interior acoustic impact of each warning sound.

## *Contributed Papers*

**2:20**

**2pEA5. Investigation of indoor soundscape indicators according to the types of occupants' activity in autonomous vehicles.** Jin Yong Jeon (Architectural Eng., Hanyang Univ., Dept. of Architectural Eng., Hanyang Univ., Seoul 04763, South Korea, jyjeon@hanyang.ac.kr), Haram Lee (Architectural Eng., Hanyang Univ., Seoul, South Korea), Juin Kim (Automotive, Res. & Development Div., Hyundai Motor Group, Hwaseong, South Korea), and Dongchul Park (Automotive, Res. & Development Div., Hyundai Motor Group, Hwaseong, South Korea)

Electric and autonomous vehicles have brought about a paradigm shift in interior space design and are now replacing their functions as spaces for living, work and leisure activities. Therefore, in order to design the optimal sound environment for the interior space of the vehicle, it is necessary to investigate the interior soundscape of the vehicle based on the type of occupant's activity and the purpose and function of the space. However, although the soundscape concept has been applied to the design of urban and natural spaces and large indoor spaces such as open plan offices, there is no study on the interior space of vehicles. In this study, soundscape indicators were set according to the activity patterns of autonomous vehicle occupants. Through the result, it is possible to derive vehicle interior soundscape design factors based on the evaluation of each type of sound environment presented.

**2:35**

**2pEA6. Investigation of the singularity under non-proportional damping in an analytical solution of brake squeal.** Caiyu Xie (The Univ. of Queensland, St. Lucia, Brisbane, Queensland 4072, Australia, caiyu.xie@uqconnect.edu.au) and Paul A. Meehan (The Univ. of Queensland, Brisbane, Queensland, Australia)

This research investigates a singularity in a published analytical solution of brake squeal due to mode coupling. Brake squeal is an annoying tonal noise that results from the slowing of a vehicle with friction brakes. Recent research has identified an efficient analytical prediction for occurrence, growth, limit cycle amplitude and mitigation of brake squeal noise. The analytical solution predicts squeal accurately, except at a singularity associated with large non-proportional damping and repeated roots. Physically this occurs at the exact transition to stiffness mode coupling, ie. complex roots ($\Delta=0$). By taking the limit of the eigenvalue expressions at $\Delta=0$, it is found that the analytical eigenvalues approach infinity. The singularity occurs because modal mass matrix goes to zero at $\Delta=0$. The eigenvalues at $\Delta=0$ can be approximated by averaging the eigenvalues that are located on both sides of the critical friction coefficient (the average of eigenvalues at $\mu_{crit}$-$\Delta\mu$ and $\mu_{crit}+\Delta\mu$). The approximations are closest to the numerical solution when $\Delta\mu$ is at the critical points of the Im($\lambda_{avg}$) vs $\Delta\mu$ plot. In summary, the analytical solution under non-proportional damping is undefined at $\Delta=0$ because of the zero modal mass matrix. Moreover, the eigenvalues at the point where singularity occurs can be approximated by taking the average of the eigenvalues located on both sides of the critical friction coefficient. It is shown that reasonable approximations can be obtained at the critical points of the Im($\lambda_{avg}$) vs $\Delta\mu$ plot.

**2:50**

**2pEA7. Active sound quality control of vehicle interior noise using a dual sampling-rate active noise equalization algorithm.** Shuai Zhang (School of Automotive Eng., Tongji Univ., 4800 Cao'an Rd., Jiading District, Shanghai, 201804, P. R. China, Shanghai 201804, China, chriszhang0903@163.com), Lijun Zhang, and Dejian Meng (School of Automotive Eng., Tongji Univ., Shanghai, China)

In-vehicle active sound quality control (ASQC) system is an advanced application of active noise control (ANC) technology, which always suppresses the interior sound pressure or improves the psychoacoustic features. However, the connection between the sound quality control state and the vehicle's working state is normally ignored, and the passenger's sound perception typically expects a linear relationship between the vehicle speed and sound loudness. This paper presents a dual sampling-rate active noise equalization algorithm. The low sampling-rate signal is used to improve system computing efficiency, the high sampling-rate signal is used to improve the ASQC effect, and their conversion communication uses a hold (low to high) and a sampler (high to low). The data relating to the ASQC system was collected, and some sound quality evaluations were conducted for determining two appropriate sampling-rate values. Combining a genetic algorithm and the corresponding ASQC simulation system, the algorithmic optimal convergence coefficients and gain coefficients were further determined at different engine speeds. The verification results of ASQC using this proposed algorithm show that the loudness of interior noise is effectively suppressed; the nonlinear index is reduced to 1.37, improving by 37% relative to the original noise and by 20% compared to the ANC system.

# Session 2pID

## Interdisciplinary: Graduate Programs in Acoustics Poster Session

Zane T. Rusk, Cochair
*The Pennsylvania State Univ., 104 Eng. Unit A, Univ. Park, PA 16802*

Miad Al Mursaline, Cochair
*Mech. Eng./ Appl. Ocean Phys. & Eng., Massachusetts Inst. of Technol./Woods Hole Oceanogr. Inst., 70 Pacific St., Cambridge, MA 02139*

All posters will be on display from 1:00 p.m. to 3:00 p.m. Authors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:00 p.m. Authors of even-numbered papers will be at their posters from 2:00 p.m. to 3:00 p.m.

### *Invited Papers*

**2pID1. Graduate studies in acoustics at Northwestern University.** Ann Bradlow (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu), Jennifer Cole, and Matthew Goldrick (Linguist., Northwestern Univ., Evanston, IL)

Northwestern University has a vibrant and interdisciplinary community of acousticians. Of the 13 ASA technical areas, 3 have strong representation at Northwestern: Speech Communication, Psychological and Physiological Acoustics, and Musical Acoustics. Sound-related work is conducted across a wide range of departments including Linguistics (in the Weinberg College of Arts and Sciences), Communication Sciences & Disorders, and Radio/Television/Film (both in the School of Communication), Electrical Engineering & Computer Science (in the McCormick School of Engineering), Music Theory & Cognition (in the Bienen School of Music), and Otolaryngology (in the Feinberg School of Medicine). In addition, *The Knowles Hearing Center* involves researchers and labs across the university dedicated to the prevention, diagnosis and treatment of hearing disorders. Acoustics research topics across the university include speech perception and production across the lifespan and across languages, dialects and socio-indexical properties of speech; sound art and design; social and cultural history of the sonic world; machine processing of music; musical communication; auditory perceptual learning; auditory aspects of conditions such as concussion, HIV, and autism; neurophysiology of hearing; and the cellular, molecular, and genetic bases of hearing function. We invite you to visit our poster to learn more about the "sonic boom" at Northwestern University!

**2pID2. Graduate programs in physical, engineering, and underwater acoustics at the Naval Postgraduate School.** Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu) and Kay L. Gemba (Phys. Dept., Naval Postgrad. School, Monterey, CA)

The Departments of Physics and of Electrical and Computer Engineering at the Naval Postgraduate School offer graduate programs in acoustics leading to MS and PhD degrees in applied physics and engineering acoustics. Engineering acoustics degrees can be completed in either traditional or distance learning modes. The departments also offer stand-alone academic certificate programs in fundamentals of engineering acoustics, underwater acoustics, and sonar system applications, with a set of three certificates leading to a MS degree in engineering acoustics. MS and PhD programs are interdisciplinary, with courses and laboratory work drawn principally from the fields of physics and electrical engineering. Subjects covered include waves and oscillations; fundamentals of physical and structural acoustics; the generation, propagation, and reception of sound in the ocean; civilian and military applications of sonar systems; and acoustic signal processing. Topics of recent theses and dissertations include development and field testing of novel sensors for atmospheric and ocean acoustics, modeling and measurements of ambient noise and sound propagation in the ocean, sound scattering in underwater waveguides, acoustic vector sensors and vector field properties, acoustic communications, noise interferometry, time reversal in acoustics, geo-acoustic inversion, acoustic remote sensing of the ocean, and acoustics of autonomous underwater and aerial vehicles.

**2pID3. Graduate acoustics at Brigham Young University.** Matthew S. Allen (Mech. Eng., Brigham Young Univ., Provo, UT), Brian E. Anderson (Phys. & Astron., Brigham Young Univ., Provo, UT), Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT), Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT), Tracianne B. Neilsen (Phys. & Astron., Brigham Young Univ., N251 ESC, Provo, UT 84602, tbn@byu.edu), Micah Shepherd (Phys. & Astron., Brigham Young Univ., State College, PA), and Scott D. Sommerfeldt (Phys. & Astron., Brigham Young Univ., Provo, UT)

Graduate studies in acoustics at Brigham Young University prepare students for industry, research, and academia by complementing in-depth coursework with publishable research. Coursework provides a solid foundation in core acoustical principles and practices and measurement skills, including a strong foundation in experimental techniques and technical writing. Labs across the curriculum cover calibration, directivity, scattering, absorption, laser Doppler vibrometry, experimental methods for dynamic structures, lumped-element mechanical systems, equivalent circuit modeling, arrays, filters, room acoustics, active noise control, and near-field acoustical

holography. Recent thesis and dissertation topics include active noise control, directivity, room acoustics, energy-based acoustics, time reversal, nondestructive evaluation, vibration and acoustics of aerospace vehicles, biomedical applications, flow-based acoustics, voice production, aeroacoustics, sound propagation modeling, nonlinear propagation, high-amplitude noise analyses, machine and deep learning applied to ambient noise level prediction, crowd noise interpretation, and underwater acoustic source localization, and ocean environment classification. Graduate students are expected to present research at professional meetings and publish in peer-reviewed acoustics journals. Additionally, graduate students often serve as peer mentors to undergraduate students on related projects and may participate in field experiments to gain additional experience.@BYUAcoustics

**2pID4. Graduate education and research in architectural acoustics at Rensselaer Polytechnic Institute.** Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180, xiangn@rpi.edu) and Jonas Braasch (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

The Graduate Program in Architectural Acoustics has been constantly advanced from its inception in 1998 with an ambitious mission of educating future experts and leaders in architectural acoustics, due to the rapid pace of change in the fields of architectural-, physical-, and psycho-acoustics, and noise-control engineering. Since years the program's pedagogy using "STEM" (science, technology, engineering, and mathematics) methods has been proven to be effective and productive, including intensive, integrative hands-on experimental components that integrate architectural acoustics theory and practice. The graduate program has recruited graduate students from a variety of disciplines including individuals with B.S., B.Arch., or B.A. degrees in Mathematics, Physics, Engineering, Architecture, Electronic Media, Sound Recording, Music and related fields. Graduate students under this pedagogy and research environment have been succeed in the rapidly changing field. RPI's Graduate Program in Architectural Acoustics has since graduated more than 120 graduates with both M.S. and Ph.D. degrees. Under the guidance of the faculty members they have also actively contributed to the program's research in architectural acoustics, communication acoustics, psycho-acoustics, signal processing in acoustics as well as our scientific exploration at the intersection of cutting edge research and traditional architecture/music culture. This paper illuminates the evolution and growth of the Graduate Program.

**2pID5. Acoustics and cognitive psychology at the University at Buffalo, the State University of New York.** Micheal Dent (Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 142260, mdent@buffalo.edu), Federica Bulgarelli, Andres Buxó-Lugo, Christopher McNorgan (Univ. at Buffalo, SUNY, Buffalo, NY), Eduardo Mercado (Univ. at Buffalo, SUNY, Clarence, NY), and Peter Pfordresher (Univ. at Buffalo, SUNY, Buffalo, NY)

The Psychology Department at the University at Buffalo, SUNY offers mentorship-based, research-intensive Masters' and Doctoral degrees in Cognitive and Behavioral Neuroscience Areas of Psychology. Faculty specialize in audition, vocal production, psycholinguistics, bioacoustics, language development, attention, music cognition, and learning. Specific expertise includes auditory and vocal processing and multisensory integration across the lifespan. Study populations include mice, whales, and human children and adults, including special populations such as musicians, bilinguals, and people with conditions like aphasia, autism, dyslexia and attention deficit hyperactivity disorder. Methodologies include, but are not limited to, eye tracking, functional magnetic resonance imaging, motion capture, animal psychophysics, electroencephalograms, and surface electromyography. The program is collaborative, collegial, and supportive, and the graduate student stipends are competitive in an area of the country with a low cost of living. Students train with other students and faculty to become experts in their chosen area of study. The combination of academic and research training allows students to thrive and leave the program prepared for independent careers in academia or applied settings such as industry. See: https://arts-sciences.buffalo.edu/psychology/graduate/overview.html for more information about the program.

**2pID6. Graduate education in acoustic engineering, transduction, and signal processing University of Massachusetts Dartmouth.** David A. Brown (ECE, Univ. Massachusetts Dartmouth, 151 Martine St., Suite 123, Fall River, MA 027230000, dbrown@umassd.edu), Paul J. Gendron, and John R. Buck (ECE, Uniiv. Massachusetts Dartmouth, Dartmouth, MA)

The University of Massachusetts Dartmouth has an established graduate program of study with a concentration in Applied Acoustics leading to the M.S. and Ph.D. degree in Electrical Engineering. The program offers courses and research opportunities in the area of electroacoustic transduction, underwater acoustics, and signal processing. Courses include the Fundamentals of Acoustics, Random Signals, Underwater Acoustics, Introduction to Transducers, Electroacoustic Transduction, Medical Ultrasonics, Digital Signal Processing, Detection Theory, and Estimation Theory. The ECE department established the university's indoor underwater acoustic test and calibration facility which is one of the largest academic facilities supporting undergraduate and graduate thesis and sponsored research. The department has collaborations with many marine acoustic related companies including nearby Naval Undersea Warfare Center in Newport, RI and Woods Hole Oceanographic Institute in Cape Cod, MA. The presentation will highlight recent theses and dissertations, course offerings, and industry and government collaborations that support acoustical engineering, transduction, and signal processing.

**2pID7. The acoustics program at Georgia Tech.** Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu), Chengzhi Shi (GWW School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Costas Arvanitis (School of Mech. Eng., Dept. of Biomedical Eng., Georgia Inst. of Technol., Atlanta, GA), Julien Meaud (Georgia Inst. of Technol., Atlanta, GA), F Levent Degertekin, Alper Erturk, Michael Leamy (GWW School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Nico Declercq (GWW School of Mech. Eng., Georgia Inst. of Technol., Metz, France)

The acoustics education program at the Georgia Institute of Technology (Georgia Tech) is run by members of the Acoustics and Dynamics research area group from the School of Mechanical Engineering. We will briefly review the scope of this program in terms of education and research activities

**2pID8. Graduate studies in acoustics at the University of Nebraska – Lincoln within the College of Engineering.** Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S 67th St, Omaha, NE 68182-0816, lwang4@unl.edu), Erica E. Ryherd (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE), and Jinying Zhu (Civil and Environ. Eng., Univ. of Nebraska - Lincoln, Omaha, NE)

Those interested in pursuing graduate studies and conducting research in acoustics are invited to learn more about opportunities at the University of Nebraska – Lincoln (UNL) within the College of Engineering. Dr. Lily Wang and Dr. Erica Ryherd work on architectural acoustics and noise control topics (http://nebraskaacousticsgroup.org ) within the Durham School of Architectural Engineering and Construction, based at UNL's Scott Campus in Omaha. Dr. Jinying Zhu in Civil and Environmental Engineering (also based on UNL's Scott Campus in Omaha) is active in structural acoustics, using ultrasonic waves for concrete evaluation (https://engineering.unl.edu/cee/faculty/jinying-zhu/). Dr. Joseph Turner in Mechanical and Materials Engineering (based at UNL's City Campus in Lincoln) focuses on ultrasound propagation through complex media for quantitative characterization of materials/microstructure (http://quisp.unl.edu ). This poster presents the graduate-level acoustics courses and lab facilities at UNL within the College of Engineering, and highlights the research interests and achievements of our faculty, graduates, and students. Extracurricular experiences are available through an Acoustical Society of America student chapter based on the Scott Campus and collaborations with Boys Town National Research Hospital.

**2pID9. Graduate acoustics education in the Cockrell School of Engineering at The University of Texas at Austin.** Megan Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Neal A. Hall (Electr. and Comput. Eng., Univ. of Texas at Austin, Austin, TX), Mark F. Hamilton (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Tyrone M. Porter (Biomedical Eng. Dept., The Univ. of Texas at Austin, TX), and Preston S. Wilson (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

While graduate study in acoustics takes place in several colleges and schools at The University of Texas at Austin (UT Austin), including Communication, Fine Arts, Geosciences, and Natural Sciences, this poster focuses on the acoustics program in Engineering. The core of this program resides in the Departments of Mechanical Engineering (ME) and Electrical and Computer Engineering (ECE). Acoustics faculty in each department supervise graduate students in both departments. One undergraduate and nine graduate acoustics courses are taught in ME and ECE. Instructors for these courses include staff at Applied Research Laboratories at UT Austin, where many of the graduate students have research assistantships. The undergraduate course, taught every fall, begins with basic physical acoustics and proceeds to draw examples from different areas of engineering acoustics. Three of the graduate courses are taught every year: a two course sequence on physical acoustics, and a transducers course. The remaining six graduate acoustics courses, taught in alternate years, are on nonlinear acoustics, underwater acoustics, ultrasonics, architectural acoustics, wave phenomena, and acoustic metamaterials. An acoustics seminar is held most Fridays during the long semesters, averaging over ten per semester since 1984. The ME and ECE departments both offer Ph.D. qualifying exams in acoustics.

**2pID10. Graduate study in physical acoustics at the University of Mississippi.** Joel Mobley (Phys. and Astron., Univ. of Mississippi, PO Box 1848, 108 Lewis Hall, Univ., MS 38677, jmobley@olemiss.edu), Cecille Labuda (Phys. and Astron., Univ. of Mississippi, Univ., MS), Likun Zhang (National Ctr. for Phys. Acoust. and Dept. of Phys. and Astron., Univ. of Mississippi, Univ., MS), and Joseph Gladden (NCPA / Office of Res., Univ. of Mississippi, Univ., MS)

The University of Mississippi is a PhD granting institution with an R1 Carnegie designation placing it among schools with the highest level of research activity. The Department of Physics and Astronomy at the university has a diverse range of research opportunities which includes Physical Acoustics. Our acoustics program is affiliated with the National Center for Physical Acoustics (NCPA). NCPA is an 85,000 square foot standalone facility on the campus of the University of Mississippi solely dedicated to the physics and engineering applications of acoustics. It has research groups dedicated to ultrasound, infrasound, aeroacoustics, atmospheric propagation, porous media, and ocean acoustics. Graduate students in both physics and engineering are pursuing PhD and MS degrees at NCPA, and four faculty members from the physics department have their research laboratories in the facility. In addition to acoustics, our department provides a broad range of research opportunities in other subfields. We have two groups associated with recent Nobel Prizes in Gravitation and High Energy Physics and offer additional programs in Computational Physics and Atmospheric Physics. Our faculty are involved with national and international collaborations including the Laser Interferometer Gravitational Wave Observatory (LIGO), The European Center for Particle Physics (CERN) and Fermilab.

**2pID11. Underwater acoustics and ocean engineering at the University of Rhode Island.** Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., 213 Sheets Lab., Narragansett, RI 02882, loravu@uri.edu), James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Underwater acoustics is one of the primary areas of emphasis in the Ocean Engineering Department at the University of Rhode Island, the first Ocean Engineering program in the United States. The program offers Bachelors, Masters (thesis and non-thesis options) and PhD degrees in Ocean Engineering. These programs are based at the Narragansett Bay campus, providing access to a living laboratory for student learning. Some key facilities of the program are an acoustics tank and a 100-foot-long wave tank. At the graduate level, students are actively involved in research focused in areas such as acoustical oceanography, propagation modeling, geoacoustic inversion, marine mammal acoustics, ocean acoustic instrumentation, and transducers. An overview of classroom learning and ongoing research will be provided, along with information regarding the requirements of entry into the program.

**2pID12. The graduate program in acoustics at Penn State.** Andrew Barnard (Acoust., Penn State, Univ. Park, PA) and Daniel A. Russell (Graduate Program in Acoust., Pennsylvania State Univ., 201 Appl. Sci. Bldg., Univ. Park, PA 16802, dar119@psu.edu)

The Graduate Program in Acoustics at Penn State offers graduate degrees (M.Eng., M.S., Ph.D.) in Acoustics, with courses and research opportunities in a wide variety of subfields. Our 820 alumni are employed around the world in a wide variety of military and government labs, academic institutions, consulting firms, and consumer audio and related industries. Our 40+ faculty from several

disciplines conduct research and teach courses in structural acoustics, nonlinear acoustics, architectural acoustics, signal processing, aeroacoustics, biomedical ultrasound, transducers, computational acoustics, noise and vibration control, acoustic metamaterials, psychoacoustics, and underwater acoustics. Course offerings include fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, experimental techniques, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, biomedical ultrasound, flow induced noise, spatial sound and three-dimensional audio, and the acoustics of musical instruments. This poster highlights faculty research areas, laboratory facilities, student demographics, successful graduates, and recent enrollment and employment trends for the Graduate Program in Acoustics at Penn State.

**2pID13. Graduate education in acoustics at a distance from Penn State.** Daniel A. Russell (Graduate Program in Acoust., Pennsylvania State Univ., 201 Appl. Sci. Bldg., Univ. Park, PA 16802, dar119@psu.edu) and Andrew Barnard (Acoust., Penn State, Univ. Park, PA)

The Graduate Program in Acoustics at Penn State has been providing access to graduate level education in Acoustics for remote students across the country and around the world for more than 35 years. This poster summarizes the distance education Acoustics program from Penn State by showcasing student demographics, capstone paper topics, enrollment statistics and trends, and the success of our graduates. Our distance education program is offered in conjunction with our resident graduate program—course lectures are broadcast as a live stream over Zoom from a hybrid multimedia classroom allowing remote students to engage with faculty and students during live lectures; archived recordings are available for offline viewing afterward. Courses offered for distance education students include: fundamentals of acoustics and vibration, electroacoustic transducers, signal processing, acoustics in fluid media, sound and structure interaction, digital signal processing, aerodynamic noise, acoustic measurements and data analysis, ocean acoustics, architectural acoustics, noise control engineering, nonlinear acoustics, outdoor sound propagation, computational acoustics, flow induced noise, spatial sound and 3D audio, marine bioacoustics, and acoustics of musical instruments. Distance Education students can earn the M.Eng. in Acoustics degree remotely by completing 30 credits of coursework and writing a capstone paper.



**2pID14. Graduate training opportunities in the hearing sciences at the University of Louisville.** Pavel Zahorik (Otolaryngol. & Comm. Dis., Univ. of Louisville, Dept. of Otolaryngol. & Comm. Dis., Louisville, KY 40292, pavel.zahorik@louisville.edu), Shae D. Morgan (Otolaryngol. & Comm. Dis., Univ. of Louisville, Louisville, KY), and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

The University of Louisville currently offers two branches of training opportunities for students interested in pursuing graduate training in the hearing sciences: A Ph.D. degree in experimental psychology with concentration in hearing science, and a clinical doctorate in audiology (Au.D.). The Ph.D. degree program offers mentored research training in areas such as psychoacoustics, speech perception, spatial hearing, and multisensory perception, and guarantees students four years of funding (tuition plus stipend). The Au.D. program is a 4-year program designed to provide students with the academic and clinical background necessary to enter audiologic practice. Both programs are affiliated with the Heuser Hearing Institute, which, along with the University of Louisville, provides laboratory facilities and clinical populations for both research and training. An accelerated Au.D./Ph.D. training program that integrates key components of both programs for training of students interested in clinically based research is under development. Additional information is available at: http://louisville.edu/medicine/degrees/audiology and https://louisville.edu/psychology/graduate/experimental.

**2pID15. Graduate programs related to acoustics at the University of Minnesota.** Kristi Oeding (Univ. of Minnesota - Twin Cities/Minnesota State Univ. - Mankato, 314 Clinical Sci. Bldg., 150 South Rd., Mankato, MN 56001, kristi.oeding@mnsu.edu), Kelly L. Whiteford (Psych., Univ. of Minnesota, Minneapolis, MN), Peggy Nelson (Ctr. for Appl./Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), Hubert H. Lim (Otolaryngol., Head and Neck Surgery, Univ. of Minnesota, Minneapolis, MN), Mark A. Bee (Ecology, Evolution & Behavior, Univ. of Minnesota, St. Paul, MN), and Andrew J. Oxenham (Psychol., Univ. of Minnesota, Minneapolis, MN)

The University of Minnesota (UMN) has graduate programs that span the areas of Animal Bioacoustics, Psychological and Physiological Acoustics, and Speech Communication. Degrees are offered in Psychology (PhD), Speech-Language-Hearing Sciences (MA in speech-language pathology, AuD, and PhD in speech-language-hearing sciences), Biomedical Engineering (MS and PhD), Ecology, Evolution, and Behavior (PhD), and Neuroscience (PhD). Faculty across departments have a shared interest in understanding how the ear and brain work together to process sound and in developing new technologies and approaches for improving hearing disorders. Located on campus is the Center for Applied and Translational Sensory Science (CATSS), which provides opportunities for interdisciplinary collaborations across departments and industry to understand how sensory impairments work. Within CATSS is the Multi-Sensory Perception Lab, which houses shared equipment, including eye trackers and electroencephalography. The Center for Magnetic Resonance Research houses several ultrahigh field magnets, while the Center for Neural Engineering and affiliated faculty labs also house multiple neuromodulation and neurorecording devices to interact with and monitor neural activity in humans and animals.

**2pID16. Graduate studies in acoustics and wave physics at Institut d'Acoustique - Graduate School, Le Mans, France.** Vincent Tournat (Inst. d'Acoustique - Grad. School, Lab. d'Acoustique de l'Université du Mans, UMR CNRS 6613, Le mans Université, CNRS, Lab. d'Acoustique de l'Université du Mans (LAUM), Le Mans, France, vincent.tournat@univ-lemans.fr)

This poster presents the graduate studies in Acoustics at Le Mans (France) offered at the "Institut d'Acoustique - Graduate School". Graduate studies in Acoustics at Le Mans University have been awarded in 2018 the excellence label "École Universitaire de Recherche" among 28 other reference centers for all fields of Science, through a highly selective national call. Our master and engineering school programs range from physical acoustics, environmental acoustics, acoustics and vibrations to international masters on electro-acoustics and on wave physics. The education through research is carried out at the LAUM, UMR CNRS, one of the largest acoustics laboratory in the world. Several details, objectives, student feedbacks and contact informations on the graduates studies will be given on the poster.

## Session 2pNS

# Noise and Psychological and Physiological Acoustics: Methods for Community Noise Testing and Analysis II

Alexandra Loubeau, Cochair
*NASA Langley Res. Ctr., Hampton, VA 34681*

William Doebler, Cochair
*NASA Langley Res. Ctr., MS 463, Hampton, VA 23681*

### *Invited Papers*

**1:00**

**2pNS1. Comparison and regression analysis of lateral sonic boom measurements and PCBoom predictions.** J. T. Durrant (Dept. of Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, taggart.durrant@gmail.com), William Doebler, Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA), Mark C. Anderson (Phys. and Astron., Brigham Young Univ., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT)

NASA plans to fly the X-59 aircraft over communities to gather data on the response to quiet supersonic flight. This data campaign could revolutionize the aerospace industry by enabling commercial, overland supersonic flight. To prepare for this campaign, NASA is developing PCBoom, a software suite of sonic boom modeling tools. PCBoom-predicted Perceived Level (PL) values were previously compared with measured PL values from a recent NASA test flight campaign, Quiet Supersonic Flights 2018 (QSF18), and were found to differ by an average of 6 dB. This work investigates the PCBoom prediction performance using data from NASA's 2020 CarpetDIEM Phase I flight test using an F-18 aircraft. PL predictions are compared using the PCBoom default F-18 F-function near-field as input versus a computational fluid dynamics near-field solution for the aircraft as input. To investigate potential sources of metric variability and differences between modeled and measured metrics, Least Absolute Shrinkage and Selection Operator (LASSO) and least-squares regression are used. Because weather has a strong influence on sonic boom variability, the regression techniques are also used to guide the necessary number of ground weather measurements to capture boom metric variability. [Work supported by NASA Langley Research Center through the National Institute of Aerospace.]

**1:20**

**2pNS2. Nonnormality of sonic boom loudness metrics in the turbulent atmospheric boundary layer at large lateral distances from the flight path.** Alexander N. Carr (Mech. and Aerosp. Eng., Univ. of Florida, NASA Langley Res Ctr., MS 461, Hampton, VA 23681, alexander.carr@nasa.gov), Joel B. Lonzaga (Structural Acoust. Branch, National Aeronautics and Space Administration, Hampton, VA), and Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

Atmospheric boundary layer (ABL) turbulence causes variability of the sonic boom waveform at the ground. Recent numerical investigations of sonic boom propagation through kinematic velocity fluctuations indicate that loudness metric distributions are positively skewed relative to a normal distribution. This skewness depends on the propagation distance and turbulence intensity. Propagation simulations of N-waves and shaped booms through inhomogeneous ABL turbulence are presented. Meteorological conditions are varied to examine different daytime ABL conditions and their effect on sonic boom loudness distributions. Two outcomes are observed: (1) the loudness metric distributions become increasingly positively skewed as the propagation distance through the ABL increases, and (2) the distributions become increasingly positively skewed at the same lateral distance from the flight path as the convection level of the daytime ABL is increased. Thus, results indicate that ground level measurements of sonic boom loudness from flight tests performed at large lateral distances from the flight path may not be normally distributed, due to turbulence present in the ABL. (This research is supported by the Commercial Supersonic Technology Project of the National Aeronautics and Space Administration under Grant No. 80NSSC19K1685.)

**1:40**

**2pNS3. Effects of dose error and sample size on sonic boom dose-response curves.** William Doebler (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, william.j.doebler@nasa.gov), Aaron B. Vaughn, Nathan B. Cruze, Kathryn Ballard, Jonathan Rathsam, and Peter A. Parker (NASA Langley Res. Ctr., Hampton, VA)

NASA will soon be collecting noise-annoyance community survey data as the X-59 aircraft flies supersonically over several communities in the USA. Sparse measurements of the X-59 sonic thumps will be used together with physics-based simulations to estimate noise doses at survey participant locations. These dose estimates have associated error that affects the accuracy of modeled dose-response curves, which can result in misestimation of annoyance. The precision in dose-response curves is also a consideration in selecting the number of survey participants. To enable pretest studies of dose error and precision, simulated dose-response data were generated

based on NASA's Quiet Supersonic Flights 2018 test. The data included various degrees of dose error and sample size. Frequentist multilevel logistic regression models were fit to the true and perturbed dose-response data. Simple proportional relationships were identified between the model parameters and the perturbation standard deviation. The summary dose-response curves illustrate the impact on accuracy if dose error is not accounted for in the model. The precision in the dose-response curves is also shown as the number of participants and degree of participation is varied. Finally, sampling variability is illustrated by showing the dose-response curves for several replicates with random draws of participants and errors.

**2:00**

**2pNS4. Exposure uncertainty effects on regression parameter values.** Richard D. Horonjeff (48 Blueberry Ln., Peterborough, NH 03458, rhoronjeff@comcast.net)

Exposure uncertainty in community noise exposure-response investigations can bias the results of regression analyses, especially when the range of sound levels is limited, the standard deviation of sound level uncertainty is more than a decibel or two and the distribution of sound levels across the sample population deviates significantly from a uniform one. The bias is manifested in the solved values of the regression coefficients, leading to incorrect representations of both the slope and intercept. Methods are available to reduce the magnitude of such bias and this paper examines the regression calibration approach. A basic overview of regression calibration is provided as well as its implementation in a logistic regression analysis. The effectiveness and limitations of the procedure are discussed, as are the requisite additional field measurements necessary to quantify situational parameters required in the calibration process. While coefficient bias can be mitigated, coefficient uncertainty that might otherwise be due only to uncertainty in the predicted variable (fraction annoyed) increases with increasing exposure uncertainty. Such uncertainty impacts the ability to observe differences between sample populations in both scalar and categorical variables.

**2:20**

**2pNS5. Overview of research-based community noise testing recommendations from Brigham Young University.** Mark C. Anderson (Dept. of Phys. and Astron., Brigham Young Univ., D-70 ASB, Provo, UT 84602, anderson.mark.az@gmail.com), Kent L. Gee, J T. Durrant (Depart. of Phys. and Astron., Brigham Young Univ., Provo, UT), and Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA)

Brigham Young University has been studying questions related to NASA X-59 community noise testing. A summary of the key findings and recommendations to date are presented. Among these are weather-resistant ground microphone setups that reduce wind noise, methods for treating the problem of ambient noise contamination on sonic boom metrics and spectra, and the turbulence-induced variability seen over relatively small-aperture arrays. Also discussed are recommendations for data post-processing, e.g., employing a digital pole-shifting filter and zero-padding to improve the fidelity and smoothness of low-frequency spectral data. Finally, the relative merits of various sonic boom metrics are considered. Although Perceived Level has been the most widely-used metric, it is relatively sensitive to ambient noise contamination and turbulence-induced variability. [Work supported by NASA Langley Research Center through the National Institute of Aerospace.]

**2:40–3:00 Break**

## *Contributed Papers*

**3:00**

**2pNS6. Standardization of modelling methodology for aircraft noise exposure contours.** Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, Ontario N9B 3P4, Canada, jovano11@uwindsor.ca) and Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, Ontario, Canada)

Aircraft noise exposure contours are important tools for land-use planning around aerodromes. They inform local governments and stakeholders as to the acoustic impacts of aircraft operations on areas surrounding the airport. This information is considered when defining municipal zoning and building regulations. While noise contours are often regarded as concrete guidance identifying the suitability of an area for noise sensitive development, their prescribed modelling methodology is imprecise and non-standardised. A vague description such as the yearly day night level or 95$^{th}$ percentile day is typically mandated by overseeing authorities which leaves ambiguity in terms of the various specific input parameters. This can result in differing noise contour outputs, which in turn can cause conflict, especially between stakeholders with competing interests. This research highlights the need for better standardization and guidance for aircraft noise exposure contour modelling. It further demonstrates how varying interpretations of modelling methodologies can alter the input parameters of an aircraft noise model and significantly impact the outputs.

**3:15**

**2pNS7. Distribution methodology for aircraft noise annoyance surveys.** Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, Ontario N9B 3P4, Canada, jovano11@uwindsor.ca) and Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, Ontario, Canada)

Annoyance is one of the most common effects of aircraft noise on individuals. The prevalence of severe annoyance within a community is a metric that informs regulatory noise exposure thresholds and guidelines. It is therefore critical that accurate annoyance data is collected through community surveys, which are typically distributed to areas affected by various levels of aircraft noise, as defined by average-day type noise exposure contours. This distribution methodology excludes segments of the population that are affected by noise but underrepresented by these types of contours. Here are presented the results of two community surveys executed around Toronto Pearson International Airport, using different distribution methodologies. The first survey identified five zones for distribution based on noise exposure contours. The second survey was distributed within a 750-meter radius around 25 noise monitoring terminals in the vicinity of the airport. The two surveys yielded different annoyance results, particularly as they relate to the locations of highly annoyed respondents. A prevalence of severe annoyance was observed in areas that were intermittently affected by aircraft noise and thus out of the range of average-day type noise contours. It was concluded that a more comprehensive approach for survey distribution is necessary to ensure unbiased annoyance results.

**3:30**

**2pNS8. Using appropriate aircraft noise metrics for various applications.** Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, Ontario, Canada) and Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, Ontario N9B 3P4, Canada, novak1@uwindsor.ca)

Aircraft operations and the resulting noise emissions have widespread implications that affect numerous stakeholders including airport authorities, airlines, navigation providers, government entities, developers, investors, community members and more. These diverse audiences require information about the degrees of impact of aircraft operations as they relate to noise. The outcome from the use of this information can affect everything from airspace design, zoning, building codes, regulatory measures, public guidance etc. To simplify the subject of aircraft noise, there has always been an effort to use a single metric for the numerous applications listed above. However, this approach more often contributes to confusion and discrepancies as multiple agencies try to force a metric to fit a purpose for which it was not intended. While cumulative noise metrics such as DNL or NEF might be appropriate for annoyance prediction and land-use planning, they are not appropriate for communicating noise data to the public. Likewise, these metrics are not appropriate for building code requirements as they cannot be applied without conversions. Different metrics are developed for different purposes, and they should be applied for their prescribed usage. This paper discusses several metrics and their appropriate applications within the study of aircraft noise and its impacts.

**3:45**

**2pNS9. Non-acoustic factors and their role in aircraft noise annoyance.** Julia Jovanovic (Mech. Automotive and Mater. Eng., Univ. of Windsor, Windsor, Ontario, Canada) and Colin Novak (Mech. Automotive and Mater. Eng., Univ. of Windsor, 401 Sunset Ave., Windsor, Ontario N9B 3P4, Canada, novak1@uwindsor.ca)

Non-acoustic factors have been acknowledged for some time as likely contributors to aircraft noise annoyance. They help explain why a given level of noise exposure can evoke severe annoyance in one person but not in another. Multiple analyses have concluded that non-acoustic factors explain more variance in annoyance results than noise exposure levels do. That begs questions as to why noise exposure levels are currently the only prescribed predictor of annoyance, and why regulating agencies continue to focus on only reducing noise exposure to combat annoyance. The subjective nature and lack of thorough understanding of non-acoustic factors has rendered them unusable for regulatory purposes, or even as topics of discussion with various stakeholders. What then is the purpose to study non-acoustic contributors to noise annoyance, other than to dismiss severe annoyance by implicating personal, attitudinal, or situational factors rather than the noise itself? This discussion suggests mechanisms by which non-acoustic factors contribute to annoyance and proposes practical ways to incorporate this knowledge in the prediction and mitigation of annoyance.

**4:00**

**2pNS10. Psychophysiological effect according to restoration factors of audio-visual environment.** Jin Yong Jeon (Medical and Digital Eng., Hanyang Univ., Seoul, South Korea), Haram Lee (Architectural Eng., Hanyang Univ., 222, Wangsimni-ro, Seongdong-gu, Seoul 04763, South Korea, haramchi7@naver.com), and Yunjin Lee (Medical and Digital Eng., Hanyang Univ., Seoul, South Korea)

This study investigated the effects of psychophysiological restoration due to environmental factors while experiencing the city, waterfront, and green space using various psychological scales and physiological measurement tools. The environment was experienced using virtual reality technology, and the subjects' responses were collected through surveys and EEG (electroencephalography) and HRV (heart rate variability) measurement. HRV responses were carried out by parameters such as total power (TP), SDNN and TSI. In case of EEG, PSD (power spectrum density) analysis indicated a relatively high restoration effect in the natural environment and increase of alpha-beta ratio. Based on functional connectivity, graph theory analysis showed that the limbic system network of non-restoration group was hyperactive. When comparing HRV responses (increase of TP, SDNN and reduction of TSI), from the resilience group, it was found to have higher global network efficiency. In fact, in urban space, there were fewer psychological restoration responses in the group with high noise sensitivity.

4:15

**2pNS11. Influence of uncertainties in noise maps on their urban planning interpretation.** Jean-Philippe Migneron (Ecole d'architecture, Universite Laval, 1, cote de la Fabrique, Quebec City, Quebec G1R 3V6, Canada, jean-philippe.migneron.1@ulaval.ca), Frédéric Hubert (Dept. des Sci. Geomatiques, Univ. Laval, Quebec City, Quebec, Canada), and Jean-Gabriel Migneron (Ecole d'architecture, Univ. Laval, Quebec City, Quebec, Canada)

The use of noise mapping in urban planning can be used to analyze different situations such as predictive studies of new developments, environmental assessments, and estimation of noise exposure for public health protection. The statistical significance of the results illustrated in the noise maps must be understood by their readers, especially if it concerns the implementation of public policies or compliance with existing regulations. Based on the examples defined by the ISO 17534 standards, some experiments are explained showing the effect of the uncertainties associated with noise maps on their interpretation for land use planning.

4:30

**2pNS12. Variability of noise prediction models in catchments featuring significant barriers and noise-enhancing meteorological conditions.** Lance Jenkin (EMM Consulting, Thornton, New South Wales, Australia), Jeffrey Peng (Dept. of Planning and Environ., Sydney, New South Wales, Australia), and Jeffrey Parnell (Ctr. for Audio, Acoust. and Vibration, Univ. of Technol. Sydney, PO Box 123, Broadway, New South Wales 2007, Australia, jeffrey.parnell@uts.edu.au)

Accurate prediction of noise propagation from industrial sources forms a vital foundation from which to determine noise pollution levels on sensitive communities, as well as informing any mitigation measures required to address unacceptable impacts. A variety of sound propagation model options are available to practitioners in commercial software platforms such as SoundPLAN and CadnaA, and the ability to design effective noise barriers is contingent on the selection of a model that is suitable for the situation under consideration. This is particularly important in noise catchments that feature noise-enhancing meteorological conditions and where significant barriers exist, or are proposed between the industrial estate and potentially noise affected residential communities. In this work, sound levels computed using CONCAWE, ISO 9613-2, Nord2000 and CNOSSOS-EU sound propagation models for homogenous and favourable conditions are compared. The cross-sectional profile of the case study featured in this work is based on a real-world situation in the built-up suburban area of Sydney, Australia. Current findings highlight some key considerations, limitations, and pitfalls associated with older empirically derived sound propagation models.

4:45

**2pNS13. Experiential design tools for acousticsi—A retrospective and look ahead at the use of sound and visualizations for transportation noise.** Ryan Biziorek (Arup, 35 E Wacker Dr., Suite 1800, Chicago, IL 60601, ryan.biziorek@arup.com), David Hiller (Arup, Manchester, United Kingdom), Vincent Jurdic (Arup, Montreal, Quebec, Canada), AnaLuisa Maldonado (Arup, Winchester, United Kingdom), Henry Harris (Arup, London, United Kingdom), Cameron Heggie, Paul Her (Arup, Amsterdam, Netherlands), Calum Sharp (Arup, London, United Kingdom), Adam Thomas, James Woodcock (Arup, Manchester, United Kingdom), devin bean, Joseph Digerness (Arup, New York, NY), Jon Swan, and Bettine Gommer (Arup, Los Angeles, CA)

Originally conceived and developed to inform the design of some of the world's best arts and culture venues, over the past 10+ years, Arup SoundLab has also been used to create sound demonstrations that simulate and gauge response to environmental sound. Sound demonstrations combine aural and visual simulations to enable clients, designers, major stakeholders and the general public to experience and better understand sound. They provide robust objective information to support decision making and help shape better outcomes for all. The SoundLab has been used to inform the design of vertiport infrastructure; to assess annoyance and possible health impacts of novel noise sources; to inform local and international policy on noise; and to provide information on the early prototyping of Advanced Air Mobility (AAM) vehicles. These applications will be described in this presentation, including recent simulations for the Los Angeles Department of Transportation (LADOT) and a human response study for the European Union Aviation Safety Agency (EASA) to provide insight into people's response to AAM noise impacts. As AAM applications broaden, auralisation and visualisation processes are being developed to facilitate understanding of planning, permitting and design processes. The immersive experience provides information that is valuable to the various parties involved.

**5:00–5:30**
**Panel Discussion**

# Session 2pPA

## Physical Acoustics and Structural Acoustics and Vibration: Frontiers of Resonant Ultrasound Spectroscopy and its Applications II

Christopher M. Kube, Cochair
*Eng. Sci. and Mech., The Pennsylvania State Univ., 212 Earth and Eng. Sci. Bldg., Univ. Park, PA 16802*

Matthew Cherry, Cochair
*Air Force Res. Lab., 2230 10th St., Fairborn, OH 45433*

Rasheed Adebisi, Cochair
*UDRI, Univ. of Dayton, 141 Firwood Dr., Shroyer Park Ctr., Dayton, OH 45419*

### *Invited Papers*

**1:00**

**2pPA1. High-temperature elastic moduli: A tool for understanding chemical bonding in thermoelectric materials.** Alexandra Zevalkink (Chem. Eng. and Mater. Sci., Michigan State Univ., 438 S Shaw Ln, East Lansing, MI 48824, alexzev@msu.edu)

In the study of thermoelectric materials, the elastic moduli are valuable for predicting and interpreting trends in lattice thermal conductivity and some aspects of electronic conductivity (e.g., electron-phonon interactions). Because elastic moduli are highly sensitive to the details of chemical bonding, they can also be used to detect subtle changes in crystal symmetry or site-disorder due to phase transitions. Consequently, measurements of elastic moduli are of fundamental value in the development of structure-property relationships in thermoelectric materials. However, experimental reports of the elastic constants of functional materials are relatively sparse, and temperature-dependent data is even less common. Further, unlike most types of functional materials, thermoelectric materials are used over an extremely wide temperature range (up to 1300 K in some cases). In our lab, we measure the elastic constants of polycrystalline and single crystalline thermoelectric materials across a wide temperature range (250–1300 K) using resonant ultrasound spectroscopy with $Al_2O_3$ buffer rods to separate the transducers from the hot zone. In this talk, I will discuss the high-temperature elastic properties of several classes of thermoelectric materials, including $Mg_3Sb_2$ and GeSe-$AgBiSe_2$ alloys, emphasizing the role that such data can play in interpreting composition- and temperature-dependent trends in lattice thermal conductivity.

**1:25**

**2pPA2. Resonant ultrasound spectroscopy for characterization of lattice instability of shape memory alloys.** Hanus Seiner (Inst. of Thermomechanics, Czech Acad. of Sci., Dolejskova 5, Prague 18200, Czechia, hseiner@it.cas.cz) and Petr Sedlak (Inst. of Thermomechanics, Czech Acad. of Sci., Prague, Czechia)

Shape memory alloys (SMAs) are materials able to undergo reversible displacive transitions between different crystal lattices. This ability is reflected by the elastic constants of the individual phases, as certain combinations of these constants exhibit anomalous softening when the crystal approaches the transformation temperature or when the lattice is destabilized by the presence of mobile interfaces. The lecture will summarize the advantages of using resonant ultrasound spectroscopy (RUS), and in particular its contact-less laser-based modification, for detecting and evaluating these elastic anomalies. Three recently studied cases will be discussed: i) single crystals of ferromagnetic shape memory alloys in which the instability-induced softening couples with magnetoelasticity; ii) fine regular laminates in tetragonal martensitic lattices in which the soft combinations of the elastic constants have to be calculated from the effective elastic response of the microstructure; iii) modulated crystals with highly mobile interfaces that act as strongly-nonlinear mechanical oscillators and where the properties of the interfaces can be achieved by analyzing the non-linearity.

1:50

**2pPA3. Resonances and mode shape imaging for characterizing the microstructure of complex metals.** Christopher M. Kube (Eng. Sci. and Mech., The Pennsylvania State Univ., 212 Earth and Eng. Sci. Bldg., Univ. Park, PA 16802, kube@psu.edu), James Hanagan (Mater. Sci. and Eng, The Pennsylvania State Univ., Univ. Park, PA), and Matthew Cherry (Air Force Res. Lab., Wright-Patterson Air Force Base, OH)

Resonant Ultrasound Spectroscopy (RUS) is a mature and reliable technique for measuring the elastic stiffness constants of anisotropic materials. In highly processed materials such as additively manufactured metal parts, the anisotropic stiffnesses arrive from microstructures that contain grains with preferred orientations, which can result from directionally dependent thermal or mechanical processes. The connection between elastic stiffnesses and microstructure suggest the potential for RUS being used as a tool to characterize microstructure. This presentation will highlight modeling and experimental measurements in this direction. The RUS model is developed from a micromechanical foundation and includes structure-property linkages between the microstructure's orientation distribution function and the macroscopic elastic constitutive behavior. The micromechanical basis allows efficient creation of a complete catalog containing all possible resonant behavior (resonant frequencies and mode shapes) of polycrystalline samples as a function of the possible textures. The second part of the presentation will focus on connecting experimentally measured resonances and mode shapes to a particular catalog entry. The inclusion of mode shape imaging in the analysis is expected to offer additional detail and potential for expanded characterization beyond texture and elastic stiffnesses.

2:05–2:20 Break

2:20

**2pPA4. Ultrasonically determined elastic constants of additively manufactured 316L stainless steel.** Mason Hayward (Univ. of Louisiana at Lafayette, 240 Hebrard Boulevard, Broussard Hall Room 103, Lafayette, LA 70503, mason.hayward1@louisiana.edu), Gabriela Petculescu (Univ. of Louisiana at Lafayette, Lafayette, LA), and Erica Murray (Louisiana Tech Univ., Ruston, LA)

We determined the effect of laser speed on the elastic constants of additively manufactured (AM) 316L stainless steel using resonant ultrasound spectroscopy (RUS). The alloy (316L) has biomechanical applications, such as medical implants. AM disks were manufactured at a constant power of 100 W and varying laser speeds of 800, 1000, and 1200 mm/s. RUS samples were extracted from the disks to determine the effects of fabrication parameters on elastic constants, as well as variations in properties across a single disk. As laser speed increases, the longitudinal ($c_{11}$) moduli decreases from 284.74 GPa to 226.84 GPa, while the shear ($c_{44}$) moduli exhibits minimal change. The measurement error in both moduli increases as laser speed decreases, which is attributed to the textured polycrystal nature of the samples built at lower speed. At lower speed, a greater amount of energy is deposited within the volume, allowing grains more time to grow. The grain structure determined by electron backscatter diffraction shows large crystallite formations in the 800 mm/s sample while the 1200 mm/s sample shows more homogenous small-grain distribution, which is expected of an ideal polycrystal. Variations of properties across the disk will also be presented.

2:35

**2pPA5. Resonant ultrasound spectroscopy measurement and modeling of additively manufactured octet truss lattice cubes.** Brian Tran (Mater. Eng. Directorate, Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550, tran64@llnl.gov), Karl A. Fisher (Mater. Eng. Directorate, Lawrence Livermore National Lab., Liveremore, CA), Jenny Wang (Phys. and Life Sci. Directorate, Lawrence Livermore National Lab., Livermore, CA), Chuck Divin, and Gabriel Balensiefer (Mater. Eng. Directorate, Lawrence Livermore National Lab., Livermore, CA)

Advancement of additive manufacturing technological readiness requires high throughput evaluation capabilities that can keep pace with the development of complex parts. Resonant ultrasound spectroscopy (RUS) is an acoustic technique that provides rapid holistic probing of a part by tracking fundamental mechanical resonance modes. In this work, the RUS responses of additively manufactured Ti-5553 octet truss lattice cubes were characterized using experimental measurements and three-dimensional finite element models. Varying percentages of missing struts were designed into the lattices as controlled defects and were verified using X-ray computed tomography. Experimental measurements of density and Young's modulus were treated as input parameters in a homogenous anisotropic continuum model. The continuum model was compared with experimental RUS measurements, thus evaluating the potential for a simplified approximation of the octet truss lattice. [This work was supported by US DOE LLNL-LDRD 20-SI-001 and was performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344.]



2:50

**2pPA6. Combined neutron scattering and resonant ultrasound spectroscopy for lattice dynamics studies.** Raphael Hermann (ORNL, 1 Bethel Valley Rd., Oak Ridge, TN 37830-8050, hermannrp@ornl.gov)

Resonant ultrasound spectroscopy (RUS) and inelastic neutron scattering afford insight into solid state structures and excitations at vastly different frequency and length scales. We will discuss the combined use of these techniques to understand phase transitions in functional and structural materials. Structural complexity, such as chirality or incommensuration leads to emerging features in both techniques and modifies energy transport channels. Novel developments in techniques at the Oak Ridge neutron sources, that now enable in situ ultrasound spectrocopy in the neutron beam, and the RUSCal analysis software that now supports all Bravais classes and implements Monte-Carlo methods, will be discussed in example materials. [Work supported by the Deparment of Energy Office of Basic Energy Science and by Laboratory Directed Research at Oak Ridge National Laboratory. My thanks go to J. Torres, V. Fanelli, Y. Shinohara, E. Cakmak, C. Hua, A. Flores-Bettancourt, M. Ruis-Rodriguez for the collaboration and A. Zevalkink, D. Mandrus, V. Keppens, E. Lara-Curzio, and T. Watkins for fruitful discussions.]

3:05

**2pPA7. Resonant ultrasound spectroscopy studies of high-entropy fluorites.** Rubayet Tanveer (The Univ. of Tennessee, Knoxville, TN 37996, rtanveer@vols.utk.edu) and Veerle M. Keppens (The University of Tennessee, Knoxville, TN)

High entropy oxides (HEOs), also referred to as multicomponent oxides or compositionally complex oxides (CCOs), have attracted attention due to the tunability of multiple cations on a single site. Since the introduction of HEOs stabilized in the rocksalt phase, the high entropy oxide concept has been expanded to various structures, offering a path for the discovery of innovative compounds with unique structure-property relations. Here, we present a study of high entropy fluorites, which have gained recognition for their low thermal conductivity and possible applications as thermal barrier coatings. We have successfully synthesized single phase samples using multiple cations on a single site, in equimolar and non-equimolar ratios. Resonant ultrasound spectroscopy was used to evaluate the elastic moduli as a function of temperature. The results obtained on these multi-component pyrochlores are compared to those of their single-component counterparts.

## Session 2pPP

# Psychological and Physiological Acoustics: Speech and Pitch Perception

Christian E. Stilp, Chair
*Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sciences Bldg., Louisville, KY 40292*

## *Contributed Papers*

**1:00**

**2pPP1. Loudness effect on pitch and timbre discrimination; a continuing study.** Wesley Bulla (Audio Eng., Belmont Univ., Curb College of Entertainment & Music Business, 1900 Belmont Blvd., Nashville, TN 37212, wesbulla@comcast.net) and Song Hui Chon (Audio Eng., Belmont Univ., Nashville, TN)

Fine discrimination in pitch and timbre is a requisite skill for musicians and audio professionals. While several trade organizations have standardized electronic-acoustic calibration levels for equipment testing and operations, conflicting and often unfounded perceptual theories along with a lack of psychophysical evidence make it difficult to pinpoint the optimal presentation level for best pitch and timbre discrimination. Results from our previous study (Bulla & Chon, FDMC, 2021) revealed higher scores for lower listening levels in conditional 2-AFC pitch and timbre discrimination tasks (15 vs 83 dBA SPL). While outcomes there indicated a consistent negative influence of the larger signal strength on the accuracy of both pitch and timbre discrimination performance, this continuing study examined changes of signal strength in order to establish whether or not observations here may apply across a variety of listening levels. Data collection is ongoing and will provide insight through observed performance metrics and help determine if there is an optimal loudness range for engaging in listening tasks such as identifying differences of pitch, discriminating fine details in timbre, or the blending and balancing of musical instruments.

**1:15**

**2pPP2. Vocal pitch perception with cochlear implants.** Emma Martin (Otolaryngol., Univ. of Illinois at Chicago, 1855 W Taylor St., STE 2.42, Chicago, IL 60612, eemarti2@uic.edu), Simin Soleimanifar (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, Champaign, IL), Keiko Ishikawa (Univ. of Kentucky, Lexington, KY), and Justin M. Aronoff (Speech and Hearing Sci., Univ. of Illinois at Urbana Champaign, Champaign, IL)

Cochlear implant (CI) users often struggle to accurately perceive vocal pitch. This suggests that they are either not able to adequately use vocal pitch cues, such as temporal modulations resulting from F0, or they are using cues that do not correspond to vocal pitch. A potentially unreliable cue they may be using is formant spacing, which is correlated with vocal tract length. There is a gross, probabilistic relationship between vocal pitch and formant spacing since males tend to have both lower F0s and longer vocal tracts, but the relationship is weak, especially within gender. To examine if CI users are using the acoustic cues resulting from vocal tract length to make pitch judgments, CI users were presented with stimuli where the F0 was held constant but the formant spacing was systematically manipulated. Participants were presented with pairs of stimuli and asked to indicate the similarity of the pitch of the two stimuli using a seven-point Likert scale. The preliminary results indicate that CI users systematically perceived the changes in formant spacing as a change in pitch.

**1:30**

**2pPP3. Perceptual effects of formant enhancement with the factors of phonetic type, listening conditions, and language experience of listeners.** Mingshuang Li (Dept. of Communication Disorders and Sci., California State Univ., Northridge, Northridge, Northridge, CA 91330, mingshuang.li@csun.edu) and Chang Liu (Univ.of Texas at Austin, Austin, TX)

The second formant (F2) enhancement is a technique that aims to improve speech perception in adverse noise by amplifying the F2 of speech signals. The current study was to investigate whether F2 enhancement would improve speech identification with the factors of phonetic type (e.g., vowel and consonant), listening conditions (e.g., speech and nonspeech noise at moderately and challenging SNRs), and language experience of listeners (e.g., native and nonnative listeners), whether the amount of perceptual benefit was dependent on these factors. Two groups of participants, English native and nonnative listeners, were recruited in this study. Identification of English vowels and consonants with and without F2 enhancement were measured in quiet, long-term speech-shaped noise (LTSSN) and six-talker babble (6-TB) at the signal-to-noise ratios (SNRs) of $-10$ dB and $-15$ dB. Overall, significant improvements from F2 enhancement were found in both vowel and consonant identification for both native and nonnative listeners in various listening conditions. Furthermore, greater improvement was found at the SNR of $-15$ dB than at the SNR of $-10$ dB, as well as for nonnative listeners than native listeners in vowel identification. Meanwhile, the amount of benefit was generally comparable in speech and nonspeech noise. These results indicate that F2 enhancement could improve phonetic identification in noise for native and nonnative listeners, showing a potential as a speech enhancement algorithm in challenging noise.

**1:45**

**2pPP4. Testing the role of primary musical instrument on context effects in music perception.** Caleb J. King (Psychol. and Brain Sci., Univ. of Louisville, 2301 S 3rd St., Louisville, KY 40292, cjking03@louisville.edu), Anya E. Shorey (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), Kelly L. Whiteford (Psychol., Univ. of Minnesota, Minneapolis, MN), and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Musicians display numerous perceptual benefits versus nonmusicians, such as better pitch and melody perception (the "musician advantage"). Recently, Shorey *et al*. (2021 ASA) investigated whether this musician advantage extended to spectral contrast effects (SCEs; categorization shifts produced by acoustic properties of surrounding sounds) in musical instrument recognition. Musicians and nonmusicians listened to a context sound (filtered string quartet passage highlighting frequencies of the horn or saxophone), then categorized a target sound (tone from a six-step series varying from horn to saxophone). Although musicians displayed superior pitch discrimination, their SCEs did not differ from those of nonmusicians. Importantly, separate research has reported that a musician's instrument of

training heavily influences musical perception, potentially improving frequency discrimination and rhythm perception/production. However, in the Shorey et al. study, musicians were recruited without respect to their primary instrument. This follow-up study uses the same methodology as Shorey et al. but recruits only musicians who play horn or saxophone (the instruments used as target sounds) as their primary instrument. It is predicted that horn and saxophone players will display larger SCEs than non-musicians due to their intimate familiarity with the instrument timbres. Preliminary data are trending in the predicted direction; full results will be discussed.

**2:00**

**2pPP5. Role of superior temporal gyrus and planum temporale in talker segregation.** Christian Herrera Ortiz (Auditory Res., LLVARE, 11201 Benton St., Loma Linda, CA 92357, christian.herreraortiz@va.gov), Nicole Whittle (Res. Services, Loma Linda Veteran's Assoc. for Res. and Educatoin, Loma Linda, CA), Marjorie R. Leek (Res., Loma Linda VA Healthcare System, Loma Linda, CA), Samuel Barnes, Barbara Holshouser (Radiology, Loma Linda Univ., Loma Linda, CA), Alex Yi (Res., Loma Linda VA Healthcare System, Loma Linda, CA), and Jonathan H. Venezia (VA Loma Linda Healthcare System, Loma Linda, CA)

Recent studies suggest the brain tracks both attended and unattended speech streams. Here, we describe the cortical mechanisms that support active talker segregation by vocal gender. Thirty-three participants with normal or near-normal hearing performed a competing speech task during fMRI scanning. The target (competing) talker was female (male). Spectro-temporal modulation filtering was applied to stochastically modulate female and male vocal pitch across trials. Using the modulation-filter patterns as predictors, spectrotemporal receptive fields (STRFs) were obtained at each voxel using coordinate descent. STRF weights associated with female- ($\sim$6 cyc/kHz) and male-talker ($\sim$12 cyc/kHz) pitch were analyzed across subjects to identify pitch-sensitive voxels (logical OR, corrected $p < 0.01$), which were then characterized by preference for female vs. male. Anterior regions in Heschl's gyrus and the superior temporal gyrus (STG) responded best to the female talker, while posterior regions in STG and planum temporale (PT) responded best to the male talker. In a control task where the talkers did not compete, the same pattern was observed but the posterior network shifted from STG to PT and responded to the acoustic boundary between talkers ($\sim$9 cyc/kHz), suggesting that acoustically coded pitch in PT becomes voice-coded in STG during active segregation.

**2:15**

**2pPP6. Inferring pitch from coarse spectral features.** Danni Ma (Univ. of Pennsylvania, 3417 Walnut St., Philadelphia, PA 19104, dannima@seas.upenn.edu), Neville Ryant, and Mark Liberman (Univ. of Pennsylvania, Philadelphia, PA)

Fundamental frequency (F0) has long been treated as the physical definition of "pitch" in phonetic analysis. But there have been many demonstrations that F0 is at best an approximation to pitch, both in production and in perception: pitch is not F0, and F0 is not pitch. Changes in the pitch involve many articulatory and acoustic covariates; pitch perception often deviates from what F0 analysis predicts; and in fact, quasi-periodic signals from a single voice source are often incompletely characterized by an attempt to define a single time-varying F0. In this paper, we find strong support for the existence of covariates for pitch in aspects of relatively coarse spectra, in which an overtone series is not available. Thus linear regression can predict the pitch of simple vocalizations, produced by an articulatory synthesizer or by human, from single frames of such coarse spectra. Across speakers, and in more complex vocalizations, our experiments indicate that the covariates are not quite so simple, though apparently still available for more sophisticated modeling. On this basis, we propose that the field needs a better way of thinking about speech pitch, just as celestial mechanics requires us to go beyond Newton's point mass approximations to heavenly bodies.

## Session 2pSA

## Structural Acoustics and Vibration and Computational Acoustics: Surrogate and Reduced-Order Modeling for Structural Acoustics Applications

Stephanie Konarski, Cochair
*Johns Hopkins Univ. Appl. Phys. Lab*

Anthony L. Bonomo, Cochair
*Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817*

Ali Kanj, Cochair
*Mech. Sci. and Eng., Univ. of Illinois at Urbana-Champaign, 105 S Mathews Ave., Mech. Eng. Lab., Urbana, IL 61801*

**Chair's Introduction—1:30**

### *Invited Papers*

**1:35**

**2pSA1. Comparing the effectiveness of structure-preserving model order reduction methods for vibro-acoustic problems.** Quirin Aumann (Max Planck Inst. for Dynamics of Complex Technical Systems, Sandtorstr. 1, Magdeburg 39106, Germany, aumann@mpi-magdeburg.mpg.de) and Steffen W. Werner (Courant Inst. of Mathematical Sci., New York Univ., New York, NY)

For the analysis of the vibration and dissipation behavior of structures, the evaluation of the systems' frequency response is essential. However, evaluating a frequency sweep on a high-fidelity model can be computationally very expensive. A remedy to this are model order reduction methods, which allow the computation of cheap-to-evaluate surrogate models. Additionally to the reduction of the internal system dimensions, model order reduction methods that preserve the matrix structure of the transfer function are preferred as the results may allow for physical reinterpretation and the use of similar analysis tools as for the original systems. In this work, we compare system-theoretic, structure-preserving model order reduction methods with a specific focus on their applicability to vibro-acoustic systems. We demonstrate their effectiveness in terms of accuracy and computational costs by applying them to numerical models of vibro-acoustic systems depicting structural vibration, sound transmission, acoustic scattering, and poroelastic-acoustic coupling. As a result, this comparative study allows us to derive some general recommendations for the choice of suitable model order reduction methods depending on the specific problem statement.

**1:55**

**2pSA2. Surrogate modeling in structural vibration problems with dynamic mode decomposition.** Frank Blubaugh (Naval Surface Warfare Ctr., 9500 MacArthur Blvd., B3 Rm 317, West Bethesda, MD 20916, francis.c.blubaugh.civ@us.navy.mil)

Solving large-scale vibration problems presents a difficult challenge for complicated geometries that is only addressable through the use of large-scale compute resources. These problems not only require extensive compute time and cost, but are also expensive in engineering hours for modeling and analysis. While optimization techniques can help limit engineering time in the loop, the computational requirements of these models have made applying traditional optimization techniques to this class of problem untenable. Dynamic Mode Decomposition (DMD) is an approach that can help bridge this gap by building high-fidelity surrogate models allowing for the inline development of a surrogate model to dramatically reduce the computational time and complexity of a problem. This approach enables fast frequency sweeping while capturing the dominant dynamics of the model. DMD has historically been applied to other high-data volume models such as computational fluid dynamics, medical imaging and controls, as well as less structured problems including sociology and market behavior. This paper will cover the fundamentals of Dynamic Mode Decomposition, the extension of the theory to apply the technique to vibration problems, the demonstration of a potential workflow, and show how this technique performs on simple test cases illustrating performance speed ups for classic problems from existing literature.

**2:15**

**2pSA3. Surrogate models of complex dynamic systems from energy distributions.** James G. McDaniel (Mech. Eng., Boston Univ., Dept. of Mech. Eng., Boston, MA 02215, jgm@bu.edu) and Allison Kaminski (Mech. Eng., Boston Univ., Boston, MA)

Engineers who design complex dynamic systems do not have the luxury of iteration. Numerical analysis of a single design, typically by finite element analysis, requires enormous amounts of CPU time. Therefore, improving a design by trying out many design modifications is impossible. Moreover, it is also impossible to identify which subsystem designs have the largest effects on the dynamic response

of the system. The present work addresses this challenge by constructing surrogate models from energy distributions in a nominal design. These surrogate models are surprisingly effective at identifying subsystems that have the largest effects on the dynamic response, and therefore guide engineers to design modifications that matter most. A significant advantage of these surrogate models is the very low cost of creating them, as they are created directly from the frequency response of a nominal design and therefore do not require additional linear solves or other time-intensive calculations. This presentation begins with an analytical justification for considering energy distributions as surrogate models. Next, Monte Carlo Simulations are presented to show strong correlations between energy distributions and the effects of design changes on dynamic responses. These simulations include random structures as well as beams. Work supported by ONR under Grant N00014-19-1-2100.

## *Contributed Paper*

**2:35**

**2pSA4. On the combination of dynamic substructuring and surrogate modeling for the design and study of complex systems.** Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, anthony.l.bonomo.civ@us.navy.mil) and Dylan D. Nguyen (Naval Surface Warfare Ctr., Carderock Div., West Bethesda, MD)

For decades, dynamic substructuring approaches have been successfully utilized for the modeling and simulation of complex systems. More recently, surrogate modeling methods have emerged as a way to reduce the computational cost of simulating for design. This talk will introduce a paradigm in which these two techniques may be combined for the design and study of complex systems. For illustration, a simple example is put forward in which a mass-spring system is broken into subsystems and parameterized. The open-source Python package SMT (Surrogate Modeling Toolbox) developed and maintained by Bouhel et al. is used to construct surrogate models for each subsystem in which the component parameters are taken as inputs and the frequency response functions (FRFs) of the interface degrees of freedom are output. These FRFs are then used to construct the response of the fully coupled system by utilizing a dual Frequency Based Substructuring technique. In many real-world applications, this manner of calculating the effect of varying design parameters on the system response is likely more practical and computationally feasible than the alternative of directly simulating the coupled system response for many parameter sets and training a surrogate model from that output for further design space exploration and optimization. [Work supported by the Office of Naval Research]



**2:50–3:05 Break**

## *Invited Papers*

**3:05**

**2pSA5. On machine learning-driven surrogates for sound transmission loss simulations.** Barbara Z. Cunha (Lab. of Tribology and System Dynamics, Ecole Ctr. de Lyon, 36 Av. Guy de Collongue, E6, Écully 69134, France, barbara.zaparoli-cunha@ec-lyon.fr), Abdel-Malek Zine (Inst. Camille Jordan, Ecole Centrale de Lyon, Ecully, France), Mohamed Ichchou (Lab. of Tribology and System Dynamics, Ecole Centrale de Lyon, Ecully, France), Christophe Droz (Univ. Gustave Eiffel, Inria, COSYS/SII, I4S team, Rennes, France), and Stéphane Foulard (Compredict GmbH, Darmstadt, Germany)

Surrogate models are data-based approximations of computationally expensive simulations that enable efficient exploration of the model's design space and informed decision-making in many physical domains. The usage of surrogate models in the vibroacoustic domain, however, is challenging due to the non-smooth, complex behavior of wave phenomena. This work investigates four Machine Learning approaches in the modeling of surrogates of Sound Transmission Loss. Feature importance and feature engineering are used to improve the models' accuracy while increasing their interpretability and physical consistency. The transfer of the proposed techniques to other problems in the vibroacoustic domain and possible limitations of the models are discussed.

**3:25**

**2pSA6. Generalized polynomial chaos based surrogate models for acoustics and vibrations.** Andrew S. Wixom (Appl. Res. Lab., Pennsylvania State Univ., P.O Box 30, Mail Stop 3220B, State College, PA 16801, axw274@psu.edu)

This work explores generalized polynomial chaos (GPC) surrogate models and their effectiveness for general acoustics and vibration applications. GPC is primarily known as an uncertainty quantification (UQ) technique and in that context the underlying polynomial-based model has been shown to be effective in mapping input probability distributions to the corresponding output probability distributions. In this study, GPC surrogate models – including those generated by both quadrature and regression methods – are evaluated for their effectiveness in non-UQ focused analyses. As points of comparison, Krylov subspace and other more traditional reduced order modeling techniques are demonstrated and compared to GPC models so that the differences may be better understood. Example problems with several different levels of complexity are used to show how the computational burden as well as the overall effectiveness of the method changes as the number of input variables that must be considered grows.

## *Contributed Papers*

**3:45**

**2pSA7. A continuous system for testing fuzzy structure concepts.** Jerry H. Ginsberg (5661 Woodsong Dr., Dunwoody, GA 30338, j.h.ginsberg@comcast.net)

Much effort had been devoted to studying fuzzy structure models in which the master structure is a large rigid body. At a time when activity in fuzzy structures began to wane, a paper by Drexel and Ginsberg [J. Vib. Acoust. **123**, 181–187 (2001)] provided a quantitative model for testing the validity of assumptions embedded in the notion of a fuzzy structure. Its model was a cantilever beam with an finite number of attached SDOF oscillators. A state space formulation in conjunction with a Ritz series led to responses in the frequency and time domains. The investigation used idealized properties in which the attached SDOFs are positioned uniformly along the span and their fixed base natural frequencies are equally spaced around a center frequency that equals an isolated natural frequency of the beam. In addition, the natural frequency of each SDOF was taken to increase linearly with distance from the fixed end of the beam. The presentation will provide the details of the model and analysis, then highlight the similarites and differences of responses obtained from the models of the master structure as a rigid body and a cantilver beam.

**4:00**

**2pSA8. Exploration of nonideal effects in a beam model of a fuzzy structure.** Jerry H. Ginsberg (5661 Woodsong Dr., Dunwoody, GA 30338, j.h.ginsberg@comcast.net)

The presentation that preceded this one analyzed a model of a fuzzy structure as a beam with multiple suspended SDOFs. It examined the degree to which phenomena encountered in a model of the master as a rigid body are replicated when the base displacements of the attachments are unequal. The conclusion was that many response features are replicated in the frequency domain, whereas contributions of higher structure modes cause time domain responses to differ. The present paper examines whether the numerous idealizations that led to that conclusion are crucial. Parameters that are examined are tuning and bandwidth of the fixed base SDOF frequencies, and randomness of the spectral and spatial distribution of the fixed base SDOF natural frequencies. Responses extracted from an “exact” state space solution of the equations of motion show that minor randomness can significantly alter, and sometimes obscure, the phenomena observed in the ideal case. The results bring to the fore the question of whether phenomena observed in idealized fuzzy models are relevant to actual structures.

## Session 2pSC

## Speech Communication: Healthcare Settings and Clinical Population (Poster Session)

Sarah Colby, Cochair
*Psychological & Brain Sci., Univ. of Iowa, Psychological & Brain Sci. Bldg., 340 Iowa Ave., Room G60, Iowa City, IA 52242*

Christopher C. Heffner, Cochair
*Communicative Disorders and Sci., Univ. at Buffalo, 122 Cary Hall, South Campus, Buffalo, NY 14214*

All posters will be on display from 1:00 p.m. to 5:00 p.m. Authors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.



### *Contributed Papers*

**2pSC1. Abstract withdrawn.**

**2pSC2. The impact on speech with different deep brain stimulation settings in Parkinson's disease.** Emily Q. Wang (Commun. Disorders & Sci., Rush Univ. Medical Ctr., 600 S. Paulina St., Suite 1017, Chicago, IL 60612, emily_wang@rush.edu), Maura Mullen (Boston VA Health System, West Roxbury, MA), Jessica A. Karl, and Leonard A. Verhagen Metman (Neurological Sci., Rush Univ. Medical Ctr., Chicago, IL)

Parkinson's disease (PD) is a progressive neurodegenerative disorder, characterized by the progressive degeneration of the nigrostriatal dopaminergic pathway which affects movement initiation and maintenance, causing hypokinetic dysarthria in PD. Deep brain stimulation (DBS) of the subthalamic nucleus (STN) has been widely used to manage rigidity, bradykinesias and drug-induced dyskinesias. While STN DBS successfully manages many symptoms of PD, speech response has been widely reported as "no or negative impact". The High-Frequency Setting (HFS) is more focused within the STN which is known to control appendicular symptoms. However, the unintentional HFS current spreading to adjacent brain areas, such as the corticospinal and corticobulbar tracts, was thought to contribute to the deterioration of axial symptoms, including speech. The Low-Frequency Setting (LFS) extends more dorsally and ventrally towards surrounding structures, hence does not show the same negative impact and may reduce the negative impact on the axial functions including speech. This double-blind, randomized perception study assesses the impact of two deep brain stimulation (DBS) settings (high-frequency stimulation and interleave-interlink dual-frequency paradigm) on speech produced by 20 patients with PD. The impact of the two testing conditions was different for different Syllable Types with significant negative correlation between the articulatory rate and articulatory accuracy.

**2pSC3. A comparison between tongue dorsum configurations during vowel production in hypernasal speech vs typical speech.** Hedieh Hashemi Hosseinabad (Communication Sci. and Disorders, Eastern Washington Univ., 310 N Riverpoint Blvd., Spokane, WA 99202, hhosseinabad@ewu.edu)

Appropriate nasalization is important for intelligible speech production and can be impacted by various disorders, including a cleft palate. Nasalization is achieved primarily through coupling the nasal and oral cavities by opening the velopharyngeal (VP) port, which introduces a variety of acoustic features. Although VP control has the primary role in nasalization, evidence suggests articulatory changes such as lip rounding (constriction) enhances acoustic features of nasality, while a downward and forward movement of the tongue dorsum in high vowels could attenuate perceptions of nasality in individuals with competent VP function. There is yet scant evidence to indicate whether such systematic changes occur in the case of VP opening. In an attempt to further understand the clinical aspect of this concept, we evaluated the tongue height and forwardness using ultrasound in three speakers with VP insufficiency (VPI) and three speakers with typical VP function to investigate the tongue configuration across two conditions. Tongue dorsum excursion and tongue positioning index were captured using Assistive Articulation Assessment (AAA) software during the production of vowels /a/ and /i/ across VCV contexts. Further results will be discussed in the meeting.

**2pSC4. Prosody analysis as a tool for differential diagnosis of cognitive impairment.** Chorong Oh (Communication Sci. and Disorders, Ohio Univ., 1 Ohio Univ. Dr., Grover Ctr. W235, Athens, OH 45701, ohc@ohio.edu), Richard J. Morris (Communication Sci. and Disorders, Florida State Univ., Tallahassee, FL), and Xianhui Wang (Otolaryngology, Univ. of California - Irvine, Athens, OH)

This study was conducted to determine whether acoustic analysis of emotional prosody can assist in differential diagnosis of cognitive impairment. Speech samples describing the Cookie Theft picture were obtained from the Dementia Talkbank and analyzed acoustically. Included in this speech dataset were 10 people with dementia of the Alzheimer's type, 9 people with mild cognitive impairment, 5 people with vascular dementia, and 10 neurotypical controls. The principal component analysis revealed five factors with two to five acoustic measures per factor for differential diagnosis of cognitive impairment.

**2pSC5. Cortical processing of speech in high-school students with different sleeping habits.** Jefford shau (Stuyvesant High School, New York, NY) and Yan H. Yu (Communication Sci. & Disorders, St. John's University, 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

Sleep plays a critical role on cognitive function especially in the developing brain. However, sleep duration and sleep timing are marked by individual differences. It is currently unknown whether and how sleep habits modulate speech perception in the developing brain. This study examined the correlation between sleep duration and brain responses to speech contrast in high-schoolers. A passive-listening oddball paradigm was used to deliver the speech contrast, and electroencephalogram (EEG)/Event-related potentials (ERPs) were used to collect cortical responses. Preliminary results indicated that there is a negative correlation between mismatch negativity responses and sleep duration in high-schoolers. The implication of the study will be discussed.

**2pSC6. Acoustic comparison and speech-based biomarkers for the detection of gastroesophageal reflux disease: Research-grade versus iPad recordings.** David Aka (Otolaryngol., Mayo Clinic, 200 1ST ST SW, Rochester, MN 55905, aka.david@mayo.edu), Diana M. Orbelo (Otolaryngol., Mayo Clinic, Rochester, MN), Mary Pietrowicz (Appl. Res. Inst., Univ. of Illinois at Urbana Champagne, Champaign, IL), Keiko Ishikawa (Communication Sci. and Disorders, Univ. of Kentucky, Lexington, KY), Manoj Yarlagadda, Kevin Buller, Amrit Kamboj, and Cadman Leggett (Gastroenterol., Mayo Clinic, Rochester, MN)

Pietrowicz *et al*. (2022) demonstrated feasibility of detecting potential speech-based biomarkers to identify gastroesophageal reflux disease (GERD) and Barrett's esophagus (BE); however, speech samples were obtained using research-grade recording equipment (TASCAM DR-40X with AKG C555L microphone). The use of more readily available equipment such as the Apple iPad with a native microphone could simplify data collection and allow access to a broader range of participants. Therefore, this study compared the research-grade recordings to iPad recordings for feasibility of detecting GERD and BE. One-hundred-fifty adults, clinically classified as GERD negative or GERD positive with or without BE, recorded a paragraph (The Rainbow Passage) simultaneously on both a TASCAM and iPad. Both data sets were analyzed manually for Alpha Ratio and Cepstral Peak Prominence (CPP) with Praat. Initial results show significantly higher alpha ratio and lower CPP in iPad recordings, indicating greater noise energy in the high-frequency range. Additionally, the capability of machine and deep learning model types to discern across GERD negative, GERD positive, BE, and normal voices were compared. Detection across these conditions and comparison of acoustic findings were explored to determine usefulness of the less controlled option of recording using an iPad with its internal microphone.

**2pSC7. The effect of contracting COVID on speech perception in high-school students: Evidence from the brain.** Shaun Karani (Bronx High School of Sci., Queens, NY) and Yan H. Yu (Communication Sci. & Disorders, St. John's Univ., 4631 216 St., Bayside, NY 11361, yanhyu@gmail.com)

Accumulating evidence suggest that some individuals who recovered from COVID experience "brain fog" aside from other organ dysfunctions. Many young individuals who contracted with COVID have relatively mild symptoms and rapid recovery. However, it is unknow whether COVID have any negative impact on their brain function. The current study compared the neural response to speech sounds in high-schooler with and without a history of contracting COVID. An event-related potential paradigm was used, and the electro-encephalogram (EEG) waves were timelocked to each speech stimulus. The mismatch negativity responses (the different between the standard and deviant sounds) were compared between the two participant groups. Preliminary analysis suggested that there are some differences between the experiment group and the control group. Further studies need to examine the long-term effect of COVID on brain and cognitive function in the young population.

**2pSC8. Quantifying continuous child speech for automated detection of speech impairment.** Marisha Speights Atkins (Communication Sci. and Disorders, Northwestern Univ., Frances Searle Bldg., 2240 Campus Dr., Evanston, IL 60208, marisha.speights@northwestern.edu) and Joel MacAuslan (Speech Technol. & Applied Res. Corp., Lexington, MA)

The SpeechMark® Automated Syllabic Cluster detection system was tested as a novel approach for analysis of continuous speech samples recorded from 4-year-old children classified as typically developing (TD, N = 44, M = 4.32 years, SD = 0.64) and with speech compromise (SC, N = 16, M = 4.14 years, SD = 0.66). The speakers were recruited in the Midwest and Southern regions of the United States. To test if the TD group produced higher syllabic clusters compared to the SC, we fit a generalized linear mixed effects model. The model adjusted for the potential influence of age and dialect in contributing to the group differences by including them as covariates. Results were interpreted using incidence rate ratios (IRR). Results showed that the IRR was dependent on age indicated by the significant interaction term group*age (p-value = 0.003). The results also showed that there was no difference between the two dialect groups. Results from linear mixed effects models showed that the speech rate was higher among speakers in SC group given all other factors held constant (effect = 0.9, p-value = 0.055). These findings are promising as we aim to automate the analysis of continuous speech samples of young children.

**2pSC9. Rapid quantitative measure of speech intelligibility through personal protective equipment facemasks using two head & torso simulators.** Daniel J. Rogers (Medical Solutions Div., 3M, Ctr., Bldg. 270-4N-09, St. Paul, MN 55144-1000, djrogers@mmm.com) and Nicholas Lee (Consumer Health & Safety Div., 3M, St. Paul, MN)

A test method has been developed that provides a rapid and quantitative measure of speech intelligibility via speech transmission index (STI) through personal protective facemasks using two Brüel & Kjær Head and Torso Simulators (HATS) under a variety of acoustical environmental conditions. STI measurements were made using this new Test Method (TM) on a collection of 3M prototype and production facemasks and respirators, a competitor facemask, and a "no mask" baseline. This new TM enables rapid differentiation among facemask designs. Comparisons among the set of face coverings shows a clear tradeoff of filtration level for STI. New 3M prototypes with 90% filtration achieved an Excellent STI of 0.85, only 0.03 below a perfect no mask baseline. There is a dramatic step from maximum STI with no filtration at all to Excellent STI with 90% filtration. Most of the 95% filtration masks had STIs clustered near the lower end of the Excellent band at 0.78. P100 filtration significantly compromised speech intelligibility in the 3M 6300 half facepiece reusable respirator with a Fair quality rating on the STI scale at 0.60.

**2pSC10. Spectral changes to sound exposure when preterm infants transition from the incubator to an open crib.** Rohit M. Ananthanarayana (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61820, rohitma2@illinois.edu) and Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Children born preterm are at risk for speech and language developmental delays and disorders. It has been proposed that adverse auditory exposures during stay in the neonatal intensive care unit (NICU) may contribute to this risk, as it is well-established that the preterm infant's developing auditory system is sensitive to acoustic input during this time. While reduced speech exposure, noxious noise levels, and excessive silence in the NICU are of concern, another potential cause for concern is abrupt changes in auditory exposure during NICU stay. Our previous data indicate that NICU incubator walls have a low-pass filtering effect, attenuating external sounds at frequencies above 200 Hz. However, internal sounds generated by the incubator or other life-saving devices still produce noise exposure. Furthermore, when infants transition to an open crib from the incubator based on improved overall health, they may be at risk of increased exposure to higher frequencies. Here we present spectral analyses of auditory exposure recordings made for several preterm infants throughout NICU stay. Our analysis reveals that for frequencies above 500 Hz, sound levels are significantly higher in the open crib; below 500 Hz, levels are generally higher in the incubator. These data point to yet another potentially disruptive effect of the NICU environment caused by an abrupt change in auditory exposure.

**2pSC11. The impact of COVID on the teenagers' brain: Changes in brain responses to music.** Jeffrey Yang (Bronx High School of Sci., 75 W 205th St., Bronx, NY 10468, yangj13@bxscience.edu) and Jungmoon Hyun (Speech Language Pathol., Hunter College, New York, NY)

A variety of cognitive- and health-related issues have been documented as post-COVID symptoms. However, it is unknown how COVID has affected young adults' brain responses to sounds, especially to musical acoustics. The current study compared brain responses to music between teenagers with a history of mild COVID and teenagers without COVID. A total of 16 teenagers, aged 14–17 participated in this ERP (event-related potential) study. Their cortical responses to the changes in six acoustical components (frequency, rhythm, duration, amplitude, location, and timbre) were recorded using 65-channel electrodes in a passive listening paradigm. Preliminary results suggested different brain response patterns between the two groups. The characteristics of group differences vary across the six acoustical changes. Detailed information on six acoustical components will be further discussed. This study's results revealed COVID contracting teenagers' neuronal changes that may not be noticeable in the activities of daily life, but it alerts the need for longitudinal studies to examine the post-COVID effect on brain functions in the developing population.

**2pSC12. Characteristics of infant-directed speech in caregivers with traumatic experiences.** Bianca Welikson (Psychological & Brain Sci., Univ. of Lousiville, 317 Life Sci. Bldg., Louisville, KY 40242, bianca.welikson@louisville.edu), Maria V. Kondaurova, Rebekah Flowers, and Lora Haynes (Psychological & Brain Sci., Univ. of Lousiville, Louisville, KY)

Mothers with depressive symptoms interact differently with their children compared to mothers without depressive symptoms. Compared with non-depressed mothers, depressed mothers use a lower mean pitch and less extended pitch range, speak less, and respond slower to their infants. Are the characteristics of infant-directed speech (IDS) affected by a caregiver's traumatic experiences? The study will examine characteristics of IDS in two groups of mothers (N=30); one group exposed to traumatic experiences and the other not exposed to traumatic experiences as assessed by the Adverse Childhood Experiences scale. The mean length of utterances (MLU, the number of morphemes per utterance), mean fundamental frequency (Hz), fundamental frequency range (Hz), speech rate (syllabus per utterance duration), and utterance duration (seconds) will be measured in maternal speech. Based on studies with depressed mothers, it is expected that mothers exposed to traumatic experiences will produce shorter MLU, lower fundamental frequency, less expanded fundamental frequency range, slower speech rate, and shorter utterance duration compared to mothers without traumatic experiences. The research will contribute to the development of early intervention strategies that will aim to strengthen language acquisition in children from families exposed to trauma.

**2pSC13. The effect of a social robot mediator on speech characteristics of children with autism spectrum disorder.** Aamira Shah (Psychological and Brain Sci., Univ. of Louisville, 2301 S 3rd St., Louisville, KY 40292, a0shah11@louisville.edu), Maria V. Kondaurova (Psychological & Brain Sci., Univ. of Louisville, Louisville, KY), Robert Pennington (Special Education and Child Development, Univ. of North Carolina Charlotte, Charlotte, NC), Karla C. Welch (Elect. and Comput. Eng., Univ. of Louisville, Louisville, KY), Grace M. Kuravackel (Norton Children's Autism Ctr., Univ. of Louisville School of Medicine, Louisville, KY), and Qi Zheng (Bioinformatics & Biostatistics, Univ. of Louisville, Louisville, KY)

Robot-mediated interventions have been investigated for the treatment of social skill deficits amongst children with Autism Spectrum Disorder (ASD). Does the use of a Nao robot as a mediator increase vocal interaction between children with ASD? The present study examined the vocalization and turn-taking rate in six children with ASD (mean age = 11.4 years, SD = 0.86 years) interacting with and without a Nao robot for 10 sessions, order counterbalanced. Each session lasted nine minutes. In the Robot condition, the robot provided vocal prompts; in the No Robot condition, children interacted freely. Child vocalization and turn-taking rate defined as the number of utterances/turns per second were measured. Results demonstrated that three children produced higher vocalization and turn-taking rates when a robot was present, and two when it was absent. One participant produced higher vocalization rates when the robot was not present, but more conversational turns when the robot was present. The findings suggest that the use of a Nao robot as a social mediator increases vocalization and turn-taking rates among children with ASD, but large individual variability is observed. The effect of the robot as a mediator on lexical diversity of child speech will also be investigated.

**2pSC14. Acoustic analysis of spatiotemporal variability in children with childhood apraxia of speech.** Janet Vuolo (The Ohio State Univ., Columbus, OH) and Alan Wisler (Utah State Univ., 3900 Old Main Hill, Logan, UT, alan.wisler@usu.edu)

Childhood apraxia of speech (CAS) is a pediatric neurological motor speech disorder characterized by impaired speech execution in the absence of neuromuscular deficits. Children with CAS show distinct speech features, including sequencing, coarticulation, and prosodic deficits, compared to children with other speech disorders. The core deficit in CAS is at the level of planning and programming the precise spatiotemporal parameters of movement sequences necessary to produce natural-sounding speech. This study will investigate spatiotemporal variability in children with CAS compared to children with typical development (TD) using the acoustic spatiotemporal index (aSTI). The aSTI measures spatiotemporal variability in the amplitude envelope of speech sounds across multiple repetitions of an utterance; higher aSTI values reflect higher movement variability. In the current project, children with CAS and TD were recruited nationally to participate in an online study investigating language and motor skills. Children produced 10 repetitions of "Buy Bobby a puppy" and "Mom pets the puppy"; we analyzed these two sentences using the aSTI. Data from 20 children (CAS = 10; TD = 10), ranging in age from 7- to 12-years-old, will be presented. We predict that children with CAS will show higher aSTI values compared to children with TD.

**2pSC15. Repeat and rephrase: Assessing implementation of communication strategies in speakers with Parkinson's disease.** Annalise Fletcher (Communicative Disorders and Deaf Education, Utah State Univ., Logan, UT), Brian Nalley (Mathematics and Statistics, Utah State Univ., Logan, UT), Samantha Budge (Communicative Disorders and Deaf Education, Utah State Univ., Logan, UT), and Alan Wisler (Mathematics and Statistics, Utah State Univ., 3900 Old Main Hill, Logan, UT, alan.wisler@usu.edu)

When encountering challenging communicative situations, speakers with dysarthria are commonly advised to rephrase their sentences, avoiding jargon and using predictable words. However, there are no standard protocols for assessing this skill. The current study explores a short educational protocol focused on rephrasing complex sentences. There are two primary research questions: 1) what (if any) acoustic and lexical changes occur when speakers are prompted to rephrase sentences, and 2) can speakers with dysarthria increase their intelligibility when repeating or rephrasing a message? Speech samples were collected from eleven speakers with Parkinson's disease. In a baseline condition, speakers read 29 sentences from the Natural Stories Corpus. Following this, speakers received verbal instructions on how to rephrase statements. In the rephrased condition, speakers were given the same stimuli and prompted to repeat the story using different words if desired. Preliminary results revealed statistically significant reductions in lexical diversity following the education protocol. Mean-segmented type-to-token (MSTTR) ratio was 7.57% lower in the rephrased condition, a difference which was significant at the $p = 0.001$ level. Although differences varied across participants, none showed increases from the original passage. Evaluations of speaking rate and intelligibility in the baseline and rephrased conditions will also be presented.

**2pSC16. Lexical effects on talker discrimination in adult cochlear implant users.** Terrin N. Tamati (Dept. of Otolaryngol., Ohio State Univ., 1608 Aschinger Blvd., Columbus, OH 43212, terrintamati@gmail.com), Almut Jebens, and Deniz Baskent (Dept. of Otorhinolaryngol./ Head and Neck Surgery, Univ. Medical Ctr. Groningen, Groningen, Netherlands)

The lexical and phonological content of an utterance impacts the processing of talker-specific details in normal-hearing (NH) listeners. For example, previous research has shown that NH listeners display stronger perception of talker-specific details for real words compared to nonwords, and for nonwords with high phonotactic probability compared to nonwords with low phonotactic probability. For adult cochlear implant (CI) users, limitations in the talker-specific details conveyed by their devices may alter the reliance on lexical information in talker discrimination. The current study examined the impact of lexical content on talker discrimination in adult CI users. In a same-different talker discrimination task, word pairs – produced

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either by the same or different talkers – were either lexically easy (high frequency, low neighborhood density) or lexically hard (low frequency, high neighborhood density). Results showed a significant interaction between lexical content and talker pair. For same-talker pairs, accuracy was higher for lexically easy words than lexically hard words. For different-talker pairs, accuracy was higher for lexically hard words than lexically easy words. Despite limitations in talker discrimination, these preliminary results suggest that adult CI users use lexical information in the processing of talker-specific details, and additionally may rely less on acoustic-phonetic details when processing is easy.

**2pSC17. A descriptive study on the acoustic and kinematic characteristics of internal open junctures: Implications for highly intelligible individuals with Parkinson's disease.** Christina Kuo (James Madison Univ., 235 Martin Luther King Jr. Way, MSC 4304 James Madison Univ., Harrisonburg, VA 22807, kuocx@jmu.edu)

This study extends previous work on the acoustic characteristics of ambiguous sentences for male speakers with Parkinson's disease (PD) [Kuo, JASA **148**(4), 2583 (2020)] with additional data. Specifically, the purpose of this study is to further understand acoustic and articulatory characteristics of internal open junctures for speakers with PD. Internal open junctures consist of set strings of phonemes that are phonetically and/or semantically ambiguous when there exist two or more ways to segment the phoneme strings [Lehiste, Phonetica 5(Suppl. 1), 5–54 (1960); Fisher & Logemann, Q. J. Speech 53(4), 365–373 (1967)]. As a part of a larger study, speech acoustic output and tongue kinematic data from four male speakers of American English with PD and two neurologically healthy males were captured simultaneously using 3D electromagnetic articulography. Vocalic segments of internal open junctures were identified acoustically from ambiguous sentence productions, in which the internal open junctures were obtained in pairs, with each phoneme string contributing to two different word combinations. The second formant (F2) and tongue kinematic trajectories are compared for these pairs. It is hypothesized that for the same phoneme string, varied linguistic boundaries elicit differences in tongue trajectory patterns and F2 changes in duration and speed.

**2pSC18. Effects of laryngeal vibratory asymmetry on voice acoustics and perception.** Yoonjeong Lee (Head and Neck Surgery, David Geffen School of Medicine at UCLA, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeol@umich.edu), Neha Reddy, Hye Rhyn Chung, Zhaoyan Zhang, and Dinesh Chhetri (Head and Neck Surgery, David Geffen School of Medicine at UCLA, Los Angeles, CA)

Laryngeal vibratory asymmetry is commonly observed in many talkers, but its acoustic and perceptual consequences remain poorly understood. This study evaluates the relationships among vibratory symmetry, acoustic measures, and voice quality perception. We first test how the degree of left-right asymmetry in thyroarytenoid (TA) muscle activation relates to changes in voice quality as quantified by cepstral peak prominence (CPP). Using an *in vivo* canine phonation model, we stimulated each of the left and right TA muscles in a graded manner. Left-right asymmetry in vocal fold vibration was visually assessed, and CPP was measured from stable phonation. Results show that CPP is the highest at conditions of symmetric vibration and significantly decreases with increasing levels of asymmetry. In the subsequent audio rating experiment, 89 listeners ranked the phonation samples based on the quality (from worst to best). Overall, listeners prefer voices with higher CPP over those with lower CPP and voices with symmetric vibration over those with asymmetries. Results further reveal that listeners do not distinguish voices that fall within certain degrees of asymmetry. The relation between neuromuscular asymmetry and phonatory sound quality identified in this study has implications for clinical evaluation of the voice. [Work supported by NIH.]

**2pSC19. Loudness and rate perception in Parkinson's disease.** Christopher C. Heffner (Communicative Disorders and Sci., Univ. at Buffalo, 122 Cary Hall, South Campus, Buffalo, NY 14214, ccheffne@buffalo.edu), Kris Tjaden (Communicative Disorders and Sci., Univ. at Buffalo, Buffalo, NY), and Eduardo Mercado (Psych., Univ. at Buffalo, Clarence, NY)

In speech communication, Parkinson's disease (PD) is typically discussed in terms of its production consequences. However, a small but growing literature indicates that PD may also result in deficits in the perception of loudness and rate information in speech. In the present study, we focus on the extent to which the perceptual deficits in PD are limited to speech or shared with other modalities, and the degree to which these deficits are dependent solely on perception or are also found in tasks of learning. To assess these questions, people with PD and age-matched controls perform two tasks: an AXB task of speech rate or intensity discrimination, and a simple learning task testing participants' ability to group versions of a sentence based on their rate or intensity properties. For the AXB task, participants perform three additional versions of the task with stimuli that vary in their speech-like properties: time-reversed speech, pure tones, or visualizations. Preliminary data indicates that for control participants, all the continua are discriminated at above-chance performance and that participants can learn to group sentences based on both rate and intensity. Performance for the people with Parkinson's Disease displays a great deal of variability.

**2pSC20. Parameterization of voice onset for automatic assessment of Parkinson's disease.** Tomas Arias-Vergara (Univ. Hospital Erlangen, Waldtrasse 1, Erlangen 91054, Germany, tomas.arias-vergara@uk-erlangen.de), Tobias Schraut (Phoniatrics and Pediatric Audiology, Univ. Hospital Erlangen, ErlangenBavaria, Germany), Juan R. Orozco-Arroyave (Univ. de Antioquia, Medellín, Colombia), and Michael Döllinger (Univ. Hospital Erlangen, Erlangen, Bavaria, Germany)

Acoustic analysis of Parkinson's disease (PD) usually focuses on sustained oscillations of the vocal folds; however, the voice onset (transition from the rest state to a saturation value) is often neglected. In this study, we investigated the parameterization of the voice onset for the objective evaluation of PD. 50 PD patients (25 females) and 50 healthy controls (25 females) performed the sustained phonation of the vowels /ah/, /ih/, and /uh/. Three experts listened to the recordings and assessed the dysarthria level according to a modified version of the Frenchay dysarthria assessment scale (mFDA). We extracted the voice onset automatically, considering the time needed for the amplitude envelope of the acoustic signal to go from 10% up to 90% of the maximum value. Then, we computed filter bank features from the voice onset and pitch, loudness, and perturbation features from the sustained phonation. We performed automatic classification and regression analyses using support vector machines. We obtained a classification accuracy of 89% (AUC: 0.93) when we used feature importance analysis to reduce the feature set. In addition, the regression analysis showed a significant correlation (r: 0.583, p-value $< 0.001$) between the acoustic features and the mFDA scale. These results show the suitability of the voice onset for objective clinical evaluation of PD.

**2pSC21. Engaging effort improves efficiency during word recognition in cochlear implant users.** Sarah Colby (Psychol. and Brain Sci., Univ. of Iowa, Psychol. & Brain Sci. Bldg., 340 Iowa Ave., Room G60, Iowa City, IA 52242, sarah-colby@uiowa.edu), Marissa Huffman, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Cochlear Implant (CI) users must cope with the degraded spectral input received through their device. This reduced quality leads to changes in how words are recognized: lexical access is delayed, leading to differences in competition between lexical candidates. CI users also report increased effort and fatigue during language processing, but effort is not clearly related to improved accuracy. We investigated the role that listening effort plays in the dynamics of word recognition. We related lexical competition and listening effort in 77 post-lingually deaf CI users with Visual World Paradigm and pupillometry tasks. Subjects also completed measures of peripheral auditory function (spectral ripple discrimination and temporal modulation detection). We used the difference in pupil size during the recognition of spoken and written words to isolate effort for spoken word recognition. Difference in pupil size predicted efficiency in the spoken word recognition

task, over and above age and peripheral fidelity. That is, CI users who engage more effort were faster to recognize words and showed lexical competition similar to normal hearing listeners. This interacted with spectral resolution: better discrimination offsets the need to engage effort. This suggests that listening effort can be engaged to improve efficiency during word recognition.

**2pSC22. Effects of spectral smearing on speech understanding and masking release in simulated bilateral cochlear implants.** Margaret Cychosz (Hearing and Speech Sci., Univ. of Maryland, 7251 Preinkert Dr., College Park, MD 20742, mcychosz@umd.edu), Kevin Xu, and Qian-Jie Fu (Dept. of Head and Neck Surgery, David Geffen School of Medicine, UCLA, Los Angeles, CA)

Cochlear implants (CIs) have limited spectral resolution due to the number of activated electrodes and current spread in the cochlea, with ramifications for speech processing. Differences in degree of channel interaction across ears may partially explain variability in outcomes among CI users. This study explores how channel interaction impacts speech understanding in simulated cochlear implants. Speech recognition thresholds (SRTs) were measured in the presence of two speech maskers by 16 normal-hearing subjects listening to simulated bilateral CIs. Speech was delivered to both ears while maskers were delivered binaurally (diotic) or monaurally (dichotic). Stimuli were 16-channel sine-vocoded speech simulating 1) limited channel interaction in both ears, 2) limited channel interaction in one and broad in the other, or 3) broad channel interaction in both. Masking release was quantified as the SRT difference between diotic and dichotic conditions. SRTs were highest with broad channel interaction in both ears; when interaction decreased in one ear, SRTs improved, and improved further with limited interaction in both. Masking release was apparent across all conditions, but especially when the interaction was similar across ears (broad: 6.45 dB; limited: 6.51 dB). These results suggest that channel interaction may impact speech recognition more than masking release.

**2pSC23. CCi-CLOUD: A framework for community based remote cochlear implant user experiments based on the CCi-MOBILE research platform.** Hazem Younis (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, 800 W Campbell Rd., Dallas, TX 75080, hazyounis@yahoo.com) and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

The speech processing field has recently developed a critical need for conducting listener experiments remotely versus traditional in-person methods due to social distancing measures enforced by COVID-19 pandemic. CCi-MOBILE is our current hardware research platform created to support the Cochlear Implant research community, facilitating the development of new strategies & algorithms for improving sound quality and scientific investigation for CI users/systems. In this study, CCi-CLOUD, a virtual platform, is proposed and developed to expand the functionality of CCi-MOBILE to allow for both subject testing and scientific studies to be performed remotely. CCi-CLOUD portal is established with three support categories: (i) *Remote Desktop*: with direct connect support for CI user remote experimentation; (ii) *Cloud Data Storage*: for data sharing among researchers & subjects; (iii) *Online Web App*: serves as a portal which connects CI subjects with the CI research community. Three multimodal sound experiments are considered in this study as a virtual experimental framework including speech recognition, speaker identification, and sound type classification for both CI and normal hearing (NH) users. Experimental evaluation is performed for both In-Lab and Remote/Online scenarios to benchmark experimental protocols and infrastructure validity for CI/NH subjects. Results on consistency between the two modalities are also discussed.

**2pSC24. Maintenance of phonological contrast in speech production of cochlear implant users.** Victoria Sevich (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, sevich.1@osu.edu), Aaron C. Moberly, and Terrin N. Tamati (Dept. of Hearing and Speech Sci., Vanderbilt Univ., Columbus, OH)

Prior work has shown that normal-hearing (NH) talkers produce more distinct vowels in words having a lower word frequency (WF) and higher neighborhood density (ND). Phonological contrast in speech production may differ between post-lingually deafened cochlear implant (CI) users and individuals with NH due to the degraded auditory input provided by CIs. The current study sought to determine whether the same pattern of vowel production exists in CI users. CI users were recorded reading monosyllabic words that were lexically “hard” (low WF, high ND) or “easy” (high WF, low ND). For each participant, vowel dispersion was calculated as the Euclidian distance from the center of the acoustic vowel space. If degraded acoustic input affects phonological contrast in CI users, we would not expect vowel dispersion to differ between vowels in “hard” and “easy” words, contrary to talkers with NH. Preliminary results revealed a significant effect of lexical difficulty on vowel dispersion, indicating that CI users produced more distinct vowels in “hard” words than in “easy” words. These preliminary findings could suggest that degraded auditory input does not substantially affect phonological contrast in vowel production in CI users, or that long-term adaptation to CI input can restore contrast after auditory deprivation.



**2pSC25. Machine learning based estimation of hoarseness severity from sustained vowels.** Tobias Schraut (Div. of Phoniatrics and Pediatric Audiol., Dept. of Otorhinolaryngol., Head and Neck Surgery, Univ. Hospital Erlangen, Waldstrasse 1, ErlangenBavaria 91054, Germany, tobias.schraut@uk-erlangen.de), Anne Schützenberger, Tomas Arias-Vergara (Div. of Phoniatrics and Pediatric Audiol., Dept. of Otorhinolaryngol., Head and Neck Surgery, Univ. Hospital Erlangen, Erlangen, Germany), Melda Kunduk (Dept. of Communication Sci. and Disorders, Louisiana State Univ., Baton Rouge, LA), Matthias Echternach, and Michael Döllinger (Div. of Phoniatrics and Pediatric Audiol., Dept. of Otorhinolaryngol., Head and Neck Surgery, Univ. Hospital Erlangen, Erlangen, Bavaria, Germany)

Acoustic assessment of voice impairment is commonly performed by subjective perceptual evaluation of continuous speech based on a grading system such as the roughness, breathiness, hoarseness (RBH) scale. Here, we present an automatic approach to objectively quantify hoarseness severity based on sustained vowels. For this study, a total of 635 recordings of the sustained vowel /a/ were collected. Temporal, spectral and cepstral features were extracted from one second of each recording. All recordings have an assigned RBH value, which was determined subjectively by an expert based on continuous speech of the respective subject. In order to account for the label noise introduced by different valuation bases, subjects were divided into two levels of hoarseness $H < 2$ and $H \geq 2$. Logistic Regression was employed as classification model, using the resulting output probabilities as continuous severity rating. Relevant features were selected using a sequence of filter methods and backward elimination. The original feature set was reduced from 50 to 5 features. A classification accuracy of 86.73% was achieved on the test set. Detailed evaluation of output probabilities shows strong correlation ($r = 0.81$) with H values, allowing to capture and quantify individual improvement, deterioration or no change of hoarseness at different points in time, e.g. before and after voice therapy. The presented method describes a promising approach for objective evaluation of hoarseness, allowing to quantify treatment progress for patients with voice disorders.

**2pSC26. Cochlear implant users show unexpected flexibility when recognizing words in challenging conditions.** John Muegge (Psychol. and Brain Sci., Univ. of Iowa, 340 Iowa Ave., Iowa City, IA 52242, jmuegge@uiowa.edu) and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

For typical listeners in quiet, lexical access occurs immediately. Multiple candidates are activated at word onset and subsequently updated as the word unfolds. For cochlear Implant (CI) users, this process changes to cope

with their degraded input: CI users delay lexical access until a substantial portion of a word is heard, a profile termed 'wait-and-see'. However, it is unclear whether this profile is adaptive or simply a result of deficient input. We addressed this using a paradigm in which neutral carrier sentences ("now click on the…") were interrupted by periodic bursts of noise, even as target words were always clean. This expected difficulty condition was compared to trials in which both sentences and targets were in quiet or in noise. We used the Visual World Paradigm to assess the timecourse of lexical access and competition. Relative to normal hearing controls, CI users (N = 22) showed a robust wait-and-see profile in noise, delaying target fixations and reducing competition. In trials where noise was expected but the target was clean, CI users initially showed a wait-and-see response but rapidly recovered to reach quiet-like performance. This indicates that wait-and-see may be adaptive for challenging conditions, providing flexibility as listeners process words in real time.

TUESDAY AFTERNOON, 6 DECEMBER 2022 RAIL YARD, 1:00 P.M. TO 3:30 P.M.

## Session 2pSP

## Signal Processing in Acoustics: General Topics in Signal Processing

Trevor Jerome, Chair
*Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., BLDG 3 #329, West Bethesda, MD 20817*

**Chair's Introduction—1:00**

### *Contributed Papers*

**1:05**

**2pSP1. Acoustics as a guide to welding quality.** Dimitra Emmanouilidou (Res., Microsoft, One Microsoft Way, Redmond, WA 98052, Dimitra.emmanouilidou@microsoft.com)

Welding, the process of melting materials with high heat and fusing them during cool-down, is a fabrication technique common across a wide range of industries including construction, automotive, aerospace. Effective monitoring of the welding process is still an open problem. Weld testing and quality control methods require trained experts and specialized equipment and - in the majority - rely on post-welding inspection for defects or irregularities. In this work, we explore the acoustic qualities of welding sounds and their reliability as an automated quality indicator of the welding process. We investigate the most reliable acoustic features for identifying the quality of welding in the presence of environmental interference and factory floor noise, and we show that the sound picked up by a microphone next to the welding station can be sufficient for identifying a bad weld.

**1:20**

**2pSP2. Using a spectral entropy criterion for improved onset picking in ultrasonic testing.** Benjamin Bühling (Dept. of Non-Destructive Testing, Bundesanstalt für Materialforschung und -prüfung (BAM), Unter den Eichen 87, Berlin 12205, Germany, benjamin.buehling@bam.de) and Stefan Maack (Dept. of Non-Destructive Testing, Bundesanstalt für Materialforschung und -prüfung (BAM), Berlin, Germany)

In ultrasonic testing, the time of flight (ToF) of a signal can be used to infer material and structural properties of a test item. In dispersive media, extracting the bulk wave velocity from a received signal is challenging as the waveform changes along its path of propagation. When using signal features such as the first peak or the envelope maximum, the calculated velocity changes with the propagation distance. This does not occur when picking the signal onset. Borrowing from seismology, researchers used the Akaike information criterion (AIC) picker to automatically obtain onset times. In addition to being dependent on arbitrarily set parameters, the AIC picker assumes no prior knowledge of the spectral properties of the signal. This is unnecessary in ultrasonic through-transmission testing, where the signal spectrum is known to differ significantly from noise. In this contribution, a novel parameter-free onset picker is proposed, that is based on a spectral entropy criterion (SEC) to model the signal using the AIC framework. Synthetic and experimental data are used to compare the performance of SEC and AIC pickers, showing an improved accuracy for densely sampled data.

**1:35**

**2pSP3. Direction of arrival estimation using Gaussian process interpolation.** Peter Gerstoft (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, gerstoft@ucsd.edu), Manual Hahmann (Dept. of Elect. Eng., Technical Univ. of Denmark, Kgs Lyngby, Denmark), William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA), Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ), Efren Fernandez-Grande (Dept. of Elect. Eng., Technical Univ. of Denmark, Kgs Lyngby, Hovedstaden, Denmark), and Christoph Mecklenbrauker (Electr. Eng., Technical Univ. of Vienna, Vienna, Austria)

Gaussian processes (GP) have been used to predict acoustic fields by interpolating under-sampled field observations. Using GP interpolation to predict fields is advantageous due to its ability to denoise measurements, and for its prediction of likely field outcomes given a certain field coherence, or in GP terminology, a kernel. While there are many design options for a coherence function, in this study we examine using the radial basis function kernel, the physically based plane wave kernel, and a composition of plane wave kernels representing a certain angular interval of directions. The composite kernel is relevant in ocean acoustics where it is often the case that arrivals can only be within a narrow direction of arrival. We demonstrate that an array sampled with spacing larger than a half wavelength can benefit from GP interpolation, giving less root mean squared error.

**1:50**

**2pSP4. Improved single snapshot beamforming in a noisy environment using the cubic autoproduct.** Nicholas J. Joslyn (Appl. Phys., Univ. of Michigan, 450 Church St., Ann Arbor, MI 48109, njoslyn@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

The presence of random noise is a challenge for remote sensing tasks as success typically diminishes with decreasing signal-to-noise ratio (SNR). To improve performance in noisy environments, signal processing techniques commonly implement snapshot averaging, requiring multiple signal samples. This presentation investigates the improvement offered by the cubic autoproduct for low SNR beamforming with a single snapshot. Prior work has demonstrated the utility of a quadratic product of complex field amplitudes within the signal bandwidth for beamforming and matched field applications. This quadratic field product, known as the frequency-difference autoproduct, is a synthetic estimate of an acoustic field at the difference frequency of the two constituent fields. That formulation is extended here to a cubic product of three complex field amplitudes within the signal bandwidth, termed the cubic autoproduct, capable of mimicking field content at frequencies below, above, and within the signal bandwidth. Important features and the mathematical description of the cubic autoproduct are reviewed before discussing the beamforming approach. Single snapshot beamforming results from experimental data, acquired in a noisy underwater environment, are directly compared between the cubic frequency-difference autoproduct and the acoustic field. [Work supported by ONR and by the US DoD through an NDSEG Fellowship.]

**2:05**

**2pSP5. Group conversation enhancement using distributed microphone arrays with adaptive binauralization.** Manan Mittal (Electr. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1308 W Main St., #119, Urbana, IL 61801, manansm2@illinois.edu), Kanad Sarkar, Ryan M. Corey, and Andrew C. Singer (Electr. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Hearing aids and other listening devices perform poorly in noisy, reverberant venues like restaurants and conference centers with numerous active sound sources. Microphone arrays can use array processing techniques like beamforming to isolate talkers from a specific region in the room while attenuating undesired sound sources. However, beamforming often removes spatial cues and is typically restricted to isolating a single talker at a time. Previous work has shown the effectiveness of remote microphones worn by talkers and adapting the signal at the earpiece to improve the intelligibility of group conversations. Due to the increase in hybrid meetings and classrooms, many spaces are equipped with high throughput, low latency devices including large microphone arrays. In this work, we present a system that aggregates information collected by microphone arrays distributed in a room to enhance the intelligibility of talkers in a group conversation. The beamformed signal from the microphone arrays is adapted to match the magnitude and phase of the earpiece microphones. The filters are continuously updated in order to track motion of both the listeners and talkers.

**2:20–2:30 Break**

**2:30**

**2pSP6. Improving the robustness of spectral estimation to loud transients with a truncated order statistics filter.** David C. Anchieta (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, danchieta@umassd.edu) and John R. Buck (Electr. and Comput. Eng., UMass Dartmouth, Dartmouth, MA)

Underwater acoustic recordings often include loud transients from human or natural sources. The transients cause a positive bias for Welch's average periodogram spectral estimator when estimating the power spectral density of the background environment. Estimators based on single order statistics (e.g., the sample median) avoid the bias caused by outliers at the cost of a higher variance than the sample mean. Schwock and Abadi (2021) showed that, for exponential random variables, the estimator based on the 80th sample percentile has the lowest variance among any unbiased estimator based on a single order statistics. This work tests a hybrid approach between Welch's and order statistics estimators by performing a weighted sum of the quietest subset of ordered samples of the periodograms. By discarding the loudest samples of the periodogram, the truncated linear order statistics filter (TLOSF) reduces the bias caused by loud transients. By combining multiple order statistics into the estimate, the TLOSF achieves a lower variance than the 80th percentile estimator. The TLOSF reduced the MSE by 1dB compared to Schwock & Abadi's 80th percentile estimator for a mixture combining an exponential distribution with 1% outliers 23 dB above the background. On periodograms of shallow water hydrophone recordings, the TLOSF yielded a lower output power in the frequency bins where both the Welch's and 80th percentile estimators had a positive bias due to loud transients. [Work supported by ONR Code 321US.]

**2:45**

**2pSP7. The Prufer transform for the calculation of acoustic normal modes.** Arthur B. Baggeroer (Mech. and Electr. Eng., Massachusetts Inst. of Technol., Room 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

The PrüferTransform for the Sturm-Liouvile (SL) equation was introduced by Prüferin 1923. An excellent reference is Numerical Solutions of Sturm-Liouville Problems by Pryce. Porter in Computational Ocean Acoustics introduced it to ocean acoustics. This author stumbled upon working on state variables at CMRE on 1978. The method derives first order equations for the phase and log-magnitude for second order equation in a phase plane. The essential feature is the phase equation is not coupled to the magnitude, so it may be solved as a nonlinear, first order equation and the surface boundary condition leads to a unique solution. New results for normal modes using the Prüfermethod are introduced. i) equations for the phase shifts and log-magnitudes at layer boundaries such as at the seabed; ii) state variable transformations based upon Ricatti equations and requiring evaluating one transcendental quantity, important for numerical methods; iii) issues of numerical stability for coupling the phase solution across the potential barriers of local sound speed maxima and poorly coupled boundaries; iv) an equation for computing the imaginary part leading to modal attenuation's. The method is fast has survived many students using it, even those writing Matlab versions.

**3:00**

**2pSP8. Alternating projections-based gridless compressive beamforming with co-prime arrays.** Yongsung Park (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, yongsungpark@ucsd.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

Direction-of-arrivals (DOAs) estimation, beamforming, retrieves the angles of several sources from the outputs of receiving a sensor array. Compressive beamforming is a sparse signal recovery approach and has shown superior performance on DOA estimation. To overcome the angular searching grid issue, gridless techniques have been proposed. Most methods require a uniform linear array and use standard convex solvers that are computationally expensive. We propose a gridless compressive beamformer based on alternating projections. This method estimates DOAs by projecting a solution matrix alternatively. One projection works for measured-data-fitting, and the other works for having sparse DOAs. Our approach improves speed and accuracy and deals with arbitrary-shaped linear arrays. We validate the method using experimental data and test the DOA performance for a single snapshot, multiple snapshots with coherent arrivals, and co-prime arrays, a well-known non-uniform array.

**3:15**

**2pSP9. Manifold learning for dynamic array geometries.** Kanad Sarkar (Electr. and Comput. Eng., Univ. of Illinois, Urbana-Champaign, B10 Coordinated Sci. Lab, 1308 West Main St., Urbana, IL 61801-2447, kanads2@illinois.edu), Manan Mittal, Ryan M. Corey (Electr. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Andrew C. Singer (Electr. and Comput. Eng., Univ. of Illinois, Urbana, IL)

Large-scale distributed arrays can obtain high spatial resolution, but they typically rely on a rigid array structure. If we want to form distributed arrays from mobile and wearable devices, our models need to account for motion. The motion of multiple microphones worn by humans can be difficult to track, but through manifold techniques we can learn the movement

through its acoustic response. We show that the mapping between the array geometry and its acoustic response is locally linear and can be exploited in a semi-supervised manner for a given acoustic environment. We will also investigate generative modelling of microphone positions based on their acoustic response to both synthetic and recorded data. Prior work has shown a similar locally linear mapping between source locations and their spatial cues, and we will attempt to combine these findings with our own to develop a localization model suitable for dynamic array geometries.

TUESDAY AFTERNOON, 6 DECEMBER 2022 NORTH COAST B, 1:00 P.M. TO 4:15 P.M.

# Session 2pUW

## Underwater Acoustics and Acoustical Oceanography: Mud Acoustics II


Charles W. Holland, Cochair
*Electr. and Comput. Eng., Portland State Univ., Portland, OR 97207*

Stan Dosso, Cochair
*School of Earth and Ocean Sci., Univ. of Victoria, Victoria V8W 2Y2, Canada*

Allan D. Pierce, Cochair
*Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537*


### *Contributed Papers*

**1:00**

**2pUW1. One track, two experiments four years apart: On the repeatability of geoacoustic inversion on the New England Mud Patch.** Julien Bonnel (Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, British Columbia, Canada), Andrew R. McNeese (ARL:UT, Austin, TX), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Our current knowledge of the geoacoustic properties of the New England Mud Patch (NEMP) is mostly driven by data collected in 2017 as part of the Seabed Characterization Experiment (SBCEX17). In 2021, a modest geoacoustic inversion experiment was performed on the NEMP using a simple and low-cost pair of experimental assets: a "TOSSIT" passive acoustic mooring and an impulsive "RIUSS" (Rupture Induced, Underwater Sound Source). The TOSSIT/RIUSS data were collected on a track that was studied intensively during SBCEX17, but with fundamental differences in oceanographic conditions: a frontal intrusion was present at the experimental site in 2021, creating a strongly stratified sound speed profile (SSP) in the water column, while the water column was essentially iso-speed in 2017. The 2021 TOSSIT/RIUSS data are used to perform geoacoustic inversion using warping and Bayesian trans-dimensional methods. The geoacoustic properties estimated for the 2021 data compare favorably to results obtained with SBCEX17 data, even when the 2021 data are inverted jointly for water-column SSP and seabed parameters. This study demonstrates inversion repeatability on the NEMP using data sets collected years apart and under different (and potentially unknown) oceanographic conditions. [Work supported by the Office of Naval Research.]

**1:15**

**2pUW2. Sound speed profile estimation in a thick mud layer from travel time data.** Charles W. Holland (Electr. and Comput. Eng., Portland State Univ., Portland, OR 97207, charles.holland@pdx.edu)

The sound speed profile in a surficial mud layer is of importance because it controls what is acoustically illuminated within and below the mud layer. A variety of estimates have been made at the New England Mud Patch (NEMP) with some near-surface sound speed gradients as large as 10 $s^{-1}$ and larger. Here, single bounce seabed reflection travel time data are used to infer what mud sound speed gradients are and are not plausible near the central NEMP region where the mud thickness is $\sim$10 m. Time series data were collected on a moored hydrophone from a broadband source with a 1 second repetition rate as the tow ship transited on a radial from the hydrophone to a distance of $\sim$1 km at a speed of 2 m/s. The water column was essentially isovelocity and grazing angles ranged from 4.5° to 84°. It is known that at the base of the mud, there is a mud-sand transition layer (of order 1 m thick) where sound speed gradients can be large. However, above this layer, the travel time data clearly indicate that a mud 10 $s^{-1}$ sound speed gradient is too large. [Research supported by ONR Ocean Acoustics Program.]

**1:30**

**2pUW3. Broadband measurements and modeling of low-grazing angle bottom reflections from muddy marine sediments.** David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu), Dajun Tang, and Peter H. Dahl (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA)

In 2017 and 2022, a series of experiments using explosive sound sources were conducted on the New England Mud Patch (NEMP), an area of the Eastern US Continental shelf defined by its thick muddy seafloor sediments. Broadband measurements from the near-bottom Intensity Vector Acoustic Recorders (IVARs) reveal resonant frequencies in the bottom reflected

signals, which are excited at very low-grazing angles. A signal model based on a layered-bottom reflection coefficient is used to examine the geoacoustic properties of the marine mud, which leads to the conclusion that the sound-speed contrast across the water-mud interface (i.e. index of refraction, n) is less than one ($n < 1$), and also an increase in sound-speed occurs within the deeper mud-sediments ($n > 1$). Measurements in May 2022, replicating the same source and receiver locations from March 2017, also show resonances in the bottom-reflected signal. Differences between the two data sets (2017 and 2022) are primarily caused by the complex oceanography encountered during the spring-time measurements, absent during the isothermal winter conditions. Comparison of the two datasets examines the stability of the sound-speed contrast, including the effect of time-varying temperature at the water-mud interface.

**1:45**

**2pUW4. Broadband acoustic wave propagation in temporally variable range dependent shallow water environment.** Mohsen Badiey (Electr. and Comput. Eng., Univ. of Delaware, Newark, DE), Lin Wan (Electr. and Comput. Eng., Univ. of Delaware, 139 the Green, 301 Evans Hall, Newark, DE 19716, wan@udel.edu), Jhon A. Castro-Correa, and Christian D. Escobar-Amado (Electr. and Comput. Eng., Univ. of Delaware, Newark, DE)

Broadband acoustic wave propagation in shallow water regions involves interactions with the sea surface, sea bottom, and the water column. The topic of shallow water acoustics in the presence of variable water column has been of interest in recent years. Experimental observations have shown the importance of the effects due to the boundaries and the sound speed profile in the water column. However, more work needs to be done to establish ground truth in numerical modeling for temporally variable water column in a range dependent environment. In this paper we examine the broadband propagation on the shelf as well as the range dependent shelf break regions to depict how temporal variability of sound speed could change the dispersion characteristics of the broadband signal. Comparison between modeled and experimental data are provided. [Work supported by ONR OA program.]

**2:00**

**2pUW5. Acoustic modeling of ducted propagation in the New England Mud Patch.** Jade F. Lopez Case (Ocean Eng., Univ. of Rhode Island, 215 S Ferry Rd., Narragansett, RI 32837, jadelcase@uri.edu), Andrew R. McNeese (Appl. Res. Lab., Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Experiments were conducted on the New England Mud Patch in 2017, 2021, and 2022. The 2017 Seabed Characterization Experiment (SBCEX17) utilized Signal Underwater Sound (SUS) charges, model Mk64, to produce an impulsive acoustic waveform. However, recent work in 2022 has additionally utilized the Rupture Induced Underwater Sound Source (RIUSS) to produce a high-amplitude, broadband waveform with minimal bubble oscillations. Results from these experiments suggested the presence of a surficial layer of mud with a sound speed lower than that of the underlying mud and overlying water. The SBCEX22 experiment included the deployment of nine RIUSS devices and the use of Ocean Bottom Recorders (OBX) to measure the acoustic pressure and three components of particle velocity at range of about 1 km. Conductivity, temperature, and depth measurements from CTD surveys were taken from several locations around the mud patch and used to generate sound speed profiles. These were input into the RAM parabolic equation model to analyze the effect of the sound speed in mud on propagation. Results from the RAM modeling indicates that at mid-frequencies (1-3 kHz) the lower sound speed at the top of the mud layer creates a duct where the transmission loss is reduced.

**2:15**

**2pUW6. Salinity, force chains, and creep in muddy sediments.** Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, chotiros@utexas.edu)

The salinity of sediments is not usually measured. Experience with sandy sediments shows that it has no significant effect on the acoustic properties. In muddy sediments, salinity is critical to the skeletal frame, because it causes the clay particles to flocculate, forming an aggregate of larger particles with significant water fraction. It behaves like a granular medium, in which stress is transmitted along random force chains. Mud is known to suffer from creep, and the force chain model fits neatly into creep theory. It may be modeled as a form of stationary creep, which is linear in many respects. No net strain-hardening is involved. The result is a creep model of the skeletal frame that naturally couples into the Biot theory of porous media. It predicts an attenuation that increases linearly with frequency at low frequencies, which is overtaken by viscous attenuation that increases as the second power of frequency and high frequencies. [Work supported by ONR, Ocean Acoustics Program.]

**2:30–2:45 Break**

**2:45**

**2pUW7. Measurements of interface waves in the New England Mud Patch and estimation of the properties of the mud layer and underlying sand.** James H. Miller (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., URI Ocean Eng., Narragansett, RI 02882, miller@uri.edu), Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying-Tsong Lin (Woods Hole Oceanogr. Inst., Woods Hole, MA), and Fin Hoyer (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

The New England Mud Patch is a 13,000 square kilometer area south of Martha's Vineyard and covered by a layer of fine-grained sediments. The water depth is about 70 meters. Below the mud is a sand layer with compressional sound speed of 1745 m/s. (Bonnel, *et al.*, 2018) Scholte and Stoneley waves were generated by the Interface Wave Sediment Profiler (iWaSP), a piezoelectric bender beam transducer which vibrates the seabed. The iWaSP was deployed in experiments in 2017 and 2022 to generate these interface waves. These waves were received at ranges of 70 to 100 meters by bottom-mounted vertical axis geophones in 2017 and Ocean Bottom Recorders (OBXs) with 3-axis geophones and a hydrophone in 2022. The speed of these interface waves is about 90% of the shear wave speed in the elastic medium. Arrivals were detected on the geophones with speeds ranging from 100 to 200 m/s for a mud thickness of about 6 meters. It is hypothesized that these waves propagated on the mud-sand interface. Based on these results, this paper will comment on the properties of the mud layer. [Sponsored by the Office of Naval Research.]

**3:00**

**2pUW8. Bayesian geoacoustic inversion of 1-2 kHz seabed reflection data for layered muddy sediments.** Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, British Columbia V8W2Y2, Canada, minj@uvic.ca), Charles W. Holland (Dept. of Electr. and Comput. Eng., Portland State Univ., Portland, OR), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, British Columbia, Canada), and Jan Dettmer (Dept. of Geoscience, Univ. of Calgary, Calgary, Alberta, Canada)

This paper presents trans-dimensional Bayesian geoacoustic inversion of seabed reflection data for sub-bottom geoacoustic profiles and associated uncertainty estimation at the New England Mud Patch. The data considered here are wide-angle seabed reflection coefficients as a function of grazing angle and frequency measured during the 2017 Seabed Characterization Experiment. Chirp pulses over a frequency band of 1–6 kHz were transmitted by an omnidirectional acoustic source towed by a research vessel at a speed of about 4 knots and recorded at a bottom-moored hydrophone. High signal-to-noise-ratio reflection coefficients from 1–2 kHz and angular coverage of $\sim$15–25° are considered here for geoacoustic inversion. This frequency range is higher than for previous reflection-coefficient data sets on the Mud Patch. The angular range, although relatively narrow, includes strong Bragg resonances which provide information on the sediment layering properties. The inversion applies the viscous grain-shearing sediment acoustics model, which provides dispersive (frequency-dependent) results for sound speeds and attenuations. The seabed structure estimated here is compared to previous inversion results and to core measurements in the vicinity. [Work supported by the Office of Naval Research.]

3:15

**2pUW9. Potential and kinetic energy of underwater noise measured below a passing ship and response to sub-bottom layering.** Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98115-7834, dahl@apl.washington.edu) and David Dall'Osto (Appl. Phys. Lab. at the Univ. of Washington, Seattle, WA)

For underwater sound energy emitted by a passing vessel and received at a relatively steep angles, such as vessel directly overhead, there can be a significant excess in potential acoustic energy relative to kinetic energy with the opposite also occurring. Observations when expressed as a ratio of kinetic to potential energy in decibels are interpretable upon inspection, yielding an estimate of the sensor height above the seabed, and information on properties of the seabed. The effect was studied as part of the recently completed experiment on the New England Mud Patch. The R/V Neil Armstrong underwent controlled transits directly overhead a vector sensor, positioned 1.45 m above the seabed. A model for kinetic and potential energy developed from method of images combined with a layered seabed, is used to invert the data. The low-speed mud-layer is identified along with higher-speed transition layer separating the mud substrate from a sediment basement, with parameters consistent with other recent studies. Similar observations made in Puget Sound are briefly discussed with the higher-impedance seabed identified being consistent with thin Holocene sediments characteristic of region. Both examples illustrate the standing wave features of such data.

3:30

**2pUW10. High-resolution transdimensional geoacoustic inversion using autonomous underwater vehicle data.** Tim Sonnemann (Dept. of Geoscience, Univ. of Calgary, 2500 Univ. Dr. NW, Calgary, Alberta T2N 1N4, Canada, tim.sonnemann@ucalgary.ca), Jan Dettmer (Dept. of Geoscience, Univ. of Calgary, Calgary, Alberta, Canada), Charles W. Holland (Electr. and Comput. Eng., Portland State Univ., Portland, OR), and Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, British Columbia, Canada)

We invert reflection coefficient measurements of muddy sediment layers along a 12-km seabed transect on the Malta Plateau in the Mediterranean Sea using 1711 source transmissions recorded on a 32-element linear hydrophone array with both source and array towed by an autonomous underwater vehicle. Trans-dimensional Bayesian inference using reversible jump Markov chain Monte Carlo sampling is applied to obtain posterior probability densities of the number of homogeneous sediment layers, their depths, and their geoacoustic parameters. The forward sediment acoustics model is based on the grain-shearing model which obeys physical causality and provides correlation between important geoacoustic properties. Each dataset was treated as one-dimensional seabed structure inversion carried out on high performance clusters, and inversion results for multiple data sets were combined to yield a two-dimensional subsurface profile including full uncertainty analysis. Comparisons of inversion results to piston and gravity core estimates show agreement in both geoacoustic parameter values and depths of discontinuities. In the range-dependent model constructed from inverting the entire data set, dipping and terminating layers are observed along the track with high vertical resolution on the order of 10 cm.

3:45

**2pUW11. Broadband sound speed and attenuation of a mud layer from acoustic propagation experiments in the New England Mud Patch.** Lin Wan (Electr. and Comput. Eng, Univ. of Delaware, 139 the Green, 301 Evans Hall, Newark, DE 19716, wan@udel.edu) and Mohsen Badiey (Electr. and Comput. Eng., Univ. of Delaware, Newark, DE)

A series of Seabed Characterization Experiments (SBCEX) have been conducted in the New England Mud Patch area, where a top mud layer overlays a sand sediment. Using broadband acoustic signals from the SBCEX, the sound speed and attenuation of the mud layer as a function of frequency (from a few tens of Hz up to around 1 kHz) have been estimated by various inversion algorithms based on (1) modal dispersion curves and transmission loss [reported at the ASA Spring meeting, 2016], (2) Airy phase [Spring, 2017], (3) modal amplitude [Fall, 2017], (4) spatial coherence measurements [Spring, 2018], (5) mid-frequency modal dispersion analysis [Fall, 2019], and (6) ground waves [Fall, 2022]. This paper summarizes these inverted sound speed and attenuation, which are obtained at various frequencies independently (without using the frequency dependent constraints from any geo-acoustic models), therefore the inverted results could be used to verify the validity of geo-acoustic models. The performances of these inversion methods are evaluated. The performance assessment is helpful for one to select appropriate inversion algorithms and to choose optimal acoustic data for geo-acoustic inversion. [Work supported by ONR Ocean Acoustics.]

4:00

**2pUW12. A forward-backward filter for estimating mud properties in SBCEX 17.** Zoi-Heleni Michalopoulou (Dept. of Mathematical Sciences, New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu), Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA), Diego Rios (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ), and William Hodgkiss (Univ. of California, San Diego, San Diego, CA)

The goal of SBCEX 17 was to investigate properties of the mud layer at the experiment site. We perform geoacoustic inversion for the mud site, integrating data and estimates obtained on a track of a moving source. The transmitted sound signals were LFM pulses in a mid-frequency range. The sound was received at a vertical array comprising 16 phones. Arrival times of multiple paths were extracted from the received signals using particle filtering and a backward-moving smoother: forward filtering and smoothing were applied to the received signals for the estimation of path arrival times generated by sound interacting with the waveguide. The smoothing step corrected for outliers that provided erroneous geoacoustic estimates for some source positions. The forward-backward filter provided accurate arrival times with a significant uncertainty reduction. The obtained arrival times were linked with a ray tracer and linearization for source localization and estimation of other parameters that affect sound propagation such as water column depth and sound speed. Sediment sound speed and thickness were then computed. Inversion was carried out at every source location within the track; the results were combined into a single set of estimates and corresponding probability density functions. [Work supported by ONR.]

## Exhibit

An instrument and equipment exhibition will be located in the Summit Foyer on the 4th floor.

The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Monday, 5 December, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including lite snacks and a complimentary beverage.

Tuesday, 6 December, 9:00 a.m. to 5:00 p.m.: Exhibit Open Hours including a.m. and p.m. breaks serving coffee and soft drinks.

Wednesday: 7 December, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. break serving coffee.

## OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. All meetings will begin at 7:30 p.m., except for Signal Processing in Acoustics (4:30 p.m.) and Engineering Acoustics (4:45 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:

| | |
|---|---|
| Signal Processing in Acoustics (4:30) | Rail Yard |
| Engineering Acoustics (4:45) | Lionel |
| Acoustical Oceanography | North Coast A |
| Animal Bioacoustics | Grand Hall A |
| Architectural Acoustics | Summit A |
| Physical Acoustics | Golden Pass |
| Psychological and Physiological Acoustics | Grand Hall C |
| Structural Acoustics and Vibration | Golden Eagle B |

### Committees meeting on Wednesday:

| | |
|---|---|
| Biomedical Acoustics | Mill Yard A |

### Committees meeting on Thursday:

| | |
|---|---|
| Computational Acoustics | Summit C |
| Musical Acoustics | Rail Head |
| Noise | Summit B |
| Speech Communication | Grand Hall B |
| Underwater Acoustics | North Coast B |

## Session 3aAA

## Architectural Acoustics, Noise, and Psychological and Physiological Acoustics: Architectural Acoustics and Audio—Even Better Than the Real Thing I

K. Anthony Hoover, Cochair
*McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362*

David A. Conant, Cochair
*McKay Conant Hoover Inc., 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362*

David S. Woolworth, Cochair
*Roland, Woolworth & Associates, 356 County Road 102, Oxford, MS 38655-8604*

**Chair's Introduction—9:00**

### *Invited Papers*

**9:05**

**3aAA1. Augmented active acoustics: Recent experiences in acoustical enhancement for modern musical performance.** Paul Henderson (Venueflex, LLC, 8620 Westmoreland Dr. NW, Concord, NC 28027, paul@venueflex.com)

Electronic acoustic enhancement systems allow the designer to implement traits not easily obtained with physical construction, or to provide variable room behavior to accommodate different program needs in a multi-purpose facility. Traditionally, most active acoustic systems have been employed for the enhancement of acoustic music, speech, and other classical program material. Here, we provide an overview of active enhancement methods particularly suited to contemporary, amplified musical performances. By accurately modeling and simulating the geometry of the performance space, naturally occurring reflection paths may be augmented with their synthetic counterparts, each rendered with high spatial fidelity but modified acoustic properties. Material characteristics may then be selected beyond what is practical in the built environment, and room signatures may be modified and changed over time. Real-time signal processing placed in-line with an audio mixing console then renders these sound fields into a collection of loudspeakers throughout the listening space. Examples are provided of auditoria where these systems have been installed, along with measured acoustical behavior demonstrating both the effectiveness of the technique and practical design constraints for implementation.

**9:25**

**3aAA2. LF electronic Architecture—Saving cost and adding acoustic flexibility in the construction of multi-purpose performing arts facilities.** Steve Barbar (E-Coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@ecousticsystems.com)

Over the course of many years, we have successfully integrated LF components into full range electronic architecture installations. These components are typically utilized when programming includes acoustic instruments such as pipe organ, or a large orchestra (or both)—or when electronic sources producing LF energy (such as effects mapping or film screening) need to be considered. However, we have also installed LF only EA systems in several concert halls in order to optimize LF performance without the high cost associated with brick and mortar construction methods. Electronic Architecture provides the added benefit of variability necessary for multi-purpose performing arts venues.

**9:45**

**3aAA3. Electronic Architecture—Solutions for archetypal coupled spaces.** Steve Barbar (E-Coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@ecousticsystems.com)

We will discuss solutions for the most common coupled spaces in performing arts facilities—Balconies and Stages. While there are exceptions, seats in balcony and under balcony spaces sometimes have less than ideal listening conditions. For electronic and close miked sources, supplemental sound reinforcement can help to provide expected perception of the direct sound. However, perceived acoustical conditions do not match that of the main volume; the sense of engagement is compromised. Electronic Architecture incorporated in these spaces can dramatically improve listening uniformity. Additionally, the loudspeaker array can be used for supplemental sound reinforcement including film surround sound. Many multi-purpose performing arts venues do not have resources to procure, deploy, strike, or store an architectural stage shell system. A virtual shell is far more cost effective, requires less time and manpower to set up or strike, and requires far less storage space. In addition, it provides far greater acoustic variability—accommodating a wider range of performances.

**10:05–10:20 Break**

**10:20**

**3aAA4. Creating an immersive audio experience at National Sawdust.** Steve Ellison (Meyer Sound Labs, Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com)

Venues designed to create immersive audio experiences may include adjustable active acoustics and spatial audio mixing. One such venue is National Sawdust in Brooklyn, New York, which has presented many spatial audio events. For one of these, the electronic music duo "Mouse on Mars," comprised of Jan St. Werner and Andi Thoma, collaborated with the late Jamaican record producer Lee "Scratch" Perry to create an immersive audio experience presented in March 2022. This audio production presented a challenge, as the content was prepared in Berlin and delivered in Brooklyn using audio systems with significantly different loudspeaker configurations. National Sawdust incorporates a Meyer Sound Constellation active acoustic system with 86 small full-range loudspeakers, 16 compact low-frequency loudspeakers, and a left/right PA system. The Berlin studio includes 20 loudspeakers. Audio sources were spatially mixed to the loudspeakers in both systems using an integrated Spacemap System. The workflow used to prepare this concert and the impact of additional production elements, including active acoustics, lighting, seating, and video, are discussed.

**10:40**

**3aAA5. Selecting a reference headphone for 3D audio reproduction.** Sean E. Olive (Harman X, Harman Int., 8500 Balboa Blvd, Northridge, CA 91377, sean.olive@harman.com)

3D audio uses generalized or personalized head-related-transfer-functions (HRTF) to provide the essential spatial and timbral cues. The headphone must reproduce these cues without introducing any errors of its own. A standard approach is to first calibrate the headphone by measuring and equalizing its frequency response at the blocked ear canal of the listener, or in the generalized case, a standard test fixture. However, the reliability of the calibration can be adversely affected by acoustical interactions between the headphone, the listener or the test fixture that can vary with headphone model, individual listener, and how the headphone is placed on their ears. The extent to which these errors affect the quality of 3D audio reproduction is not well understood. This paper discusses some recent investigations into the acoustical interactions between headphones, listeners, and standard test fixtures, and the effect on the quality of 3D audio reproduction through headphones. Several different models of headphones were measured on both human subjects and test fixtures with multiple placements to quantify the errors and their reproducibility. From these data, the headphone with the most linear and consistent response was selected for 3D audio reproduction.

**11:00**

**3aAA6. How real is Virtual Acoustics?** Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, mvo@akustik.rwth-aachen.de) and Lukas Aspoeck (IHTA, RWTH Aachen Univ., Aachen, NRW, Germany)

Recent comparisons of room-acoustic simulations and auralizations demonstrated that a high degree of realism can be achieved under certain conditions. How close the simulation can get to reality depends on the definition of the objective in general terms (authenticity or plausibility). The question is also what is the application scenario of the simulation. The goal can be to create an exact replica of a real space, a blind simulation of a future or lost space, or a demonstration of consequences of decisions during a design process. This work discusses the sources of uncertainties in Virtual Room Acoustics, the effort to be taken for achieving high realism, and suggestions for use of Virtual Acoustics in practice.

# Session 3aAO

## Acoustical Oceanography: Acoustical Sensing of Ocean Turbulence, Mixing, and Stratification (Hybrid Session)

John A. Colosi, Cochair
*Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943*

Christopher Bassett, Cochair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th St., Seattle, NJ 98105*

### *Invited Papers*

**8:30**

**3aAO1. Measuring ocean mixing: From observing processes to quantifying impacts.** Caitlin Whalen (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Box 355640, Seattle, WA 98105-6698, cbwhalen@uw.edu)

The impacts of ocean mixing are varied, ranging from the local across-isopycnal transport of heat, salt, and nutrients, to the global overturning circulation with implications for climate. A full understanding of turbulent mixing, from driving processes to impacts, spans all oceanic time and length scales. Turbulent mixing in the ocean occur on scales less than centimeters and timescales less than hours, yet the processes that drive this turbulence occurs on meters to 100s of km length scales and the impact of the turbulent mixing spans the full range of oceanic spatiotemporal scales. Here, we will discuss current approaches to measuring ocean mixing and explore how to bridge this scale gap to link the turbulence measurements to the processes that drive ocean mixing and the subsequent impacts. Intriguing examples of ocean mixing processes and their influence on ocean dynamics will be discussed throughout.

**9:00**

**3aAO2. Sound propagation and scattering in turbulent media.** Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, vladimir.ostashev@colorado.edu) and D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

This presentation overviews sound propagation and scattering in turbulent media and pertinent acoustic remote sensing of turbulence. Although sound propagation through a turbulent atmosphere is mainly considered, the presented results apply to other media such as oceanic turbulence. Formulations are provided for various statistical characteristics of acoustic signals such as the scattering cross section, the variances of the phase and log-amplitude fluctuations, the spatial, temporal, and cross-frequency coherences, and the probability density functions. Monte Carlo simulation of sound propagation in a random medium is explained. Specifics of acoustic signals in turbulent media as compared to electromagnetic waves are highlighted. For example, sound waves are scattered by both the sound speed and medium velocity fluctuations, which are scalar and vector random fields, respectively, with different statistical properties. Also, fluctuations in the acoustic refractive index are much stronger than those in electromagnetic propagation, and sound can be scattered by humidity (or salinity) fluctuations. Finally, the Markov approximation, which is widely used for electromagnetic waves, might not be applicable for some statistical characteristics of acoustic signals. It is explained how these specifics of sound propagation are addressed in atmospheric acoustics.

**9:20**

**3aAO3. Acoustic tomography measurement and large-eddy simulation of macroturbulence in high flow tidal channels.** Alex E. Hay (Oceanogr., Dalhousie Univ., 1355 Oxford St., Halifax, NS B3H 4R2, Canada, alex.hay@dal.ca), Len Zedel (Phys. and Physical Oceanogr., Memorial Univ. of NF, St. John's, NF, Canada), Angus Creech (Dalhousie Univ., Halifax, NS, Canada), Mahdi Razaz (Univ. of Southern Mississipi, Stennis Space Ctr., MS), and Kiyosi Kawanisi (Hiroshima Univ., Higashi–Hiroshima, Japan)

Velocity spectra from a pilot tomography experiment in Grand Passage, Nova Scotia, exhibit the structure at 10 m to several 100 m scales similar to that of the spectra of vertically integrated velocity registered by bottom-mounted acoustic Doppler profilers. The experiment was motivated in part by the need for flow and turbulence measurement at sites being targeted for in-stream tidal power development, and by the potential of acoustic tomography for providing this information from shore-based locations in near real time. The measurements were made using a single pair of acoustic transceivers operating at 7 kHz and separated by 1.5 km at an oblique angle to the channel axis. Limited to a single cross-channel path, the experiment provided no information on the spatial structure of the turbulence. In order to investigate this structure, virtual tomographic experiments were carried out by propagating pulses through with meter-scale resolution velocity fields from a non-hydrostatic large-eddy simulation (LES) model of flow in Grand Passage. Results from these virtual experiments are presented, and their implications are discussed in relation to the potential of multi-transceiver tomography for macro-turbulence measurement and the role of horizontal shear in macro-turbulence production in coastal environments.

**9:40**

**3aAO4. How prevalent is acoustic scattering from oceanic microstructure?** Andone C. Lavery (AOPE, Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, alavery@whoi.edu), Christopher Bassett (Appl. Phys. Lab., Univ. of Washington, Seattle, NJ), and Scott Loranger (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Any oceanic environment with spatial gradients in sound speed and density can result in acoustic scattering hot spots. These acoustic hotspots can be rapidly evolving and vary in their spatial heterogeneity. Here, we review acoustic scattering models for turbulence and present data collected with a variety of split-beam and multi-beam echosounders illustrating the range of environments and spatial scales associated with scattering from physical microstructure, including shear instabilities in estuarine environments, non-linear internal waves on the continental shelf, strong interface scattering due to double-diffusion, scattering from strong gradients or interfaces, and turbulent microstructure at the New England shelf break front. The theoretical acoustic scattering formulations for different types of physical microstructure are applied to these different environments, and recommendations are made for optimal frequency bands to sample the different types of physical microstructure and the optimal measurements for inference of parameters that describe the physical microstructure. Limitations imposed on detection and quantification of scattering from the physical microstructure due to sources including suspended sediment, bubbles, and biological targets, are also discussed. [Work supported by the ONR.]

**10:00–10:15 Break**

**10:15**

**3aAO5. Sensing of ocean mixing via forward scattering acoustic scintillation.** John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943, jacolosi@nps.edu) and Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Ocean mixing process observations show complex spatio-temporal patterns as well as energy and dissipation levels that vary over six or more orders of magnitude, creating a challenging sampling problem. Acoustic imaging of ocean turbulent structures is informative, but extraction of quantities like spectra and dissipation rates of kinetic energy, ε, and temperature variance, X, requires geometric sampling constraints. Here, a remote sensing approach is proposed in the spirit of Di Iorio and Farmer (1994), and Duda *et al.* (1988), based on weak multiple-forward scattering physics. In the correct scattering regime, observations of the wavenumber spectrum of acoustic intensity obtained on a vertical array gives a direct measure of the range average turbulence index of the refraction spectrum in a plane transverse to the acoustic path. In this talk, the theory will be reviewed and some illustrative simulations will be presented; importantly, one-way acoustic transmission measures a combination of both ε and X. Experimental arrangements (frequency/range) where the theory is expected to apply will be outlined with an eye towards observing bottom boundary layer mixing. With a suitable vertical array, one could resolve scales from the smallest internal waves (∼several meters), through the transition regime, and down to scales of turbulence (∼tens of cm).

## *Contributed Papers*

**10:35**

**3aAO6. Broadband acoustic characterization of backscattering from a rough stratification surface.** Elizabeth Weidner (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, eweidner@ccom.unh.edu) and Thomas C. Weber (Univ. of New Hampshire, Durham, NH)

This work aims to elucidate the utility and limitations of broadband scientific echo sounders in characterizing the frequency-dependent scattering from stratification interfaces. A recently published 1D acoustic backscattering model for the oceanic stratification structure was expanded for this work to consider echo sounding applications and the importance of roughness (vertical and spatial variability) along the stratification interface. The expanded model was tested in utilizing data from a recent acoustic experiment in the Kattegat basin, a region with well-documented, strong, salinity-driven stratification. A suite of broadband split-beam echosounders (45–270 kHz) provided broadband backscattering data, which were validated using high resolution *in situ* observations of water column structure, collected with a CTD (Conductivity, Temperature, Depth) sensor and MVP (Moving Vessel Profiler). Overall, the frequency dependence of backscattering agrees with model predictions of a gradual decay in the reflected wave amplitude from stratification structure with increasing frequency. Stratification interface roughness drives changes in frequency-dependent scattering away from a smooth surface solution as a function of different roughness regimes, primarily defined by the root-mean-squared slope of the interface. Results suggest a path to remote estimations of physical medium properties through broadband acoustic inversion.

**10:50**

**3aAO7. Acoustic backscatter from bubbles as a (quasi)passive tracer of turbulent mixing in high-flow tidal channels.** Maricarmen Guerra (Universidad de Concepcion, Concepcion, Chile) and Alex E. Hay (Dalhousie Univ., Dalhousie University, 1355 Oxford St., Halifax, NS B3H 4R2, Canada, alex.hay@dal.ca)

In high-flow tidal channels, the water column tends to be well-mixed vertically due to the high levels of turbulence. Under favorable circumstances, such as those in which wind waves incident at the channel entrance from the adjoining open ocean or large embayment are opposed by the tidal flow, the waves steepen and break as they propagate upstream, resulting in bubble injection at the sea surface. These bubbles are then mixed downward by the tidally generated turbulence, resulting in pronounced, surface-connected, downward-propagating plumes of high backscatter in records from bottom-mounted upward-looking acoustic Doppler current profilers, typically operating at 100s of kHz. Letting the backscatter amplitude represent a pseudo-concentration, C, we demonstrate that the vertical turbulent diffusivity, K, can be estimated from the vertical turbulent flux $i < c'w' >$ along the axis of the vertical beam, and the mean vertical gradient, dC/dz. Intriguingly, and despite the fact that bubbles are positively buoyant, the resulting estimates of K are very close to the values of momentum diffusivity required to obtain good agreement between the observed tidal velocities and a data-validated numerical tidal circulation model for the particular channel in question

**11:05**

**3aAO8. Ambient sound directionality and rapid estimation of empirical Green's functions in a coastal ocean.** Tsu Wei Tan (Dept. of Marine Sci., ROC Naval Acad., Kaohsiung, Taiwan) and Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu)

Noise interferometry offers an opportunity for passive ocean remote sensing through retrieval of empirical Green's functions (EGFs) from cross-correlations of ambient sound. At ranges of tens of ocean depths, obtaining stable and accurate EGF estimates usually requires noise averaging periods of hours or days. Decreasing the necessary averaging times without compromising the EGF accuracy is critically important for operational applications. Using data acquired in the Shallow Water 2006 experiment on the continental shelf off New Jersey, it has been found that occasionally averaging periods as short as a few minutes are sufficient. The phenomenon is observed for various receiver pairs but does not occur simultaneously in all azimuthal directions. Short averaging periods give EGFs in a broader frequency band and with a richer mode content. These are better suited to monitor physical processes in the water column. The conditions conducive to rapid EGF emergence are studied using *in situ* temperature measurements, meteorological information, and the acoustic data acquired with a horizontal line array. Strong intermittency is observed in the horizontal directionality of ambient sound. For various receiver pairs, EGFs are found to emerge rapidly when ambient sound directionality favors propagation in the direction from one receiver to the other.

WEDNESDAY MORNING, 7 DECEMBER 2022 — MILL YARD A, 8:30 A.M. TO 12:00 NOON

## Session 3aBA

## Biomedical Acoustics: Ultrasound Therapy in the Brain

Charles F. Caskey, Chair
*Radiology, Vanderbilt University Medical Center, 1161 21st Ave S., Nashville, TN 37232*



### *Invited Papers*

**8:30**

**3aBA1. Acoustic droplet vaporization for nonthermal ablation of brain tumors.** Tyrone M. Porter (Biomedical Eng., Univ. of Texas at Austin, 110 Cummington Mall, Boston, MA 02215, tmp@bu.edu), Chenguang Peng (Boston Univ., Boston, MA), Tao Sun, Chanikarn Power, Yongzhi Zhang, and Nathan J. McDannold (Radiology, Brigham and Women's Hospital; Harvard Med. School; Tufts Univ., Boston, MA)

Acoustic vaporization of perfluorobutane-based phase-shift nanoemulsions (PSNE) can be used to nucleate inertial cavitation (IC) *in vivo*. The acoustic pressure amplitude must exceed a threshold for vaporization of PFB-based nanoemulsions. Two focused ultrasound transducers with a center frequency of 837 kHz were oriented such that their focal volumes overlapped and the acoustic pressure amplitude was amplified. In this study, the dual transducer system was combined with circulating PSNE to nucleate IC in established brain tumors, leading to nonthermal ablation of the tumors. For comparison, microbubble ultrasound agents (UCA) were used as IC nuclei for ultrasound-mediated tumor ablation. The ablation volume was confined to focal volume when PSNE were used to nucleate IC, whereas pre-focal damage was observed when UCA were used as IC nuclei. Additionally, PSNE-nucleated IC ablated a larger percentage of the brain tumors on average than MB-nucleated IC (89.46.8% vs 11.17.7%, respectively). These results suggest that PFB-based PSNE may be used to significantly reduce the inertial cavitation threshold in the cerebrovasculature, and when combined with transcranial focused ultrasound, enable efficient focal intracranial nonthermal ablation.

**8:55**

**3aBA2. Ultrasound-guided microbubble-mediated brain therapy with a modular transmit/receive phased array.** Ryan M. Jones (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, rmjones@sri.utoronto.ca), Dallan McMahon, Dallas Leavitt, Rohan Ramdoyal, Kang Lee, Wai Meng Kan, Steven Yang, Yi-Shiuan Chen, and Kullervo Hynynen (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada)

The development of low-cost, ultrasound-guided focused ultrasound (USgFUS) treatment platforms is expected to advance the adoption of microbubble (MB)-mediated brain therapy by improving access to the technology. Our group has designed and fabricated clinical-scale transmit/receive phased array systems for MB-mediated USgFUS brain therapy. Acoustic field simulations were carried out to optimize array element placement, and transducer scaffolds were constructed using 3D printing techniques. These devices have been employed for skull computed tomography-array registration, 3D spatial mapping of MB activity *in vivo* through *ex-vivo* human skullcaps via noninvasive aberration correction methods, and we have harnessed this spatiotemporal cavitation information to calibrate exposure levels for safe volumetric blood-brain barrier opening. At higher exposure levels, we have demonstrated the ability of 3D MB imaging

data to predict the tissue damage volume distributions induced during nonthermal brain ablation. Ultrafast processing of acoustic emissions data has been shown to uncover MB dynamics hidden by conventional whole-burst temporal averaging, and can inform temporal undersampling strategies when millisecond-long tone bursts are applied. Machine learning approaches can assist with image-based classification of MB activity, which may result in finer control of the induced bioeffects. This talk will focus on our recent results obtained with a novel modular USgFUS phased array system.

**9:20**

**3aBA3. Neuronavigation-guided transcranial histotripsy, results in a cadaveric model.** Jonathan R. Sukovich (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd, Ann Arbor, MI 48109, jsukes@umich.edu), Timothy L. Hall, Sang-Won Choi, Mahmoud Komaiha, Ning Lu, Dave Choi, and Zhen Xu (Univ. of Michigan, Ann Arbor, MI)

Results from neuronavigation-guided transcranial histotripsy treatments in cadaveric models are presented. Histotripsy treatments were delivered transcranially to the brains of 3 cadavers (<96-h post-mortem) using a 30-cm diameter, 700-kHz, 360-element, transmit-receive capable transcranial histotripsy array. Pre-treatment CT and MRI scans of the head were acquired. A clinical neuronavigation system was used to coregister the histotripsy array with the cadaver head and guide treatments. Two-step aberration correction, combining pre-treatment CT-based correction and intra-procedure acoustic cavitation emission (ACE)-based correction, was employed to correct for skull-induced acoustic aberrations. Cavitation was generated at rates up to 200-Hz, and steered through 1-cm wide cubic targets in the brain. ACE signals were acquired throughout treatments using the array elements as receivers and used to localize cavitation and monitor treatment.Transcranial histotripsy was successfully applied to generate lesions in the cadaver brains. Two-step aberration correction resulted in significant improvements in focal quality and pressure. ACE-based cavitation localization could be achieved at rates up to 120-Hz. Neuronavigation-based coregistration/targeting errors ranged from 3 to 9-mm, but workflow issues have been identified and refinements have since reduced targeting errors to $\leq$1.5-mm. Changes in ACE signal features throughout treatments were observed to correlate with morphological observations of induced damage.

**9:45**

**3aBA4. Transcranial focused ultrasound as a treatment for hypertension.** Harriet Lea-Banks (Physical Sci., Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, harriet.lea-banks@sri.utoronto.ca), Neha Chauhan (Physical Sci., Sunnybrook Res. Inst., Toronto, ON, Canada), and Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Pharmaceuticals currently used for reducing blood pressure prove ineffective for many patients. As well, although hypertension is the most common disorder in pregnancy, many anti-hypertension drugs pose significant risks associated with restricted fetal growth, decreased uteroplacental blood flow, and fetal loss. Deep brain stimulation of the periaqueductal gray (PAG) region has shown promise in treating drug-resistant hypertension in patients. However, a non-invasive approach would carry far fewer risks, particularly during pregnancy. Our work has explored the potential of focused ultrasound (FUS), targeted to the PAG region, to reduce blood pressure. In a rodent model of hypertension, FUS delivered to the PAG successfully reduced high blood pressure to within the normal range. By comparing treatment regimes, we have also investigated the mechanisms underlying this reduction in blood pressure and ways to prolong the effect. This approach could point towards a new treatment for hypertension that has greater safety and broader applicability for vulnerable patient populations.

**10:10**

**3aBA5. Emerging applications of image-guided therapeutic ultrasound for brain tumor-directed immunomodulation and immunotherapy.** Natasha D. Sheybani (Biomedical Eng., Univ. of Virginia, Medical Res. Bldg. 5 (MR-5), 415 Ln. Rd., Charlottesville, VA 22908, nds3sa@virginia.edu)

Immunotherapy has been established as a disruptive pillar of cancer treatment. However, the impact of immunotherapy has yet to be fully realized in neuro-oncology owing to the unique physical barriers (i.e., blood brain/tumor barriers [BBB/BTB]) and distinct immune microenvironment within brain tumors. These obstacles may be overcome through the alliance of image-guided interventions, such as therapeutic ultrasound, with immunotherapies. Specifically, transient disruption of the BBB/BTB via focused ultrasound (FUS) and circulating microbubbles is a promising strategy for potentiating immunotherapy through both immuno-modulation and drug delivery – notably, in a non-invasive, non-ionizing, and precisely targeted manner. We have demonstrated that FUS BBB/BTB disruption invokes mild, transient shifts in the immune milieu of brain tumor-bearing mice. Additionally, we have leveraged immuno-PET imaging to rationally deploy a therapeutic approach combining innate immune checkpoint inhibitor delivery with FUS. We show that this immuno-PET-informed paradigm achieves superlative targeted antibody delivery and improved therapeutic outcomes in murine glioblastomas, with markedly reduced systemic antibody dose. Our ongoing investigations seek to advance the nexus of FUS, molecular imaging, and biomarker discovery with the goal of enabling non-invasive, precision immunotherapy paradigms for brain tumors.

**10:35–10:50 Break**

**10:50**

**3aBA6. Image guidance and beam localization for transcranial focused ultrasound therapy.** Marshal A. Phipps (Vanderbilt University Inst. of Imaging Sci., Vanderbilt Univ. Medical Ctr., 1161 21st Ave., S AA-1105, Nashville, TN 37232, m.anthony.phipps@vumc.org), Thomas J. Manuel, Michelle K. Sigona, Huiwen Luo (Biomedical Eng., Vanderbilt Univ., Nashville, TN), Pai-Feng Yang (Vanderbilt University Inst. of Imaging Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN), Allen Newton, Li Min Chen (Radiology, Vanderbilt Univ. Medical Ctr., Nashville, TN), Will Grissom (Biomedical Eng., Vanderbilt Univ., Nashville, TN), and Charles F. Caskey (Radiology, Vanderbilt Univ. Medical Ctr., Nashville, TN)

Therapeutic uses of transcranial focused ultrasound (tFUS) in the brain are being widely explored in both clinical and research settings from tissue destruction to blood brain barrier opening (BBBO) to neuromodulation. The appeal of tFUS is the ability to target structures throughout the brain with a millimeter scale focus. In order to understand the outcome of the treatment or experiment it is

important to ensure the tFUS beam is targeted to and located at the region of interest. Image guidance allows for targeting of anatomical or functional structures within the brain. Here we present a targeting and localization scheme where optical tracking of the transducer is used to target the tFUS focus to a brain region within a previously acquired image and MR acoustic radiation force imaging is used to localize the beam in the brain to ensure the focus is at the target. By combining this targeting and localization scheme with a phased array transducer we are able to then steer the focus to ensure accurate sonications of specific brain regions using tFUS. This talk will discuss the application of this method to two different phased array transducers used for neuromodulation and BBBO in nonhuman primates.

## *Contributed Papers*

**11:15**

**3aBA7. Generating patient-specific acoustic simulations for transcranial focused ultrasound procedures based on optical tracking information.** Michelle K. Sigona (Biomedical Eng., Vanderbilt Univ., 1161 21st Ave., S AA-1105, Nashville, TN 37232, michelle.k.sigona@vanderbilt.edu), Thomas J. Manuel, Huiwen Luo (Biomedical Eng., Vanderbilt Univ., Nashville, TN), Marshal A. Phipps, Pai-Feng Yang (Vanderbilt University Inst. of Imaging Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN), Kianoush Banaie Boroujeni (Psych., Vanderbilt Univ., Nashville, TN), Robert L. Treuting (Biomedical Eng., Vanderbilt Univ., Nashville, TN), Thilo Womelsdorf (Psych., Vanderbilt Univ., Nashville, TN), Allen Newton (Vanderbilt University Inst. of Imaging Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN), Will Grissom (Biomedical Eng., Vanderbilt Univ., Nashville, TN), Li Min Chen, and Charles F. Caskey (Radiology, Vanderbilt Univ. Medical Ctr., Nashville, TN)

During transcranial focused ultrasound (FUS) procedures, accurate targeting is important and neuronavigation with optically tracked tools is used to estimate the free-field focal location on pre-acquired images. Offline neuronavigation systems do not typically incorporate aberrating effects of the skull known to displace and distort the focus. Here, we developed a pipeline that integrated patient-specific acoustic simulations informed by transformations from optically tracked FUS procedures as a tool to evaluate transcranial pressure fields and demonstrated its use in three FUS scenarios: magnetic resonance imaging-guided (MR-guided) phantom experiments, MR-guided non-human primate (NHP) experiments, an offline behaving NHP experiments. Distance vectors between the estimated focus from optical tracking and peak intracranial location from simulations were less than 1 mm for all groups (Phantom: $0.6 \pm 0.3$ mm, NHP: $0.7 \pm 0.3$ mm, Behaving NHP: $0.5 \pm 0.2$ mm). Comparisons of the target registration error of MR measurements with the optically tracked focus ($TRE_{Tracked}$) and simulated focus ($TRE_{Simulated}$) suggest that focal location errors are dominated by optical tracking errors rather than aberration through the skull in the NHP (Phantom: $TRE_{Tracked}$: $3.3 \pm 1.4$ mm, Phantom $TRE_{Simulated}$: $3.3 \pm 1.9$ mm, NHP $TRE_{Tracked}$: $3.9 \pm 1.9$ mm, NHP $TRE_{Simulated}$: $4.1 \pm 1.6$ mm). Our software pipeline provides patient-specific estimates of the acoustic field during transcranial FUS procedures.

**11:30**

**3aBA8. Synergistic effects of microbubble-mediated focused ultrasound and radiotherapy in a F98 glioma model.** Stecia-Marie Fletcher (Brigham and Women's Hospital/Harvard Med. School, 221 Longwood Ave., EBRC 515b, Boston, MA 02115, sfletcher4@bwh.harvard.edu), Yongzhi Zhang, and Nathan J. McDannold (Brigham and Women's Hospital/Harvard Med. School, Boston, MA)

Combined microbubble-mediated focused ultrasound (FUS) and radiation therapy (RT) has been shown to improve outcomes in tumors outside the brain. Here, we study the effects of FUS + RT in a F98 glioma model. Tumor cells were implanted into the brains of 45 Fischer rats (n = 4–8 per group): Controls, FUS, RT (4,8,15-Gy), and FUS + RT (4,8,15-Gy). 9 days after implantation, tumors were targeted using FUS (1–2 W, 220 kHz, 5 ms bursts, 1 Hz PRF, 180 s, 20 $\mu$L/kg Definity microbubbles), followed by RT. Tumor progression was monitored using MRI. At 4Gy, FUS + RT increased tumor doubling time by 11% compared to RT only and Controls ($P < 0.01$). At higher doses, where RT alone had a significant effect, there were no significant differences in doubling time compared to RT alone. Tumor volumes were reduced by 21–57% (4Gy) and 20–48% (15Gy) compared with RT alone ($P < 0.05$). No significant benefit was observed at 8Gy. A moderate, but not significant, increase in median survival was observed: 28 versus 27days (4Gy), 30 versus 29days (8Gy), and 35 vs. 33 days (15Gy). However, at 4Gy, FUS + RT significantly improved survival by 6% compared to the control group (median survival = 27days), while RT only offered no improvement. This study indicates that FUS + RT may improve therapeutic outcomes in glioblastoma, particularly at low RT doses, where RT alone has no therapeutic benefit.

**11:45**

**3aBA9. Acoustic coupling pads for the control of ultrasound neuromodulation exposure.** Samantha Schafer (School of Biomedical Eng., Sci. and Health Systems, Drexel Univ., 3141 Chestnut St., Philadelphia, PA 19104, sfs77@drexel.edu) and Mark Schafer (School of Biomedical Eng., Sci. and Health Systems, Drexel Univ., Philadelphia, PA)

We have developed acoustic coupling pads that facilitate single-blind and double-blind neuromodulation experiments by selectively transmitting ultrasound without affecting the audible sound. The pads were made from a skin-safe two-part silicone with little to no ultrasonic attenuation, that also provided the necessary acoustic properties, stability and flexibility. To inhibit the transmission of ultrasound, a foam disk was imbedded into the pad during the manufacturing process. The design goal for acoustic attenuation was −40 dB. 20 sets of transmit pads and non-transmit pads were fabricated. Acoustic transmission loss was measured in a water tank with a 60 mm diameter, 80 mm focus circular disk transducer operated at 650 kHz and a standard hydrophone (Reson TC4038). Transmit pads had an average of −0.4 dB loss; non-transmit pads met the required −40 dB loss (average: −47.4 dB). An operator experienced with ultrasound treatments was unable to distinguish them by visual inspection or casual physical manipulation. An experienced subject exposed to a typical treatment regimen was not able to distinguish any audible difference. The acoustic coupling pads create identical testing situations for single-blind and double-blind studies for neuromodulation treatments so that neither the patient nor the operator administering the treatment can distinguish which patient group received the ultrasound treatment and which did not.

# Session 3aCA

## Computational Acoustics: Learning and Stochastic Modeling in Computational Acoustics I

Pierre F. Lermusiaux, Cochair
*MIT, 77 Mass Ave., Cambridge, MA 02139*

Wael H. Ali, Cochair
*Mechanical Engineering, Massachusetts Institute of Technology, 77 Massachusetts Avenue, Cambridge, MA 02139*

Aaron Charous, Cochair
*Mechanical Engineering, Massachusetts Institute of Technology, 77 Massachusetts Avenue, Cambridge, MA 02139*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**3aCA1. Higher-order perturbative parabolic-equation solutions for reduced-order sound propagation modeling.** Ying-Tsong Lin (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, ytlin@whoi.edu)

The parabolic-equation (PE) approximation to the Helmholtz equation has been shown to be one of the most effective and efficient numerical methods for sound propagation modeling. In this talk, a recursive and iterative algorithm will be introduced to compute the PE solutions due to perturbations in the medium wavenumber, which is a function of frequency and sound speed. This algorithm is developed based on perturbation theory with higher-order nonlinear terms, and it can efficiently compute higher-order derivatives of the sound pressure field with respect to changes in the local wavenumber. This higher-order perturbative PE (HOPPE) method can eventually lead to model order reduction for sound propagation simulations. An example of broadband computations with nonlinear interpolations between sampled frequencies will be presented to demonstrate the application of reduced-order modeling. A sensitivity kernel with higher-order nonlinear terms will also be introduced. Applications to data assimilation modeling and sensitivity tracking will be discussed. [Work supported by the Office of Naval Research.]

**8:25**

**3aCA2. Non-intrusive mode based analyses to predict uncertainty in sound propagation in the ocean.** Emanuel F. Coelho (Sci., Appl. Ocean Sci., LLC, 5242 Port Royal Rd., 1032, Springfield, VA 22151, emanuel.coelho@appliedoceansciences.com), Paul Hursky (Sci., Appl. Ocean Sci., LLC, San Diego, CA), and Kevin D. Heaney (Appl. Ocean Sci., LLC, Fairfax Station, VA)

Ocean dynamics can result in significant sound speed variability that are not well captured by deterministic numerical models and can impact differently high to low frequency acoustic propagation and bottom interactions by constraining horizontal and vertical paths. Uncertainty in the sound speed can, therefore, cause inaccuracies in acoustic based tactical decision tools and severely impact the accuracy of algorithms using sound propagation to estimate ocean volume states (e.g., ocean tomography or data assimilation). In this work, we propose to use a non-intrusive Reduced Order Modelling solution that consists of building a dictionary of static modes from historical ocean simulations with different settings and resolutions and climatology to derive an expedite uncertainty model along a central deterministic forecast. The system will then, through Orthogonal Matching Pursuit along the dictionary, use the available ocean model-data mismatches and/or acoustic innovations (e.g., arrival numbers and times, acoustic energy distribution, etc.) to define an ensemble of possible sound speed volume distributions and associated acoustic propagation outcomes. We will show results using a simplified simulation experiment that extracts "observations" from a nature reference field to document the procedure and benchmark results to reconstruct 3D sound speed fields from ocean-acoustic measurements.

**8:45**

**3aCA3. Dynamic mode decomposition for underwater acoustics on autonomous platforms.** Tony Ryu (Mech. Eng., MIT, 476 Windsor St., Cambridge, MA 02141, tonyryu@mit.edu), Wael H. Ali (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Pierre F. Lermusiaux (MIT, Cambridge, MA)

A sufficiently accurate representation of the ocean physics is needed for reliable underwater acoustics forecasting. Ocean physics are often simulated by high-dimensional data-assimilative numerical ocean simulations. Due to operational constraints such as computing, memory, communication-bandwidth, etc., it is infeasible to run these numerical simulations onboard the platforms. In this work, we employ the onboard data-assimilative incremental, Low-Rank Dynamic Mode Decomposition (iLRDMD) for forecast transmission, onboard computation, and data assimilation. The iLRDMD is an adaptive reduced-order model (ROM) that uses proper orthogonal

decomposition (POD) and DMD for compression and transmission of high-fidelity ocean forecast to communication-disadvantaged platforms. It provides reduced-order onboard forecasting and also enables efficient updates of the DMD model itself from inputs from remote centers. Finally, when sparse observations are made by the platform, the iLRDM uses Bayesian data assimilation to correct both the forecasts and the DMD model. We demonstrate the use of deterministic and data-assimilative iLRDMD forecasted ocean fields for underwater acoustics computations in real ocean examples. We further explore the use of our adaptive ROM for joint reduction of the ocean physics and acoustics fields.

**9:05**

**3aCA4. Inclusion of environmental stochasticity in navy system performance modeling.** Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com) and Emanuel F. Coelho (Sci., Appl. Ocean Sci., LLC, Springfield, VA)

The performance of all acoustic navy systems is criticallly dependent upon the environmental conditions for propagation, primarily the ocean sound speed structure, the seafloor topography, the sediment type and the sea surface characteristics. Advances in computational power and improvements in hydrodynamic modeling have led to the opportunity of including ocean varaibiltiy and uncertainty into the acoustic propagation portion of system performance and, therefore, through the application of straight forward statistics into the performance prediction product. The first application of stochastic processing is in the computation of ambient noise (the shipping and noise sounsdcape) for comparison between modeled noise and measurements from long-term buoy deployments. Uncertainty in ship source level, position, wind speed, and environmental variabiility is key drivers in the model-data comparison. A second example of inclusion of environmental stochasticity in the passive anti-submarine warfare scenario of a passive surveillance barrier. A primary challenge here is in the communication of the uncertainty to a community that is used to thinking in simple performance metrics, such as range of the day. The last example is the use of an ensemble ocean forecast in the computation of the spread in eigenray arrivals for a real-time tomography system utilizing an active ASW sonobuoy field.

## *Contributed Paper*

**9:25**

**3aCA5. Adaptive variational Hamiltonian Monte Carlo sampling with application to ocean acoustics.** Brian A. Whiteaker (Elec. and Comput. Eng. (ECE), UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, bwhiteak@ucsd.edu), William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA), Jeremy Schmitt, and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

The Hamiltonian Monte Carlo (HMC) algorithm is a variant Markov Chain Monte Carlo (MCMC) excelling at sampling high dimensional target distributions. The HMC explores a distribution as a particle trajectory in a Hamiltonian system by augmenting distribution parameters with auxiliary momentum variables. A particle trajectory is numerically integrated from initial point to some distant low-correlation point. Projecting back to parameter space yields a proposal point with high acceptance probability leading to faster exploration of the target distribution. The Leap-frog integrator is commonly used for its volume preservation (symplecticity) and reversibility traits that maintain the detailed balance condition of MCMC. Although effective, Leap-frog has sensitivity to choice of integration step-size and number of integration steps. Poor choices cause slow exploration or bias when regions of the distribution are not explored. Adaptive step methods could remove step size issues and explore regions of a distribution where Leap-frog fails. Existing adaptive methods self-adjust step-size while maintaining symplecticity and reversibility. In this work, we apply adaptive step methods to simulated target distributions and real world ocean acoustic data and compare performance. [Work supported by Office of Naval Research.]



**9:40–9:55 Break**

## *Invited Papers*

**9:55**

**3aCA6. Applications of Bayesian optimization with a Gaussian process surrogate model in ocean acoustics.** William F. Jenkins (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, wjenkins@ucsd.edu) and Peter Gerstoft (Univ. of California, San Diego, La Jolla, CA)

In this study, we present a method that samples geoacoustic parameter space with a Bayesian approach that uses a Gaussian process as a surrogate model of the objective function. The objective function is defined as a Bartlett processor whose output measures the match between a received and replica pressure field on a vertical line array. Replica fields are obtained using a normal mode propagation model whose geoacoustic parameters are selected from the parameter search space. The surrogate model represents the posterior on the objective function and is updated with each model evaluation. Optimization is performed with sequential model evaluations, with an acquisition function guiding the next point in parameter space to be evaluated. Various use cases and parameterizations are discussed, including the effects of the choice of acquisition function and covariance function of the Gaussian process. Results indicate that Bayesian optimization using a Gaussian process surrogate model converges rapidly on an approximated optimal solution.

**10:15**

**3aCA7. Neural network predictions of acoustic transmission loss uncertainty.** Brandon M. Lee (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, leebm@umich.edu), Jay R. Johnson (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Many applications rely on predictions of acoustic transmission loss (TL) computed with numerical tools given expected environmental properties and potential circumstances. Because environmental uncertainty limits the quality of these TL predictions, methods to quantify the corresponding TL uncertainty are critical for robust decision-making and uncertainty reduction. Given the variety of needs across relevant applications, a desirable TL-uncertainty quantification method should be compatible with realistic (database-precise) descriptions of environments and uncertainties, computationally non-intrusive for the many possible Helmholtz equation solvers, and fast enough for real-time applications. Supervised machine learning provides such a method where a neural network (NN) is trained to relate TL uncertainty to spatial patterns in the existing nominal TL-field solution across a dataset of examples generated from Monte Carlo simulations. While this method has been successfully implemented for long-range, underwater propagation simulated with a 2D Parabolic Equation solver, this study will assess its capability on different computational models, environmental scenarios, and sources and degrees of uncertainties. Examples will be given, which demonstrate how data-driven learning, physical understanding, and hyper-parameter optimization can be used to design the key features of this method: the NN inputs, the NN architecture, and the training data-set(s). [Work supported by the NDSEG fellowship program]

## *Contributed Paper*

**10:35**

**3aCA8. Improved acoustic situation awareness using reduced order models of ocean circulation.** Paul Hursky (Appl. Ocean Sci., LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com), Emanuel F. Coelho (Appl. Ocean Sci., LLC, Springfield, VA), and Pierre F. Lermusiaux (MIT, Cambridge, MA)

Acoustic situation awareness consists of knowing how far own sensors can detect other platforms and how detectable your platform is to their sensors. Since acoustic propagation is reciprocal, differences are down to source levels produced by each platform, transmission loss between them (reciprocal), the relative sensitivity of sensors, and the different directional noise observed at each platform. Noise fields are due to sources present both in the neighborhood (nearby ships) and in distant hubs (harbor shipping). Acoustic propagation models are used to predict transmission loss and received levels at all locations. Such models use sound speed for inputs. Platforms with limited communications can only be updated with the best available sound speed forecasts and measurements on a sporadic basis. We have developed reduced order models for sound speed, which consist of pre-loaded basis sets that capture anticipated ocean state fluctuations, for which very small (easily communicated) sets of coefficients can be used to reconstruct 3D and 4D ocean fields (sound speed and currents) on remote or edge platforms. We will present examples in the simulation of how our reduced order models contribute to improved acoustic situational awareness on such platforms.

## *Invited Papers*

**10:50**

**3aCA9. Optimal stochastic modeling in random media propagation: Dynamically orthogonal parabolic equations?** Wael H. Ali (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, whajjali@mit.edu), Chris Mirabito, Patrick J. Haley Jr., and Pierre F. Lermusiaux (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

Reliable underwater acoustic propagation is challenging due to complex ocean dynamics such as internal-waves and to the uncertain larger-scale ocean physics, acoustics, bathymetry, and seabed fields. For accurate acoustic propagation, capturing the important environmental uncertainties and variabilities and predicting the probability distributions of the acoustic pressure field is then what matters. Prior works towards addressing this goal include (i) wave propagation in random media techniques such as perturbation methods, path integral theory, and coupled-mode transport theory, and (ii) probabilistic modeling techniques such as Monte Carlo sampling and Polynomial Chaos expansions. Recently, we developed a novel technique called the Dynamically Orthogonal Parabolic Equations (DO-ParEq) which represent the sound speed, density, bathymetry, and acoustic pressure fields using optimal dynamic Karhunen-Loeve decompositions. The DO-ParEq are range-evolving partial and stochastic differential equations preserving acoustic nonlinearities and non-Gaussian properties. In this presentation, we showcase the theoretical and computational advantages of the DO-ParEq framework compared to the state-of-the-art techniques in the Pekeris waveguide and wedge benchmark problems, in addition to a realistic ocean example in the New York Bight region.

**11:10**

**3aCA10. Efficient reversible-jump Markov-chain Monte Carlo sampling in trans-dimensional Bayesian geoacoustic inversion.** Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, British Columbia V8W 2Y2, Canada, sdosso@uvic.ca)

This paper considers efficient computational approaches to estimate the posterior probability density (PPD) of seabed geoacoustic profiles in the Bayesian inversion of ocean acoustic data, a numerically intensive problem. Trans-dimensional (trans-D) inversion is applied, which samples probabilistically over an unknown number of seabed layers as well as the layer geoacoustic properties and parameters of the error model (variances and autoregressive coefficients). Sampling is based on the reversible-jump Markov-chain Monte Carlo algorithm, the efficiency of which depends strongly on the formulation of the proposal density by which new candidate models are generated for probabilistic acceptance/rejection. A highly efficient proposal density is presented which combines principal-component (PC) reparameterization with parallel tempering. PC reparameterization applies an adaptive linearized approximation to the PPD as the

proposal density, which provides effective directions and length scales for model perturbations in high-dimensional parameter spaces. Parallel tempering considers a series of interacting Markov chains with successively relaxed likelihood functions, which greatly improves the sampling of multi-modal parameter spaces and trans-D transitions. These approaches are combined by computing different PC reparameterizations for each Markov chain in the parallel tempering formulation. Inversion results are presented as marginal probability profiles for geoacoustic properties, marginalized over the number of layers.

**11:30**

**3aCA11. On the use of machine learning for ray-based ocean acoustic tomography.** Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu), Jihui Jin, Priyabrata Saha (ECE, Georgia Inst. of Technol., Atlanta, GA), Nicholas Durofchalk (School of Phys., Naval Post-Graduate School, Monterey, CA), Justin Romberg, and Saibal Mukhopadhyay (ECE, Georgia Inst. of Technol., Atlanta, GA)

Underwater sound propagation is primarily driven by a non-linear forward model relating variability of the ocean sound speed profile (SSP) to the acoustic observations (e.g., eigenray arrival times). Ocean acoustic tomography (OAT) methods aim at reconstructing SSPs variations (with respect to a reference environment) from changes of the acoustic measurements between multiple source-receiver pairs. We investigated the performance of three different OAT methods: (1) model-based methods (i.e., classical ray-based OAT using a linearized forward model), (2) data-driven methods (such as deep learning) to directly learn the inverse model, and (3) a hybrid solution (i.e., the Neural Adjoint-NA- method) which combines deep learning of the forward model followed by a standard recursive optimization to estimate SSPs. Additionally, synthetic SSPs were generated to augment the variability of the training set. We tested these methods with modeled ray arrivals calculated for a downward refracting environment with mild fluctuations of the thermocline using idealized towed and fixed source configurations. Results indicate that merging data-driven and model-based methods can benefit OAT predictions depending on the selected sensing configurations and actual ray coverage of the water column. But ultimately, the robustness of the OAT predictions depends on the actual dynamics of the SSP variations.

## Session 3aNS

# Noise, Psychological and Physiological Acoustics, and Architectural Acoustics: Acoustics Design of Indoor and Outdoor Firing Ranges and Protection from High Level Impulse Noise

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*Division of Field Studies and Engineering, Noise and Bioacoustics Team, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998*

Steven C. Campbell, Cochair
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Cameron J. Fackler, Cochair
*3M Personal Safety Division, 7911 Zionsville Rd., Indianapolis, IN 46268*

**Chair's Introduction—8:15**

### *Invited Papers*

**8:20**

**3aNS1. Working towards soundscape compatibility of indoor and outdoor shooting ranges with surrounding properties.** Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Keely M. Siebein, Marylin Roa, Jennifer Miller, Gary Siebein Jr., and Matthew Vetterick (Siebein Assoc., Inc., Gainesville, FL)

This paper explores methods to help make shooting ranges compatible with surrounding properties. Sounds from outdoor firing ranges can propagate over 2 miles from the range depending upon topography, vegetation, background sound levels, numbers of shooters, weapon types, and mitigation systems employed at the range. Many communities have regulations for maximum sound levels that can be propagated from one property to another. The impulsive nature of gun shots produces sounds that are not easily measured using conventional acoustical metrics and sound level meters. These items can be studied using three-dimensional computer models. Acoustical data for different weapon types are used as the sound sources. Mitigation options such as shooting sheds, berms, and other strategies can be studied as part of the design process to optimize sonic compatibility with neighboring properties. Similar processes are used for partially and fully enclosed ranges with the addition of the walls, roofs, doors, and HVAC systems for the range included in the models. An architect, engineer, and other design team members work to design specific systems to provide the required mitigation methods. Consultants can evaluate the cost of implementing the mitigation measures so that sonic compatibility is addressed prior to using the range.

**8:40**

**3aNS2. Acoustic treatments for indoor and outdoor firing ranges.** Willaim L. Bergiadis (2580 Sydney Lanier Dr., Brunswick, GA 31525, bill.bergiadis@troyacoustics.com)

Small-caliber firearms can produce impulse noises that frequently exceed 150 dB peak sound pressure level (dB pSPL) and can approach 185 dB pSPL. These impulse noises can present a significant risk for noise-induced hearing loss for the unprotected ear and pose a risk for persons wearing hearing protection that is possibly poorly fitted or insufficient. For range safety officers and personnel who work in the firing range on a regular basis, the daily cumulative effects of noise exposure can lead to increased fatigue and stress. Acoustic treatments of the reflective surfaces can mitigate these health risks. This paper will review some community noise guidelines as well as health and safety regulations. As a manufacturer of acoustic range treatments, the Troy System materials will be reviewed with regards to their laboratory performance and their capabilities to reduce noise in various firing ranges. One aspect of performance that may be overlooked is the safety features of the materials which Troy Acoustics provides, such as flammability, ability to be cleaned, and the resistance to moisture and mold.

9:00

**3aNS3. Analysis of an acoustic propagation model for sources of noise with directivity in indoor environments.** Steven C. Campbell (Ball Aerosp., 4398 Coach Light Trail, Dayton, OH 45424, steven.campbell.28.ctr@us.af.mil), Alan T. Wall (711 Human Performance Wing, Air Force Reseach Lab., Wright-Patterson AFB, OH), Frank S. Mobley (711 Human Performance Wing, Air Force Reseach Lab., Dayton, OH), Reese D. Rasband (Ball Aerosp., Provo, UT), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Small caliber firearm (SCF) noise sources are typically impulsive in nature, possess a large amount of acoustic energy (consequently a large hearing damage risk), and are not omnidirectional. These sources are often operated in indoor shooting ranges where the potential for noise exposure risk is greater due to the reflective surfaces in the room. Indoor sound propagation models require inputs such as geometry, wall material properties, and some quantified source level description. One such room acoustic modeling technique is the Image-Source Method (ISM). ISM typically assumes specular reflections off the walls and represents those reflections as image sources. Many ISM algorithms can operate with high computational efficiency for simple omnidirectional source models, usually represented by a single quantity: sound power. However, ISM models using an omnidirectional source assumption can produce high errors in some scenarios involving highly directional sources, such as SCFs. In this work, an ISM algorithm has been modified to predict listener exposure levels from non-omnidirectional sources in complicated room designs, and has been validated against measured data from an SCF on an indoor Air Force shooting range.

9:20

**3aNS4. Revision of ANSI S12.7 methods for measurements of Impulse Noise.** Richard L. McKinley (4366 OSBORN Rd., Medwat, OH 45341, rich3audio@aol.com)

ANSI S12.7 Methods For Measurements Of Impulse Noise 1986 (R2006) applies to all kinds of impulsive noise, whether discrete event sources, such as quarry and mining explosions or sonic booms, or from multiple event sources such as pile drivers, riveting, nail guns, or rifle or gun firing. The working has been working on proposed changes to the measurement methods, especially high speed digital waveform recording techniques to improve the accuracy of capturing the peak of the waveform as well as the details of the structure. The presentation will describe the proposed changes to the measurement methods and measurement microphones as well as the applications of the revised standard once it is approved.

9:40

**3aNS5. Assessing effect of ground reflections on gunshot noise exposure.** Avery K. Sorrell (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 Eyring Sci. Ctr., Provo, UT 84602, averysorrell137@gmail.com), Jacob R. Smith, Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Reese D. Rasband (Ball Aerosp., Provo, UT), Steven C. Campbell (Ball Aerosp., Dayton, OH), and Alan T. Wall (Ball Aerosp., Wright Patterson Airforce Base, OH)

Investigation of weapons noise exposure outdoors necessitates including the impact of ground reflections. When analyzing the reflections' impact on A-weighted equivalent level (LAeq), the contribution of the reflected waveform to the total level can often be acceptably modeled using a Friedlander equation-based source model and an ideal reflection based on the measurement geometry. However, the Friedlander model does not adequately model the negative phase of a gunshot waveform, making it inaccurate for locations where the waveform of the direct blast and the waveform of the reflection begin to overlap temporally. This merging of the waveforms for locations where the direct and ground-reflected path length difference is relatively small also makes it difficult to separate the waveforms and calculate the reflected wave's contribution to noise exposure. As such, this paper investigates different approaches to estimating the contribution of the ground reflected wave to the LAeq, using direct and reflected waveforms in regions where they are well separated and then adjusting timing and amplitude for different geometries where merging takes place. Results reveal regions where the ground reflection contributes negligibly or even negatively to the total LAeq, which is not predicted by the Friedlander model nor by simple energy-addition approaches. [Work supported by Air Force Research Laboratory through Ball Aerospace Technology Corporation.]

**10:00–10:15 Break**

**Chair's Introduction—10:15**

## *Invited Papers*

10:20

**3aNS6. Comparisons of auditory risk metrics with suppressed and unsuppressed civilian firearms.** William J. Murphy (Stephenson and Stephenson, Res. and Consulting, 2264 Heather Way, Forest Grove, OR 97116, wmurphy@sasrac.com), Gregory Flamme, Stephen M. Tasko (SASRAC, Forest Grove, OR), Donald S. Finan, Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), James E. Lankford (Speech-Language-Hearing Clinic, Northern Illinois Univ., Dekalb, IL), and Michael Stewart (Commun. Disord., Central Michigan Univ., Mount Pleasant, MI)

The assessment of the risk posed by firearm noise is complicated by the divergence in the allowable numbers of rounds (ANORs) returned by the available damage-risk criteria (DRCs) for impulsive noises. The ANORs returned by DRCs are strongly correlated but wide differences exist in absolute ANORs. Some firearms have suppressors and/or discharge low-velocity ammunition, both of which increase the ANORs. The extent to which suppressors and low-velocity ammunition alter the relationships among ANORs has not been determined. In this presentation, we present ANORs across many DRCs for 14 civilian firearms (3 handguns, 10 rifles, 1 shotgun) as a

3a WED. AM

function of suppressor and ammunition. Results will be compared against DRC correlations observed across 54 unsuppressed firearm conditions.

10:40

**3aNS7. When is a firearm suppressor like a hearing protector?** Stephen M. Tasko (Stephenson and Stephenson Res. and Consulting, 2264 Heather Way, Forest Grove, OR 97116, stasko@sasrac.com), William J. Murphy, Gregory Flamme, Kristy K. Deiters (Stephenson and Stephenson Res. and Consulting, Forest Grove, OR), Donald S. Finan, Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), James E. Lankford (Northern Illinois Univ., Dekalb, IL), and Michael Stewart (Central Michigan Univ., Mount Pleasant, MI)

High-level impulse noise exposure from small caliber firearms presents a significant risk of noise induced hearing loss (NIHL) for an unprotected ear. The most common method to reduce the level of noise is to provide hearing protection to both the shooter(s) and potential observers such as firing range instructors or bystanders. Not all hunters and shooters consistently use correctly fit hearing protection. Firearm suppressors provide an engineering noise control that can mitigate a significant portion of the NIHL risk. In this paper, potential analyses for the noise reduction of a firearm suppressor will be presented. The impulse insertion loss as a function of the measurement analysis time window for A-weighted, C-weighted, and unweighted sound pressure levels will be presented. This approach will be contrasted with the ANSI S12-42 standard for measuring impulse peak insertion loss for hearing protection devices.

11:00

**3aNS8. Proposed updates for ANSI/ASA S12.42 impulsive noise test methods.** Cameron J. Fackler (3M Personal Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268, cameron.fackler@mmm.com), William J. Murphy (Stephenson and Stephenson, Res. and Consulting, Forest Grove, OR), Dan Gauger, and Elliott H. Berger (Berger Acoust. Consulting, Indianapolis, IN)

ANSI/ASA S12.42-2010 contains the first standardized test method for measuring the performance of hearing protection devices (HPDs) with high-level impulsive noise. This standard defines the impulsive peak insertion loss (IPIL) as a time-domain metric that quantifies the reduction in peak sound pressure level provided by an HPD for impulsive noises. However, IPIL as defined in S12.42-2010 is dependent on the spectrum of the impulsive noise source used for measurements of HPDs. Recent studies of HPDs with impulsive noise have led to the investigation of frequency-domain metrics and calculation methods. Using frequency-domain calculations allows IPIL to be computed for a wide range of impulsive noises, not only those used for the measurement. Consequently, significant revisions to S12.42 are under consideration. This presentation highlights select proposals including: definition and calculation of the frequency-dependent impulsive insertion loss, incorporation of frequency-domain aspects such as bone-conduction limits to attenuation, and redefinition of IPIL to be calculated from the impulsive insertion loss.

## *Contributed Papers*

11:20

**3aNS9. Measuring impulse insertion loss of hearing protection device using high-level impulse noise.** William J. Murphy (Stephenson and Stephenson, Res. and Consulting, 2264 Heather Way, Forest Grove, OR 97116, wmurphy@sasrac.com), Gregory Flamme, Stephen M. Tasko, Kristy K. Deiters (SASRAC, Forest Grove, OR), Donald S. Finan, Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), James E. Lankford (Speech-Language-Hearing Clinic, Northern Illinois Univ., Dekalb, IL), and Michael Stewart (Commun. Disord., Central Michigan Univ., Mount Pleasant, MI)

The American National Standard, ANSI S12.42-2010, specifies the impulse peak insertion loss (IPIL) for a hearing protection device (HPD) using an acoustic test fixture (ATF) over a range of impulse levels nominally at 132, 150, and 168 dB peak sound pressure level (dB pSPL). The GRAS 45CB ATFs have ear simulators fitted with GRAS 40BP microphones (6.35 mm diameter) that have amplitude limits of about 169 dB SPL (172 dB pSPL). The open ear transfer function measured at the 168-dB level can cause the peak amplitude to exceed the design specification of the microphone. Accordingly, the standard directs the user to measure the 150-dB transfer function between the field microphone and unoccluded ear ($H_{FF,TM}$) and apply it to the field microphone measurement at 168 dB. We have measured the $H_{FF,TM}$ with GRAS 45CB ATF and with the GRAS 45CB-S2 ATF equipped with GRAS 40BH microphones (193 dB/196 dB pSPL). Impulse peak insertion loss measured with both ATFs will be reported and compared.

11:35

**3aNS10. Silencing a free field blast simulator to meet residential standards.** Scott B. Harvey (Phoenix Noise & Vib., LLCC, 5216 Chairmans Ct, Ste. 107, Frederick, MD 21703, sharvey@phoenixnv.com)

The Naval Medical Research Center in Silver Spring, Maryland, plans to install a new free field blast simulator in an existing building on campus. The new blast simulator will be an addition to two current blast simulators and is capable of generating higher dynamic pressure levels, up to 35 psi, than the existing simulators by igniting a mixture of oxygen and acetylene to create the simulated blast. In order to accurately simulate the blast, the outlet of the simulator must be exhausted to the outdoors which will produce noise levels in excess of 140 dBA at five feet. The campus lies within 1800 feet of the nearest residential neighborhood and 950 feet of the nearest commercial neighborhood. The nearest outdoor occupied area on campus, a parking lot, is only 90 feet away. Sound pressure levels produced by the simulator as well as the exhaust requirements were evaluated in order to design a mitigation scheme to reduce noise to meet acceptable residential levels in the parking lot area. Phoenix Noise and Vibration worked with Vibro Acoustics to develop a silencer design to meet these criteria which will be reviewed in the presentation.

## Session 3aPAa

# Physical Acoustics and Computational Acoustics: Infrasound I

Philip S. Blom, Cochair
*Earth & Environmental Sciences, Los Alamos National Laboratory, PO Box 1663, M/S F665, Los Alamos, NM 87545*

Gil Averbuch, Cochair
*Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 31 Wareham Avenue, Onset, MA 02558*

Roberto Sabatini, Cochair
*Embry-Riddle Aeronautical University, 1 Aerospace Blvd, Daytona Beach, FL 32114*

### *Invited Papers*

**8:00**

**3aPAa1. The Multi-Channel Maximum-Likelihood (MCML) method for infrasound detection and wave parameter estimation: Evaluation, implementation, and first applications.** Alexis Le Pichon (CEA, DAM, DIF, Arpajon F-91297, France, alexis.le-pichon@cea.fr), Benjamin Poste, Julien Vergoz, Constantino Listowski (CEA, DAM, DIF, Arpajon, France), and Marine De Carlo (LOPS, Univ. of Brest, CNRS, Brest, France)

The detection capability of the International Monitoring System (IMS) infrasound network is a key concern since ubiquitous coherent noise interfere with the identification of explosive events. The need to distinguish between incoherent wind noise and real coherent infrasonic wave signals has motivated the development of a new parametric stochastic model. We establish the expressions of both the Generalized Likelihood Ratio Test (GLRT) and the Maximum Likelihood Estimator (MLE). The major results are the expression of the asymptotic distribution of the GLRT under null hypothesis, leading to the p-value computation, and the expression of the asymptotic covariance. The Multi-Channel Maximum-Likelihood (MCML) detection is implemented in the time-frequency domain in order to discriminate between interfering signals. Extensive simulations with synthetic signals show that MCML outperforms the state-of-the-art multi-channel correlation detector algorithms in poor signal-to-noise ratio scenarios. We illustrate the performance of MCML on real IMS data, highlighting its capability to characterize the coherent ocean ambient infrasound noise in high resolution through comparisons with theoretical state-of-the-art numerical wave models as implemented at the LOPS research unit of IFREMER in the DATARMOR HPC center.

**8:20**

**3aPAa2. Localization and quantification of the acoustical power of lightning flashes.** Damien Bestard (Institut Jean Le Rond d'Alembert, Sorbonne Université, Paris, France), Thomas Farges (CEA, Arpajon, France), and Francois Coulouvrat (CNRS, Sorbonne Universite - 4 Pl. Jussieu, Institut Jean Le Rond d'Alembert, Paris 75005, France, francois.coulouvrat@upmc.fr)

Lightning is an ubiquitous source of infrasound, and an essential climate variable. To observe lightning flashes, thunder measurement efficiently complements electromagnetic methods. Using acoustical arrays, time delays between sensors inform on the direction of sound arrival, while the difference between emission time and sound arrival provides the source distance. Combining two allows a geometrical reconstruction of lightning flashes viewed as sets of sound sources. The measured sound amplitude can also be back-propagated, compensating for absorption and density stratification. This allows us to evaluate the acoustical power of each detected source and the total power of an individual flash. This methodology has been carried out to analyse data from two campaigns in Southern continental France in 2012 and in Corsica in 2018. In Corsica, power from reconstructed sources could also be forward-propagated towards several isolated microphones and compared to measurement there, providing an additional validation of the method. A large number of events from the two campaigns has been analysed, including negative and positive cloud to ground discharges and intra-cloud ones. The analysis outlines the method efficiency, and the strong variability of lightning as sound sources, in terms of both power spatial distribution and global values.

**8:40**

**3aPAa3. Infrasound generated by meteorological fronts as a background for infrasound monitoring of explosions.** Igor P. Chunchuzov (Atmospheric Dynam., Obukhov Inst. of Atmospheric Phys., 3 Pyzhevsky Per, Moscow 119017, Russian Federation, igor.chunchuzov@gmail.com), Sergey Kulichkov (Atmospheric Dynam., Obukhov Inst. of Atmospheric Phys., Moscow, Russian Federation), Oleg Popov (Atmospheric Dynam., Obukhov Inst. of Atmospheric Phys., Moscow, Russian Federation), and Vitaly Perepelkin (Atmospheric Dynam., Obukhov Inst. of Atmospheric Phys., Moscow, Russian Federation)

The results of study of temporal variations of the characteristics of infrasound (amplitudes, coherences, grazing angles, azimuths and horizontal phase speeds) detected during a passage of warm and cold fronts through the networks of microbarometers in the Moscow region are presented. .Infrasound radiated during periods of weather changes is an almost continuous background against which infrasound monitoring of explosions in the atmosphere is carried out. The significant differences observed in the characteristics of infrasound from warm and cold fronts are found. Such differences must be taken into account when detecting infrasound precursors of atmospheric storms. A possible aerodynamic mechanism for the generation of infrasound caused by the turbulent air flow around the geometric irregularities of the surface of meteorological front is proposed. [Work supported by RSF Grant No. 21-17-00021.]

**9:00**

**3aPAa4. Investigating infrasonic signal amplitudes at the lateral edges of propagation ducts.** David N. Green (AWE Blacknest, Brimpton, United Kingdom, dgreen@blacknest.gov.uk) and Alexandra Nippress (AWE Blacknest, Reading, United Kingdom)

Azimuthal variation in expected infrasonic signal strength is often modelled using Nx2D finite-frequency acoustic propagation models. Such simulations frequently exhibit rapid changes in transmission loss ($>$30 dB across 5°) at the lateral edges of stratospheric propagation ducts, due to the sensitivity of acoustic ducting to the along-path windspeed. The inclusion of microbarometers in the USArray Transportable Array, with an inter-station separation of $\sim$70 km, has provided improved resolution across the lateral extent of tropospheric and stratospheric ducts within which infrasound is propagated over local and near-regional distances (10s to 100s km). We analyse signals from two explosions that generated infrasound across a broad swath of USArray microbarometers. Signals from the October 2012 Camp Minden Ammunition Plant explosion, Louisiana, show smoothly varying amplitudes across the stratospheric duct edge while those from the October 2011 Atchison Grain Elevator explosion, Kansas, exhibit less azimuthal variation. These signals provide a basis for comparison with current numerical modelling methods. Understanding infrasonic amplitudes at the lateral duct edge is important for both accurate signal interpretation from events of interest and for detection capability assessments of infrasound sensor networks. UK Ministry of Defence © Crown Owned Copyright 2022/AWE

**9:20**

**3aPAa5. Measurement of infrasound sensor self-noise.** Thomas B. Gabrielson (Penn State Univ., State College, PA), B. J. Merchant (Sandia National Labs., Albuquerque, NM), Dominique Rodrigues (Laboratoire National de métrologie et d'Essais, Trappes, France), and Chad M. Smith (Penn State Univ., PO Box 30, State College, PA 16804, cms561@psu.edu)

Self-noise is a critical performance characteristic of sensors intended for weak-signal detection and localization; however, the lower the self-noise, the more challenging the measurement. A recent international sensor-characterization exercise coordinated by the Provisional Technical Secretariat of the Comprehensive Nuclear-Test-Ban Treaty Organization included self-noise measurement of several infrasound sensors from 0.01 Hz to 10 Hz. Of three common methods for self-noise assessment—isolation of the sensor from external excitation, subtraction of common (coherent) components among several sensors, or de-activation of the sense mechanism—the first two were used in this exercise and the results highlight important measurement issues. In the infrasonic frequency range, isolation from external excitation is challenging. A sealed, thick-walled chamber attenuates ambient noise but heating from dissipation of power in the chamber interior can induce convection with large, low-frequency pressure fluctuations in the chamber. Capping the inlet(s) of a sensor creates a small, closed volume with strong coupling between temperature and pressure fluctuations. Subtraction of coherent components shared by co-located sensors can be effective in reducing the influence of ambient excitation; however, the process may be frustrated by errors in subtraction of large, nearly equal components or by unexpected electrical coupling.

**9:40**

**3aPAa6. Automated detection of dust-devil-induced pressure signatures.** Louis Urtecho (NASA JPL/ California Inst. of Technol., 1905 Liverpool Dr., Plano, TX 75025, lurtecho@smu.edu), Siddharth Krishnamoorthy (NASA JPL/ California Inst. of Technol., Pasadena, CA), Elizabeth Berg, Elizabeth Silber, Daniel Bowman, Andrew Sparks (Sandia National Labs., Albuquerque, NM), Miro Giannone (Sandia National Labs., Dallas, TX), Leo Martire, and Attila Komjathy (NASA JPL/ California Inst. of Technol., Pasadena, CA)

Dust devils are common on Mars and in Earth's arid regions. On Mars, studying the dust devil population is important for understanding dust loading of the Martian climate and has important consequences for robotic and future human exploration. Studying dust devils on Mars is difficult due to insufficient spatio-temporal coverage. Our team is analyzing an infrasound dataset encompassing 7 years of data recorded at the Nevada Nuclear Security Site in the Mojave desert, to identify and characterize terrestrial dust devils as analogues for Martian dust devils. However, the size of this long-duration dataset mandates the use of automated techniques for the detection of dust devils.The automated detection scheme that we have developed follows a two-step process. The first step comprises of significance testing in the time-frequency domain using wavelet transforms and an empirical background red noise model. The second step is a correlation-based, template-matching detector, which utilizes the "heartbeat" pressure signal produced by the dust devil convolved with the microbarometer's impulse response. In this presentation, we will discuss our automated dust devil detection algorithm, error characterization in the identification of signatures using Monte Carlo analysis and the application of the automated detector to the long-term monitoring dataset.

**10:00–10:15 Break**

**10:15**

**3aPAa7. Modeling of infrasonic and acoustic-gravity wave propagation, nonlinear evolution, and observable effects.** Jonathan B. Snively (Dept. of Physical Sci., Embry-Riddle Aeronautical Univ., 1 Aerosp. Blvd, Daytona Beach, FL 32114, snivelyj@erau.edu), Donna A. Calhoun (Dept. of Mathematics, Boise State Univ., Boise, ID), Pavel A. Inchin (Ctr. for Space and Atmospheric Res. (CSAR), Embry-Riddle Aeronautical Univ., Daytona Beach, FL), Roberto Sabatini (Dept. of Physical Sci., Embry-Riddle Aeronautical Univ., Daytona Beach, FL), Christopher J. Heale (Ctr. for Space and Atmospheric Res. (CSAR), Embry-Riddle Aeronautical Univ., Daytona Beach, FL), and Matthew D. Zettergren (Dept. of Physical Sci., Embry-Riddle Aeronautical Univ., Daytona Beach, FL)

Mechanical disturbances associated with hazardous events—e.g., earthquakes, explosions (volcanic or man-made)—and severe weather – generate broad spectra of infrasound and acoustic-gravity waves (AGWs). These wave signals may provide diagnostic insight into the processes that generated them. They are routinely detected as fluctuations in atmospheric pressure, measured at ground or from balloon-borne platforms; at lower frequencies ($<$1 Hz), and where they may attain sufficient amplitudes at high altitudes, they may also be measured via the fluctuations that they impose in densities of layered species throughout the atmosphere and ionosphere. Thus, they provide complementary remote sensing opportunities, where waves and their effects, especially above and surrounding larger sources, may be diagnosed as they propagate. We review recent progress and techniques for high-fidelity modeling and simulation, to capture the propagation and evolution of low-frequency infrasound and AGWs throughout the atmosphere, from 0–500 km altitude (from surface to exobase). Strategies to (1) efficiently extend model simulation domains well into the diffusive thermosphere, to (2) connect models of atmospheric dynamics to those for other measurable processes (e.g., the ionosphere), and to (3) construct simulations that extend from source processes to specific remote sensing methodologies are discussed.

**10:35**

**3aPAa8. Numerical modeling of infrasound propagation in a stratified atmosphere with small-scale inhomogeneities.** Codor Khodr (LMFA, École Centrale Lyon, 36 Ave., Guy de Collongue, Écully 69130, France, codor.khodr@ec-lyon.fr), Didier Dragna, Philippe Blanc-Benon (LMFA, École Centrale Lyon, Ecully, France), Régis Marchiano (Sorbonne Université, Paris, France), Ludovic Aubry (CEA, DAM, DIF, Bruyères-le-Châtel, France), Olaf Gainville, and Christophe Millet (CEA, DAM, DIF, Bruyères-le-Châtel, France)

A major focus in infrasound research is the inclusion of realistic atmospheric flows in numerical propagation models. This usually encompasses temperature gradients and horizontal wind components, which create atmospheric waveguides. Additionally, gravity waves lead to small-scale variations of the ambient wind in the middle atmosphere, between 10 and 100 km in altitude. Using an internal gravity wave model from the literature [Gardner, *J. Atmos. Terr. Phys.* (1996)], we parametrize random fluctuations of the wind profiles, where the initial background state is based on a published case study [Sabatini *et al.*, *J. Acoust. Soc. Am.* (2016)]. Acoustic propagation from an impulsive point source located at the ground is then modelled with solvers of the PLetma package [Khodr *et al.*, *J. Acoust. Soc. Am.* (2021)], namely, ray tracing, normal modes, one-way equation and FDTD, for frequencies higher than 0.01 Hz. Cases with and without wind are considered, and propagation effects are investigated by a sensitivity analysis on source parameters and characteristics of the gravity wave. [This work was conducted within the framework of LETMA, a Contractual Research Laboratory shared between CEA, CNRS, Ecole Centrale de Lyon, C-Innov, and Sorbonne Université.]

**10:55**

**3aPAa9. Local seismo-acoustic waves, but from the other side of the world.** Jelle D. Assink (R&D Seismology and Acoust., KNMI, Utrechtseweg 297, Utrecht 3731GA, Netherlands, jelle.assink@knmi.nl)

It is well understood that shallow seismic events near the Earth surface couple readily to the atmosphere where they can be observed as infrasonic waves. Typically, the observed sequence of infrasound arrivals includes a first arrival (i.e., "local infrasound"), which corresponds to the coupling of seismic body and surface waves to atmospheric infrasound, near an infrasound array. Under favorable atmospheric ducting conditions, additional coupled arrivals arrive later in time, having propagated throughout the (slower) atmosphere for part of the propagation path. In this presentation, we show that seismo-acoustic waves can couple from very deep earthquakes that occur on the other side of the world. This is demonstrated using seismo-acoustic observations from a M8.2 earthquake in Fiji that generated infrasound throughout Europe. Our analysis shows that various phases that reflected off the inner core boundary (ICB) were detected as pressure waves. This suggests that signals that are detectable on microbarometer arrays may have a surprising origin.

**11:15**

**3aPAa10. The Las Vegas infrasound array: Long term deployments for the characterization of urban environments.** Fransiska Dannemann Dugick (Sandia National Labs., 3600 Calle del Ranchero, Albuquerque, NM 87110, fkdanne@sandia.gov), Nora Wynn, Elijah Bird, Daniel Bowman (Sandia National Labs., Albuquerque, NM), Melissa Wright, Douglas Seastrand (Mission Support & Test Services, Las Vegas, NV), and Jonathan Lees (Univ. of North Carolina, Chapel Hill, NC)

The Las Vegas Infrasound Array (LVIA) is a network of eleven infrasound sensors deployed from November 2019 through September 2022. While ambient infrasound noise in high and low-noise rural environments has been well characterized, little attention has focused on similar characterization in urban areas with presumed higher background noise levels. The LVIA long-term deployment provides an unprecedented opportunity to study urban infrasound and low frequency audio (20–500 Hz). In addition, large scale shutdowns due to the COVID-19 pandemic provide the ability to discriminate between background noise sources as closures reduced human-generated noise while natural signals remained stable. Within this presentation we will provide an overview of the LVIA installation, focusing on data quality. In addition, we will discuss comprehensive background noise models in urban regions, focusing on presenting probability density functions (PDFs) and median, 5$^{th}$ percentile, and 95$^{th}$ percentile amplitude values to evaluate variations in frequency and amplitude. We will summarize observed trends in background noise over time, highlighting sharp declines in acoustic power following COVID-19 shutdowns. Both sets of analyses will be combined to evaluate periodicities in urban acoustics throughout the city of Las Vegas. [ SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

11:35

**3aPAa11. From atmospheric profile statistics to transmission loss statistics.** Roger M. Waxler (NCPA, Univ. of MS, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu), Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., Los Alamos, NM), Adedotun Lawal (NCPA, Univ. of MS, University, MS), Garth Frazier (NCPA, Univ. of MS, Oxford, MS), and Claus Hetzer (NCPA, Univ. of MS, Tempe, AZ)

Infrasound propagation depends critically on the temperature and wind velocity profiles at the time of signal propagation. These vary wildly, showing at best qualitative systematic behavior. It follows that any predictions of signal detection capability are necessarily statistical in nature. One approach is to collect a large number of historical temperature and wind velocity profiles, for a specific time and location, and use this as a sample space from which to generate a statistical model for the atmosphere as a propagation medium. To generate a statistical model for the expected transmission loss one must run a propagation model through a sampling of atmospheric profiles sufficient to reproduce the statistical behavior. The sampling is done using an Empirical Orthogonal Function (EOF) decomposition. The advantages of the use of an EOF decomposition will be discussed and examples from transition zone (ranges less than 100 km) and regional (ranges of several hundreds of kilometers) will be presented. Given a model for the turbulent pressure fluctuation levels near a receiving array, signal detection probability functions can be estimated.

WEDNESDAY MORNING, 7 DECEMBER 2022 LIONEL, 8:20 A.M. TO 11:25 A.M.

## Session 3aPAb

## Physical Acoustics and Education in Acoustics: My Favorite Homework Problems (Based on Measurements, Demonstrations, or Experimental Data)

Daniel A. Russell, Cochair
*Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802*

Thomas L. Szabo, Cochair
*Biomedical Engineering, Boston University, 44 Cummington Mall, Boston, MA 02215*

Preston S. Wilson, Cochair
*Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 East Dean Keeton Street, Mail Stop: C2200, Austin, TX 78712-0292*

### *Invited Papers*

8:20

**3aPAb1. Homework problems inspired by one of humanity's loudest achievements: The Saturn V.** Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu), Grant W. Hart, Mark C. Anderson, and Logan T. Mathews (Dept. of Phys. and Astronomy, Phys. and Astronomy, Brigham Young Univ., Provo, UT)

This paper discusses advanced undergraduate homework problems related to one of humanity's greatest technological (and loudest) achievements—the Saturn V rocket. The homework problems are provided as supplementary material to a manuscript by the same authors, presently accepted to the Education in Acoustics Special Issue of The Journal of the Acoustical Society of America. Entitled "Saturn-V Sound Levels: A letter to the Redditor," the article discusses how much of the information found online about Saturn-V acoustics is inaccurate and seeks to correct misconceptions and errors using available data and models. The supplementary homework problems increase pedagogical value of the article, which can be used as a teaching tool for concepts such as sound levels, power versus pressure, radiation efficiency, importance of decibel references, etc. In this presentation, an overview of the different homework problems will be given and specific problems solved with help from the audience.

8:40

**3aPAb2. Blackstock 9-3: Spherical waves, reflection, and attenuation.** Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu)

One of my favorite homework problems for waves in fluids or a general acoustics class is Blackstock 9-3, which involves a submarine positioned 50 m below the ocean surface and 500 m above the seafloor. The problem asks students to calculate the impedance of the seafloor given a frequency and known sound pressure levels received by the submarine after bouncing off the ocean surface or the seafloor. In part a, the calculation is made assuming a lossless medium and then in part b, using attenuation. The reasons this is one of my favorite problems are that it is real-world derived, combines many concepts, and requires understanding of wave propagation physics to solve; there is no simple equation (that I have found at least) that will immediately provide the correct answer. Students often struggle with what type of wave is emitted (plane versus spherical), then, if spherical waves, how to account for the boundary—especially on the seafloor where one must calculate the reflection coefficient. Every year this problem allows for good discussion in class and homework review sections.

9:00

**3aPAb3. Favorite problems on acoustic transducers and linear systems theory.** Preston S. Wilson (Walker Dept. Mech. Eng., The Univ. of Texas at Austin, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

Practitioners of acoustics sometimes forget how lucky we are that the vast majority of acoustic phenomena and devices we encounter in daily life are linear and solutions using linear systems theory work very well for predicting behavior and designing devices. Typical examples are microphones, loudspeakers and acoustic propagation at sufficiently low excitation levels. One might contrast this to a viscous dashpot, which is also commonly modeled as a linear device. The sliding seals in all viscous dashpots render them essentially-nonlinear from the start. One favorite problem I give year after year, in various forms, is reported here. A generic acoustic transducer, based on a linear simple harmonic oscillator is presented and often the system response function is given in the problem statement. Derivation of the transfer function could be a problem from earlier or later in the class. The students are asked to predict the time domain output of the transducer, subject to a periodic forcing composed of a sawtooth wave, using frequency-domain convolution and Fourier series. Various sub-questions explore the effect of transducer bandwidth and damping on the output signal. A variation of this problem using time-domain convolution is also a favorite. Predictions can be compared to measurements.

9:20

**3aPAb4. When is a Pepsi® can more than just a beverage container? When it is the inspiration for an acoustics homework problem!.** Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

When teaching acoustics to graduate students, some of most effective homework problems require students to compare multiples approaches, combine concepts from related topics, and evaluate numerical answers using experimental data. A problem from David T. Blackstock's *Fundamentals of Physical Acoustics* textbook asks students to model an empty beverage can as a Helmholtz resonator, first using the simple formula and the measured dimensions of a real can. Then, students are asked to evaluate whether the simple formula is valid and to provide a more accurate solution if it is not. Most students find an improved impedance approach solution which changes the calculated frequency by approximately 9%. However, most students fail to realize that this problem is exactly analogous to the problem of elastic longitudinal waves in a fixed, mass-loaded bar, with the related first approximation of a simple mass-spring system in which 1/3 of the spring's mass must be included. Further challenges arise when students attempt to measure the resonance of the empty can and obtain a measured frequency (with damping) that does not match calculated values. The pop-can resonator problem, its analogues, and some experimental challenges will be discussed in detail.

9:40

**3aPAb5. Imaging three dimensional objects with ultrasound.** Thomas L. Szabo (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

The ultrasound imaging laboratory provided students with the opportunity of determining what an unknown three dimensional object was from two-dimensional images. Armed with a portable diagnostic ultrasound imaging system, and internal calipers for quantification, students were given unknown objects from the creepy crawly collection. Each object was immersed in a small tub of opaque fluid. Students could adjust the imaging system to give different cross-sections or cut planes through the object. From this information and linear calipers, they were to determine what the object was and provide a quantitative three dimensional sketch. This experience gave them a taste of the chief difficulty in diagnostic imaging of the body: deciphering and recognizing tissue structures and organs from partial views. Other imaging exercises included the use of imaging phantoms to measure spatial and temporal resolution as a function of depth and other controls.

**10:00–10:15 Break**

10:15

**3aPAb6. But how does this work in the real world?** Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@utexas.edu)

One common student critique of homework problems in lecture courses is that the complexities of the real-world are removed in order to create problems that reinforce a specific physical phenomenon or mathematical principle that is relevant to the content discussed in class. This tends to produce problems that are effective in teaching specific topics but may seem too distant from applications to provide an intuitive understanding of the subject matter. Fortunately, some topics in acoustics are very well-suited to use experimental data.

This talk will highlight a selection of homework problems that make use of experimental data to illustrate physical principles covered in graduate courses in the acoustics program at The University of Texas at Austin. The first problem-set uses data from ultrasonic measurements to estimate the phase and group velocities in various materials and the second problem-set uses measurements of broadband exponential chirps played in a concert hall to estimate a range of performance metrics of the space. The talk will discuss how these exercises fit into the overall course structure, present student solutions and feedback, discuss learning outcomes, and make suggestions on how successfully integrate these kinds of problems into a lecture course.

**10:35**

**3aPAb7. A potpourri of homework and test problems inspired by published research and/or classroom demonstrations.** Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

Fundamental engineering and physics topics in an acoustics course can be made more relevant and interesting when applied to problems involving real-world scenarios, both from technical apparatus and from the natural world. This presentation will highlight several "fun" homework and test problems inspired by reading acoustics journal papers or data collected during a classroom demonstration. For example: tree crickets cut holes in leaves to convert the ineffective sound radiation from a dipole source into a much more effective baffled monopole. Spiders locate prey on their webs by comparing arrival times between transverse and longitudinal waves in web fibers; a related problem can be modeled as a fixed-fixed string with a point-mass load (originally posed by Lord Rayleigh). A compound string presents a challenging boundary value problem, with theoretical results easily compared to animations and physical demonstrations and application to musical instruments. Deceptively "simple" demonstrations of standing waves in open pipes and Helmholtz resonators can be used to estimate damping losses and mass loading. Measured data from a tuning fork spectrum can lead students to evaluate appropriate boundary condition choices. Additional examples will be discussed as time allows.

## *Contributed Papers*

**10:55**

**3aPAb8. Physical acoustics homework problems written by students: Undisciplined, irreverent, and original.** Chirag Gokani (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78758, chiragokani@utexas.edu), Michael R. Haberman, and Mark F. Hamilton (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

"Study hard what interests you the most in the most undisciplined, irreverent and original manner possible" [*Perfectly Reasonable Deviations from the Beaten Track*, Richard P. Feynman, Basic Books, 2005]. Writing original homework problems is a powerful way students of physical acoustics can practice Feynman's advice. Three problems that involve a broad range of concepts covered in introductory graduate-level physical acoustics courses illustrate how the student-author unleashed his creativity in an undisciplined manner, injected his problems with an irreverent sense of humor, and derived a great sense of originality and ownership over physical acoustics. The problems synthesize David T. Blackstock's problems 1D-2, 1E-3, 1G-1, 1G-3, 7-6, 10-10, and 10-11 [*Fundamentals of Physical Acoustics*, David T. Blackstock, Wiley, 2000], addressing concepts including acoustic intensity, impedance, horns, enclosures, and radiation. [CAG was supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

**11:10**

**3aPAb9. Students are sitting in a room.** Andrew A. Piacsek (Dept. of Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926-7422, andy.piacsek@cwu.edu)

Alvin Lucier's seminal 1969 composition and recording, "I Am Sitting in a Room" provides a fascinating and engaging tool for students to improve their understanding of room acoustics, as well as their skills in analyzing spectral data. This meta-composition, in which Lucier's description of the process plays an essential role, involves speech being recorded in a particular room, then repeatedly played back and re-recorded in the same room. Some frequencies in Lucier's voice excite room modes; these are reinforced with each iteration, eventually to the exclusion of all other frequencies. This source material can be the basis for either a lab activity or a homework problem. For the latter, students are provided with high resolution spectrograms created by the instructor from two excerpts from Lucier's 1981 recording (available online), one from an early iteration, and the other from a later one. Students are tasked with identifying the frequencies that correspond to room modes, then inferring the dimensions of the room used in the recording ("…different from the one you are in now.") A variety of learning objectives at the undergraduate level, as well as challenges and successes from prior experience, will be discussed.

# Session 3aSA

## Structural Acoustics and Vibration, Computational Acoustics, Engineering Acoustics, and Physical Acoustics: Acoustic Metamaterials

Christina Naify, Cochair
*Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758*

Alexey Titovich, Cochair
*Naval Surface Warfare Center, Carderock Division,*

Bogdan-Ioan Popa, Cochair
*Univ. of Michigan, 2350 Hayward St, Ann Arbor, MI 48109*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**3aSA1. Non-reciprocity and roton dispersion in non-local piezoelectric metamaterials.** Christopher Sugino (Mech. Eng., Stevens Inst. of Technol., 1 Castle Point, Hoboken, NJ 07030, csugino@stevens.edu)

This talk will investigate the behavior of piezoelectric acoustic metamaterials with non-local circuit interactions, demonstrating that unbalanced, directional interactions yield non-reciprocal, roton-type dispersion behavior. The coupling between mechanical and electrical domains introduces additional dispersion branches whose periodicity is determined by the non-local circuit spacing. When these branches intersect the original dispersion curves of the system, non-reciprocal band gaps are opened in the mechanical domain. Furthermore, although the system is non-Hermitian, the appropriate use of local electrical resistance ensures that all dispersion branches are stable. This offers a practical alternative to realize non-reciprocal propagation that does not rely on nonlinearity or time-varying components, but instead on well-established directional electrical circuits. Numerical simulations show that the strength and spacing of the non-local interactions determines the frequency of non-reciprocity, providing a straightforward mechanism to program the dispersion properties of the system. The talk will conclude with preliminary experimental results and a discussion of system stability.

**8:25**

**3aSA2. Complete and detailed inverse design for any arbitrary dispersion relations via non-local metamaterials.** Arash Kazemi (Mech. Eng., Univ. of Utah, Salt Lake City, UT), Kshiteej Deshmukh (Dept. of Mathematics, Univ. of Utah, Salt Lake City, UT), Yunya Liu (Mech. Eng., Univ. of Utah, Salt Lake City, UT), Bolei Deng (MIT, Salt Lake City, UT), Henry Fu (Mech. Eng., Univ. of Utah, Salt Lake City, UT), and Pai Wang (Mech. Eng., Univ. of Utah, 1495 E 100 S, MEK Bldg., Salt Lake City, UT 84112, pai.wang@utah.edu)

Phononic crystals and vibro-elastic metamaterials are characterized by their dispersion relations—how frequency depends on wave number/vector. While there are many existing methods to solve the forward problem of obtaining the dispersion relation from any arbitrarily given design. The inverse problem of obtaining a design for any arbitrarily given dispersion bands have only had very limited success so far. Here, we report a new design scheme capable of leading to arbitrary dispersion relations by incorporating non-local interactions between unit cells. Considering discrete models of one-dimensional mass-spring chain, we investigate the effects of both local (i.e., springs between the nearest neighbors) and non-local (i.e., springs between the next nearest neighbors and other longer range springs) interactions. First, we derive the general governing equations of non-local phononic chains. Next, we examine all design constraints for a linear, periodic, passive, statically stable, non-gyroscopic, and free-standing system. Finally, we perform analytical calculations and numerical simulations to solve the inverse problems. The results illuminate a new path toward novel wave manipulation functionalities, such as sophisticated combinations of roton-like, maxon-like, undulation-point and other zero-group-velocity (ZGV) modes, as well as multi-wavelength and multi-speed propagations of the same mode at the same frequency.

8:45

**3aSA3. Deep reinforcement learning-based framework for the design of broadband acoustic metamaterials.** Tristan A. Shah (San Jose State Univ., San Jose, CA) and Feruza Amirkulova (Mech. Engineering, San Jose State Univ., 1 Washington Sq., San Jose, CA 95192, feruza.amirkulova@sjsu.edu)

This talk presents a general framework for discovering optimal designs of metamaterials with deep reinforcement learning (RL). In our previous research [1], we applied RL to acoustic cloak design and optimized the design parameters of a planar configuration of up to 12 cylindrical scatterers to minimize the scattering of an incident acoustic wave. The designs produced by RL algorithms are comparable and in some cases superior to those produced by *fmincon* solver. However, the challenge of finding an efficient approach for inverse design, is still in its emergent stage, requiring further modifications of these models and significant computational resources to increase the complexity of a design. This creates a significant challenge when attempting to enable model scaling since training times surge to unfeasible levels. We will discuss methods to counteract this challenge and present an RL-based framework for discovering optimal designs of meta-devices. We expanded our research [1] to large numbers of scatterers, and new designs with variable cylinder radii, position, and material. We noticed a dramatic improvement in the training times of RL agents due to the speed increase provided by the Julia programming language which accelerates episode runtimes 10-times compared to our previous models in python [1]. [1] T. Shah *et al.*, "Reinforcement learning applied to metamaterial design," *J. Acoust. Soc. Am.* **150**(1), 321–338 (2021).

## *Contributed Papers*

9:05

**3aSA4. Experimental demonstration of scalable active metamaterials with interacting cells of tunable bulk modulus.** Dylan Kovacevich (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, dkovac@umich.edu) and Bogdan-Ioan Popa (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Active metamaterials can be designed to exhibit acoustic properties that are challenging to obtain using passive structures without relying on narrowband resonant behavior. Arrangements of unit cells that independently sense the local acoustic conditions and produce a coherent response, which is determined by the gain of electronics between the sensor and driver, act as a material with programmable properties. Such an active metamaterial has been previously demonstrated, but only for a single or a few non-interacting cells due to a limited understanding of how the programmed gains relate to the effective properties. Recently we developed a polarized source model to address this challenge, and based on that work we now demonstrate experimentally an active metamaterial composed of interacting cells with programmable bulk modulus. Each unit cell consists of a speaker and microphone mounted in the bottom plate of a 2D waveguide, with electronics of adjustable gain stored underneath. The field generated by a $2 \times 3$ arrangement of cells in response to an incident pulse was compared to the simulated scattered field of the equivalent continuous material. By tuning the gain, we accurately reproduced the behavior of a medium with negative, fractional, and large relative bulk modulus.

9:20

**3aSA5. Time-reversal in phononic crystal-based water filled pipe.** Utban Ahmed (Dept. of Civil and Environ. Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, New Territories, Hong Kong, uahmed@connect.ust.hk), Saber Nasroui, Moez Louati, and Mohamed S. Ghidaoui (Dept. of Civil and Environ. Eng., The Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Time-reversal based source localization in a conventional water-filled pipe using acoustic waves is hamstrung by the diffraction limit ($\lambda/2$). In this paper, time reversal in a novel metamaterial-based water pipeline is studied where a conventional rigid pipe is modified to have periodic square corrugations. Based on the choice of periodicity, such a novel waveguide can support subwavelength resonance in the low frequency regime while exhibiting Bragg resonance, avoided crossing of modes and forbidden bands in diffraction regime. This paper focuses on the latter such that a band structure is obtained. Full wave simulation performed on COMSOL verifies the phononic crystal-like properties in the diffraction regime. Time reversal refocusing is performed by placing a point source in the middle of the corrugated section while a receiver outside the said section records the response of the system. The receiver acts as a Time Reversal Mirror (TRM) such that it flips and reinjects the recorded signal into the system which retraces its path and focuses on its source. Time-reversal refocusing is found to be sensitive to the recorded frequency grains in addition to the energy propagated from the source such that if the source encompasses a band gap, the time reversal quality deteriorates.

9:35

**3aSA6. Optimization-based designs for communicating through impenetrable barriers with mechanical waves.** Chandler Smith (Simulation Modeling Sci., Sandia National Labs., 1515 Eubank Blvd SE, Albuquerque, NM 87123, chasmit@sandia.gov), Timothy Walsh, and Volkan Akcelik (Simulation Modeling Sci., Sandia National Labs., Albuquerque, NM)

Piezoelectric-based acoustic energy transmission is an emerging alternative communication method that allows signal transmission across a barrier without wires or physical penetration. A major challenge in the design of an electrical-mechanical coupled communication channel is how to resolve the undesirable signal transformation known as acoustic multipath as the elastic wave propagates through a multi-layered deformable solid. Solutions largely investigated by the communications community are typically based on sophisticated signal processing methods. However, we approach the problem through the lens of computational mechanics and design optimization. Proposed method is to design a piezoelectric stack that mitigates acoustic multipath by optimizing the thickness of various stack layers and the material properties of the epoxy-based backing. We optimize the design parameters of a 3-D coupled electrical-mechanical finite element model using partial differential equation (PDE) constrained optimization. We leverage higher order polynomial elements (P-elements) and high-performance computing to solve the computationally expensive 3-D wave propagation problem in the MHz frequency range. The results demonstrate how finite element modelling and optimization can be used to achieve optimal physical channel designs for mechanical acoustic-based communication. [SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525.]

**9:50–10:05 Break**

10:05

**3aSA7. Ultra-stiff and ultra-light architected metamaterial for vibration mitigation.** Edward Huang (Enloe Magnet High School, University Park, State College, PA 16801, erhuang23@gmail.com), Mourad Oudich, Hyeonu Heo, and Yun Jing (Graduate Program in Acoust., Penn State Univ., State College, PA)

We designed, fabricated, and experimentally validated an architected mechanical metamaterial vibration shielding while ensuring high effective stiffness with low density. The structure is made of a periodic succession of octet truss units with alternating circular strut diameters. The octet truss endows the meta-structure with a high effective stiffness and a low effective density, while alternating the units with different struts radii enables the opening of a wide band-gap for elastic waves. Samples were 3D printed and experimentally validated using a shaker and accelerometer. Unidirectional

compression test was also performed to characterize the mechanical properties of the metamaterial. The results are in good agreement with the numerical prediction.

**10:20**

**3aSA8. Meta-interface textures for increased frictional damping in elastic structures.** Iyabo G. Lawal (Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712, iyabo@utexas.edu) and Michael R. Haberman (Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

The unique behavior of acoustic and elastic metamaterials results from engineered subwavelength material composition and geometry. Though there has been a profusion of metamaterial research over the last two decades, relatively little focus has been on the dissipation of vibro-acoustic energy. Most research on metamaterials for acoustic damping have focused on viscous or viscoelastic effects [1]. However, the design of interfaces within materials or at connecting surfaces between elastic structures to amplify micro-friction events due to the relative motion of contacting surfaces is of interest for the reduction of vibro-acoustic energy. Dissipation due to friction is best represented as a nonlinear constitutive response. Here, we report on the use of friction via meta-interface textures to create interfaces with elevated damping, leading to increased macroscopic dissipation. We employ the five-parameter Bouc-Wen model to describe the complex behavior at these interfaces and study the influence of subwavelength interface texture on the parameters of the friction model. This increased dissipation from this approach can be used to reduce structural vibrations and the transmission of acoustical energy in structural components. [1] C. L. Bacquet *et al*., "Metadamping: Dissipation emergence in elastic metamaterials." *Adv. App. Mech.* **51**, 115i–164, (2018).

**10:35**

**3aSA9. Topology optimization design of elastodynamic metasurfaces based on resonance gaps and antiresonance matching.** Daniel Giraldo Guzman (Mech. Eng., Penn State, 132 Reber Bldg., Universtiy Park, PA 16802, dzg5526@psu.edu), Mary Frecker (Mech. Eng., Penn State, University Park, PA), and Parisa Shokouhi (Eng. Sci. and Mech., The Pennsylvania State Univ., University Park, PA)

Elastodynamic locally resonant metasurfaces have emerged as effective means to control low-frequency Lamb and Rayleigh wave propagation. Currently, the design of these metasurfaces is typically achieved through experimentation and parameter tuning; no systematic design methodology has been established. In this work, a topology optimization-based design methodology is developed to tailor resonance and antiresonance frequencies of local resonators which compose locally resonant metasurfaces. The optimization seeks to match the antiresonance frequency to the bandgap's desired central frequency while a resonance gap around that target frequency is sought to widen the bandgap. Both the antiresonance matching and the resonance gap are included in the optimization problem formulation. The objective function minimizes the difference between antiresonance eigenfrequencies and a target frequency while maximizing the difference between resonance eigenfrequencies and the target. A numerical case study is presented, in which an array of topology-optimized resonators placed over a surface is shown to effectively prevent the propagation of Rayleigh waves over a wide range of frequencies around the target design frequency.

**10:50**

**3aSA10. Sonic black hole perforation study.** Kayla Petrover (NSWC, 9500 MacArthur Blvd, Apt. 8B, Bethesda, MD 20817, kayla.petrover@gmail.com)

A powerful and under-explored method to target unwanted reflection and transmission of sound is the sonic black hole. It is the application of the acoustic black hole effect to acoustic waves. The interaction between the propagating acoustic wave and the power-law tapering geometry of the sonic black hole effectively reduces the wave speed, so that it never reaches the termination and, as a result, cannot be reflected or transmitted. One method to improve the overall attenuation created by the ideal phenomenon of the acoustic black hole effect is by adding perforations to the sonic black hole. This study explores the effect that varying the perforation size will have on the sonic black hole's transmission loss and reflection coefficient across a frequency range of 0 to 5 kHz. Approved for Public Release Distribution Statement A.

**11:05**

**3aSA11. Broadband acoustic lens design using gradient-based optimization and adjusting radii and positions of scatterers.** Feruza Amirkulova (Mech. Engieering, San Jose State Univ., 1 Washington Sq, San Jose, CA 95192, feruza.amirkulova@sjsu.edu), Samer Gerges (Mech. Engieering, San Jose State Univ., San Jose, CA), and Vaishnavi Dabhade (Mech. Eng., San Jose State Univ., San Jose, CA)

In this talk, we present the semi-analytical gradient-based optimization (GBO) technique for an efficient design of the broadband acoustic lens. This idea differs from earlier inverse designs that use topology optimization, shape optimization, and generic algorithms. We derived a formula for the gradients of the absolute pressure at the focal point with respect to positions and radii of a set of cylindrical scatterers. The derived analytic form of gradients of absolute pressure complements the modeling when combined with optimization and parallel computing. The GBO algorithm maximizes the sound amplification at focal point by evaluating pressure derivative with respect to the cylinder positions and radii, and then perturbatively optimizing the position and radius of each cylinder in the lens while considering acoustic multiple scattering between the cylinders. Computations are performed using the MultiStart Global optimization solver with fmincon while supplying the gradient of pressure at the focus and the gradient of nonlinear geometrical constraints. The GBO of the broadband acoustic lens is illustrated performing several performance measures for the dependency on the wavenumber and the incidence angle. The method is presented giving numerical examples for non-uniform configurations of the cylindrical structures of various radii and material submerged in water.

**11:20**

**3aSA12. Multi-electrode piezoelectric plates as non-reciprocal metamaterials.** David Schipf (Eng. and Phys., Whitworth Univ., Washington, DC), Peter Le (Naval Res. Lab, Washington, DC), Matthew D. Guild (Naval Res. Lab, 4555 Overlook Ave., SW, Acoust. Div., Code 7160, Washington, DC 20375, matthew.guild@nrl.navy.mil), Gregory Yesner, and Caleb F. Sieck (Naval Res. Lab, Washington, DC)

Non-reciprocal transmission of elastic waves through a solid is an exciting phenomenon that could open up new applications for elastic metamaterials. Non-reciprocal transmission of elastic waves can be achieved with spatio-temporally modulated boundary conditions. Some previous studies of non-reciprocal metamaterials have used uniformly modulated piezoelectric patches on metal host beams. Those beams have the advantage of electrically tunable boundary conditions due to the electro-mechanical coupling in piezoelectric patches, but are multi-material composites that are not easy to scale down. Instead, we show multi-electrode (patterned) piezoelectric plates, which are practical for a large range of geometric scales and frequencies, can exhibit non-reciprocal transmission with spatio-temporally modulated shunted circuits connected to the electrodes. In this work a study of sub-wavelength, ultrasonic band gaps that arise from shunted circuits on piezoelectric plates will be presented. The achievement of spatio-temporal modulation by using time-modulated capacitor-inductor circuits that vary in phase with their neighbors will be discussed. The results of frequency domain simulations of non-reciprocal transmission will be presented, and future directions for this research will be discussed. [Work funded by the Office of Naval Research.]

**11:35**

**3aSA13. Inverse design of direction independent metamaterials to control longitudinal and transverse waves.** Pravinkumar R. Ghodake (Dept. of Mech. Eng., Indian Inst. of Technol. Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

Direction-independent elastic metamaterials that can control longitudinal and transverse elastic waves independently for nonlinear ultrasonic testing are designed using an inverse design approach by defining the time-dependent inverse problem. Forward finite element wave propagation models are solved inversely as multiple multi-objective design optimization

problems using gradient-free algorithms. Suppressing the 2nd harmonics ($2f$) generated due to instrumentation during nonlinear ultrasonic testing is obtained by keeping the first objective as reducing only second harmonics without concentrating on the amplitude of the 1st harmonics, which is a necessary condition for ultrasonic testing, whereas almost all the inversely designed metamaterials take care of this condition due to intelligent selection of material combinations. The second objective is defined as reducing error due to the difference between a pulse with 1st ($f$) and 2nd ($2f$) harmonics and a delayed pulse with 1st harmonics ($f$), as well as maximizing the power of the output pulse by defining min-max multi-objective optimization problems.

WEDNESDAY MORNING, 7 DECEMBER 2022 SUMMIT E, 8:00 A.M. TO 12:00 NOON

# Session 3aSC

## Speech Communication: Speech Perception (Poster Session)

Shiloh Drake, Chair
*Department of Linguistics, University of Oregon, 1290 University of Oregon, Eugene, OR 97403*

All posters will be on display from 8:00 a.m. to 12:00 noon. Authors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even numbers papers will be at their posters from 10:00 a.m. to 12:00 noon.

### *Contributed Papers*

**3aSC1. A cross-linguistic study of audio-aerotactile perceptual integration using voicing continua.** Haruka Saito (Université du Québec à Montréal, 522 Ave., des Pins, Rm. #5, Montreal, QC H2W 1S6, Canada, saito.haruka@courrier.uqam.ca), Lucie Ménard (Université du Québec à Montréal, Montréal, QC, Canada), Mark Tiede (Haskins Labs., New Haven, CT), and D. H. Whalen (CUNY Graduate Ctr., New York, NY)

Previous studies on multimodal integration in speech perception have found that not only auditory and visual cues, but also tactical sensation—such as an air-puff on skin that simulates aspiration—can be integrated in the perception of speech sounds (Gick & Derrick, 2009). However, most previous investigations have been conducted with English listeners, and it remains uncertain whether such multisensory integration can be shaped by linguistic experience. The current study investigates audio-aerotactile integration in phoneme perception for three groups: English, French monolingual and English-French bilingual listeners. Six step VOT continua of labial (/ba/—/pa/) and alveolar (/da/—/ta/) stops constructed from both English and French endpoint models were presented to listeners who performed a forced-choice identification task. Air-puffs synchronized to syllable onset and applied to the hand at random increased the number of 'voiceless' responses for the /da/—/ta/ continuum by both English and French listeners, which suggests that audio-aerotactile integration can occur even though some of the listeners did not have aspiration/non-aspiration contrasts in their native language. Furthermore, bilingual speakers showed larger air-puff effects for English stimuli compared to English monolinguals, which suggests a complex relationship between linguistic experience and multisensory integration in perception.

**3aSC2. Auditory free classification of gender diverse speakers.** Brandon Merritt (Rehabilitation Sci., The Univ. of Texas at El Paso, 734 S. Mesa Hills Dr., #72, El Paso, TX 79912, bmmerritt@utep.edu) and Tessa Bent (Speech, Lang., and Hearing Sci., Indiana Univ., Bloomington, IN)

A central issue in speech perception is how listeners resolve variability. Gender diversity presents an opportunity to examine how listeners learn and represent one dimension of sociophonetic variability arising from an evolving social category, speaker gender. To assess listeners' perceptual organization of speaker gender, recordings were obtained from 30 cisgender (15 men, 15 women) and 30 transgender (1 agender, 15 non-binary, 7 transgender men, 7 transgender women) speakers. In an auditory free classification paradigm, listeners categorized speakers reading the same or unique sentences by perceived general similarity and gender identity. Multidimensional scaling revealed gradient organization of speaker gender with masculinity/femininity and gender prototypicality as the two most salient dimensions. Cluster analyses showed fewer perceptual speaker groups and attention to fewer indexical speech features when speakers produced unique sentences. Instructions to explicitly categorize speakers by perceived gender identity substantially simplified the hierarchical structure of speakers and shifted listeners' attention towards masculinity/femininity over gender prototypicality. Results suggest that listeners engage in fine-grained, multivalent analysis of speaker gender that cannot be adequately captured by a "male" vs "female" dichotomy. Assumptions of a gender binary in speech communication research may require a critical re-examination to accommodate multidimensional and gradient representation of speaker gender.

**3aSC3. Low tone bias during perception of period doubling.** Yaqian Huang (Dept. of Linguist.,Univ. of California San Diego, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, yah101@ucsd.edu)

Period-doubled voice consists of two alternating periods with multiple frequencies and is often perceived as rough with an indeterminate pitch. Past pitch-matching studies in period-doubled voice found that the perceived pitch was lower as the degree of amplitude and frequency modulation between the two alternating periods increased. The perceptual outcome also differed across f0s and modulation types: a lower f0 prompted earlier identification of a lower pitch, and the matched pitch dropped more quickly in frequency- than amplitude-modulated tokens (Sun & Xu, 2002; Bergan & Titze, 2001). However, it is unclear how listeners perceive period doubling when identifying linguistic tones. In an artificial language learning paradigm, this study used resynthesized stimuli with alternating amplitudes and/

or frequencies of varying degrees, based on a production study of period-doubled voice (Huang, 2022). Listeners were native speakers of English and Mandarin. We confirm the positive relationship between the modulation degree and the proportion of low tones heard, and find that frequency modulation biased listeners to choose more low-tone options than amplitude modulation. However, a higher f0 (300 Hz) leads to a low-tone percept in more amplitude-modulated tokens than a lower f0 (200 Hz). Both English and Mandarin listeners behaved similarly, suggesting that pitch perception during period doubling is not language-specific. Furthermore, period doubling is predicted to signal low tones in languages, even when the f0 is high.

**3aSC4. Learning from a single cue: Is phonetic learning dimension-based?** Kaori Idemaru (Linguist, Univ. of Oregon, Eugene, OR), Adam A. Bramlett (Second Lang. Acquisition, Carnegie Mellon Univ., 1314 Fox Hunt Dr., Cheswick, PA 15024, abramlet@andrew.cmu.edu), and Vsevolod Kapatsinski (Linguist, Univ. of Oregon, Eugene, OR)

Phonetic cue-weighting, the process of altering the weights of certain dimensions (e.g., F0) in the speech signal, is a fundamental process in speech perception. Cue-reweighting is the process of adaptation required for understanding new accents and learning second language speech contrasts; however, little is understood about the underlying mechanisms. Harmon *et al*. (2019) examined three candidate mechanisms (distributional, supervised, and reinforcement learning) showing evidence for reinforcement learning. The current study investigates Harmon *et al*.'s (2019) assumed phonetic dimensions by asking how a single cue in a phonetic dimension (e.g., a single voice onset time (VOT) value) of a phonological contrast ([b]/[p]) generalizes to other values of the phonetic dimension. Said simpler, is phonetic learning dimension-based? Native English listeners (N = 270) participated in an online perceptual training experiment in which participants were asked to identify word contrasts like pear and bear. Results suggest that learning to downweigh a cue (e.g., VOT = 5 ms) for [b]/[p] generalizes across new VOT values (e.g., VOT = 15 ms). However, the generalization did not extend to the most distant value (e.g., VOT = 35 ms). That is, cue-reweighting can affect a single phonetic category but does not extend to the entire phonetic dimension across category boundaries.

**3aSC5. Examining the effectiveness of multi-talker training on morphophonological learning.** Shiloh Drake (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, sdrake@uoregon.edu), Isabel Preligera, and Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, Eugene, OR)

This study explores how auditory stimuli from multiple talkers affects learning of morphophonological alternations in an artificial grammar. Hearing words from multiple voices creates more robust lexical representations (e.g., Davis & Gerken, 2013, 2014; Rost & McMurray, 2009, 2010) and aids generalization to new grammatical structures (Gonzales *et al*., 2018) and accents (Baese-Berk *et al*., 2013; Bradlow & Bent, 2008). In ongoing work, participants hear either four different voices or four repetitions from the same voice when learning singular and plural forms in two artificial grammars. As found previously, we expect that hearing from multiple talkers will result in more accurate responses when participants are tested when compared to the participants who heard the same number of repetitions from the same voice. Furthermore, we expect a more pronounced difference in accuracy between the two talker conditions in the Arabic-like grammar than in the English-like grammar.

**3aSC6. Perception of Bengali stops: The role of Markedness.** Sreeparna Sarkar (Linguist and Cognit. Sci., Univ. of Delaware, 260 South Main St., Newark, DE 19711, sree@udel.edu)

Marked segments, by definition, bear additional properties with respect to unmarked ones, in principle requiring more articulatory effort. At the same time, however, these additional properties may make marked sounds more audible and identifiable. This study investigates how voicing and aspiration, as mark properties, affect the perception of Bengali stops, exhibiting the common Indo-Aryan four-way contrast: voiceless unaspirated, voiceless aspirated, voiced unaspirated, voiced aspirated. Specifically, I test whether the properties of voicing and aspiration result in improved perceptibility of stops separately and whether their combination shows an increased cumulative effect.50 native Bengali speakers each listened to 270 CV stimuli, the first syllable extracted from real 3-syllable words. The C was one of the four stop types; the V was /a/. The participants identified the syllable they heard by selecting one of six options (the four stops and two fillers) shown in Bengali script. The results reveal significantly higher perceptual accuracy in the stops marked by voicing vs. voiceless categories (83.78% vs 80.48%, respectively) and higher perceptual accuracy of aspirated versus unaspirated stops ( 84.04% vs 78.49%, respectively). The two "marks" combined in voiced aspirated stops do not increase accuracy more than each alone (83.89%) but overall differ most from plain unaspirated stops (75.8%).

**3aSC7. Intelligibility of synthetic words generated by transformation of a sequence of discrete acoustic events into modulation of the vocal tract shape.** Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu) and Kate Bunton (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Within the paradigm of a recent model of speech production [Story and Bunton, *JASA* **146**(4), 2522–2528], an utterance is specified as a sequence of relative acoustic events along a time axis. These events consist of directional changes of the vocal tract resonance frequencies called resonance deflection patterns (RDPs) that, when associated with a temporal event function, are transformed via acoustic sensitivity functions, into time-varying modulations of the vocal tract shape. RDPs specifying the targeted directional shift of the first three resonances for bilabial, alveolar, and velar consonants would be coded as [−1 −1 −1], [−1 1 1], and [−1 1 −1], respectively. In this study, these RDPs were combined with four vowels ("ih, ae, eh, uh") to construct a set of 40 American English words (CVCs). A word intelligibility test was conducted in which listeners heard a synthesized target word and were asked to indicate what they heard by choosing a word from a matrix that included the target and seven near-neighbor words. Results indicate listener word recognition was aligned with the RDP settings, suggesting that they are an effective discrete representation of phonetic segments that can be transformed into speech by modulation of the vocal tract shape.

**3aSC8. The perception-production link in two types of cross-language mismatches.** Corentin Mazet (French and Italian, Indiana Univ., 355 North Eagleson Ave., Bloomington, IN 47406, cmazet@iu.edu)

Previous data on the perception-production link have allowed for a variety of analyses, from the traditional straight-forward correlation between the two competencies (Flege 1995) to more nuanced interpretations of the data (Shultz *et al*. 2012). Where the traditional approach suggests that better perception is linked to better production, recent work challenges this conclusion and reveals a complex relationship. Here, we compare native French speakers' perception and production of English diphthongs (/a͡ʊ/, /e͡ɪ/ and /a͡ɪ/) and interdental fricatives (/θ/ and /ð/). These specific segments were chosen as they are absent from the French phonological system, they show identifiable realizations, and they behave differently from one another. The competence in perception and production of 21 native hexagonal French speakers were collected through a repetition task and an AXB matching task. Consistent with Flege (1995), we show that for any given segment, greater perception accuracy predicts greater production accuracy. However, at the speaker level, the relationship between the two competences is a complex one, as the link between perception and production is not apparent in our data. These results would confirm the traditional approach at the segment level, but individual variation might prevents us from seeing it at the speaker level.

**3aSC9. Acoustic cues to the perception of Danish stød.** Jailyn M. Pena (Linguist, New York Univ., 3 E. 124th St., Unit 2, New York City, NY 10035, jmp987@nyu.edu)

This study examines the effect of F0 and creak (raspiness) on Danish stød perception. Stød is typically characterized by high fundamental frequency (F0) at the beginning of the syllable and low F0, often with creaky phonation, at the end of the syllable (Fischer-Jørgensen, 1989). However, it is not clear how these acoustic cues, high F0 at rhyme onset, low F0 at rhyme offset, and creak, contribute to the percept of stød for listeners. A nonce word ABX task and a perceptual rhyming task were conducted. Results from the ABX task indicate that nonce words with high-mid or high-low F0 contours and added raspiness (Corretge, 2012) or a sustained high F0 with or without raspiness were more likely to be considered the "same" as naturally produced nonce words with stød. In the rhyming task, nonce words with high F0 and raspiness were significantly more likely to be considered as rhyming with a naturally produced real word with stød for all tonal contours tested. Overall, the results indicate that both high F0 onset{Citation} and creaky phonation contribute to the stød percept. Furthermore, both cues were almost always needed to increase stød responses in both tasks, suggesting little evidence of cue trading.

**3aSC10. Extraction of indexical and linguistic information as a function of duration in the older population.** David R. Smith (Psych., Univ. of Hull, Cottingham Rd., Hull HU6 7RX, United Kingdom, d.r.smith@hull.ac.uk) and Chiara Guerrini (Psych., Univ. of Hull, Hull, United Kingdom)

When someone speaks information is present in a number of forms. The most obvious information is what the person has said (linguistic message). However, indexical information relating to the physical characteristics of the speaker is also embedded in the speech sound wave and influence judgements about the speaker such as whether they are a man or woman. Previous research studied how vowel recognition and speaker-sex discrimination information accumulates as a function of speech duration in the normal-hearing young population [D. R. R. Smith, *Acta Psychol.* **148**, 81–90 (2014)]. We extended this research by investigating the build-up of linguistic and indexical information in the older population ($M_{\mathrm{age}}$ 72-years). Psychometric functions were collected plotting percent correct speaker sex discrimination and vowel recognition, as a function of vowel duration (5–40 ms), for older listeners compared to younger listeners ($M_{\mathrm{age}}$ 22-years). Older listeners' performance on both tasks was markedly impaired compared to younger listeners' performance. It is argued that deficits in the accumulation of indexical information at short durations might partially underlie the problems the older population experience in understanding speech and following conversations, especially where who is speaking is constantly changing such as in a noisy restaurant where several people are conducting a conversation.

**3aSC11. Perceptual training enhances Seoul Korean listeners' use of vowel quality and pitch cues to English lexical stress.** Annie C. Tremblay (Lang. and Linguist, Univ. of Texas at El Paso, 500 West University Ave., Liberal Arts 137, El Paso, TX 79912, actremblay@utep.edu), Hyoju Kim, Keira Dobbs (Linguist, Univ. of Kansas, Lawrence, KS), Sahyang Kim (English Education, Hongik Univ., Seoul, Korea (the Democratic People's Republic of)), and Taehong Cho (English Lang. and Lit., Hanyang Univ., Seoul, Korea (the Democratic People's Republic of))

This study investigates whether high-variability (multi-talker) perceptual training is superior to low-variability (single-talker) perceptual training for enhancing Seoul-Korean listeners' use of vowel quality and pitch cues to English lexical stress. Vowel quality and pitch are the two most important cues to English lexical stress in accented words (Tremblay *et al.*, 2021). Seoul Korean does not have lexical stress, lexical pitch accents, or lexical tones. Seoul-Korean listeners completed a pre-/post-test sequence-recall experiment where they heard and recalled four English words uttered by different talkers. Experimental conditions: The words differed in lexical stress, which was cued by: (i) vowel quality, pitch, and duration; (ii) vowel quality and duration; (iii) pitch and duration; or (iv) duration. Control condition: The words differed in their initial consonant (Seoul-Korean contrast). Listeners completed one of two training types ($8 \times 30$-mins). The words in the training followed a cue distribution that mimicked naturalistic spoken English (Im *et al.*, 2018). Results: Both training types improved listeners' recall for (i)–(iii), suggesting enhanced use of vowel quality and pitch. Learning gains were greater with high-variability training, but training type did not interact with cue. Thus, high variability provides a global rather than selective enhancement of listeners' use of cues to lexical stress.

**3aSC12. Digging into sentence intelligibility: Interactions with talker.** Benjamin V. Tucker (Linguist, Univ. of AB, 4-32 Assiniboia Hall, University of AB, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca), Matthew C. Kelley, Marina Oganyan, and Richard A. Wright (Linguist, Univ. of Washington, Seattle, WA)

An implicit feature found in studies of intelligibility and perception is the assumption that noise masks all spoken-sentence stimuli equivalently regardless of the stimulus-talker. For example, when using controlled and normed sentences, such as the IEEE Harvard set, the mean intelligibility across talkers should be affected by masker-noise equivalently. In other words, any particular sentence spoken by one talker should be equivalently intelligible to when it is spoken by another talker using the same style and rate. However, previous studies have shown that there is significant intelligibility variation between talkers in controlled stimuli. Therefore, we predict talker-masker interactions in the IEEE stimuli. We investigated the UAW speech intelligibility dataset which contains typed responses from over 900 native listeners of English to over 604 sentences from the UWNU IEEE sentence corpus presented in three signal-to-noise ratios ($+2$, 0, $-2$ dB). We calculated the Levenshtein Distance for the listeners' transcriptions of the sentences. We find that different talkers have different intrinsic intelligibility even when reading the same set of sentences. Moreover, we find an interaction with SNR level and individual talker intelligibility. The present results suggest that individual talker effects in speech stimuli may influence the outcomes of perception and intelligibility studies.

**3aSC13. Digging into sentence intelligibility: Interactions with noise.** Richard A. Wright (Dept. of Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195-2425, rawright@uw.edu), Matthew C. Kelley, Marina Oganyan (Linguist, Univ. of Washington, Seattle, WA), and Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada)

Implicit in the design of many intelligibility and perception studies is the assumption that noise masks all sentence stimuli equivalently, at least when they are controlled for length and structure. For example, controlled and normed sentences, such as the IEEE Harvard set, are typically thought to be equally affected by masker-noise. However, previous research has established that spoken stimuli of different structures, lengths, or complexities, are differentially masked (e.g. nonsense versus real words, words with different usage frequencies), therefore it is possible that masking affects even controlled sentences differentially . Using the UAW speech intelligibility dataset we analyzed Levenshtein Distance values based on transcriptions from over 900 native listeners of English to over 604 sentences from the UWNU IEEE sentence corpus. Each sentence was presented in noise at three SNRs ($+2$, 0, $-2$ dB). We find that the sentences are not equally intelligible. Moreover, there is an interaction with SNR level and sentence. In other words, the intelligibility of the sentences is impacted differentially by masker-noise. The results of this study suggest that, as researchers using speech stimuli, we should recognize that there are many sentence level factors that may introduce variance or otherwise affect the outcomes of our studies.

**3aSC14. "Please say what this word is," in the US and now again in the UK: Dialectal differences in a replication of Ladefoged and Broadbent (1957).** Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu) and Eleanor Chodroff (Lang. and Linguistic Sci., Univ. of York, York, United Kingdom)

One of the most famous speech perception studies is Ladefoged and Broadbent (1957), which demonstrated that acoustic properties of earlier sounds (in a context sentence) alter perception of subsequent sounds (in a target word). In their study, a context sentence with a lower first formant (F1) frequency promoted perception of a higher F1 in the target word, and vice versa. The decades since this seminal finding have seen myriad indirect replications, but none matched the incredibly large effect size of Ladefoged and Broadbent's original study. Here, replication of Ladefoged and Broadbent (1957) was pursued using digitized versions of their original stimuli. Listeners were normal-hearing students in the UK (matching Ladefoged and Broadbent's original sample) or the US (where most of the follow-up research has been conducted). Like the original study, perception of the

target word shifted as a function of acoustic properties in the context sentences as in the example above. Perceptual shifts were larger for UK listeners than US listeners, but neither sample replicated the large effect sizes of the original study. While these results agree with Ladefoged and Broadbent's conclusions, they invite consideration of how listener dialect and language background may relate to the magnitude of these effects.

**3aSC15. Speaking rate normalization with and without segregation of simultaneous context sentences.** Dawson Stephens (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY), Caleb J. King (Psychol. and Brain Sci., Univ. of Louisville, 2301 S 3rd St., Louisville, KY 40292, cjking03@louisville.edu), Anya E. Shorey, and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Speech perception is shaped by acoustic properties of earlier sounds influencing recognition of later speech sounds. For example, when a context sentence is spoken at a faster rate, the following target word (varying from "deer" to "tier") is perceived as "tier" (longer VOT) more often; when a slower context sentence is spoken, the following target word is perceived as "deer" (shorter VOT) more often. This is known as a temporal contrast effect (TCE, a.k.a. speaking rate normalization). Recently, Bosker, Sjerps, and Reinisch (2020 Scientific Reports) concluded that selective attention (to one of two simultaneous talkers) had no impact on TCEs. However, their paradigm was not an ideal test of this question; the voices heard were different talkers presented to each ear, and thus relatively easy to perceptually separate. Here, on each trial, the same talker spoke one sentence to both ears (no segregation), two sentences simultaneously to both ears (poor segregation), or a different sentence to each ear (easier segregation). Fast or slow context sentences preceded target words varying from "deer" to "tier." TCE magnitudes were similar across all presentation modes. Results are consistent with the claims set forth by Bosker *et al*. – TCEs are automatic and low-level, not modulated by selective attention.

**3aSC16. Japanese pitch-accent perception of noise-vocoded sine-wave speech.** Yasuaki Shinohara (Faculty of Commerce, Waseda Univ., 1-6-1 Nishiwaseda, Shinjuku-ku, Tokyo 169-8050, Japan, y.shinohara@waseda.jp)

A previous study has demonstrated that speech intelligibility is improved for a tone language when sine-wave speech is noise-vocoded, because noise-vocoding eliminates the quasi-periodicity of sine-wave speech. This study examined whether identification accuracy of Japanese pitch-accent words increases after sine-wave speech is noise-vocoded. The results showed that the Japanese listeners' identification accuracy significantly increased, but their discrimination accuracy did not show a significant difference between the sine-wave speech and noise-vocoded sine-wave speech conditions. These results suggest that Japanese listeners can auditorily discriminate minimal-pair words using any acoustic cues in both conditions, but quasi-periodicity is eliminated by noise-vocoding so that the Japanese listeners' identification accuracy increases in the noise-vocoded sine-wave speech condition. The same results were not observed when another way of noise-vocoding was used in a previous study, suggesting that the quasi-periodicity of sine-wave speech needs to be adequately eliminated by a noise-vocoder to show a significant difference in identification.

**3aSC17. Is emotion expressed by people with and without cognitive impairment perceived differently?** Chorong Oh (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W235, Athens, OH 45701, ohc@ohio.edu), Richard J. Morris (Commun. Sci. and Disord., Florida State Univ., Tallahassee, FL), and Xianhui Wang (Otolaryngol., Univ. of California Irvine, Athens, OH)

This study was designed to explore whether neurotypical listeners perceive emotion expressed by people with cognitive impairment differently, compared to that expressed by neurotypical older adults. For this study, speech samples describing the Cookie Theft picture, of 11 people with dementia of the Alzheimer's type, 9 individuals with mild cognitive impairment, 5 people with vascular dementia, and 10 healthy age- and education-matched older adults were obtained from the Dementia TalkBank. Then, 28 listeners evaluated emotion expressed in each utterance of the speech samples. The listeners differently perceived emotion expressed by the four groups. The outcomes of this study implies possible communication breakdowns among people with cognitive impairments.

**3aSC18. The effect of the number of items on the reliability of speech intelligibility results for WIPI test.** Mary M. Flaherty, Charlie Nudelman (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), and Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, pb81@illinois.edu)

The Word Intelligibility by Picture Identification (WIPI) test is a widely used test to assess speech recognition in the pediatric population. The test consists of four 25-item monosyllabic word lists that are within the vocabulary of preschool children. The child responds to each item by selecting one of six pictures on a page, one being the test item. The purpose of this study was to understand if the final score of a test can be obtained accurately with a fewer number of items in a word list. Ten monolingual children (8–12 yrs) with normal hearing were tested using a revised version of the WIPI (2nd ed.), presented in a soundbooth over headphones. Target words were presented in a classroom-noise masker at a fixed signal-to-noise ratio. The results suggest that a minimum of seven items is needed to obtain results that are not statistically different from the one complete test (25 items). This implies that the WIPI can be administered in a shorter time period, which could be particularly beneficial when testing multiple acoustic conditions on experimental tasks.

**3aSC19. Adapting to talkers in speech perception: Balancing generalization and individuation.** Samantha Chiu (Psychol. Brain Sci., Univ. of Iowa, 340 Iowa Ave., Iowa City, IA 52242, samantha-chiu@uiowa.edu), Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), Joseph C. Toscano (Villanova Univ., Villanova, PA), and Cheyenne M. Toscano (Psychol. and Brain Sci., Villanova Univ., Villanova, PA)

Listeners can account for systematic variability between talkers, which is learned over exposure to multiple talkers. Previous research suggests that listeners can both generalize prior knowledge of phoneme categories from a familiar to a novel talker (Eisner & McQueen, 2005; Kraljic & Samuel, 2006, 2007) and individuate talkers, preventing generalization (Luthra, Mechtenberg, & Myers, 2021; Tamminga, Wilder, Lai, & Wade, 2020). It is unclear how listeners balance these competing demands. Participants ($n = 160$) learned two novel talkers (one male and one female voice) with a unique voice onset time (VOT) boundary across two days. On each day, participants were passively exposed to a bimodal distribution of VOTs from one talker, then tested on a second talker (uniform distribution). Day 1 assessed how listeners generalize to a novel talker while Day 2 assessed how the talker that was learned on Day 1 is individuated from the new talker. We found evidence for generalization after Day 1 but little evidence of learning after learning both talkers on Day 2. Two follow-up experiments using interleaved designs and supervised learning also showed little evidence for individuation. This suggests that listeners are likely accounting for variability by shifting their VOT boundary to match current input.

**3aSC20. Lombard effect and speech intelligibility by frequency content.** Pasquale Bottalico (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 South Sixth St., Champaign, IL 61820, pb81@illinois.edu) and Silvia Murgia (Speech and Hearing Sci., Univ. of Illinois - Urbana Champaign, Champaign, IL)

The Lombard Effect is an increase in vocal effort in response to rising noise levels and disturbance in the communication environment. The objective of this study is to evaluate the Lombard Effect during four intensity levels from 45dB(A) to 75dB(A) of low-frequency, medium-frequency, and high-frequency energy noise to measure the effect of disturbance and vocal discomfort on the speaker's intent to communicate, as well as the effect on speech intelligibility. Twelve conditions were randomly presented and recorded for each participant with the three types of noise and levels. At each condition, 20 participants were asked to read a passage. Immediately following each reading, participants were asked to rate the amount of noise disturbance and vocal comfort they had experienced. After, the speech

intelligibility was evaluated by asking the participants to repeat the sentences of the QuickSin test emitted by a Head and Torso Simulator. The medium-frequency energy noise showed the highest Lombard Effect and the stronger decrease in intelligibility. In the conditions with noise with mid-frequency energetic content, the decrease in intelligibility was drastic with the increase in noise level. Low-frequency noises minimally impact speech intelligibility. High-frequency noises show little change in intelligibility with the increase in noise level.

**3aSC21. "Sounds like home": The effect of listener on matching guise in regional accent on syntactic acceptability.** Nour Kayali (Linguist, Univ. of Kentucky, 337 Linden Walk, Apt. 5, Lexington, KY 40508, nourkayali@uky.edu)

This study unites sociophonetic speech perception and syntax research by matching or mismatching social expectations of participants during a syntactic grammaticality judgement task. I hypothesize that regional syntactic structures are more acceptable when heard in their corresponding local, regional accent compared to a nonlocal accent. The experiment focuses on Southern American English as spoken in western Kentucky as the local accent and personal datives and double modals as matching, regional grammatical structures, along with a set of nonlocal filler sentences. 81 University of Kentucky participants completed a between-subjects matched guise survey. Regional structures are consistently rated more acceptable in the local accent than the nonlocal. Self-identified Kentuckian Participants rated local structures higher than non-Kentuckian participants, regardless of which accent they heard. This is prominent in ratings of the nonlocal audio; Kentuckian participants rated both local structures in the nonlocal accent approximately fifteen points higher than non-Kentuckians. In contrast, non-Kentuckian participants mark filler sentences in the local audio lower than the nonlocal, a variation absent from Kentuckian participants' ratings. These results suggest that grammaticality judgements result from an interplay of sociocultural expectations with the phonological and syntactic structure of an utterance. Judgement of structural grammaticality is not independent of social expectation.

**3aSC22. Clear speech promotes speaking rate normalization.** Lilah Kahloon (Univ. of Louisville, Louisville, KY), Anya E. Shorey (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., University of Louisville, Louisville, KY 40292, anya.shorey@louisville.edu), Caleb J. King (Univ. of Louisville, Louisville, KY), and Christian E. Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

When speaking in noisy conditions or to a hearing-impaired listener, talkers often use clear speech which is slower, louder, and hyperarticulated relative to conversational speech. In other research, changes in speaking rate are known to affect speech perception (called temporal contrast effects, or speaking rate normalization). For example, when a sentence is spoken quickly, the voice onset time (VOT) in the next word sounds longer by comparison (e.g., more like the /t/ of "tier"); a sentence spoken slowly makes the VOT in the next word sound shorter (e.g., like the /d/ of "deer"). Typically, one sentence is manipulated to produce fast and slow versions. We tested whether naturally produced clear and conversational speaking styles would also produce these temporal contrast effects. On each trial, listeners heard either a clear (slow) sentence or a conversational (fast) sentence followed by a target word to be categorized as "deer" or "tier." Temporal contrast effects were observed both for conversational relative to clear speech and for conversational relative to a slowed version of the conversational speech. Changing speaking styles aids speech intelligibility but may produce other consequences such as contrast effects that alter sound/word recognition.

**3aSC23. Linguistic fine-tuning in the spectra of the frequency-following responses to Mandarin tones.** Jianjing Kuang (Linguist, Univ. of Pennsylvania, Ste. 300C, 3401 Walnut St., Philadelphia, PA 19104, kuangj@upenn.edu), Tian C. Zhao (Dept. of Speech and Hearing Sciences/Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA), and Fernando Llanos (Linguist, The Univ. of Texas at Austin, Austin, TX)

Our understanding of the meaningful linguistic information encoded in neural signals such as FFR is still rather limited. Previous FFR tone studies mostly have focused on F0, and it is unclear how the other rich phonetic cues are encoded in the spectra. In this study, we analyzed the spectra of the frequency following responses to four Mandarin lexical tones from 13 native Mandarin listeners and 22 native English listeners. The stimuli consisted of four 250 ms synthetic syllable /yi/ with Mandarin four lexical tones minimally distinguished by F0 contours. In general, the harmonic structure of the FFR signals for native Mandarin listeners is much better defined than for non-native listeners, and the advantage is especially clear for higher-frequency harmonics. This result suggests that native listeners can hear the pitch contours more clearly and precisely not only because of a clearer f0 representation but also because of better-defined overtones. Moreover, for native listeners, harmonic representation varies by tonal targets. For a higher-pitched tone (Tone 1), H2 is the most prominent harmonic; for a lower-pitched tone (e.g., Tone 3 and Tone 2), H1 is the most prominent harmonic. These results suggest that pitch tracking strategies may vary by tonal targets.

**3aSC24. The effect of listener dialect experience on generalization of a novel vowel shift.** Marie Bissell (Ohio State Univ., Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, bissell.43@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

When adapting to novel dialects, listeners rely on the systematicity of novel variants across vowel categories, even when only part of the novel vowel system is presented. I examined how a listener's dialect experience with components of a novel front lax vowel shift impacts perceptual adaptation and generalization. Three listener groups were exposed to different portions of a novel English dialect with front lax vowel backing. Listener groups varied in their dialect experience: Westerners were experienced with /ɪ ɛ æ/ backing in the California Vowel Shift, Southerners were experienced with parallel movements of /ɪ ɛ/ in the Southern Vowel Shift, and New Englanders were minimally experienced with parallel front lax vowel shifts. Listeners were exposed to either no front lax vowels, /ɪ/, /ɪ æ/, or /ɪ ɛ æ/. Then, listeners completed a lexical decision task for items with /ɪ ɛ æ/, requiring variable amounts of adaptation and generalization by condition. Westerners and New Englanders endorsed more words in the /ɪ æ/ exposure condition than in the /ɪ/ exposure condition, but Southerners endorsed fewer. Southerners' unfamiliarity with /æ/ shifting may have inhibited adaptation and generalization. A listener's dialect experience can affect perceptual adaptation to and generalization of a novel dialect.

**3aSC25. A potential approach for reducing hearing difficulty by improving perception of intensity changes in speech.** Aisling M. Maguire (Psychol. and Brain Sci., Villanova Univ., 800 E Lancaster Ave., Villanova, PA 19085, amaguir8@villanova.edu) and Joseph C. Toscano (Psychol. and Brain Sci., Villanova Univ., Villanova, PA)

Many listeners report hearing difficulty, especially for speech in noisy environments, despite having normal audiometric thresholds. Recent work suggests that such cases may be caused by disruptions in coding of suprathreshold sounds at early stages of auditory processing. Specifically, differences in the function of auditory nerve fibers with lower spontaneous firing

rates could disrupt coding of intensity changes for sounds in the typical range used for speech communication. In turn, this could affect perception of acoustic cues in speech that are dependent on intensity changes over time, such as voice onset time (VOT). The current study investigated a speech sound manipulation designed to counteract the effects of such disruptions by making intensity differences in the signal more pronounced. Listeners heard speech sounds varying along two acoustic dimensions that provide information about word-initial voicing, VOT and f0 onset, and categorized sounds as either voiced or voiceless. Results showed that the intensity manipulation reduced listeners' use of f0 for voicing categorization and also affected their use of VOT. This suggests that intensity manipulations can make acoustic cues like VOT more salient, providing a potential treatment approach for listeners who have difficulty coding intensity differences in speech.

**3aSC26. Capturing vocal fry: Understanding the acoustic and perceptual differences in elicited and naturally produced vocal fry.** Jessica Alexander (Psych., Centenary College of Louisiana, 2911 Centenary Blvd, Shreveport, LA 71104-3335, alexander.jed@gmail.com) and Sarah T. Irons (Psych., Centenary College of Louisiana, Shreveport, LA)

Vocal fry, used frequently by both men and women, is often associated with negative stereotypes of young women. This study examines how different ways of eliciting vocal fry affects acoustic properties as well as listener perceptions of speech. Stimuli in previous studies have been obtained from actors who were presented with obvious examples of fry from media sources and were then instructed to produce speech with vocal fry. This method of eliciting fry may introduce other vocal characteristics that activate stereotypes. We obtained natural stimuli by asking speakers to read passages with no instructions about vocal fry and created a corpus of acoustic and prosodic data about naturalistic fry. We then recorded actors who listened to naturally-produced examples of vocal fry. We compared the acoustic characteristics of low and high fry speech from our naturally-produced utterances, our actors, and stimuli collected by Anderson *et al*. (2014). Elicited fry (from both studies) differed when compared to naturally-produced fry. Additionally, elicited fry in our study differed from that of Anderson et al on jitter and other measures. The three sets of stimuli were used to replicate and extend Anderson and colleagues' study on listener perceptions of men and women using vocal fry.

**3aSC27. Predicting talker intelligibility: Contrasting talkers' racialized identities and listeners' racializations.** Alayo Tripp (Speech Lang. Hearing Sci., Univ. of Minnesota, Twin Cities, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, tripp158@umn.edu), Tatiana Lyons, and Benjamin Munson (Speech Lang. Hearing Sci., Univ. of Minnesota, Twin Cities, Minneapolis, MN)

Recent studies have found that speech intelligibility can be manipulated by changing the race of a person whose face is used as a visual prime presented with a spoken sentence (Babel & Russell, 2015; McGowan, 2015; Kutlu *et al*., 2022). These findings suggest that the differences in *racializing* faces—the process whereby people ascribe a race to an individual an explanation for observed traits and behaviors and to predict future behavior–may mediate speech intelligibility. However, other data suggests that the contrastive effect arises more from perceived differences in vocal behavior (McLaughlin *et al*., 2022). In this study, we address this discrepancy in previous research. We examine variability in the process of racialization from faces and voices by contrasting the predictive power of talkers' self-reported racialized identities on speech intelligibility with that of racializations reported by listeners. We examine this in an experiment assessing the audio-only and audio-visual intelligibility in noise of 28 talkers of various racialized identities by 242 listeners. The listeners both reported the speech they heard, and indicated what they believed the talkers' racialized identities to be. Data collection is complete, and analysis is ongoing. [Funded by NIH grant R21 DC018070]

**3aSC28. Perceived linguistic marginalization and speech intelligibility in a racially diverse cohort of talkers.** Alayo Tripp (Speech Lang. Hearing Sci., Univ. of Minnesota, Twin Cities, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, tripp158@umn.edu) and Benjamin Munson (Speech Lang. Hearing Sci., Univ. of Minnesota, Twin Cities, Minneapolis, MN)

Studies suggest that the listeners' social categorizations of talkers may mediate how intelligible those talkers' speech is perceived to be (Babel & Russel, 2015; McGowan, 2015; Kutlu *et al*., 2021; Kutlu *et al*., 2022; McLaughlin *et al*., 2022; Yi *et al*., 2013). One interpretation of these findings is that talkers' attitudes toward different racialized groups affects intelligibility. Studies have used implicit association tests (Greenwald *et al*., 1998) to test this hypothesis. Some studies have found that negative implicit associations with specific groups (i.e., Asian Americans) predict intelligibility (Yi *et al*.) while others have found no association (McLaughlin *et al*.). In this presentation, we present a new tool to examine listeners' attitudes toward different racialized groups. The *Perceived Linguistic Marginalization Scale* (PLMS, based on the *Perceived Societal Marginalization Scale*, Bulwerk *et al*., 2022) directly examines individuals' beliefs about the significance of language variation in marking groups of people as marginalized. In this poster, we present the results of 242 participants' performance on the PLMS. We examine whether performance on the PLMS predicts audio-only and audio-visual intelligibility of 28 racially diverse talkers in a speech intelligibility experiment. Data collection is complete and analysis is ongoing. [Funded by NIH grant R21 DC018070]

3a WED. AM

**3aSC29. Prosodic features conveying doubt and trust.** Abbey L. Thomas (Brain and Behavioral Sci., The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, abbey.thomas@utdallas.edu)

Research in affective prosody has shown that doubt (specifically, incredulity) is communicated through distinctive features like a slower speech rate, greater variability in fundamental frequency (F0), and a characteristic F0 pattern including an utterance final rise in F0. However, the opposite of doubt, trust (or belief), has not been examined as a prosodically distinct attitude. In response, this study explores the full range of doubt and trust native speakers of American English can express through prosody. In a web-based production experiment, we asked 30 participants to produce 12 utterances in neutral, trusting, and doubting prosody. A multinomial logistic regression analysis revealed that incredulous prosody differed significantly from neutral prosody primarily in its lower initial F0 and longer duration. Trust also differed from neutrality, with a significantly higher utterance initial F0 and less amplitude variability than was observed in neutral utterances. In a perception experiment, 115 participants listened to 36 tokens collected from the production experiment and were asked whether the talker expressed doubt or trust. Participants were generally accurate in their judgement of talker attitude, and acoustic variables of utterance-initial F0 and speech rate significantly predicted participant ratings of the talker's doubt or trust.

## Session 3aSP

# Signal Processing in Acoustics: Signal Processing for Musical Audio Production

Scott H. Hawley, Chair
*Department of Chemistry and Physics, Belmont University, Chemistry & Physics Dept., 1900 Belmont Blvd, Nashville, TN 37212*

**Chair's Introduction—9:00**

### *Invited Paper*

**9:05**

**3aSP1. Artificial intelligence in music production: Controversy and opportunity.** Josh Reiss (Queen Mary Univ. of London, Mile End Rd., London, Greater London E1 4NS, United Kingdom, joshua.reiss@qmul.ac.uk)

Could a robot replace the sound engineer? Recent years have seen the emergence of intelligent systems aimed at algorithmic approaches to autonomous mixing and mastering of audio content. This talk will give an overview of the field, explanations and demonstrations of the technology, and a discussion of the challenges and directions currently being explored. We further discuss the role of artificial intelligence in audio production, and what it means for sound engineering in future.

### *Contributed Papers*

**9:30**

**3aSP2. Analysis-by-synthesis paradigm evolved into a new concept.** Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

This work aims at showing how the well-known analysis-by-synthesis paradigm has recently been evolved into a new concept. However, in contrast to the original idea stating that the created sound should not fail to pass the foolproof synthesis test, the recent development is a consequence of the need to create new data. Deep learning models are greedy algorithms requiring a vast amount of data that, in addition, should be correctly annotated. Annotation is a bottleneck to getting quality-reliable data as the process relies on annotating a person's experience and, in many cases, personality-related issues. So, the new approach is to create synthesized data based on a thorough analytical examination of a musical/speech signal resulting in cues for a deep model of how to populate data to overcome this problem. Typically, a 2D feature space is employed, e.g., mel spectrograms, cepstrograms, chromagrams, etc., or a wave-based representation with the counterpart on the algorithmic side called wavenet. In this paper, examples of 2D musical/speech signal representation are presented, along with deep models applied. Creating new data in the context of applications is also shown. In conclusion, further possible directions of this paradigm development which is now beyond the conceptual phase, are presented.

**9:45**

**3aSP3. Musical audio samples generated from joint text embeddings.** Zach Evans (Harmonai130 E Sunset Way, Issaquah, WA, zach@stability.ai), Scott H. Hawley (Dept. of Chemistry and Phys., Belmont Univ., Nashville, TN), and Katherine Crowson (EluetherAI, Portland, OR)

The field of machine learning has benefited from the appearance of diffusion-based generative models for images and audio. While text-to-image models have become increasingly prevalent, text-to-audio generative models are currently an active area of research. We present work on generating short samples of musical instrument sounds generated by a model which was conditioned on text descriptions and the file structure labels of large sample libraries. Preliminary findings indicate that generation of wide-spectrum sounds such as percussion are not difficult, while the generation of harmonic musical sounds presents challenges for audio diffusion models.

**10:00**

**3aSP4. Audio (vector) algebra: Vector space operations on neural audio embeddings.** Scott H. Hawley (Dept. of Chemistry and Phys., Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, scott.hawley@belmont.edu), Zach Evans (Harmonai, Issaquah, WA), and Joe Baldridge (Audio Eng. Technol., Curb College of Entertainment and Music Business, Belmont Univ., Nashville, TN)

Ever since the work of Castellon, Donahue, and Liang (ISMIR 2021) showed that latent space "embedding" representations encoded by OpenAI's Jukebox model contain semantically meaningful information about the music, many have wondered whether such embeddings support vector relations akin to the famous "king—man + woman = queen" result seen in word vector embeddings. Such an "audio (vector) algebra" would provide a way to perform operations on the audio by displacing the embeddings in certain directions, and then decoding them to new sounds. The nonlinear aspects of the encoding process suggest that this may not be possible in general, however, for certain kinds of operations in finite regions of embedding spaces, such embedding vector transformations may indeed have musically relevant counterparts. In this talk we investigate the feasibility of such schemes for the cases of mixing and audio effects.

**10:15–10:25 Break**

10:25

**3aSP5. Digital models of analog circuits for musical audio production: A review of techniques and library for automated circuit solving.** Eric Tarr (Audio Eng. Technol., Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, eric.tarr@belmont.edu)

In music production, many recording and mixing engineers prefer to use analog equipment as a matter of perceptual preference. Digital models of analog circuits have the potential to achieve similar perceptual qualities as hardware without the drawbacks of cost, maintenance, and availability. Many different techniques of system modeling are used in music production software, ranging on a spectrum from "black-box modeling" to "white-box modeling." In black-box modeling, the analog system is modeled as a processing block which maps an input signal to an output signal. Examples include the linear impulse response, adaptive filters, Volterra series, and Weiner-Hammerstein models. In white-box modeling, each individual component of the analog circuit is modeled as part of the overall system. Examples include wave digital filters, state-space modeling, and modified nodal analysis. Various other techniques exist on the spectrum between these two types, using some combination of each. One example is Virtual Analog Filtering based on the Topology Preserving Transform. Machine Learning techniques have also had an important role in advancing the accuracy of digital modeling. Lastly, the "Point to Point Library," developed by the author, will be demonstrated. This MATLAB and C++ library performs automated circuit solving for modeling audio effects.

10:40

**3aSP6. Abstract withdrawn.**

10:55

**3aSP7. Textsetting as sequence alignment.** Hassan Munshi (Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104, hm679@sas.upenn.edu), Mark Liberman, and Jianjing Kuang (Linguist, Univ. of Pennsylvania, Philadelphia, PA)

We argue that textsetting is better accounted for under the framework of sequence comparison, whereby textsetting is taken to be an instance of sequence alignment. Given a text, a tune, and an alignment procedure, we argue that all one needs is to align strong elements of one sequence (the text) to elements of the other (the tune). We show that, in many cases, the only principle needed to account for textsetting in English folk music is Strength Match, i.e., matching stressed syllables to strong beats. Many constraints proposed by previous treatments are to be discarded if we allow our textsetting model to compare between different possible textsettings and then choose the optimal one. The optimization problem is automated under the framework of dynamic programming, which allows us to create a distance metric to score all possible alignments. Moreover, we show that preferred textsettings always have the highest score, when compared to other textsettings that are either ill-formed or less preferred.

## Exhibit

An instrument and equipment exhibition will be located in the Summit Foyer on the 4th floor The Exhibit will include computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems, and other exhibits on acoustics.

Schedule

Monday, 5 December, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including lite snacks and a complimentary beverage.

Tuesday, 6 December, 9:00 a.m. to 5:00 p.m.: Exhibit Open Hours including a.m. and p.m. breaks serving coffee and soft drinks.

Wednesday: 7 December, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. break serving coffee.

## Session 3pAA

# Architectural Acoustics, Noise and Psychological and Physiological Acoustics: Architectural Acoustics and Audio—Even Better Than the Real Thing II

K. Anthony Hoover, Cochair
*McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362*

David A. Conant, Cochair
*McKay Conant Hoover inc, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362*

David S. Woolworth, Cochair
*Roland, Woolworth & Associates, 356 County Road 102, Oxford, MS 38655-8604*

### *Invited Papers*

**1:00**

**3pAA1. Modern movie sound: reality and simulated reality.** Jeffrey Reed (Owner/Engineer, Taproot Audio Design, 108 County Rd. 411, Oxford, MS 38655, info@taprootaudiodesign.com) and David s. Woolworth (Roland, Woolworth & Assoc., Oxford, MS)

Modern movie sound mixing utilizes impulse responses and convolution to provide a usable reproduction of dialogue and other sounds to align the visual and sonic impression while keeping the viewer engaged in the story. This paper will outline the process by which movie audio is processed to get to the final product, and provide and overview of playback platforms. Furthermore short clips of film with audio will be presented to compare the actual *in situ* recording to the studio mixed product, and other experimental/virtual approaches to creating impulse responses for use in the studio mix approach.

**1:20**

**3pAA2. Virtual Acoustics, better than the real thing? Considering the creative side.** Wieslaw Woszczyk (Music Res., McGill Univ., 527 Sherbrooke St., Rm. A-636, Montreal, QC H3K 3G9, Canada, Wieslaw.Woszczyk@mcgill.ca), Aybar Aydin, and Ying-Ying Zhang (Music Res., McGill Univ., Montreal, QC, Canada)

Once room acoustical reflections data are extracted from a physical space or a model, and are encapsulated in a 3D impulse response, they can be used to render immersive sound fields in real time. A range of possibilities then opens for creative use of acoustics in music. A skilled virtual acoustics designer-engineer may rebalance digital signals representing the room response to situate player and listener on the stage or at the back of the auditorium, may modify and arrange temporal segments to re-imagine the aural dimensions of the space, and apply gain and directional placement to shape the impression of immersive presence, adapting acoustics to musicians' creative needs. In the process of building an idealized acoustical environment for the music, techniques of sound reinforcement and of rendering room acoustics are combined to balance presence with ambience and to deliver a sensation of acoustical power with lift-off. The means exist to move beyond acoustical realism into fictionalized acoustics.

**1:40**

**3pAA3. Concert hall reputations versus reality over time, for better or worse?** David A. Conant (McKay Conant Hoover inc, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, dconant@mchinc.com)

The acoustical reputations of many concert halls provide a curious study of human nature as illuminating as their acoustical attributes. This paper reveals how the reputations of 7 prominent halls that opened between 1880 and 1973 have morphed and offers conjecture as to the reasons. Discussed briefly, are Troy Music Hall (1875), Carnegie Hall (1891), Boston Symphony Hall (1900), Philharmonic Hall (1962), and Phoenix Symphony Hall (1973) among others. Each has undergone renovations and met with varying public reaction.

## *Contributed Papers*

**2:00**

**3pAA4. A new variable acoustics technology and the research behind it.**
Niels W. Adelman-Larsen (Flex Acoust., Ny Carlsbergvej 27, Copenhagen 1760, Denmark, nwl@flexac.com)

The paper presents the room acoustic research behind the choice of design of a new variable acoustics technology. It discusses measurements from installations in labs and in a concert hall. Performing arts centres face the challenge of providing a good sound-experience for performers and audiences for a wide variety of types of performances. From theatre or amplified music one night to concerts of quiet chamber music the next. This calls, according to for instance ISO 23591, "*Acoustic quality criteria for music rehearsal rooms and spaces*", for a variability of the reverberation time of a factor of almost 2 at lower frequencies. This standard, as well as recent research by the author is also debated.

**2:15**

**3pAA5. Realism analysis of synthesized healthcare sound environments.**
selen okcu (Architecture, Kennesaw State Univ., 1100 South Mariette Rd., Marietta, GA 30060, sokcu@kennesaw.edu)

The acoustic qualities of healthcare sound environments can have significant influences on care providers' ability to conduct critical sound tasks. Healthcare professionals constantly listen to the aural cues (e.g., alarms) and make critical decisions based on them. This study aims to assess the reliability of an open-source acoustical simulation tool in rendering healthcare providers' auditory experience to enable the development of effective healthcare sound environments. The Pachyderm is developed as an open-source acoustical modeling and simulation plug-in for the Rhinoceros 3D-modeling platform commonly used by designers. The plug-in can compute various room acoustics metrics and generate auralizations by convolving anechoic recordings with the predicted acoustic signature of the architectural settings. While reliable predicted acoustic metrics are critical for iterative room-acoustics design studies, high fidelity auralizations can enable acoustic evaluations mainly based on critical listening and encourage collaboration between acoustical consultants, care providers and designers. The study introduces the findings of acoustic assessments including simulations and field measurements conducted in a new 14-bed adult medical-surgical inpatient unit at Milton S. Hershey Medical Center. The effectiveness is evaluated based on the deviation between the predicted and measured objective measures, and perceived differences between the auralizations and audio recordings assessed via listening tests.

**2:30–3:00**
**Panel Discussion**

## Session 3pBA

## Biomedical Acoustics and Signal Processing in Acoustics: Modern Image Quality Assessment

Brett Byram, Chair
*Vanderbilt University, 2301 Vanderbilt Pl, Nashville, TN 37235*

**Chair's Introduction—1:00**

### *Invited Papers*

**1:05**

**3pBA1. The Myriad of metrics in medical ultrasound imaging; Which should we use?** Ole M. Rindal (Univ. of Oslo, Gaustadalleen 23B, Oslo 0373, Norway, omrindal@ifi.uio.no)

Software beamforming allows flexible adaptive beamformers with great popularity and exponential growth in the number of papers published. Software beamforming can be *a blessing* to the image quality since access to raw channel data allows algorithms that better exploit the data. However, software beamforming can also be considered *a curse*, since the flexibility leads to non-linear algorithms that invalidate the conventional metrics used to evaluate the image quality. A review of the contrast metrics used in ultrasound imaging shows no consensus on the metrics used in the research literature. Multiple metrics exist and different types of data, in different scales, from different stages of the processing chain, are used as input to the metrics.We have demonstrated that many adaptive algorithms alter the dynamic range of the ultrasound images, effectively breaking the conventional metrics both for contrast and resolution. Therefore, we have introduced a new improved contrast metric, the generalized contrast-to-noise ratio (gCNR) immune to dynamic range alternations. The gCNR can be estimated on all kinds of images, regardless of compression, scale, or output units. A major drawback of the conventional contrast metrics, as well as the gCNR, is that they require localization of specific regions and targets in the image. Such objects can be difficult to identify in *in-vivo* images. We recently introduced Global Image Coherence (GIC), as an *in-vivo* image quality metric that does not require any identified regions.



**1:35**

**3pBA2. A mathematical perspective of ultrasound image representations and image quality criteria.** Dongwoon Hyun (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, dongwoon.hyun@stanford.edu)

What is an ultrasound "image"? Where an engineer or physicist may see a **quantitative** map of physical properties, a physician may see **qualitative** anatomical features and pathologies. In the past, each image type was evaluated appropriately within its respective context: quantitative image quality was measured with criteria like the contrast-to-noise ratio (CNR), and qualitative images underwent dynamic range transformations to optimize human perception of contrast. Recent non-linear beamforming methods have blurred this line: the produced images are transformed like qualitative images but evaluated with quantitative criteria like CNR. Here, we identify the mathematical flaws in reasoning and suggest rigorous alternatives.An image assigns values to spatial coordinates. Let $X$ denote the set of all possible image values. The mathematical structure endowed on $X$ determines the set of equivalence-preserving transformations. For instance, qualitative images treat $X$ as a general topological space, where images are equivalent under homeomorphisms of $X$ (dynamic range transformations). Quantitative images treat $X$ as a metric space and have fewer equivalence-preserving transformations. These structures determine when criteria like contrast, CNR, generalized CNR, and spatial resolution can be used. To extrapolate these criteria to other contexts, one must use relevant structure-preserving isomorphisms (e.g., histogram matching to preserve information content).

**2:05**

**3pBA3. The impact of loss function domain for optimizing ultrasound imaging.** Jacob Spainhour, Stephen Becker (Appl. Mathematics, Univ. of Colorado Boulder, Boulder, CO), and Nick Bottenus (Mech. Eng., Univ. of Colorado Boulder, 1111 Eng. Dr., ECME 107, 427 UCB, Boulder, CO 80309, nick.bottenus@colorado.edu)

The conventional view of ultrasound pulse-echo data—focused sound sent in a direction and bounced back towards the transducer—does not capture the rich information available. REFoCUS beamforming abandons this limitation, combining the ideas of spatially coded excitation and synthetic aperture imaging. Each transmission is an encoding of elements with various weights and time delays, and the multistatic data set (all transmit/receive element pairs) is estimated from the received data. We can, therefore, apply a single focusing operation across different choices of transmit pulse sequencing and ask questions about how to best optimize that sequence.The multistatic data set is a useful mathematical model since it is tied to individual array elements. Several works have used such raw echo data in optimization problems, for instance training neural networks to improve data quality from limited transmissions. However, we have found that the image formation process (e.g., focusing, beamforming) can change the appearance of errors in the data set. Even small errors in the raw data can result in significant image artifacts due to the inherent ill-conditioning of the beamforming operation. We demonstrate improvements in optimization using loss functions defined in the image domain compared to the raw multistatic data.

**2:35**

**3pBA4. A spatiotemporal coherence-based approach for optimizing transcranial ultrasound imaging sequences.** Emelina P. Vienneau (Biomedical Eng., Vanderbilt Univ., 5824 Stevenson Ctr., Nashville, TN 37232, emelina.p.vienneau@vanderbilt.edu) and Brett Byram (Biomedical Eng., Vanderbilt Univ., Nashville, TN)

Functional ultrasound imaging (fUSI) is a promising new neuroimaging modality for preclinical and niche clinical applications, but translation to noninvasive use in adults remains unresolved due to skull-induced image degradation. Attenuation leads to low signal-to-noise ratio (SNR), whereas clutter leads to low signal-to-clutter ratio (SCR). Coded excitation or microbubbles could increase SNR, whereas adaptive beamforming, phase aberration correction, or harmonic imaging could increase SCR. However, it's unknown how the relative importance of noise and clutter varies across demographics. To optimize a demographic-specific transcranial imaging sequence, a tool to measure SNR and SCR and assess the impact of sequence modifications is needed. We propose using a spatiotemporal coherence technique [DOI:10.1109/TUFFC.2022.3152717] to measure SNR and SCR with coded excitation [DOI:10.1109/IUS46767.2020.9251650] for different parameters like code length and voltage. We conducted a transcranial imaging study in five volunteers (median age 25, Caucasian, three males, two females) and used focused M-Mode data to measure SNR and SCR. With optimally chosen parameters, we improved SNR by $16.87 \pm 2.18$ dB and $13.18 \pm 1.14$ dB with 65 bit and 25 bit codes without introducing additional clutter (SCR gains were $-0.29 \pm 0.67$ dB and $0.13 \pm 0.84$ dB with 65 and 25 bit codes). These improvements will enable clinical translation of transcranial fUSI.

**2:50**

**3pBA5. High-order singular value decomposition of contrast pulsing sequences for improved clutter suppression.** Geraldi Wahyulaksana (Erasmus MC, Wytemaweg 80, Rotterdam, Zuid Holland 3015CN, the Netherlands, g.wahyulaksana@erasmusmc.nl), Luxi Wei (Erasmus MC, Rotterdam, the Netherlands), Jason Voorneveld (Erasmus MC, Rotterdam, Zuid Holland, the Netherlands), Nico de Jong, Antonius F. van der Steen, and Hendrik Vos (Erasmus MC, Rotterdam, the Netherlands)

Contrast-enhanced ultrasound is a diagnostic tool used to visualize blood flow in the cardiovascular system. The use of ultrasound contrast agent (microbubbles) in combination with contrast pulsing scheme (CPS) improves the sensitivity and specificity of ultrasound flow imaging by enhancing the signal in the blood compartment. The commonly used CPS are pulse inversion (PI), amplitude modulation (AM), and amplitude-modulated pulse inversion (AMPI). Using differences in phase or amplitude of multiple pulses, the linear tissue clutter signal can be suppressed. However, this process can be degraded by motion or non-linear propagation of the ultrasound wave. These effects cause the cancellation of linear clutter signal to be ineffective We propose using higher-order singular value decomposition (HOSVD) with spatial, temporal, and pulsing dimensions as the input to improve clutter suppression under the conditions of motion and non-linear propagation. We performed systematic *in-vitro-* experiment emulating these conditions, as well as *in-vivo* cardiac measurements. The results showed that HOSVD increases the clutter suppression of all the 3 CPS compared to the conventional linear processing. The improvement of clutter reduction could be beneficial to various cardiac evaluation like myocardial perfusion or intra ventricular flow assessment.

**3:05**

**3pBA6. Evaluating a modified delay-multiply-and-sum reconstruction algorithm to improve detection of osteochondritis dissecans.** Philip M. Holmes (Mayo Clinic Graduate School of Biomedical Sci., 200 1st St. SW, RO_OS_02_2008, Rochester, MN 55902, holmes.philip@mayo.edu), Kun-Hui Chen (Dept. of Orthopedic Surgery, Mayo Clinic, Rochester, MN), Hyungkyi Lee (Dept. of Radiology, Mayo Clinic, Rochester, MN), James Fitzsimmons (Dept. of Orthopedic Surgery, Mayo Clinic, Rochester, MN), Shawn O'Driscoll (Dept. of Orthopedic Surgery, Mayo Clinic, Rochester, MN), and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Osteochondritis dissecans (OCD) is a focal joint defect that is prevalent among youth athletes. With the application of medical ultrasound, OCD of the humeral capitellum could be detected earlier and prevent surgery. In this work, we tested a modified Delay-Multiply-and-Sum (DMAS) reconstruction algorithm to evaluate how it affects medical ultrasound's ability to detect OCD. Starting with the DMAS reconstruction algorithm described by Matrone *et al*. (2015), we modified the implementation of filtering and envelope detection steps. Delay-and-Sum (DAS), DMAS, and modified DMAS algorithms were tested on phantom and cadaveric models of capitellar OCD. The DMAS and modified DMAS images were histogram matched to the DAS image for quantitative comparison. By taking several profiles across the images of the artificial OCD lesions, we compared the lesion contrast and bone surface clarity produced by each algorithm. We found that the unmodified DMAS algorithm showed little improvement over the DAS algorithm, particularly after histogram matching. The modified DMAS algorithm showed a much greater improvement in the detection of OCD lesions compared to the DMAS algorithm. This modified DMAS algorithm could be used for other bone surface imaging applications. Future work includes evaluating these algorithms *in vivo* with patients diagnosed with OCD.

# Session 3pCA

## Computational Acoustics: Learning and Stochastic Modeling in Computational Acoustics II

Pierre F. Lermusiaux, Cochair
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Wael H. Ali, Cochair
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Aaron Charous, Cochair
*Mechanical Engineering, Massachusetts Institute of Technology, 77 Massachusetts Avenue, Cambridge, MA 02139*

### *Contributed Papers*

**1:00**

**3pCA1. Trans-dimensional Inversion in two spatial dimensions for geoacoustic parameters.** Tim Sonnemann (Dept. of Geoscience, Univ. of Calgary, 2500 University Dr. NW, Calgary, AB T2N 1N4, Canada, tim.sonnemann@ucalgary.ca), Jan Dettmer (Dept. of Geoscience, Univ. of Calgary, Calgary, AB, Canada), Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR), and Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

We test a series of model parametrizations to invert a two-dimensional (2-D) seabed transect going from simple fixed-dimensional to more complex trans-dimensional configurations. Inverting spherical-wave reflection coefficient datasets independently as one-dimensional layered models for a 2-D section is less efficient than applying a more parsimonious 2-D parametrization while also carrying out full uncertainty quantification. We approach the problem by proposing different fixed and dynamically inferred parametrization schemes, and discuss implementation, computational cost and resulting accuracy. We demonstrate the application in geoacoustics using a dataset of 1711 source transmissions recorded on a 32-element linear hydrophone array with both source and array towed by an autonomous underwater vehicle along a 12 km transect on the Malta Plateau in the Mediterranean Sea.

**1:15**

**3pCA2. High frequency stochastic acoustic wavefront propagation and joint ocean-acoustic inference: The GMM-DO wavefront.** Manmeet S. Bhabra, Wael H. Ali (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Pierre F. Lermusiaux (MIT, 77 Mass Ave., Cambridge, MA 02139, pierrel@mit.edu)

In marine applications, the value of accurate modeling and learning for stochastic acoustic propagation in uncertain ocean environments cannot be overstated. In this work, we derive stochastic theory and schemes for (i) modeling of high frequency acoustic propagation in uncertain ocean environments and (ii) joint Bayesian assimilation of ocean-acoustic measurements to infer fields, parameters, and uncertain model functions. We first obtain the Dynamically Orthogonal (DO) wavefront equations to solve for the stochastic extension of the Liouville Equation that governs the dynamics of acoustic wavefront in an augmented phase space. These DO wavefront equations provide the prior for the Gaussian Mixture Model—DO (GMM-DO) filter that completes joint physics-acoustics Bayesian inference using sparse observations. Specifically, given a set of receivers, the Eulerian nature of the DO wavefront equations allows for the efficient extraction of arrival time prior probability distributions. The GMM-DO Waverfront filter then combines these joint priors with arrival time measurements using Bayes rule, jointly inferring environmental properties (e.g., unknown source location and/or sound speed field), the acoustic wavefront distribution, and the arrival time distribution itself. We evaluate results using high-frequency applications, illustrating the estimation of mean fields and properties, but also of probability density distributions and model parameterizations.

**1:30**

**3pCA3. Simultaneous detection and ranging of baleen whale impulsive vocalizations using a temporal convolutional network.** Mark Goldwater (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Mail Stop 55, Woods Hole, MA 02543-1050, mgoldwater@whoi.edu), Daniel P. Zitterbart (Woods Hole Oceanographic Inst., Woods Hole, MA), Dana Wright (Duke Marine Lab., Beaufort, NC), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

In shallow-water environments, low-frequency acoustic signals exhibit dispersive propagation due to interactions with the sea surface and seabed. The received signal can then be modeled as a set of propagating modes. Single-hydrophone modal dispersion has been used to range baleen whale vocalizations and estimate shallow-water geoacoustic properties. However, these algorithms require preliminary signal detection and human labor to estimate the modal dispersion. Here, we apply a temporal convolutional network (TCN) to time-frequency representations of baleen whale gunshots (impulsive calls) to simultaneously detect and range them in large single-hydrophone passive acoustic monitoring datasets. The TCN jointly learns ranging and detection by training using both synthetic gunshots simulated across multiple environments/ranges and experimental noise. The synthetic data is informed by only the experimental dataset's water column depth, sound speed, and density, while other waveguide parameters vary within empirically observed bounds. The method is applied to an experimental North Pacific right whale dataset collected in the Bering Sea using a single hydrophone. To evaluate model performance, 50 calls are manually ranged using a state-of-the-art physics-based inversion method. The TCN closely matches the physics-based range estimations and detects dispersive gunshots among noise-only examples with high precision and recall. [Work supported by the ONR].

**1:45–2:05 Break**

**2:05**

**3pCA4. Predicting cross-barrier communication disruption using adaptive Support Vector Machines.** Cameron A. McCormick (Sandia National Labs., 1611 Innovation Pkway SE, Albuquerque, NM 87123, camccor@sandia.gov), Wilkins Aquino (Duke Univ., Durham, NC), Chandler Smith, and Timothy Walsh (Sandia National Labs., Albuquerque, NM)

Certain applications may exist where it is necessary to transmit data between two sides of a physical barrier without compromising the structure of the barrier itself. This may be the case when the barrier separates personnel from dangerous conditions or when sensitive electronics are hermetically sealed. One solution is to use mechanical waves to transmit data across the barrier directly. However, this solution is susceptible to new environmental disruptions that can compromise the data stream, such as mechanical vibrations from components in contact with the barrier. Comprehensively evaluating the entire range of expected environmental disruptions can be prohibitive. This work will present the application of Support Vector Machines (SVMs) to adaptively predict environments in which data disruption is expected to occur. The problem is cast as a binary classification problem, for which SVMs are one of the leading machine learning algorithms. The adaptive SVM approach is able to accurately predict disruptive environmental conditions while exploring only a small subset of the entire range. Knowledge of the environmental conditions that lead to data disruption will then enable the design of experiments for validation.SNL is managed and operated by NTESS under DOE NNSA contract DE-NA0003525

**2:20**

**3pCA5. Developing a latent space to describe full-sphere head-related transfer functions.** Eric M. Sumner (Háskóli Íslands, Bjargargata 1, #318, Reykjavík 102, Iceland, ems36@hi.is), Rúnar Unnþórsson, and Morris Riedel (Háskóli Íslands, Reykjavík, Iceland)

Several prior studies have used machine-learning techniques, such as Principal Component Analysis (PCA), to investigate the information content in unidirectional Head-Related Transfer Funcions (HRTFs). This paper builds upon this prior work to investigate the information content in full-sphere, free-field HRTFs. An artificial neural network is trained to reproduce the PCA coefficients of an HRTF from the combination of a direction vector and latent space vector identifying the HRTF. The training data comes from several different HRTF databases; the methods used to combine these databases into a single coherent dataset are described. Hyperparameter optimization techniques are then used to minimize the dimensionality of the latent space vector, providing new information about the information content required to accruately describe a full-sphere HRTF.

**2:35**

**3pCA6. Stochastic multiscale simulation of porous sound absorbing material based on adaptive bayesian quadrature.** Yosuke Komatsu (Mech. Eng., Kogakuin Univ., 2665-1 Nakano-machi, Hachioji-shi, Tokyo 192-0015, Japan, ad20001@ns.kogakuin.ac.jp) and Takashi Yamamoto (Mech. Eng., Kogakuin Univ., Hachioji-shi, Tokyo, Japan)

This presentation proposes a method to analyze porous sound absorbing materials by multiscale simulation with the adaptive Bayesian quadrature method. The homogenization method calculates equivalent physical acoustic properties from the microstructure of a porous material. This method is powerful in designing new porous materials because it can predict acoustic properties from arbitrary microstructures. On the other hand, it is difficult to consider the randomness of actual materials because of the assumption of periodic microstructures. We propose a method for quantifying uncertainty in acoustic properties by incorporating a Bayesian perspective into Gaussian process regression. Assuming the microscopic structure of porous material as random variables, the response obtained by the homogenization method is approximated by Gaussian process regression. Bayesian quadrature is used to calculate the statistical moments of the integral of the characteristic functions.An additional integration point that minimizes the variance of the integral is adaptively selected, and further responses are obtained using the homogenization method. Repeating this process shows that the integral of characteristic functions can be calculated accurately and efficiently, and their uncertainties can be quantitatively evaluated.

## Session 3pED

## Education in Acoustics: Acoustics Education Prize Lecture

**Chair's Introduction—1:00**

### *Invited Paper*

**1:05**

**3pED1. DJ Prof: Reflections on teaching.** Kathleen E. Wage (George Mason Univ., 4400 University Dr., Fairfax, VA 22151, kwage@gmu.edu)

In a short-lived career as a cartoonist for Acoustics Today [Summer 2016], I sketched DJ Prof, an acoustics professor who mixes multiple modes of instruction to engage and excite students. DJ Prof illustrates the concept of active learning, which Freeman *et al*. called the "preferred empirically validated teaching practice" [PNAS, 2014]. Numerous studies show that active learning courses improve student learning, increase retention rates, and reduce performance gaps in STEM for economically disadvantaged students and females in male-dominated classes. This lecture reflects on my evolution as DJ Prof using examples from acoustic signal processing courses. I will provide a brief overview of the literature on active learning, share a few favorites from my pedagogical playlist, and highlight open questions for the acoustics education community. The session will include interactive exercises. Please bring an open mind, a sense of humor, and a willingness to meet the people sitting near you.

## Session 3pID

## Interdisciplinary: Hot Topics in Acoustics

Christina Naify, Cochair
*Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78758*

Zane T. Rusk, Cochair
*The Pennsylvania State University, 104 Engineering Unit A, University Park, PA 16802*

A. J. Lawrence, Cochair
*Walker Department of Mechanical Engineering, University of Texas at Austin, 204 E. Dean Keeton Street, Stop C2200, Austin, TX 78712-1591*

**Chair's Introduction—2:15**

### *Invited Papers*

**2:20**

**3pID1. On the potential use of infrasound for tornado detection and monitoring.** Roger M. Waxler (NCPA, Univ. of MS, P.O. Box 1848, University, MS 38677, rwax@olemiss.edu), Garth Frazier, Carrick Talmadge (NCPA, Univ. of MS, Oxford, MS), Claus Hetzer (NCPA, Univ. of MS, Tempe, AZ), Bin Liang, and Hank Buchanan (NCPA, Univ. of MS, Oxford, MS)

The notion that tornadoes emit an infrasonic signal that can be detected from great distances; and thus be used to aid in the detection, monitoring, and potentially tracking of active tornadoes; has been discussed for decades. To test this notion, we deployed a network of infrasound sensor arrays in northern Alabama for the 2017, 2018, and 2019 tornado seasons. Infrasound propagation is complicated by vertical temperature gradients and wind shear. Efficient propagation depends on ducts caused by positive temperature and wind speed gradients. Infrasound signal detection depends on a signal arriving at the detection array with amplitude larger than the turbulent pressure amplitudes (wind noise) in the neighborhood of the array. Focusing on the 1 Hz to 10 Hz frequency band, we show that from each tornado that passed through northern Alabama during that period for which propagation and wind nose analysis suggested that a signal should be detected, a signal was detected. To make a truly convincing argument that the observed signals are produced by active tornadoes two things are needed. One needs to demonstrate that storms from which no tornadoes were spawned do not produce such signals and one must show that there is a physical mechanism related directly to tornadoes through which the infrasound is emitted. We will report on progress on both these fronts.

**2:45**

**3pID2. Acoustics for in-process melt pool monitoring during metal additive manufacturing.** Christopher M. Kube (Eng. Sci. and Mech., The Pennsylvania State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, kube@psu.edu)

Intrinsic to most metal additive manufacturing (AM) processes are melt pools generated from directed energy sources like lasers. Melt pools are critical as they function to join powder layers to previous layers during the process. Their criticality extends deeper as most porosity in AM parts stems from melt pool behavior while melt pool solidification dictates the parts' microstructure. Significant breakthroughs in metal AM are severely hindered by the lack of access to experimental tools to study melt pools and modeling that do not fully capture the complex physics. Thus, melt pool related defects are often difficult to predict in occurrence and location while determining optimal process parameters to eliminate defects is extremely challenging and costly. Furthermore, the many exciting opportunities such as realizing new AM alloys, developing gradient materials/structure, and tailoring microstructure to intended applications are possible only with further understanding of melt pool behavior. With these clear needs, in-process acoustics have been proposed as plausible experimental tools for studying melt pool behavior. This presentation will provide an overview of the current activity in this area in addition to the specific needs the acoustics community can potentially address.

3:10

**3pID3. The roar of the rocket: A hot topics discussion.** Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N281 ESC, Provo, UT 84602, kentgee@byu.edu)

An increasing number of companies and countries are building and launching space vehicles of different sizes and configurations for diverse purposes. The goal of this Hot Topics presentation is two-fold: first, to introduce a wide-ranging audience of Acoustical Society of America members to principles of rocket noise radiation, propagation, and reception; second, to discuss needs and challenges related to future launch vehicle acoustics research. If attendees leave both mildly entertained and better informed, the presentation will have been a success.

WEDNESDAY AFTERNOON, 7 DECEMBER 2022 SUMMIT B, 1:00 P.M. TO 2:15 P.M.

## Session 3pNS

## Noise, Architectural Acoustics, Computational Acoustics, and Physical Acoustics: Topics on Noise: Noise Induced Hearing Loss and Community Noise

S. Hales Swift, Chair
*Sandia National Laboratories, P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082*

### *Contributed Papers*

1:00

**3pNS1. What is the safe noise level to prevent noise-induced hearing loss?** Daniel Fink (The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733, DJFink@thequietcoalition.org)

Noise causes short-term mechanical and chemical changes in the cochlea, transduced into electrical signals transmitted to the brain. Excessive noise exposure overwhelms the cochlea's ability to reverse these changes, causing noise-induced hearing loss (NIHL). The Centers for Disease Control and Prevention state that NIHL is the only completely preventable type of hearing loss. What is the evidence-based safe noise exposure level to prevent NIHL? The National Institute for Occupational Safety and Health (NIOSH) recommended exposure level (REL), 85 dBA, does not prevent NIHL. Occupational exposures are calculated for 8 hours/day, 240 days/year, for 40 years. To calculate the noise exposure level to prevent NIHL in the public, 70 dB for 24 hours ($L_{eq(24)}=70$), the Environmental Protection Agency (EPA) adjusted the REL for 24 hours/day, 365 days/year exposure. The actual safe noise exposure must be lower than 70 dB because 1) the EPA did not adjust for lifetime noise exposure, 2) NIOSH based its REL on limited-frequency pure tone audiometry, an insensitive measure of auditory damage, and 3) non-occupational noise exposure has increased. The safe noise level may be as low as 55 dBA, the effective quiet level required for humans to recover from noise-induced temporary threshold shift.

1:15

**3pNS2. The National Institute for Occupational Safety and Health recommended exposure level for noise needs to be revised downwards.** Daniel Fink (The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733, DJFink@thequietcoalition.org)

Noise-induced hearing loss (NIHL) is caused by cumulative occupational and non-occupational noise exposure. The National Institute for Occupational Safety and Health (NIOSH) developed recommendations for occupational noise exposure in 1972, revised in 1998. Fifty years after the NIOSH 85 dBA recommended exposure level (REL) was first published, it needs to be revised downwards. In its calculations, NIOSH assumed that workers have quiet (<60 dBA) when not at work, something no longer true. Everyday public noise exposure now exceeds the Environmental Protection Agency's safe noise level, a time-weighted average of 70 dB for 24 hours, with multiple studies reporting average exposures near 75 dBA. Also, NIOSH calculations were based on occupational studies of material hearing loss measured by limited frequency pure tone audiometry, an insensitive measure of noise-induced auditory damage. Modern audiology techniques include extended range audiometry, speech in noise testing, diffusion product otoacoustic emissions testing, and questions about tinnitus and hyperacusis. Doing the needed research to support downward REL revision, detecting auditory damage in noise-exposed workers with these newer techniques, is a daunting task, but this is necessary both to protect workers and because the NIOSH REL forms the basis for the world's recommendations for non-occupational noise exposure.

1:30

**3pNS3. Urban noise.** Juvanely Y. Landa (8266 Swanston Ln., Gilroy, CA 95020, juvanelylanda15@gmail.com) and Kimberly A. Riegel (Phys., Queensborough Community College, Pelham, NY)

People in urban areas are exposed to high noise levels, leading to an increased number of health issues. A mobile phone application called Auditive, was developed to allow the user to report health history, sound level annoyance, and record sound levels on their mobile phones. This allows for real-time sound level data to be directly correlated with the annoyance response. The data recorded with the auditive app provides opportunities for researchers to find correlations between noise exposure, health, and annoyance. A major concern with mobile phone recorded noise is the accuracy of the absolute sound levels without calibration. This study is focused on calibrating mobile phones to provide more accurate data. Two methods for calibration were considered. Technique one calibrates, using the same correction for each type of phone. MATLAB was used to generate pink

3p WED. PM

noise at different sound levels to test a variety of iPhone models. The correction for each frequency level ranging from 63 Hz to 8000 Hz, was determined. Technique two calibrated individual phones. We tested consistent sounds that allow Auditive users to calibrate their phones. These two different techniques will be compared, to establish the best way to calibrate different mobile phones.

**1:45**

**3pNS4. Noise Pollution in Hospitals and its Impacts on the Health Care Community and Patients.** Olivia Coiado (Univ. of Illinois Urbana-Champaign, 1914 Max Run Dr., Champaign, IL 61822-3449, coiado@illinois.edu), Felipe Vergara, and Lizandra Vergara (Federal Univ. of Santa Catarina, Florianopolis, Brazil)

Noise pollution in hospitals is known to affect the health of patients, but it also impacts the staff. Most of a hospital's environment is affected by the sounds of equipment and machines with high sound pressure levels (SPL). We directed the study of both quantitative aspects to reduce SPL and qualitative research which considers the soundscapes of hospitals and people's perceptions. The main goal of this study was to do an assessment of the noise pollution in hospitals in Brazil and USA to investigate the effects on the health care community and patients. The objectives were: 1) Implement a sound mapping, day and night, in different units of the hospital; 2) Characterize the variations of the SPL of the various noise sources in the hospital's care units; 3) Develop and apply a qualitative assessment based on the opinion of users of the hospital in relation to the noise perceived by them; 4) Establish/propose an analytical-experimental model based on correlations of objective data and subjective data. This study identified metrics that can be applied as an intervention plan and prevention to reduce noise pollution in hospitals in Brazil and in the USA that could be implemented by other institutions, locally and internationally.

**2:00**

**3pNS5. European policy for port noise control and mitigation.** Corrado Schenone (Dept. of Mech. Eng., Univ. of Genoa, Via Opera Pia 15/A, Genova I-16145, Italy, corrado.schenone@unige.it), Tomaso Gaggero (DITEN, Univ. of Genoa, Genova, Italy), Francesco D'Alessandro (IIA, CNR, Rome, Italy), Giorgio Baldinelli (Dipartimento di Ingegneria, Univ. of Perugia, Perugia, Italy), Samuele Schiavoni (Metexis, Perugia, Italy), and Davide Borelli (Dept. of Mech. Eng., Univ. of Genoa, Genova, Italy)

Port noise affects the development of harbour cities on EU shores. Tourism, goods transportation and passenger traffic are increasingly limited by noise impact on port cities, which annoys residents and disturbs the sleep of the exposed population. Therefore, the EU has promoted coordinated actions and programmes aiming to control and reduce port noise in the Europena area. Port authorities, shipping companies, public orgnizations, shipyard owners, terminal operators, municipalities are developing an ambitious programme pursuing harbour noise control. Several projects have been funded to develop a recovery strategy and promote collaboration between port stake-holders. These projects cover different aspects of port noise in the framework of a coordinated policy over a specific area of the European shores. Each project addresses a specific goal, in accordance with an overall coordinated strategy: RUMBLE deals with large commercial ports, DECIBELwith small touristic harbours, LIST with noise from traf-fic generated by the ports, MONCUMEN with measurements and characterisation, REPORT with modelling and impact prediction, TRIPLO with reaction of the exposed population. The outcomes of these projects are expected to create a decisive turning point, both at territorial and technical level, with relevant benefits for the whole Europe.

# Session 3pPAa

## Physical Acoustics and Computational Acoustics: Infrasound II

Philip S. Blom, Cochair
*Earth & Environmental Sciences, Los Alamos National Laboratory, PO Box 1663, M/S F665, Los Alamos, NM 87545*

Gil Averbuch, Cochair
*Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 31 Wareham Avenue, Onset, MA 02558*

Roberto Sabatini, Cochair
*Embry-Riddle Aeronautical University, 1 Aerospace Blvd, Daytona Beach, FL 32114*

### *Contributed Papers*

1:00

**3pPAa1. Modeling regional propagation of infrasonic Mach cone energy from a supersonic source.** Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., PO Box 1663, M/S F665, Los Alamos, NM 87545, pblom@lanl.gov)

Supersonic sources such as bolides, aircraft, and rocket launches are known to produce a wide range of acoustic and infrasonic signals. The lower frequency infrasonic component of these signals can propagate large distances through the atmosphere due to the decreased thermo-viscous losses at low frequency and presence of temperature and wind induced infrasonic waveguides so that signals are often detected at regional distances of hundreds of kilometers. The dominant radiator of energy from such sources is the Mach cone signal that emanates from the source with very specific geometry. Ray tracing analysis using a Mach cone source model to predict infrasonic signals at regional distances will be presented and several parametric studies and comparisons with observed signals from several representative supersonic sources will be presented.

1:15

**3pPAa2. Updates to global empirical models for infrasonic signal celerity and backazimuth from ground truth data.** Alexandra Nippress (AWE Blacknest, AWE Blacknest, Brimpton, Reading RG7 4RS, United Kingdom, alex@blacknest.gov.uk) and David N. Green (AWE Blacknest, Brimpton, United Kingdom)

Global empirical models for infrasonic signal celerity (the epicentral distance divided by the total travel time) and backazimuth deviation (the difference between the measured and predicted backazimuth assuming great circle propagation), are used for the association of infrasound automatic detections, event location and acoustic propagation simulation validation. Following a previous methodology to develop a regional celerity-range model (Nippress *et al.*, 2014), we developed a software suite for consistent analysis of a global ground truth database, allowing estimation of empirical models for celerity and backazimuth. We observe 304 detections in the 0.32–1.28 Hz passband, with propagation path lengths of between 25 and 6280 km. Models derived from these observations suggest the backazimuth deviation distribution is range-independent, 92% of the detections studied have a deviation $\leq \pm 5°$. However, the celerity model, produced through fitting the travel-times with a linear regression model, is range-dependent. The celerity model bounds are determined using a quantile regression fit to the travel-time residuals, and are consistent with the current understanding of infrasound propagation. At 10+ years since the publication of the last global celerity-range model, this study provides a timely update.UK Ministry of Defence © Crown Owned Copyright 2022/AWE

1:30

**3pPAa3. Coherent infrasound generation using an air-propane burner.** Chad M. Smith (Appl. Res. Lab., The Pennsylvania State Univ., State College, PA 16804, chad.smith@psu.edu), Thomas B. Gabrielson (Appl. Res. Lab., The Pennsylvania State Univ., State College, PA), and B. J. Merchant (Sandia National Labs., Albuquerque, NM)

An invaluable tool in characterization of any receiver, propagation path, or detection system, is a source with known and repeatable signal characteristics. This talk will discuss development and evaluation of a coherent (non-explosive, periodic, with controlled duration) infrasound source with frequency capabilities in the sub-hertz to several hertz band. Design of a practical sound source within this band is a difficult engineering challenge. The simple source equation, which will govern any portable human-fabricated infrasound source due to the long wavelengths, shows this fundamental difficulty. As frequency decreases volume displacement must increase by the squared inverse factor of frequency in order to maintain an equal pressure amplitude at equal range. For this reason, the authors investigate utilizing the high energy density available in gas combustion to periodically displace large volumes of air within the open atmosphere. Prototype testing has verified the capability of generating continuous signals at a fundamental frequency of 0.25–1.5 Hz in the farfield—or ranges from the source where pressure and particle velocity are roughly in-phase. Harmonics of this fundamental are also generated throughout the 0.25–4.0 Hz band with reasonable signal-to-noise ratio. Development of the infrasound source prototype as well as experimental testing and results will be discussed.

1:45

**3pPAa4. On the correlation between microbaroms and microseism, and what can we learn from it.** Gil Averbuch (Earth Sci., Southern Methodist Univ., 515 W 10th St., apt 116, Dallas, TX 75208, gil.averbuch@whoi.edu), Jelle D. Assink (R&D Seismology and Acoust., KNMI, Utrecht, the Netherlands), and Stephen Arrowsmith (Southern Methodist Univ., Dallas, TX)

Wind-driven ocean swells act as the source for both atmospheric microbaroms and solid earth's microseisms. Since they share the same source, microbaroms and microseisms' power are expected to display similar characteristics in collocated seismo-acoustic stations. Any deviation from this hypothesis can provide information about the source mechanism, the state of the atmosphere, and the seismo-acoustic energy partitioning at the source region. In addition, prolonged observations may show evidence of climatic trends. This study presents a multi-year (~20 years) comparison between microbaroms and microseisms' power in collocated seismo-acoustic stations. Initial results show that sporadically, microbaroms and microseisms' power can be in and out of phase up to 180 degrees. Moreover, stations'

spatial location seems to affect the observations and will be further investigated.

**2:00–2:15 Break**

**2:15**

**3pPAa5. Numerical modeling of seismically-induced infrasound impacts on ionospheric plasma densities and mesospheric airglow emissions.**
Pavel A. Inchin (Dept. of Physical Sci., Embry-Riddle Aeronautical Univ., 1 Aerosp. Blvd., Daytona Beach, FL 32114, inchinp@erau.edu), Yoshito Nozuka (Dept. of Geophys., Kyoto Univ., Kyoto, Japan), Jonathan B. Snively (Dept. of Physical Sci., Embry-Riddle Aeronautical Univ., Daytona Beach, FL), Yoshihiro Kaneko (Dept. of Geophys., Kyoto Univ., Kyoto, Japan), Roberto Sabatini, and Matthew D. Zettergren (Dept. of Physical Sci., Embry-Riddle Aeronautical Univ., Daytona Beach, FL)

Seismically-induced infrasonic waves (IWs) are known sources of disturbances in the upper layers of the atmosphere. These waves experience growth and potential for nonlinear evolution with height and may ultimately exhibit amplitudes of ten(s) of % of local Mach number following events of sufficient strength and scale. Along with a routine detection of their impacts on the ionospheric plasma, e.g., in GPS signal-based measurements of integrated total electron content (TEC), studies demonstrate opportunities to exploit these observations for the investigation of earthquake sources.We present the results of specific case and parametric studies of the feasibility to infer earthquake source characteristics based on observations of IW impacts on mesospheric airglow emissions and ionospheric densities. Studies include numerical simulations performed with three-dimensional coupled seismic wave propagation, and neutral atmospheric and ionospheric models, covering the chain of processes from earthquake sources to observable signatures. The results suggest that upper-atmospheric/ionospheric measurements of IWs may supplement classical seismic and geodetic observations over complex ruptures or undersea earthquakes, by providing additional independent information. They also reveal key dependencies on model specifications – from physical processes to propagation environments in multiple media – and thus require comprehensive validation to establish their quantitative utility.

**2:30**

**3pPAa6. Impact of explosion-generated acoustic waves on atmospheric layers.**
Roberto Sabatini (Embry-Riddle Aeronautical Univ., 1 Aerosp. Blvd, Daytona Beach, FL 32114, sabatinr@erau.edu), Pavel A. Inchin (Physical Sci., Embry-Riddle Aeronautical Univ., Daytona Beach, FL), Benedict Pineyro (Embry-Riddle Aeronautical Univ., Daytona Beach, FL), and Jonathan B. Snively (Dept. of Physical Sci., Embry-Riddle Aeronautical Univ., Daytona Beach, FL)

Explosive events, such as artificial or accidental explosions, and volcanic eruptions, among others, generate low-frequency acoustic waves that propagate through and perturb atmospheric and ionospheric layers (e.g., the hydroxyl and oxygen airglow layers, the sodium layer, and the ionospheric D-region). The subsequent disturbances (i.e., airglow emission intensity, sodium or other minor species density, or electron density fluctuations) are potentially detectable by optical or radio remote sensing methods. Understanding and quantifying the impact of acoustic waves on atmospheric layers are, therefore, crucial steps for establishing detectability thresholds (e.g., relative to source scales and effective yields). In this work, we investigate the propagation of upwardly-traveling acoustic perturbations induced by 1t to-1 kt of TNT-equivalent ground explosions and their signatures on different atmospheric layers. Specifically, we estimate the amplitudes and periods of the induced fluctuations of hydroxyl, sodium, and electron densities at mesospheric through lower-thermospheric altitudes. This work investigates the potential of such layers to serve as sensors for characterizing lower-atmospheric explosive events and the complementarity of such indirect measurements with direct sensing, e.g., of pressure fluctuations *in situ*.

**2:45**

**3pPAa7. Numerical modeling of *T*-wave generation in a horizontally inhomogeneous ocean.**
Oleg A. Godin (Phys. Dept., Naval Postgrad. School, Monterey, CA 93943, oagodin@nps.edu), Paul Cristini (LMA, CNRS UPR 7051, Aix-Marseille Univ., Marseille, France), Alexis Bottero (Scripps Inst. of Oceanogr., UCSD, Nice, France), and Vadim Monteiller (LMA, CNRS UPR 7051, Aix-Marseille Univ., Marseille, France)

*T*-waves are generated by earthquakes and travel horizontally with the speed of sound. They are the most common earthquake sounds in the ocean and make strong contributions to the ambient sound field. However, the mechanisms of their generation remain poorly understood. Recent theoretical paper [O. A. Godin, *J. Acoust. Soc. Am.* **150**, 3999–4017 (2021)] argues that scattering of ballistic waves from the earthquake focus by the ocean surface is a strong source of *T*-waves, which should be considered on par with previously identified generation mechanisms of wave diffraction on large bathymetric features and scattering by the rough seafloor. This paper compares the *T*-waves due to ocean surface roughness and a seamount in deep water. 3-D surface scattering and wave interaction with large-scale bathymetric features are modeled rigorously using time-domain spectral-element code SPECFEM3D, with the ocean surface being shaped by wind waves and sea swell and a large seamount located off the earthquake epicenter. These simulations are complemented by a normal-mode code to propagate the sound field from the generation region to long ranges. A mode projection technique is developed to efficiently couple SPECFEM3D to the normal mode code. The two mechanisms are found to generate *T*-waves of comparable amplitudes.

**3:00**

**3pPAa8. Vector properties of the low-frequency acoustic waves generated by earthquakes on the New England Bight.**
David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu), Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Gopu R. Potty, and James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

During a recent ONR experiment on the Northwest Atlantic continental shelf (i.e. New England Bight), a series of earthquakes occurred roughly 120 km from the 40 km $\times$ 20 km experimental area. A strong signal from these earthquakes was received by near-bottom vector sensors (the intensity vector acoustic recorders, IVARs) and high-bandwidth ocean-bottom seismometers (OBXs) embedded in the thick muddy sediments characteristic of the area. The events are roughly 30 seconds in duration, and are low-frequency (from 3 to 40 Hz). As such, energy is observed well below the acoustic cut-off frequency of the continental shelf. Correlating the arrivals to the USGS epicenter show these arrivals are supersonic in water, with group velocity $>$5 km/s. These arrivals appear to be leaky waves propagating within the rock below the ocean seafloor, distinct from the well-known T-phase of oceanic earthquakes that propagate at speeds close to the sound speed in seawater. The frequency dependence of the potential and kinetic energy of these arrivals suggest the field is evanescent, in both the water column and upper sediments. Spectral properties of the arrivals are shown to correspond to the water depth and sediment layering.

# Session 3pPAb

## Physical Acoustics and Signal Processing in Acoustics: Particle Velocity Sensing and Associated Signal Processing

W. C. K. Alberts, Cochair
*CCDC-Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783*

Ning Xiang, Cochair
*School of Architecture, Rensselaer Polytechnic Institute, 110 Eighth Street, Troy, NY 12180*

**Chair's Introduction—1:00**

### *Invited Papers*

**1:05**

**3pPAb1. Assessment of road surface state with acoustic vector sensor.** Jozef Kotus, Grzegorz Szwoch (The Faculty of Electronics, Telecommunications and Informatics, Multimedia Systems, Gdansk Univ. of Technol., Gdansk, pomorskie, Poland), Andrzej Czyzewski (Multimedia Systems, Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, andczyz@gmail.com), and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Gdansk, Poland)

A method of determining the road surface state based on the sound intensity analysis is presented. The proposed method is designed for the passive, non-contact assessment of a road surface state, especially in dry/wet conditions. The proposed method is intended for monitoring stations utilizing low-cost hardware. The road surface state is determined from the analysis of sound intensity emitted by road vehicles passing by the sensor and recorded with an acoustic vector sensor (AVS). A frequency domain sound intensity analysis included spatial filtering to reduce the environmental interference using the designed amplitude and phase correction algorithms. A test installation in a real-world scenario, consisting of a small AVS constructed from MEMS microphones and a state-of-art optic-based sensor, was used to evaluate the proposed method. A dataset representing many road vehicles moving at different speeds through the observed road section in varying conditions (dry and wet surface) was collected. Compared with the reference data, an evaluation of the proposed method and its accuracy in determining the road surface state is presented and discussed.The project has been subsidized by the Polish National Centre for Research and Development (NCBR) from the European Regional Development Fund No. POIR.04.01.04/2019 entitled: INFOLIGHT—"Cloud-based lighting system for smart cities."

**1:25**

**3pPAb2. Measuring the effect of ground impedance on the vector field, both in air and underwater.** David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu), Peter H. Dahl (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA), and Jim Waite (AIVS Inc., East Sound, WA)

One direct advantage to characterizing the acoustic field with a vector sensor is the ability to measure both components of acoustic energy, potential (pressure) and kinetic (particle velocity). While it is well known that the kinetic energy exceeds potential in the near-field of a source, equivalent and opposite imbalances are prevalent in the far-field due to propagation effects. For example, at normal incidence the field exhibits a distinct frequency dependent relationship that depends explicitly on the impedance of the ground reflector, including any internal reflections from sediment layers. In this paper, we examine the measured vector field, both in-air and underwater, with a specific focus the energy balance when a source is directly overhead. While the construction of a neutrally buoyant volume for use in dense media (such as water) is fairly straightforward, the application to airborne acoustics requires extremely lightweight materials. Airborne measurements are provided by an Accelerometer-based Intensity Vector Sensor (AIVS) system, which is constructed from a lightweight MEMS accelerometer and microphone embedded in a spherical, expanded polystyrene (EPS) foam volume.

**1:45**

**3pPAb3. Methods for accurate acoustic characterization with ultra-low noise and minimal effect from reflection wave.** Junpeng Lai (Mech. Eng., Binghamton Univ., 85 Murray Hill Rd., Eng. and Sci. Bldg., Vestal, NY 13850, jlai16@binghamton.edu), Morteza Karimi (Mech. Eng., Binghamton Univ., Vestal, NY), and Ronald Miles (Mech. Eng., Binghamton Univ., Binghamton, NY)

It is common to measure the response of devices and structures to sound due to an imposed sound source. Unfortunately, acoustic reflections from walls and/or instruments often contaminate the results. In this presentation, methods of acoustic characterization are described to minimize the influence of acoustic reflections. It is shown that this process results in clean and smooth data. A simple time-domain window is implemented for diminishing the contribution from reflection waves. Moreover, a single frequency curve fitting approach is employed for better parameter identification and noise reduction, compared to traditional fast Fourier Transform analysis. Results obtained from a theoretical acoustic model with a reflection source are compared with measured results. Different cases of data acquisition with time and frequency analysis are experimentally demonstrated and validated. All experimental measurements are performed in an anechoic chamber. Results show that the approach presented here significantly reduces noise and also the influences of reflection waves in experimental data acquisition outcomes.

**2:00–2:15 Break**

**2:15**

**3pPAb4. Infrasonic horn and ribbon microphone.** Thomas G. Muir (Appl. Res. Labs, Univ. of Texas, P.O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu) and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Infrasound measurements normally utilize seismic sensors (geophones and accelerometers) with deployment on the ground surface, the seafloor, ice packs, etc. Here, the infrasound propagating in an overlying fluid medium such as air or water, interacts with the solid medium, thereby imparting its signature. These sensors primarily detect ground motion in terms of the particle velocity or acceleration (force) of the ground in contact with the sensor. Pressure microphone sensing of infrasound in the atmosphere has also been explored. We examine here the atmospheric sensing of infrasound, using the well-established technique of the ribbon microphone, which employs a thin metallic ribbon, suspended between two magnets to detect the particle velocity of the sound propagating in a fluid medium. Movement of the ribbon in the sound field generates a voltage according to Faraday's Law. Although ribbon microphones are fixtures in audio band acoustics, their application to infrasound is not, and is explored here. Ribbon microphones made from suspension of conductive mylar tape between two neodymium magnets have been fabricated and tested in an infrasound band around a few Hertz. The results show a viable approach to vector sensing in this band. Work supported by ARL:UT Austin.

**2:30**

**3pPAb5. Holographic signal processing for estimation of sound source direction by a vector receiver in shallow water.** Sergey A. Pereselkov (Mathematical Phys. and Information Technol. Dept., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Hydrophysics Lab., Prokhorov General Phys. Inst. of the Russian Acad. of Sci., Moscow, Russian Federation), Ilya Kaznacheev, Sergey Tkachenko, and Pavel Rybyanets (Mathematical Phys. and Information Technol. Dept., Voronezh State Univ., Voronezh / Russia, Russian Federation)

A holographic signal processing method for estimation of sound source direction of broadband moving sound source is proposed in the paper. The holographic estimation of sound source direction is based on signal processing of single vector receiver (VR) channels: x-th and y-th components of the oscillatory velocity. The sound field of a moving sound source creates a stable interference pattern of the intensity distribution (interferogram) in the frequency-time domain on VR channels. The two-dimensional Fourier transformation (2D-FT) is applied to analyze the channels interferograms. The result of the 2D-FT are holograms of the x-th and y-th oscillatory velocity components. The angle distributions of channels holograms are calculated. The offered holographic method is based on the ratio of the absolute values of the holograms angle distributions maxima. The expressions for VR channels holograms are derived. The results of the numerical experiment for source direction reconstruction are presented. A significant advantage of the proposed holographic method that there is no need for information about noise signal source, the additive noise obstacle and waveguide transfer function. [This study was supported by Russian Science Foundation grant no. 22-79-10233. Tkachenko's numerical experiment was supported by grant no. MK-4846.2022.4.]

**2:45**

**3pPAb6. Standing wave tube for the calibration of acoustic vector sensors.** James S. Martin (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Love Bldg Rm 130, Atlanta, GA 30332-0405, james.martin@me.gatech.edu), Andrew Lawrence, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Acoustic vector sensors are useful in a variety of applications as they provide directional information for ambient noise by sensing the acoustic particle motion in addition to the acoustic pressure. Traditionally, the motion sensor is an accelerometer that is calibrated separately from the associated hydrophone. There are certain advantages that can be realized by calibrating the accelerometer and hydrophone concurrently in a standing wave tube at low frequencies. First, since the pressure and acceleration are simply related in this environment both sensors can be calibrated against a single reference hydrophone. Second, the vector sensor can be calibrated along with the suspension that will be used to mount it thereby taking into account any effects of the suspension's dynamics. Third, the vector sensor is calibrated to the fluid acceleration rather than its own acceleration thereby accounting for any mismatch in its density relative to the fluid as well as the suspension dynamics. And fourth, the phase response of the acceleration output relative to the pressure output is measured in the calibration. This technique was implemented using a travelling wave tube that was developed for other applications and found to be accurate and convenient.

**3:00**

**3pPAb7. Vector acoustic properties of the underwater sound field from multiple, simultaneously observed ships.** Hanah A. Choice (Underwater Acoust., Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, hchoice@uw.edu), David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA), and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Vector acoustic properties of the underwater sound field simultaneously observed from multiple surface vessels during the recently completed (May–June 2022) experiment on the New England Mud Patch (NEMP) approximately 100 km south of Cape Cod are discussed. The properties, such as bearing, degree of polarization, and other non-dimensional parameters are derived from a 4 by 4 field matrix composed of pressure and 3-component velocity, and a $3 \times 3$ matrix composed of only velocity components. The measurements were made with two sensing systems, separated by 21 km, referred to as an Intensity Vector Autonomous Recorder (IVAR), and identified as IVAR-1 and IVAR-2. Individual ship CPAs tracks are identified with automated ship identification (AIS) data. In this presentation, the two sources of measured vector acoustic field data combined with the AIS data are exploited to study the effect of interaction of multiple acoustic sources on a received vector field.

## Session 3pPP

# Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Poster Session I

Virginia Best, Chair

*Speech, Language and Hearing Sciences, Boston University, 635 Commonwealth Ave., Boston, MA 02215*

All posters will be on display from 1:00 p.m. to 3:00 p.m. Authors of odd numbered papers will be at their posters from 1:00 p.m. to 2:00 p.m. and authors of even-numbered papers will be at their posters from 2:00 p.m. to 3:00 p.m.

### *Contributed Papers*

**3pPP1. A noninvasive auditory prosthesis: Evaluation of a safety margin of the transtympanic laser stimulation.** Aya Okamoto (Doshisha Univ., 1-3 Tataratuya, Kyotanabe Campus, The Ishinkan Bldg. GF, Kyotanabe-shi, Japan, ctug1029@mail4.doshisha.ac.jp), Miku Uenaka, Kohta I. Kobayasi (Doshisha Univ., Kyotanabe, Kyoto, Japan), and Yuta Tamai (Eberhard Karl Univ. of Tübingen, Tübingen, Germany)

Cochlear implants are widely used for compensating the sensorineural hearing loss. Surgical intervention for implanting electrodes is one of the most significant obstacle for the further widespread of these devices. We proposed a non-invasive auditory prosthesis using infrared laser because the laser can stimulate cochlea nerves from the outer ear. The purpose of this study was to evaluate the safety margin of the transtympanic laser stimulation. Head-fixed classical conditioning was performed on Mongolian gerbils. Subjects were trained using auditory stimulus, a bandpass noise, as a conditioned stimulus for a water reward, and licking behavior was recorded as a conditioned response. After the training, each subject was exposed to continuous pulsed laser irradiation of 1.6, 3.3, 6.6, 26.4, 52.8, or 105.6 W/cm$^2$ for 15 hours. The bandpass noise with various intensities was presented without the reward before and after one-day of the laser exposure. As a result, the licking rate did not change after laser exposure of 6.6 W/cm$^2$ or weaker, but drastically decreased after 26.4 W/cm$^2$or above. The result suggests that the injury threshold in Mongolian gerbils for transtympanic laser stimulation is between 6.6 and 26.4 W/cm$^2$ and the exposure over 6.6 W/cm$^2$ could be out of the safety margin.

**3pPP2. An analysis of "speech glimpses" in realistic environments.** Virginia Best (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu), Jorg M. Buchholz (Dept. of Linguist, Macquarie Univ., Sydney, New South Wales, Australia), S. Theo Goverts (Amsterdam UMC, Vrije Universiteit Amsterdam, Amsterdam, the Netherlands), and H. Steven Colburn (Dept. of Biomedical Eng., Boston Univ., Boston, MA)

A great deal of effort is currently going into recreating everyday acoustic environments in laboratories and clinics, with the goal of obtaining more relevant measurements of real-world listening abilities and intervention benefits. Related work is focused on generating naturalistic speech stimuli that capture important features of real conversations, and on estimating real-world signal-to-noise ratios. Here we make use of a framework that brings together all of these approaches to arrive at highly realistic speech-in-noise stimuli. Using ideal time-frequency segregation, we characterized the "speech glimpses" that are available in these real-world stimuli. By speech glimpses, we refer to distributed time-frequency regions in which the speech of interest dominates the acoustic mixture. One goal was to compare the glimpses that are available in highly realistic stimuli to those available in simpler, commonly used, laboratory stimuli. A second goal was to analyze these glimpses in detail, to provide a new perspective on the many sources of disruption that may hinder the understanding of conversational speech. These include masking (which reduces the number of available glimpses) and reverberation (which reduces the quality of glimpses), as well as interactions between sounds that cause distortions of the spatial information present in the target glimpses.

**3pPP3. Estimating the contribution of central noise from composite performance across multiple tasks.** Jonathan H. Venezia (VA Loma Linda Healthcare System, 11201 Benton St., Loma Linda, CA 92357, Jonathan.Venezia@va.gov), Nicole Whittle (Portland VA Healthcare System, Loma Linda, CA), Marjorie R. Leek (Res., Loma Linda VA Healthcare System, Loma Linda, CA), and Christian Herrera Ortiz (VA Loma Linda Healthcare System, Loma Linda, CA)

According to signal detection theory, the ability to detect a signal is limited only by internal noise, which comprises peripheral and central sources. Here, we develop a statistical approach to parse central from peripheral noise. Fifty-two Veterans (mean age = 47.8, range = 30–60) with normal or near-normal hearing performed AXB discrimination for several temporal processing tasks: gap duration discrimination, forward masking, frequency modulation detection, and interaural phase modulation detection. After training, a single adaptive run (40 reversals) was completed for each task. Subjects also completed speech-in-noise testing ("Theo-Victor-Michael) with four masker types (48 trials ea.): speech-shaped noise, speech-envelope modulated noise, one and two competing talkers. Composite speech performance was estimated using principal component analysis. Bayesian hierarchical regression was used to estimate two-parameter psychometric functions (threshold, slope) simultaneously for all temporal tasks and subjects. Crucially, fixed (group-level) thresholds were estimated per task but only a single random (subject-level) intercept was estimated (mean across-task deviation from the group thresholds). We assume central noise is the primary factor limiting across-task performance. The principal speech scores were entered as regressors on this "central threshold." Indeed, central threshold was correlated with the principal speech scores, suggesting that central noise limits temporal processing and speech-in-noise.

**3pPP4. Effect of electric-acoustic cochlear implant stimulation and coding strategies on spatial cues of speech signals in reverberant room.** Muhammad A. Asyraf (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya 60111, Indonesia, 6009211006@mhs.its.ac.id) and Dhany Arifianto (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia)

The comparison of spatial cues changes in different setups and coding strategies used in cochlear implants (CI) is investigated. In this experiment, we implement three voice coder setups, such as bilateral CI, bimodal CI, and electro-acoustic stimulation (EAS). Two well-known coding strategies are used, which are continuous interleaved sampling (CIS) and spectral peak (SPEAK). Speech signals are convoluted with appropriate binaural room impulse response (BRIR), creating reverberant spatial stimuli. Five different reverberant conditions (including anechoic) were applied to the

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stimuli. Interaural level and time differences (ILD and ITD) are evaluated objectively and subjectively, and their relationship with the intelligibility of speech is observed. Prior objective evaluation with CIS reveals that clarity (C50) becomes a more important factor in spatial cue change than reverberation time. Vocoded conditions (bilateral CI) show an increment in ILD value (compression has not been implemented yet on the vocoder processing), when the value of ITD gets more different (decreased) from the middle point. Reverberation degrades the intelligibility rate at various rates depending on the C50 value, both in unvocoded and vocoded conditions. In the vocoded condition, decrement on spatial cues was also followed by the decreement on the intelligibility of spatial stimuli.

**3pPP5. Skin conductance during remote collaborative communication under noisy, reverberant and delayed communication transmission conditions.** Vishakha Rawool (Commun. Sci. & Disord., Univ. of MS, 1006 Briarwood Dr., Oxford, MS 38655, vishakharawool8@gmail.com)

This study, which is part of a larger study, was designed to monitor skin conductance during remote collaborative communication. Ninety (45 men and 45 women) participants were divided into 30 teams with three members in each team. Each team built different LEGO models under three communication transmission delay conditions in the following order: No delay, 750 ms delay and 5000 ms delay. One individual served as the LEGO builder, who did not have the instruction manual. The remaining two team members provided instructions for building the models. Participants were in separate rooms and communicated via earphones. Background white noise (50 dB SPL) was introduced in the ear canals and reverberation was applied to the transmitted communication. Participants were fitted with Empatica E4 wristbands. The ANOVA revealed that the Sum of non-specific skin conductance response (NS-SCR) amplitude/minute was significantly reduced in the 5000 ms delay condition compared to the No delay and 750 ms delay conditions with a medium effect size. These results suggest that participants tend to adapt to the task yielding reduced stress in the last delay condition. Acknowledgement: Graduate students Sydney Osbarn and Melissa Reyes assisted in data collection. Sydney Osbarn also assisted in pre-processing the data.

**3pPP6. Cocktail party effect in team sports: A perceptual advantage.** Reethee Antony (Misericordia Univ., 301 Lake St., Dallas, PA 18612, rantony@misericordia.edu), Isabella Fredo, Emma Schaedler, Meghan Dunne, Erica Scheinberg, and Stephanie Fazio (Misericordia Univ., Dallas, PA)

This study focuses on possible perceptual advantage in speech perception in team sports players. The overall aim was to compare the speech perception in team sports persons versus non-team sports persons, in quiet and in noise. Twenty adults including team sports players and non-team sports players performed speech discrimination task and a speech identification task. Natural digitized speech stimuli including /a/-/a/, /a/-/s/, /s/-/a/, /s/-/s/ were presented binaurally at comfortable loudness level in two conditions—in quiet and in background noise. For noise, talker babble was used at a signal-to noise ratio (SNR) of 0. Percent correct scores and reaction times were measured. Statistical analyses included a mixed model ANOVA. Results were considered significant when pi$<$0.05. Significant group differences were present between team sports persons versus non-team sports persons specifically for the speech identification task. The team sports persons had higher speech identification scores and faster reaction times relative to non-team sports persons, specifically in noise ($p < 0.05$), thus illustrating better auditory separation of signal from noise. The findings from this research serve as a frontier to many studies in expanding our knowledge about speech perception and application of auditory training in team sports players.

**3pPP7. Categorical perception of Mandarin tones in neurotypical adults with subclinical autistic traits.** Yu Li (Dept. of Life Sci., BNU-HKBU United Int. College, 2000 Jintong Rd., Tangjiawan, Zhuhai, Guangdong 519087, China, yuli@uic.edu.cn), Sabrina Y. Jiang (Macau Univ. of Sci. and Technol., Macau, China), Yuanyuan Wang (Qufu Normal Univ., Jining, Shandong, China), and Linjun Zhang (Peking Univ., Beijing, China)

Early studies have revealed that categorical speech perception is impaired in individuals with autism spectrum disorders. However, it remains unclear whether categorical speech perception ability is associated with subclinical autistic traits in neurotypical populations. The current study sought to address this question by examining categorical perception of Mandarin tones (T1, high-level; T4, falling) in young adults. One hundred and thirty-one college students were recruited to participate in two classic categorical perception task, identification and discrimination, and to fill in the Autism-Spectrum Quotient (AQ; Baron-Cohen *et al*., 2001) for measuring subclinical autistic traits. The results showed that identification slope and categorical boundary width were not significantly correlated with the overall AQ score and the scores of the five subscales. However, between-category discrimination accuracy was significantly correlated with the social subscale score, indicating that individuals with less social skills have heightened between-category discrimination ability. Possible theoretical and practical implications of this association in subclinical populations are discussed. The current study adds to a better understanding of the associations between subclinical autistic traits and categorical speech perception in neurotypical individuals. [Work supported by the Humanities and Social Sciences Foundation of Ministry of Education of China 20YJCZH079.]

**3pPP8. Medial olivocochlear reflex strength in ears with low-to-moderate annual noise exposure.** Donguk Lee (Audiol. and Speech Pathol., The Univ. of Tennessee Health Sci. Ctr., 1303 white Ave., 13, Knoxville, TN 37916, dlee88@uthsc.edu), Morgaine Goettl-Meyer (Dept. of Physiol. and Biophys., Univ. of Colorado Anshutz Medical Campus, Aurora, CO), and James D. Lewis (Audiol. and Speech Pathol., The Univ. of Tennessee Health Sci. Ctr., Knoxville, TN)

Recent work suggests enhanced medial-olivocochlear reflex (MOCR) strength in noise exposed individuals. The current study aimed to replicate this finding using stricter signal-to-noise ratio (SNR) criteria in MOCR estimates and a larger subject population. Ninety-eight normal-hearing young adults participated. The MOCR test included measurement of click-evoked otoacoustic emissions (CEOAE) with and without contralateral noise. MOCR-induced magnitude and phase shifts were calculated. The MOCR test was performed twice to provide a within-session replication. Annual noise exposure history was estimated using the Noise Exposure Questionnaire. Annual noise exposure was not a statistically significant predictor of the MOCR-induced CEOAE magnitude shift. A statistically significant negative correlation was found between annual noise exposure and the MOCR-induced CEOAE phase shift. A significant negative correlation was also identified between annual noise exposure and CEOAE level. Results suggest diminished MOCR strength in noise-exposed individuals. This finding contrasts with recent work using similar methods. Compared to earlier work, the current study relied on higher OAE SNRs in MOCR estimates and sampled a larger population of subjects with broader noise exposure histories. The concurrent finding of reduced OAE levels in noise exposed individuals suggests acoustic trauma may be an underlying cause of reduced MOCR strength.

**3pPP9. Individual factors that impact noise tolerance.** Kristi Oeding (Univ. of Minnesota - Twin Cities/Minnesota State Univ. - Mankato, 314 Clinical Sci. Bldg., 150 South Rd., Mankato, MN 56001, kristi.oeding@mnsu.edu), Evelyn Davies-Venn (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota - Twin Cities, Minneapolis, MN), and Peggy Nelson (Ctr for Applied/Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN)

One of the most common complaints of patients wearing hearing aids is difficulty hearing speech in noise. Limited information is available on how to program speech in noise programs as fitting formulas are based on quiet environments. In order to understand what settings could optimize programs in background noise, it is important to determine what objective and subjective factors impact speech understanding in noise. This study sought to evaluate what individual factors (personality, working memory, noise tolerance, and speech in noise abilities) might impact person's tolerance in background noise. Data was collected remotely using questionnaires and equipment that was dropped off at the participant's home due to covid restrictions at the time. A principal component analysis was used to determine clusters of patients based on the examined factors. The goal of finding group factors will be used to determine the best settings to increase a hearing aid user's tolerance for background noise so they can wear their hearing aids in these environments and hear speech better. Benefits and disadvantages of at home testing will also be discussed.

**3pPP10. Middle-ear muscle reflex: Sensitivity and reliability of thresholds and amplitude-intensity functions in normal-hearing adults across measurement systems.** Sarah L. Bochat (Commun. Sci. & Disord., Univ. of South Florida, 4202 E Fowler Ave., Tampa, FL 33620, bochat@usf.edu), Nathan C. Higgins (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL), Shawn S. Goodman (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), David A. Eddins, and Ann C. Eddins (Commun. Sci. & Disord., Univ. of South Florida, Tampa, FL)

Standard clinical middle-ear muscle reflexes (MEMRs) are typically measured as a change in admittance for a single, 226-Hz probe tone in the presence of an eliciting stimulus. Alternatively, wideband MEMRs are measured as a change in reflectance for a wideband click probe in the presence of an elicitor. This study evaluated test-retest measures of MEMR thresholds and amplitude-intensity (A-I) functions across four measurement systems based on differences in sensitivity (i.e., thresholds, A-I slopes) and reliability. Ten normal-hearing adults participated in two sessions. MEMR thresholds and A-I functions were measured using two clinical systems (GSI, Titan) with two elicitors (2 kHz, broadband noise-BBN), and two wideband systems (Titan Research module, Custom Etymotic-based system). The 2-kHz elicitor yielded the highest thresholds overall. BBN elicitors produced similar thresholds across clinical and wideband systems. The broadband elicitor with the clinical systems produced the steepest A-I slopes, and the slopes were steeper overall for clinical versus wideband systems. In most conditions, no significant test-retest differences were observed for thresholds or AI-functions, but intraclass correlations were highest for the clinical system measures. MEMR measures made with current clinical systems using BBN elicitors may optimize sensitivity, reliability, and overall diagnostic value of such measures.

**3pPP11. Exploring the need for language-specific hearing aid signal processing.** Abhijit Roy (Commun. Sci. and Disord., Northwestern Univ., 124 Callan Ave., Apt. 2B, Evanston, IL 60202, abhijitroy2025@u.northwestern.edu), Pamela E. Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), and Ann Bradlow (Linguist, Northwestern Univ., Evanston, IL)

Speech phonemes in higher frequencies have varying acoustic characteristics in different languages. For example, /s/ and /ʃ/, fricatives commonly used in English, have spectral peaks at higher frequencies than /ʂ/ and /x/, fricatives commonly used in Mandarin. Data on the relationship between the acoustic characteristics of a listener's native language and their hearing aid signal prescription requirements could provide important clinical guidance. Non-linear frequency compression is a digital signal processing tool that compresses auditory information above a pre-determined cutoff frequency. In hearing aids, this compression is applied with the intention of making higher-frequency speech sounds more audible to listeners with high-frequency hearing loss. Here, we studied the effect of different frequency compression settings on fricative perception between Mandarin and English listeners. Normal-hearing participants between age 18 to 50 years were presented with fricative identification and fricative discrimination tasks under various frequency compression settings. Participants were grouped between those with exclusive English language background birth to 6 years to those with exclusive Mandarin language background during the same period. Results display different responses for presented phonemes and frequency compression settings, suggesting that language specificity could be considered for hearing aid signal prescription.

**3pPP12. Envelope following responses following partial cochlear deafferentation in guinea pigs.** JoAnn McGee (VA Loma Linda Healthcare System, VA Loma Linda Healthcare System, Loma Linda, CA 92357, mcgeej@umn.edu), Xiaohui Lin, Ashley Vazquez, Hongzhe Li, Jonathan H. Venezia, Marjorie R. Leek, and Edward J. Walsh (VA Loma Linda Healthcare System, Loma Linda, CA)

The sound-induced loss of ribbon synapses connecting inner hair cells to auditory nerve fibers primarily exhibiting high thresholds has been the subject of numerous investigations. The condition resulting from this inner ear abnormality is commonly referred to as hidden hearing loss, a name that reflects the relative invulnerability of low threshold auditory nerve fibers that survive noise-exposure and function normally. Although auditory sensitivity recovers completely among animals experiencing hidden hearing loss, recovery of auditory brainstem response amplitudes is incomplete. The residual loss of function reflected in diminished response amplitudes to transient stimuli serves as a highly reliable indicator of the condition. The extent to which responses to sustained stimuli might serve as indicators of pathology is less clear. To that end, we will review findings related to differences in spectral magnitudes of envelope following responses to a battery of sinusoidally-amplitude modulated test conditions that include level dependent response growth, modulation transfer functions under a variety of carrier conditions, the influence of varying modulation depths, as well as the influence of maskers on responses acquired from control and noise-exposed animals. [Work supported by the Department of Defense Award #W81XWH-19-1-0862.]

**3pPP13. Bilateral cochlear implants or bimodal hearing: A comparison of quality of life.** Jessica Lewis (Otolaryngol., The Ohio State Univ., 4811 Leap Court, Hilliard, OH 43026, lewis.1792@osu.edu), Irina Castellanos, and Aaron C. Moberly (Otolaryngol., The Ohio State Univ., Columbus, OH)

Current literature suggests that bimodal users (i.e., individuals using a cochlear implant—CI—and a contralateral hearing aid) may report better outcomes in some listening situations (e.g., pitch perception and music appreciation) than bilateral CI users; however, several studies report conflicting results. Currently, there is no widely accepted criterion for bilateral simultaneous or sequential CI candidacy in adults, and clinicians typically rely on the patient's experience to determine whether bilateral or bimodal hearing is best for that individual. Therefore, in this study we compare responses from twenty-four bilateral and bimodal users on the Cochlear Implant Quality of Life (CIQOL) questionnaire. Results demonstrated that bilateral CI users report better scores for the emotional, environmental, social, listening effort, and global scales than bimodal users. No significant difference in age, speech recognition abilities, and duration of hearing loss between the two groups were observed; however, the two groups did differ on their duration of CI use, such that the bimodal users had less experience listening with their CIs than the bilateral CI users. These findings do not replicate previous in-lab results, which demonstrate a bimodal user advantage and stresses the need to further explore the self-reported benefits of bimodal versus bilateral CI hearing.

**3pPP14. The possible range of interaural cross correlation after cochlear implant processing.** Prajna BK (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S 6th St., Champaign, IL 61820, prajnab2@illinois.edu), Pasquale Bottalico, and Justin M. Aronoff (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Normal hearing listeners rely on their binaural auditory system to combine signals from both ears into single auditory objects, which aids source segregation and sound localization. The ability to combine signals from the two ears is influenced by interaural symmetry - the similarity of the place of stimulation along the left and right cochlea—and the interaural cross-correlation (ICC) of the signal, calculated as the peak correlation between the signal for the two ears across unilateral temporal offsets. The ability of bilateral cochlear implant (CI) users to combine signals from the two ears can be worse than that of NH listeners. This is partly attributed to decreases in interaural symmetry related to factors such as differences in electrode array insertion depth. A secondary effect of interaural asymmetry is that it

can result in reduced ICC at interaurally symmetric locations, mediated by aspects of CI processing. This study aims to investigate how CI processing and interaural asymmetry affect the ICC available to CI users for interaurally symmetric locations, using electrodograms recorded from a CI processor placed on a Knowles Electronics Manikin for Acoustic Research (KEMAR). Inferences on how ICC is altered with interaural asymmetry for CI users will be discussed.

**3pPP15. Differential relations of auditory and cognitive functions to speech recognition tasks in adult cochlear implant users.** Aaron C. Moberly (Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, Aaron.Moberly@osumc.edu) and Terrin N. Tamati (Otolaryngol., The Ohio State Univ., Columbus, OH)

Clinical testing for assessing adult cochlear implant (CI) candidacy and outcomes relies primarily on tasks of open-set recognition for words and sentences. However, it is not clear in what ways performance on these different tasks relates differentially to auditory and cognitive functions. We tested the hypothesis that auditory and cognitive functions would contribute differentially to performance on recognition of isolated words (CID-W22 words), meaningful sentences (Harvard Standard), nonsense sentences (Harvard Anomalous), and sentences with high talker variability (PRESTO) in adult CI users. Sixty-five experienced CI users completed a battery of auditory and cognitive tests to assess auditory spectro-temporal processing, vocabulary size, working memory capacity, inhibition-concentration, nonverbal reasoning, and speed of lexical access. Findings were that spectro-temporal processing, lexical access speed, and nonverbal reasoning predicted scores across all word and sentence recognition tests. In contrast, inhibition-concentration contributed to nonsense sentence recognition, while vocabulary knowledge contributed to high talker variability sentence recognition. Results suggest that there is a core set of auditory and cognitive functions that contribute broadly to speech recognition, while more specific cognitive functions contribute to performance on some speech recognition tasks. Clinical implications are that individual cognitive abilities should be considered when evaluating CI candidacy and outcomes.

**3pPP16. Evaluation omn three-forced choice audiometry for hearing threshold measurement.** Naomi Ashilah (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya 60111, Indonesia, naomiashilah1@gmail.com), Yuniar Syahadhatin, Ainun Nadiroh (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia), Nyilo Purnami (School of Medicine, Airlangga Univ., Surabaya, Indonesia), and Dhany Arifianto (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia)

Common audiometry used in hospitals uses the 2AFC (Two Force Choice) method, which has a large and predictable bias. In this research, the three-force choice (3AFC) method is proposed for a smaller bias to measure hearing threshold. A hearing test was conducted on 50 participants. Three kinds of audiometric tests are used, including conventional, Pychoacoustic, and portable audiometry. A validation test was carried out by comparing the results of Pychoacoustic and portable audiometric tests using the 3AFC method with the golden standard (conventional audiometry). Pychoacoustic audiometry is unable to display the hearing threshold value in accordance with that shown by golden standard audiometry (conventional audiometry), where the mean value is 18.0–43.0, and standard deviation is 7–12 . However, the portable audiometry test is able to display a threshold value that is close to the golden standard audiometric results, with a mean value of 11—28, and standard deviation value of 3—6 . Based on the statistical calculation of the Wilcoxon test and Bonferroni's correction, it can be concluded that data collection using Pychoacoustic outdoors and Portable in the relevant room has performance that is parallel to the golden standard test (sig value >0.05).

# Session 3pSC

## Speech Communication: Topics in Speech Production

Matthew Masapollo, Chair
*University of Florida, 1225 Center Dr., Gainesville, FL 32610*

### *Contributed Papers*

**1:00**

**3pSC1. Inter-articulator coordination in speech production: Timing is of the essence.** Matthew Masapollo (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Gainesville, FL 32610, mmasapollo@phhp.ufl.edu) and Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

In skilled speech production, sets of vocal-tract articulators work cooperatively to achieve task-specific goals, in spite of contextual variation. Efforts to understand these functional units have focused on identifying control variables responsible for allowing articulators to achieve these goals, with some research focusing on temporal variables (relative timing of movements) and other research focusing on spatiotemporal variables (phase angle of movement onset for one articulator, relative to another). Here, both types of variables were examined. Ten talkers recorded /tV#Cat/ utterances using electromagnetic articulography, with alternative V (/ɑ/-/ɛ) and C (/t/-/d/), across variation in rate (fast-slow) and stress (first syllable stressed-unstressed). Two measures were obtained: (1) timing of tongue-tip raising onset for medial C, relative to jaw opening-closing; (2) angle of tongue-tip raising onset, relative to the jaw phase plane. Results showed that any manipulation that shortened the jaw opening-closing cycle reduced both the relative timing and phase angle of the tongue-tip movement onset, but relative timing of tongue-tip movement onset scaled more consistently with jaw opening-closing across rate and stress variation. This finding supports the hypothesis that an intrinsic timing mechanism is the control variable for interarticulatory relations, with immediate compensation then allowing these structures to achieve their goals spatially.

**1:15**

**3pSC2. Effects of prosodic structure on the temporal organization of speech and co-speech gestures.** Yoonjeong Lee (Linguist, Univ. of Michigan, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeol@umich.edu), Jelena Krivokapić, and Ruaridh Purse (Linguist, Univ. of Michigan, Ann Arbor, MI)

Although many studies have observed a close relationship between prosodic structure and co-speech gestures, little is understood about cross-modal gestural coordination. The present study examines the relationship between articulatory and co-speech gestures at prosodic boundaries and under prominence, focusing on non-referential manual and eyebrow beat gestures in Korean, a language in which co-speech gestures are virtually unexplored. This study hypothesizes that prosodic structure systematically governs the production of both speech and co-speech gestures and their temporal organization. Multimodal signals of a story reading were collected from eight speakers (5F, 3M). The lips, tongue, and eyebrows were point-tracked using EMA, and the vertical manual movements obtained from a video recording were auto-tracked using a geometrical centroid tracking method. Measurements taken included the duration of intervals from the timepoint of concurrent beat gesture onset and target to 1) consonant gesture onset and target, 2) vowel gesture onset and target, 3) pitch gesture (F0) onset and target, and 4) phrasal boundaries. Results reveal systematic inter-articulator coordination patterns, suggesting that beat gestures co-occurring with speech gestures are recruited to signal information grouping and highlighting. The findings are discussed with reference to the nature of prosodic representation and models of speech planning. [Work supported by NSF.]

**1:30**

**3pSC3. Respiratory and articulatory influences on inhalation noise during speech.** Laura Koenig (Commun. Sci. and Disord., Adelphi Univ., 300 George St., New Haven, NY 06511, koenig@haskins.yale.edu) and Susanne Fuchs (Leibniz Ctr. for General Linguist, Berlin, Germany)

During speech production, individuals inhale more deeply and rapidly compared to non-speaking inspiration. Such inspirations are also more frequently audible than their non-speech counterparts. To what extent do the acoustic characteristics of inhalation noise reflect respiratory contributions versus other articulatory mechanisms? To gain insight into this question, here we relate acoustics, captured using a head-mounted microphone, with kinematic data collected via respiratory inductance plethysmography (RIP) and optopalatography (OPG). The OPG system permits measurement of lip movements (displacement and velocity), tongue-palate contacts, and midsagittal distance when the tongue has no contact with the palate. Custom-made palates, including both contact sensors as well as optical sensors, were obtained for eight typical female speakers of German. Speech was recorded in normal and loud conditions. In this exploratory analysis, we anticipate that (a) louder speech will be accompanied by deeper, more rapid inhalations; and (b) greater articulatory opening during the inhalation phase; (c) the magnitude of lip aperture will be correlated with inspiration-excited first formant values; and (d) both supralaryngeal and respiratory displacements and velocities will be reflected in the inhalation noise characteristics. That is, speech inspiration, like speech itself, will demonstrate regular correspondences between physiological actions and the acoustic output.

**1:45**

**3pSC4. Articulatory response to delayed and real-time feedback based on regional tongue displacements.** Sarah Dugan (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, 406 Hickory St., Dayton, OH 45410, hamilsm@ucmail.uc.edu), Sarah R. Li, Kathryn Eary (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), AnnaKate Spotts (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Nicholas S. Schoenleb, Ben Connolly (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Renee Seward (Design, Univ. of Cincinnati, Cincinnati, OH), Michael A. Riley (Rehabilitation, Exercise, and Nutrition Sci., Univ. of Cincinnati, Cincinnati, OH), T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Speech is one of the most complex motor tasks, due to its rapid timing and necessary precision. The difficulty of measuring articulatory movement in real time has made it difficult to investigate motion-based biofeedback for speech. Previously, we demonstrated the use of an automatic measure of tongue movement accuracy from ultrasound imaging. Using this measure for articulatory biofeedback in a simplified, game-like display may benefit the learning of speech movement patterns. To better understand real-time articulatory biofeedback and improve the design of this display, this study

3p WED. PM

presented articulatory biofeedback for the target word /ɑr/ (“are”) in a game with two conditions for feedback timing (delayed and concurrent, indicating whether the game object started moving after or during speech production) and for difficulty level (easy and hard target width, indicating the articulatory precision necessary for achieving the target). For each participant, two blocks of biofeedback for 20–50 productions were presented (randomizing whether the delayed or concurrent block was presented first) in one collection session, with the difficulty level randomized for each production within each block. Data from nine children with typical speech or residual speech sound disorder were analyzed, showing that response and preference of feedback condition vary among individuals.

**2:00**

**3pSC5. Monitoring laryngeal and respiratory adjustments during voice production using a simulation-based voice production inversion neural network.** Zhaoyan Zhang (Dept. of Head and Neck Surgery, Univ. of California, Los Angeles, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu) and John Kegelmeyer (Dept. of Head and Neck Surgery, Univ. of California, Los Angeles, CA)

Previously we reported a simulation-based neural network for estimating vocal fold properties and subglottal pressure from the produced voice. In this study, we evaluate the feasibility of using this neural network to monitor laryngeal and respiratory adjustments during speech production in individual speakers. Acoustic and aerodynamic data were collected in human subjects while producing utterances of five repetitions of the syllable /pa/ at different loudness levels. Voice features were then extracted and used as input to the neural network to estimate changes in vocal physiology and subglottal pressure. The results showed all subjects increased the subglottal pressure and vocal fold adduction when producing a louder voice, although the degrees of laryngeal and respiratory adjustments were speaker-specific. The neural network also estimated on average shorter and thinner vocal folds in female subjects than male subjects. These results demonstrate the potential of this neural network toward monitoring and identifying potentially unhealthy vocal behaviors.

WEDNESDAY AFTERNOON, 7 DECEMBER 2022 NORTH COAST B, 1:00 P.M. TO 2:55 P.M.

# Session 3pUW

## Underwater Acoustics, Acoustical Oceanography and Computational Acoustics: Updating Ocean Acoustic Situational Awareness with *In situ* Measurements

Carolyn Binder, Chair

*Defence Research and Development Canada, 9 Grove St, Dartmouth, B2Y3Z7, Canada*

### *Invited Papers*

**1:00**

**3pUW1. Ship noise radiation characteristics observed from an Arctic acoustic array.** Aaron Webstey (DRDC, Halifax, NS, Canada), Dugald Thomson (Dept of Oceanogr., Dalhousie Univ., Rm. 3635 - 1355 Oxford St., Halifax, NS B3H 4R2, Canada, dugald@dal.ca), David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada), and Jessica Topple (DRDC, Halifax, NS, Canada)

A 48-hydrophone bottom-mounted array in the Arctic provides an opportunity to test a novel acoustic frequency tracker on ships of opportunity. The Arctic is is currently experiencing an historic increase in shipping traffic, resulting in an acoustic environmental that is changing rapidly. Through the ice-free study period, ship passes were observed in the multi-channel acoustic recordings, providing an opportunity to compare spectral fluctuations, directionality, and correlation with ship tracks.

**1:20**

**3pUW2. Evaluating the impact of intermediate and deep sound speed uncertainty in the Irminger Sea on acoustic situational awareness.** EeShan Bhatt (Woods Hole Oceanographic Inst., 77 Massachusetts Ave., 5-223, Cambridge, MA 02139, ebhatt@whoi.edu), Robert A. Weller (Woods Hole Oceanographic Inst., Woods Hole, MA), and Arthur B. Baggeroer (Massachusetts Inst. of Technol., Cambridge, MA)

High-resolution observations from moorings and transect data can provide new insights into ocean structures, their variability, and the downstream effects on acoustic operations. Previously, we established a framework to maintain optimal communication and navigation for small-scale, under-ice operations that focused on spatio-temporal variability in the upper water column and around an autonomous underwater vehicle. However, mid- and deep-water sound speed fidelity are crucial for larger operations that exploit convergence

zone ranging. The Irminger Sea is a useful case study for understanding how sound speed uncertainty, whether from partial data or models, obscures the informational content from acoustics. Generally, the density gradient dominates the sound speed at depth. In the Irminger Sea, hydrographic variability comes from the advection of different water masses through the region at all depths and also from water mass modification by local air-sea interaction. Instead of a uniform gradient as a result of hydrostatic loading, a shifted gradient in the mid-water column and a negative gradient near the bottom are often observed in mooring and transect data. This talk presents initial results for understanding the impacts of the environment through the lens of impact on acoustic propagation. [Work supported by the Office of Naval Research]

## *Contributed Papers*

**1:40**

**3pUW3. Modes selection based on holographic interferometry in presence of intense internal waves in shallow water.** Sergey A. Pereselkov (Mathematical Phys. and Information Technol. Dept., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Hydrophysics Lab., Prokhorov General Phys. Inst. of the Russian Acad. of Sci., Moscow, Russian Federation), Elena Kaznacheeva (M), Sergey Tkachenko, and Pavel Rybyanets (Mathematical Phys. and Information Technol. Dept., Voronezh State Univ., Voronezh/Russia, Russian Federation)

A holographic method is proposed for sound field modes selection and estimating their parameters (amplitude, horizontal wavenumber, group velocity, absorption coefficient) in shallow water by single scalar receiver in presence of intense internal waves (IIW). It is taken into account that IIW leads to horizontal refraction and modes coupling. The offered holographic method is based on the two-dimensional Fourier transform (2D-FT) of the real part of the moving-source sound field in time–frequency domain. It is shown in the paper that different sound field modes create different focal spots in the hologram domain in presence of IIW. The relationship of the mode phases and group velocities with the coordinates of focal-spot peaks is derived. The hologram domain allows to reconstruct the selected mode field and mode parameters in presence of IIW. The results of modes selection within framework of the numerical experiments are presented in the paper. The modal parameters are estimated. Analysis of influence of IIW on mode selection and mode parameters estimations is performed. It is shown that the effects of horizontal refraction and modes coupling do not lead to loss of the identity of mode parameters of the received signal. [This study was supported by Russian Science Foundation grant no. 22-79-10233. E.S. Kaznacheeva's numerical experiment was supported grant no. MK-6144.2021.4]

**1:55**

**3pUW4. Joint ocean physics-acoustics inversion and mutual-information based experiment design in realistic ocean fields.** Wael H. Ali (Mech. Eng., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, whajjali@mit.edu), Chris Mirabito, Patrick J. Haley Jr., and Pierre F. Lermusiaux (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

Reliable acoustic exploration and navigation in the ocean requires precise knowledge of the environmental fields (e.g. ocean physics, bathymetry, seabed) and acoustic parameters (e.g. source location and frequencies). However, such knowledge is typically incomplete due to the sparse and heterogeneous observations, and to the complex high-dimensional multiscale dynamics. In this work, we use our stochastic Dynamically Orthogonal Parabolic Equation (DO-ParEq) framework to: (i) model the ocean environment uncertainties computed from a large realistic ocean ensemble forecast, and (ii) predict the resulting stochastic acoustic fields and their probability distributions. We then employ new nonlinear Bayesian learning algorithms that use Gaussian Mixture Models to assimilate ocean-acoustic measurements and jointly infer the unknown properties of the ocean environment, the bathymetry, the acoustic field and parameters, and even the model parameterizations themselves. We showcase the developed techniques in realistic data-assimilative sea experiments in the New York Bight Region and the Mediterranean Sea. Finally, we demonstrate the impact of selecting measurement locations using a mutual-information-based algorithm that maximizes the information content in the learned fields based on the measurement constraints.

**2:10**

**3pUW5. Detecting hydro-acoustic signals using Distributed Acoustics Sensing technology.** Shima Abadi (School of Oceanogr., Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, abadi@uw.edu), William S. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA), and Brad P. Lipovsky (Dept. of Earth and Space Sci., Univ. of Washington, Seattle, WA)

Distributed Acoustic Sensing (DAS) is a relatively new technology that transforms fiber optic cables, typically used for telecommunications, into dense sensor arrays, capable of meter-scale recordings up to $\sim$100 km. The interest in these technologies for ocean exploration and monitoring has risen in recent years. These systems enable continuous and highly sensitive measurements of both temporal and spatial acoustic data. In this presentation, we use data recorded during a 4-day DAS experiment on the twin cables of the Ocean Observatories Initiative (OOI) Regional Cabled Array (RCA) extending off central Oregon. We demonstrate the capabilities of DAS in recording a wide range of acoustic signals including the 20-Hz call of fin whales, the 15-Hz calls and harmonics of the Northeast Pacific blue whale, and ship noises. We use beamforming and the time difference of arrival (TDOA) algorithm to find the bearing and the location of the signal of interest. We also explain the DAS array response and its sensitivity to paths arriving parallel or perpendicular to the cable and discuss the best practices to overcome the challenges in analyzing this large data set.

**2:25**

**3pUW6. Modal-MUSIC extensions.** Franklin H. Akins (Scripps, UCSD, 562 Arenas St., La Jolla, CA 92037, fakins@ucsd.edu) and William Kuperman (Scripps, UCSD, La Jolla, CA)

Modal-MUSIC [Akins and Kuperman, *JASA Express Lett*. **2**(7), 074802 (2022)] is a variant of the well-known MUSIC plane wave algorithm that extracts direction of arrivals (DOA's) from data containing multiple sources. Modal-MUSIC, rather than extract the DOA's of the multiple sources, extracts the acoustic normal modes from incoherently summed data from multiple and/or moving sources received on a short vertical array. The method uses the local sound speed profile but does not require bottom geophysical information. We discuss extensions of the method that include using shorter arrays, geophysical-inversion and the additional application of synthetic source aperture [Akins and Kuperman, *J. Acoust. Soc. Am.* **150**(1), 270–289 (2021)] to enhance source localization.

**2:40**

**3pUW7. Low-cost assimilation for sound speed fields in the PCA framework.** Gary Marple (SRI Int., 2100 Commonwealth Blvd, Ann Arbor, MI 48105, gary.marple@sri.com) and David Walker (SRI Int., Ann Arbor, MI)

Forecast ocean sound speed fields are often inaccurate and need to be reconciled with observation data. Conventional data-assimilation methods used for this are generally quite computationally intensive. A compressed representation of the forecast sound speed fields can be obtained using principal component analysis (PCA), where the forecast fields are represented by a linear combination fo PCA modes. We develop a low-cost assimilation approach that updates the PCA compressed representation of the background forecast field, based on observation data.The approach uses Bayes‘ rule to obtain the maximum likelihood estimate for the update in the space spanned by the PCA modes. Significant cost savings come from the dimensionality reduction that is provided by PCA. Results are presented for sound speed fields derived from HYCOM ocean forecast data and expendable bathythermograph data obtained from the Scipps Institution of Oceanography.

## Plenary Session and Awards Ceremony

Peggy B. Nelson
*President, Acoustical Society of America*

**Annual Membership Meeting**

**Introduction of Recipients of ASA Scholarships**

**Presentation of Certificates to New Fellows**

Brian E. Anderson – For contributions to applications of acoustic time-reversal and acoustics education

Julien Bonnel – For advances in time-frequency analysis of underwater sound

John J. Galvin, III – For research into speech and music perception with cochlear implants

Murray S. Korman – For advancing acoustics education through mentoring young acousticians and developing innovative demonstrations

Lori Holt – For improving understanding of neural processing and perception of complex auditory phenomena over the lifespan

Steven M. Lulich – Studies of speech production using 3D ultrasound imaging

Andrew A. Piacsek – For exceptional service to the Society including leadership in scientific communication of acoustics

Lina Reiss – For studies of combined electric and acoustic hearing

Yue Wang – For studies of the behavioral and neural mechanisms underlying speech learning and processing

**Introduction of Prize Recipient**

2022 Rossing Prize in Acoustics Education to Kathleen E. Wage

**Presentation of Awards**

Gold Medal to Michael J. Buckingham

## OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. All meetings will begin at 7:30 p.m., except for Signal Processing in Acoustics (4:30 p.m.) and Engineering Acoustics (4:45 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

| | |
|---|---|
| Signal Processing in Acoustics (4:30) | Rail Yard |
| Engineering Acoustics (4:45) | Lionel |
| Acoustical Oceanography | North Coast A |
| Animal Bioacoustics | Grand Hall A |
| Architectural Acoustics | Summit A |
| Physical Acoustics | Golden Pass |
| Psychological and Physiological Acoustics | Grand Hall C |
| Structural Acoustics and Vibration | Golden Eagle B |

### Committees meeting on Wednesday

| | |
|---|---|
| Biomedical Acoustics | Mill Yard A |

### Committees meeting on Thursday

| | |
|---|---|
| Computational Acoustics | Summit C |
| Musical Acoustics | Rail Head |
| Noise | Summit B |
| Speech Communication | Grand Hall B |
| Underwater Acoustics | North Coast B |

# Gold Medal


## Michael J. Buckingham
## 2022

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

| | |
|---|---|
| Wallace Waterfall | 1954 |
| Floyd A. Firestone | 1955 |
| Harvey Fletcher | 1957 |
| Edward C. Wente | 1959 |
| Georg von Békésy | 1961 |
| R. Bruce Lindsay | 1963 |
| Hallowell Davis | 1965 |
| Vern O. Knudsen | 1967 |
| Frederick V. Hunt | 1969 |
| Warren P. Mason | 1971 |
| Philip M. Morse | 1973 |
| Leo L. Beranek | 1975 |
| Raymond W. B. Stephens | 1977 |
| Richard H. Bolt | 1979 |
| Harry F. Olson | 1981 |
| Isadore Rudnick | 1982 |
| Martin Greenspan | 1983 |
| Robert T. Beyer | 1984 |
| Laurence Batchelder | 1985 |
| James L. Flanagan | 1986 |
| Cyril M. Harris | 1987 |
| Arthur H. Benade | 1988 |
| Richard K. Cook | 1988 |
| Lothar W. Cremer | 1989 |
| Eugen J. Skudrzyk | 1990 |
| Manfred R. Schroeder | 1991 |
| Ira J. Hirsh | 1992 |
| David T. Blackstock | 1993 |
| David M. Green | 1994 |
| Kenneth N. Stevens | 1995 |
| Ira Dyer | 1996 |
| K. Uno Ingard | 1997 |
| Floyd Dunn | 1998 |
| Henning E. von Gierke | 1999 |
| Murray Strasberg | 2000 |
| Herman Medwin | 2001 |
| Robert E. Apfel | 2002 |
| Tony F. W. Embleton | 2002 |
| Richard H. Lyon | 2003 |
| Chester M. McKinney | 2004 |
| Allan D. Pierce | 2005 |
| James E. West | 2006 |
| Katherine S. Harris | 2007 |
| Patricia K. Kuhl | 2008 |
| Thomas D. Rossing | 2009 |
| Jiri Tichy | 2010 |
| Eric E. Ungar | 2011 |
| William A. Kuperman | 2012 |
| Lawrence A. Crum | 2013 |
| Brian C. J. Moore | 2014 |
| Gerhard M. Sessler | 2015 |
| Whitlow W. L. Au | 2016 |
| William M. Hartmann | 2017 |
| William A. Yost | 2018 |
| William J. Cavanaugh | 2019 |
| Judy R. Dubno | 2020 |
| James F. Lynch | 2021 |

## CITATION FOR MICHAEL J. BUCKINGHAM

*...for theoretical and experimental contributions to ocean acoustics and for service to the society*

### 7 DECEMBER 2022 • NASHVILLE, TENNESSEE

Mike Buckingham was born in Oxford, England and attended the Orange Hill Grammar School for Boys where he was deputy head boy. Mike studied at the University of Reading and obtained his B.Sc with Honors in 1967 and Ph.D. in 1971, both in physics. He is a Chartered Engineer in the U.K. and Fellow of the Institute of Acoustics (U.K.), Institution of Engineering and Technology (U.K.), and Acoustical Society of America. Mike currently holds the position of Distinguished Professor in Ocean Acoustics at Scripps Institution of Oceanography, UC San Diego and was awarded the Pioneers of Underwater Acoustics Medal from the Acoustical Society of America in 2017. He holds 5 patents for acoustic innovations.

Mike has had a lively career in underwater acoustics, full of innovative experiments, fundamental theoretical advances, and enormous contributions to our scientific culture, including mentoring a generation of young minds and dedicated service to our society. Early in his career, Mike studied ambient sound in the Arctic, deploying vertical line arrays of his own design from Royal Air Force (RAF) flights across the marginal ice zone. He describes these campaigns in colorful terms; RAF pilots of the time had their own view of what constituted a “safe vertical distance” above the ocean and skirting with the waves on these flights left an indelible impression. Not at all put off, Mike’s use of aircraft as deployment platforms for experiments and sources of sound has spanned his entire career. Indeed, Mike is himself a private pilot, with airplane and glider ratings and qualifications in aerobatics. More than one graduate student, postdoctoral scholar, or visiting colleague has experienced both the joy and terror of aerobatic glider flights with Mike!

Mike’s career is built on the foundation of innovative experimental approaches integrated with new and rigorous theoretical constructs. This powerful approach has advanced the field of underwater acoustics in topics spanning an enormous range, from volcano and seafloor acoustics to the acoustics of bubble plumes and ambient sound in the deep ocean trenches. Mike introduced the name and concept of “Acoustic Daylight” to underwater acoustics, which refers to the use of ambient sound to image the ocean environment. This innovation, published in the journal Nature, prompted the development of Acoustic Daylight monitoring sonar systems in the United States, Australia, and Singapore roughly two decades ago and, more recently, has spurred studies of how marine life may use something analogous to acoustic daylight to sense their underwater environment. Mike developed a new model of wave propagation in marine sediments, the Viscous-Grain-Shearing theory. This elegant model is easy to implement yet predicts wave properties that are in close agreement with data and has been widely accepted by the underwater acoustics community. Such sweeping developments, stemming from a single transformative concept developed with imaginative physical insight and elegant mathematical analysis, can also be seen in Mike’s analysis of seafloor acoustics and his early work on the three-dimensional propagation of sound in shallow water environments.

Mike has mentored a generation of graduate students and postdoctoral scholars, teaching them the discipline of underwater acoustics. His approach is rigorous, with full expectation of excellence in scholarship, but guided by kindness and good humor. Mike’s mentoring style has proven very effective, providing young minds with a roadmap for a fulfilling and productive career in ocean acoustics. Many of Mike’s mentees have gone on to have their own scintillating academic careers and winning their own accolades. In addition to passing on his knack for seeing into the heart of a problem and tackling it with physical and mathematical analysis, one of Mike’s hallmarks is his patent ability to train young scientists in the art of scientific communication. A successful academic career requires the effective communication of ideas and many of Mike’s students have won Best Student Paper awards from international conferences over the years. The appreciation, gratitude, and affection of these scholars toward Mike was abundantly clear at the winter 2019 Acoustical Oceanography special session in his honor, which was a well-attended and cheerful celebration of Mike’s career.

Mike has made significant and lasting contributions to our society over the past 30 or more years, including election to the Executive Council for a 3-year term in 2010 and election to the position of Vice President (VP), which is also essentially a 3-year term as VP-elect, VP, and past-VP. His participation on committees includes the Membership Committee, two 3-year terms on the Medals and Awards Committee, and numerous other commitments. Mike, along with Hank Medwin, David Farmer and Van Holiday, formed the "Gang of Four" that created the Acoustical Oceanography Technical Committee (AOTC). Formed as a specialty group in 1989, AOTC was conferred full status as a Technical Committee in 1992 and Mike was its inaugural Chairman. The early days of AOTC were both tumultuous and exciting; Mike's calm and insightful leadership during this time were essential to the committee's continued success.

Mike's career in marine acoustics has been bountiful. We have all benefited from his leadership within the society. Those of us fortunate enough to be considered friends know Mike to be insightful and rigorous in his scholarship, generous with his time and energy, and encouraging in his mentorship. His contributions to us all and to the science we pursue mark him as an outstanding leader of acoustical oceanography and contributor to the mission of the Acoustical Society of America.

GRANT DEANE
DAVID FARMER

# Session 4aAA

## Architectural Acoustics and ASA Committee on Standards: Show Your Data: Architectural Acoustics Metrics

Ana M. Jaramillo, Cochair
*Ahnert Feistel Media Group, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444*

Bruce Olson, Cochair
*Olson Sound Design LLC, 8717 Humboldt Avenue N, Brooklyn Park, MN 55444-1320*

### *Invited Papers*

**8:30**

**4aAA1. Examination of reverberation time predictions and measurements.** Joseph Keefe (Ostergaard Acoust. Assoc., 1460 US Hwy. 9 North, Ste. 209, Woodbridge, NJ 07095, jkeefe@acousticalconsultant.com)

This presentation will discuss reverberation time data, including measured versus predicted, occupied versus unoccupied, and calculated based on different predictive techniques. Some predicted data agree well with observed conditions, while others differ significantly. This presentation will facilitate a discussion of techniques, methods, and approaches.

**8:50**

**4aAA2. Simulation versus measurement of acoustical banners.** Ana M. Jaramillo (Ahnert Feistel Media Group, 8717 Humboldt Ave., N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu), Brandon Cudequest (Threshold Acoust., Chicago, IL), and Bruce Olson (Ahnert Feistel Media Group, Brooklyn Park, MN)

This study compares the results of measurements and simulations for variable acoustical banners in several venues. Reverse calculations of real venues reveal a wide spread of achievable absorption coefficients for banners. When modeling these spaces and materials in pre-design, it is critical to use the correct absorption coefficients for accurate results. This presentation will explore the different ways of modeling banners using classical ray tracing acoustic simulation software and offer recommendations for interpreting absorption coefficients provided by banner manufacturers.

**9:10**

**4aAA3. Show your scattering coefficients.** Michael Vorlaender (IHTA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, mvo@akustik.rwth-aachen.de) and Stefan Feistel (AFMG, Berlin, Germany)

Absorption and scattering coefficients of surfaces are crucial components of room-acoustic simulations. The random-incidence scattering coefficient is the most uncertain metric applied in geometrical acoustics. Databases of scattering coefficients are rare. Their impact on auditory perception in auralizations, however, is relatively well studied. Research directions in developing improved scattering metrics can focus on their relevance for auditory effects, their influence on the spatial sound energy flow, their characterization in angular decomposition, and their estimation with simple rules of thumb. This paper will highlight the state of the art in determining and using scattering coefficients, and it will discuss possible improvements.

**9:30**

**4aAA4. Retooling results of a massive database of home theater measurements.** Tomlinson Holman (7648 Butts Canyon Rd., Pope Valley, CA 94567, napkin.assets_0t@icloud.com)

In 2010 the first results of a large-scale measurement program on home theaters was reported in an AES presentation. More than 1,000 rooms were measured for impulse response from each of a minimum of six loudspeakers to at least four (usually more) listening locations. However, computer processing capacity limited the initial report to 275 rooms. Room acoustics and sound system information was reported: RT and its std. dev. with frequency, room volume statistics for those rooms, and transient and steady-state frequency response for some example rooms were shown. From these few rooms an important item of interest emerged: the Schroeder frequency did not show as a boundary of response deviations between lower and higher frequencies—the average deviation from the average response was more constant with frequency than expected. This work will be briefly reviewd. Now in 2022 a new look at this data plus that which was gathered in the intervening years may be studied with much more advanced math tools such as machine learning so much more can be understood. This presentation is interim describing work to be done and soliciting comments from practitioners.

**9:50–10:05 Break**

**10:05**

**4aAA5. One acoustician's defect is another artist's feature: Simulating real flutter for an art installation.** Marcus R. Mayell, Nicolaus T. Dulworth (Threshold Acoust., Chicago, IL), Brandon Cudequest (Threshold Acoust., 141 W Jackson Blvd., Ste. 2080, Chicago, IL 60604, bcudequest@gmail.com), and Robin Glosemeyer Petrone (Threshold Acoust., Chicago, IL)

Flutter is generally considered an acoustical defect – the timbral distortions and rhythmic nature can be distracting, annoying, or downright disruptive. The paper recontextualizes flutter as a compositional tool for sound art, particularly when the flutter of existing rooms is used for site-specific installations. Due to the global COVID-19 pandemic, the artist was unable to travel, allowing us to visit the site and take several room impulse responses. The measured spaces were parallelpiped, sound reflective, and fluttery. To give the artist creative flexibility, we simulated the rooms for an extended range of source and receiver locations informed by the *in-situ* measurements. This paper will discuss our calibration and modeling techniques to simulate flutter and reverberation coloration in real rooms, which is non-trivial for image source methods or ray-based software.

**10:25**

**4aAA6. Applying unsupervised machine learning clustering techniques to early childcare soundscapes.** Kenton Hummel (Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182, khummel@huskers.unl.edu), Erica E. Ryherd, Lason Konstantzos (Architectural Eng., Univ. of Nebraska - Lincoln, Omaha, NE), and Abbie Raikes (Dept. of Health Promotion, Univ. of Nebraska Medical Ctr., Omaha, NE)

Early childhood is a critical time period for language, brain, cognitive, and social/emotional development. Out-of-home childcare is a normative, typical experience for millions of young children. Although Indoor Environmental Quality (IEQ) in K-12 settings has received recent, significant attention, the links between IEQ and children's learning and development in early childcare settings is a less understood topic. This work focuses specifically on the sound aspect of IEQ in early childcare settings to better understand typical noise levels and occupant experience. Standard approaches to analyzing background noise will be presented alongside more detailed statistical analyses utilizing unsupervised machine learning clustering techniques. Noise data collected in three daycares will be presented using typical acoustic metrics and clustering techniques to better understand room activity conditions and support new metrics. Overall, this study can lead to a better understanding of daycare soundscapes and pave the way towards a better childcare for young children.

## *Contributed Papers*

**10:45**

**4aAA7. Measuring decay time in non-diffuse rooms.** Arthur W. van der Harten (Acoust., Acoust. Distinctions/Open Res. in Acoust. Sci. and Education, 400 Main St. Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com) and Matthew Azevedo (Acentech, Cambridge, MA)

Measurements in non-diffuse rooms present challenges in determining decay time. A room with a severe flutter echo may exhibit a double slope decay without the incorporation of a coupled chamber. The measured decay time may differ greatly depending on the software used to measure, the choice of Schroeder integral sampling range, and how noise correction is implemented. A standardized approach to characterizing decay time under non-diffuse conditions is needed. In this talk, the authors discuss possible methods for determining the optimal noise correction and sampling of the Schroeder integral, including varying the IR truncation and sampling start and stop range. Computational optimization methods are employed to objectively determine the most reasonable settings for interpreting a particular impulse response.

**11:00**

**4aAA8. Calibrating geometrical acoustics models of non-diffuse rooms.** Arthur W. van der Harten (Acoust., Acoust. Distinctions/Open Res. in Acoust. Sci. and Education, 400 Main St. Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com) and Matthew Azevedo (Acentech, Cambridge, MA)

A case study is presented for the challenging case in which a conference room with a strongly non-diffuse impulse response is measured, and geometrical acoustics models are built and calibrated using two separate software packages simultaneously. Both software packages are able to reproduce the measured flutter echoes, although with surprising input scattering coefficients, particularly for parallel stainless steel wall finishes. Caution is advised concerning common assumptions about scattering coefficients for operable partitions. Furthermore, the authors suggest that benchmarking for acoustics simulation software be performed in strongly non-diffuse rooms, in addition to the existing round robin studies that have traditionally been held using more normal or diffuse rooms.

**11:15**

**4aAA9. Assessing cognitive effects of transportation noise on office workers with electroencephalography and performance.** Heui Young Park (The Pennsylvania State Univ., Graduate Program in Acoust., 201 Appl. Sci. Bldg., University Park, PA 16802, hkp5188@psu.edu), Michelle C. Vigeant (The Pennsylvania State Univ., University Park, PA), and Andrew Dittberner (GN Adv. Sci., GN Group, Bloomington, MN)

Open offices are prone to not only internal noise but also external traffic noise. Noise has been shown to have detrimental effects on cognition and task performance, yet current literature on the effects of noise in open offices lack clarity and the findings are often inconclusive. This study aims to explore the effects of different acoustic conditions in an office: (1) with traffic noise and (2) in quiet, on task performance, brain activity, and subjective ratings. Subjects were given cognitive tasks, such as grammatical reasoning, while wearing an electroencephalography (EEG) headset. Brain activity and performance were recorded after a training session. Recorded traffic noise was auralized in an anechoic chamber with a 32-loudspeaker array to simulate road, rail, and subsonic aircraft noise heard when sitting in an elevated office with open windows. After completion of the tasks in the two acoustic conditions, subjective surveys were conducted for several items including perceived performance and annoyance. A statistical analysis was conducted to study the effects of different acoustic conditions on cognitive task performance, EEG data, and subjective ratings. Study details and results will be presented.

**11:30**

**4aAA10. Spatial release from masking in anechoic and reverberant environments.** Drake A. Hintz (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 9431 U Plaza #8, Omaha, NE 68127, Drake.A.Hintz@gmail.com), Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), and Z. Ellen Peng (Boys Town National Res. Hospital, Omaha, NE)

Listening with both ears provides children with access to binaural and monaural cues that are helpful for understanding speech in competing babbles. Specifically, when the target and masker are spatially separated, children can gain an intelligibility benefit which is known as spatial release from masking (SRM). Recent work [Peng *et al.*, 2021 JASA] suggested that school-age children demonstrated immature SRM using binaural cues that are distorted by reverberation. In this follow-up study, we further investigate the effect of reverberant distortion on individual auditory spatial cues, namely binaural and monaural head shadow cues. We compare SRM between adults and school-age children with typical hearing using the novel measure of minimum angle of separation (MAS) between target and masker, for which individual achieves a 20% intelligibility gain, in both virtually simulated anechoic and reverberant environments. MAS was measured in both binaural and monaural hearing conditions, as well asymmetric versus asymmetric masker displacement to probe access to various auditory cues of interest. Preliminary results show that adults demonstrate similar SRM in anechoic and in reverberation of $T60 = {\sim}0.6$ s, but better (smaller MAS) when additional spatial cues become available. Reverberation effect on children's SRM will be discussed. [Work supported by NIH-NIDCD.]

THURSDAY MORNING, 8 DECEMBER 2022 NORTH COAST A, 8:00 A.M. TO 11:30 A.M.

## Session 4aAO

## Acoustical Oceanography: Topics in Acoustical Oceanography

Alexander S. Douglass, Cochair
*Oceanography, University of Washington, 1501 NE Boat St., MSB 206, Seattle, WA 98195*

Elizabeth Weidner, Cochair
*University of New Hampshire, 24 Colovos Road, Durham, NH 03824*

### *Contributed Papers*

**8:00**

**4aAO1. Improved characterization of ambient sound levels in a coastal environment via use of an unmanned Wave Glider.** Joseph Lafrate (Dept. of Navy, 1176 Howell St., Newport, RI 02841, joseph.iafrate@navy.mil), Georges Dossot, Stephanie Watwood (Dept. of Navy, Newport, RI), Eric Reyier, and Bonnie Ahr (Herndon Solutions/Kennedy Space Ctr., Kennedy Space Ctr., FL)

In this project, we use an unmanned Liquid Robotics SV3 wave glider as a tool for basic research studying ambient acoustics in shallower, coastal waters off the coast of East-Central Florida. The primary objective is to characterize sound levels in varying conditions via collection of high-resolution environmental data. The SV3 is used to systematically characterize the acoustic environment in coastal waters, while simultaneously measuring oceanographic parameters such as wind speed, wind direction, and wave height. Quantifying the influence of these fundamental oceanographic influences on shallow water acoustics will provide improved characterization of surface noise and comparison with traditional ambient noise spectra. Ambient noise spectra will be presented for various sea states and wind scenarios, as measured at 25 foot depth using a towfish attached to the SV3. Some of these measurements are also compared to fixed station bottom recordings. Finally, this effort also includes characterization of unique sources of biological sounds recorded by the SV3, including not previously documented loud fish chorusing in this region.

**8:15**

**4aAO2. Acoustic rainfall estimation with support vector machines and error correcting output codes.** C. J. Berg (Elec. and Comput. Eng., UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, cberg@umassd.edu), C. Mallary, John R. Buck (Elec. and Comput. Eng., UMass Dartmouth, Dartmouth, MA), Amit Tandon (School for Marine Sci. and Technol., UMass Dartmouth, Dartmouth, MA), and Alan Andonian (Mech. Eng., UMass Dartmouth, Dartmouth, MA)

Ma and Nystuen (2005) successfully detected and estimated rainfall at sea from passive acoustics. They detected rain from three narrowband frequencies and then estimated log rainfall rate via a regression with energy in the 5 kHz band. Mallary *et al.* (2022) improved rainfall detection by exploiting broadband spectra while reducing the dimensionality through principal component analysis (PCA). This project builds upon Mallary's work moving beyond detection to estimate the rainfall by quantization into discrete ranges based on PCA-reduced acoustic power spectra. The classification scheme combines multiple binary support-vector machine (SVM) classifiers (Boser *et al.* 1992) with Dietterich and Bakiri's error-correcting output codes (1995) to classify acoustic PSDs into one of 6 rainfall rate classes. Evaluating the PCA/SVM classifier on 4 months of acoustic recordings and meteorological data collected from a shallow water pier in New Bedford, MA found the hourly accumulations from the rain gauge and acoustic estimates had a correlation of $0.97 \pm 0.01$. Emulating Ma & Nystuen's estimator on the same data set yields a correlation of $0.76 \pm 0.02$. [Work supported by ONR.]

8:30

**4aAO3. The soundscape of two deep-sea hydrothermal vent sites.** Brendan Smith (Oceanogr., Dalhousie Univ., Halifax, NS B3H 4R2, Canada, brendan.smith@dal.ca) and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Hydrothermal vents are sea floor structures where geothermally heated seawater is discharged. The high-temperature, chemically rich environment is host to uniquely adapted marine life. Vent soundscapes may contain important bioacoustic cues as well as signals enabling passive acoustic monitoring of hydrothermal vent dynamics. Proposals for deep-sea mining of seafloor massive sulfides near hydrothermal vents have elicited concern over potential environmental impacts due to disturbance from industrial activity, including changes to the soundscape. This study assesses the baseline soundscape at two sites, the Main Endeavour Field on the Juan de Fuca Ridge and the Lucky Strike vent field on the Mid-Atlantic Ridge over 12 months and 3 months respectively. To facilitate comparison with future studies at other sites, the most recently proposed standard soundscape analysis methodologies are employed, including terminology in alignment with ISO 18405:2017. In accordance with the latest soundscape standard literature, metrics quantifying the amplitude, impulsiveness, periodicity, and uniformity are reported. Spectral probability densities, percentiles, and long-term spectrograms are computed in hybrid millidecade frequency bands. Finally, a qualitative analysis is included to describe the source types contributing to the hydrothermal vent sound field.

8:45

**4aAO4. Real-time instantaneous wide-area ocean acoustic monitoring in the shallow Great South Channel and deep ocean south of Rhode Island with Northeastern University coherent hydrophone array.** Sai Geetha Seri (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, seri.s@northeastern.edu), Hamed Mohebbi-Kalkhoran, Max Radermacher, Matthew E. Schinault (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), Chenyang Zhu, Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

A large-aperture 160-element coherent hydrophone array system, developed in-house at Northeastern University, was deployed and tested at sea in shallow water Great South Channel and deep ocean south of Rhode Island during September 2021. The system implements the passive ocean acoustic waveguide remote sensing technology for real-time instantaneous wide area ocean monitoring. The array data sampled at 100 kHz per element were processed in real-time at sea to provide beamformed spectrogram imagery in multiple frequency subbands from 10 Hz to 50 kHz and spanning 360 degree horizontal azimuths about the receiver array. A wide variety of acoustic detections were made in real time, including marine mammal vocalizations and motion-related signals, fish grunts and other fish-generated signals, tonal and broadband signals from distant ships and own tow ship. Real-time detections of marine mammal vocalizations include fin whale 20 Hz pulses, humpback whale songs and social sounds, minke whale buzz sequences, sperm whale echolocation and social clicks, dolphin whistles and echolocation clicks, as well as calls from unidentified baleen and toothed whale species. A database of automatically detected signals with bearing-time information along with extracted time-frequency characteristics was generated in real-time at sea.

9:00

**4aAO5. Observations of Noise due to Nonlinear Internal Waves in the ASIAEX Experiment in the South China Sea.** Yanyu Jiang (Marine Geosciences, Univ. of Haifa, Haifa, Israel), Boris Katsnelson (Marine Geosciences, Univ. of Haifa, 199 Adda Khouchy Ave., Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il), and Oleg A. Godin (Phys., Naval Postgrad. School, Monterey, CA)

Large, transient increases in underwater noise intensity (noise bursts) were repeatedly observed during the ASIAEX experiment on a 48-hydrophone array consisting of vertical and horizontal line arrays. The arrays were located in the South China Sea in the area known for exceptionally strong nonlinear internal waves (NIWs). The ASIAEX experiment was intended to study sound propagation in the presence of NIWs and featured extensive, concurrent acoustic and oceanographic observations. The NIWs propagating past the acoustic array site were characterized using water temperature measurements with a nearby thermistor chain. Remarkable correlation is found between noise intensity increases on the vertical array hydrophones and the NIW presence. Intensity of the noise bursts strongly depends on the hydrophone depth. The low acoustic frequencies below 30–40 Hz are primarily responsible for the noise enhancement during the NIW passage. Analysis of the spectral properties and the depth dependence of the noise intensity suggests the flow noise due to NIW-induced currents as the physical mechanism of the noise bursts in the South China Sea. [Work was supported, part, by ISF award 946/20.]

9:15

**4aAO6. Dictionary learning for classification of sound speed profile in the ocean.** Jhon A. Castro-Correa (Elec. and Comput. Eng., Univ. of Delaware, 139 The Green, 3rd Fl., Newark, DE 19716, jcastro@udel.edu), Mohsen Badiey, and Lin Wan (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

Internal waves (IWs) are waves that propagate along density gradients within the water column and alter the isotropic properties of sound speed profiles (SSPs). Changes in the SSPs modify the acoustic channel ducts and affect underwater acoustic propagation, causing most of the energy to be dissipated into the seabed due to the downward refraction of sound waves. Therefore, variations in the SSP must be considered when modeling acoustic propagation in the ocean. It has been proved that Dictionary learning (DL), an unsupervised machine learning method, succeeds in sparsely representing signals by employing a few non-orthogonal basis functions (atoms) learned from data. In this work, we use the ability of learned dictionaries (LDs) for data representation and thus train class-specific dictionaries to capture relevant features from labeled data within a supervised learning setting. We developed an LD-based supervised framework for SSP classification and compared it with state-of-the-art models. The algorithms presented in this work are trained and tested on data collected from the shallow water experiment 2006. Results show that overcomplete DL is a robust method to classify SSPs during IW activity, reporting comparable and higher accuracy than standard supervised classification methods. [Work supported by ONR, Grants No. N00014-21-1-2424 and N00014-21-1-2760.]

**9:30–9:45 Break**

9:45

**4aAO7. Underwater low frequency Helmholtz bubble resonator.** Andrey K. Morozov (Marine Systems, Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

Low-frequency sound sources have found application in oceanology and geoacoustic methods of remote sensing. An underwater low-frequency sound source with a pneumatically driven bubble resonator covered with an elastic membrane effectively provides a very high source level. However, it has a narrow bandwidth, and its resonant frequency is difficult to change without changing the size of the original system. Internal acoustic resonators included in a bubble filled with gas can change the frequency response of the entire source and expand its bandwidth. Internal resonant systems can be designed so that the bubble resonator can be tuned over a wide frequency range. Other systems may add one or more internal resonances and spread the emitted spectrum over a very wide frequency band. It is possible to consider various multipole resonant systems in combination with an underwater bubble. A simple and efficient system consists of a bubble resonator and an internal Helmholtz resonator. The addition of a Helmholtz resonator converts the single resonant bubble into a double resonant system and extends its bandwidth. The theory of the underwater bubble Helmholtz resonator and various applications of these resonators for practical systems are considered. The results of the experimental verification are discussed.

**10:00**

**4aAO8. Measuring marine gas flux via passive acoustic signals—New field and lab observations.** Ben Roche (Univ. of Southampton, National Oceanogr. Ctr., Southampton SO14 3ZH, United Kingdom, bjr1e21@soton.ac.uk), Paul White (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom), Jonathan Bull (Univ. of Southampton, Southampton, United Kingdom), and Timothy G. Leighton (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

In recent years there has been a growing need to refine methods for quantifying the volume of gas released into the water column from marine sediments via seeps/vents. Traditional methods of flux quantification are flawed, being energy intensive techniques that provide single snapshot measurements. As a result, emission estimates across larger areas are very poorly constrained, often ranging across several orders of magnitude, are rarely cross validated and almost never account for temporal variation. Passive acoustic flux inversion techniques, whereby the acoustic signal of such seeps can be used to measure the gas flux, have been identified as a key tool for overcoming these limitations. Several field studies have already demonstrated the potential of such techniques, granting fresh insights into the variability of gas release with tides, surface seiches and even day/night temperature cycles. Here, we discuss our work to further advance passive acoustic flux inversion in order to enable large scale non-specialist adoption. Presenting field observations and our recent work studying the excitation of gas bubbles and the acoustic signature of gas propagating through unconsolidated sediment. Achieved with a combination of high-speed photography, CT imaging, radiography, geophones, fibre optic cables and traditional marine hydrophones.

**10:15**

**4aAO9. Volume scattering of sound signals in gassy bottom and estimation of bubbles concentration in Lake Kinneret.** Ernst Uzhansky (Marine Geosciences, Univ. of Haifa, Izhaq Greenboim Str., Apt. 12, (Koren Family), Haifa 3498793, Israel, ernstuzhansky@gmail.com), Boris Katsnelson (Marine Geosciences, Univ. of Haifa, Haifa, Israel), Andrey Lunkov (General Phys. Inst., Moscow, Russian Federation), Regina Katsman (Marine Geosciences, Univ. of Haifa, Haifa, Israel), and Anatoliy N. Ivakin (Univ. of Washington, Seattle, WA)

In the paper, results of experiments in shallow Lake Kinneret (Israel) and the corresponding data processing are presented. Wideband LFM sound signals (300 Hz–3.50 kHz and 300 Hz–15 kHz) were radiated by the source mounted directly on vertical line array (VLA) of the length 20 m placed at several locations with different water depths (from 20 to 35 m). Received sound field timeseries constitutes a sequence of pulse arrivals comprised of specular reflections from interfaces followed by reverberation codas caused by non-specular scattering from the interface roughness and volume inhomogeneity. Properties of received signals are analyzed assuming that the sediment can be modeled as a thin layer containing bubbles over a homogeneous fluid half-space. Using a set of hydrophones of VLA allowed isolation of the volume scattering component, estimating frequency-angle dependences of scattered field, and hypothesizing some properties of the gassy sediment, such as concentration of bubbles and their effective size. [Work was supported by BSF grant 2018150.].

**10:30**

**4aAO10. Effects of sub-seabed characteristics on acoustic transmission loss in seismic reflection surveys.** Alexander S. Douglass (School of Oceanogr., Univ. of Washington, 1501 NE Boat St., MSB 206, Seattle, WA 98195, asd21@uw.edu), Warren T. Wood, Benjamin J. Phrampus (Ocean Sci. Div., US Naval Res. Lab., Hancock County, MS), and Shima Abadi (School of Oceanogr., Univ. of Washington, Seattle, WA)

Marine seismic reflection surveys are used to study the composition of the seabed and sub-seabed by recording reflections of the field generated by an airgun array from the seafloor with a towed horizontal streamer. The abundance of data within a single survey and the number of publicly available surveys provides substantial opportunity for data-driven acoustic analysis. Here, we consider data from surveys that were conducted off the coasts of Oregon, Washington, British Columbia, and Alaska, covering portions of the Cascadia Subduction Zone and the Queen Charlotte Fault. Throughout the experiments, data was collected at depths ranging from 200 m to 3 km, typically using a 15 km long towed streamer containing 1200 hydrophone groups with 12.5 m spacing. With these datasets, there is a breadth of opportunity for studying the impacts of various seabed features on acoustic propagation. This talk will explore the impacts of seabed and sub-seabed characteristics, bathymetry, reflectivity, etc., on acoustic transmission loss. The data from these experiments, corrected for array design impacts, will be compared directly with computational propagation models. [Work supported by ONR.]

**10:45**

**4aAO11. Low-order acoustic mode arrivals in the Beaufort Duct.** Jessica Desrochers (Ocean Eng., The Univ. of Rhode Island, 13 Gilroy St. Apt. 2, Newport, RI 02840, jfothergill@uri.edu), Lora Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Sarah E. Webster (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Alexander P. Muniz (Naval Undersea Warfare Ctr., Div. Newport, Newport, RI), Cristian E. Graupe (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), and Luis O. Pomales Velázquez (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI)

Significant changes in the stratification of the Beaufort Sea over the last few decades have produced a subsurface duct located between 100- and 300-meters depth, known as the Beaufort Duct. This subsurface duct allows for long-range acoustic transmission with little to no interaction with the sea surface or seafloor. In August and September of 2017, acoustic transmissions from five active moored tomography sources were collected at ranges up to 530 km by two Seagliders along with *in-situ* environmental measurements. Sound-speed profiles from the Seaglider data were used as input for parabolic equation and normal mode predictions. Both the predictions and recorded acoustic data show a peak acoustic arrival prior to the final cutoff. We refer to this as a "foldover" feature in the acoustic timefront, and it can be connected back to the unique ducting features in the input sound speed profiles. The relationship between the extent of the foldover and the shape of the sound-speed profile in the duct is explored using normal modes. Modal group speed predictions for the low-order modes are used to understand which modes make up the foldover feature present in the acoustic timefront and to interpret the acoustic arrival patterns measured on the Seagliders.

**11:00**

**4aAO12. Robust processing for multipath in acoustic navigation, tomography, and source tracking.** Paul Hursky (Appl. Ocean Sci. LLC, 4825 Fairport Way, San Diego, CA 92130, paul.hursky@gmail.com), Christopher Verlinden (Appl. Ocean Sci. LLC, Fairfax Station, VA), John Boyle (Appl. Ocean Sci. LLC, Seattle, WA), Emanuel F. Coelho (Appl. Ocean Sci. LLC, Springfield, VA), and Kevin D. Heaney (Appl. Ocean Sci. LLC, Fairfax Station, VA)

Because sound travels along many refracted paths through the ocean, passing through many horizontal sound speed layers, it is often challenging to unravel the received multipath to convert arrival times to ranges (e.g., by assigning them to eigenrays calculated by a ray tracer), particularly when it must be automated. This problem is exacerbated when the source and/or receivers are in motion. Assigning the wrong eigenray to an arrival results in the arrival time being modeled for the wrong acoustic path and the wrong range assigned to that arrival, resulting in degraded navigation, tomography, or localization. We will show how to take advantage of redundant measurements to ensure that we avoid spurious eigenray assignments. We use an initial "robust" optimization process whose cost function is less sensitive to spurious measurements or outliers than processes using least squared errors. Such processes both enable outliers to be identified and produce better estimates (than if spurious measurements were admitted). Better estimates allow the assignment of eigenrays to arrivals to be better constrained, often leading to corrected assignments. This virtuous cycle results in fewer spurious assignments and improved forward modeling for applications like navigation, tomography, and source tracking.

11:15

**4aAO13. Enhancing underwater Acoustic Identification Tag design using concurrent wireless ultrasonic power transfer and backscatter communication.** Ananya Bhardwaj (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, J. Erskine Love Bldg., Rm. 131, Atlanta, GA 30332, ab22@gatech.edu), Ahmed Allam, Alper Erturk, and Karim G. Sabra (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Improvements in undersea localization and navigation of Autonomous Underwater Vehicles (AUVs) is critical. However using conventional active transponders as navigation aids is limited by powering constraints. Passive backscatter devices such as Acoustic Identification (AID)Tags are an alternate solution to this energy limitation [Satish *et al.*, JASA 149 (2021)], but offer limited information content. To address this limitation, adding a wireless ultrasonic power transfer capability to these passive AID tags can be used to enable backscatterer uplink communication between the AID tag and an interrogating compact high-frequency SONAR system (which can be mounted on an AUV). Our design comprises a piezoelectric transducer impedance matched to a connected electrical load across two frequency ranges: a narrowband energy harvesting range, and a broadband data range. Onboard electronics harvest acoustic energy in the narrowband to power an onboard microcontroller, which can be used to simultaneously modulate the transducer broadband acoustic impedance by switching the electrical load connected to transducer during backscatter uplink communication. Water tank experiment using custom transducers, operating either around 1 MHz or 350 kHz, were conducted to quantify-amongst others-the operational source to tag distance, system power requirements and achievable communication data rates using the proposed concurrent ultrasonic power and data transfer approach.

THURSDAY MORNING, 8 DECEMBER 2022 MILL YARD A, 9:00 A.M. TO 11:55 A.M.

## Session 4aBAa

## Biomedical Acoustics: Detection and Quantification of Bubble Activity in Therapeutic Ultrasound I

Adam Maxwell, Cochair
*Department of Urology, University of Washington, Seattle, WA 98195*

Eli Vlaisavljevich, Cochair
*Dept. of Engineering Mechanics, Virginia Tech, Blacksburg, VA 24061*

**Chair's Introduction—9:00**

### *Invited Papers*

9:05

**4aBAa1. Finding the bubbles in therapeutic ultrasound—For better or for worse.** J. Brian Fowlkes (Radiology, Univ. of Michigan, 3226C Medical Sci. Bldg. I, 1301 Catherine St., Ann Arbor, MI 48109-5667, fowlkes@umich.edu)

Cavitation has played a variety of roles in therapeutic ultrasound since the field began. Bubble activity has been seen as both a harmful side effect and the means to the therapeutic end. Throughout this range, the monitoring of bubble activity has been an important goal for numerous reasons including safety considerations, progression of treatment, and determining therapeutic efficacy. Methods of cavitation monitoring have been summarized in various contexts including the ASA effort led by Wesley Nyborg, PhD that generated the American National Standards Institute (ANSI) S1.24 TR-2002 (R2007) entitled ANSI Technical Report—Bubble Detection and Cavitation Monitoring. More recently a six-part virtual series, The Detecting, Mapping, and Quantifying Bubble Activity in Therapeutic Ultrasound workshop (https://www.fusfoundation.org/posts/bubble-activity-in-therapeutic-ultrasound-workshop-series/) was held in 2021, sponsored by The Focused Ultrasound Foundation, in partnership with the American Institute of Ultrasound in Medicine's Future Fund. A summary was released as a white paper. This presentation will provide an introduction to the field of bubble and cavitation detection and monitoring and a summary of this most recent workshop and associated follow-on efforts.

**9:25**

**4aBAa2. Sub-megahertz cavitation detection and characterization in mouse brains.** Gregory T. Clement (Office of Sci. and Eng. Labs., US Food and Drug Administration, 10903 New Hampshire Ave., Silver Spring, MD 44195-0001, gregory.clement@fda.gov), Asis Lopez, Guangying Wu, and Meijun Ye (Office of Sci. and Eng. Labs., US Food and Drug Administration, Silver Spring, MD)

A key consideration in ultrasound medical device safety is the potential for cavitation related bioeffects. Device testing should address possible cavitation pathways, such as bubble formation, either in the focal region or at the device-tissue interface along with possible consequences regarding both safety and effectiveness. This has been particularly important in a number of emerging transcranial therapies including ablation, neuromodulation, and targeted opening of the blood-brain barrier (BBB) in conjunction with microbubbles. Subharmonic, and ultraharmonic emissions have been identified as indicators of treatment outcome and are a basis for safety and feedback control in BBB disruption. However, there remains a scarcity of baseline cavitation threshold data in the living mammalian brain, particularly at lower ultrasound frequencies. In our ongoing study, we are measuring and evaluating *in vivo* cavitation response in mice using a passive cavitation detector (PCD) for ultrasound frequencies as low as 40 kHz. Additional measurements are being carried out using a hydrophone imbedded in the brain tissue of mouse cadavers to characterize absolute pressure values as well as evaluate the sensitivity and reliability of the PCD measurements. Analyzed data will be presented along with detailed methodology and potential implications in future product evaluation will be discussed.

**9:45**

**4aBAa3. Passive and active Doppler methods and metrics to quantify inertial cavitation induced by pulsed focused ultrasound.** Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, 1013 NE 40th St., CIMU, Portage Bay Bldg., Seattle, WA 98105, tdk7@uw.edu), Minho Song (Mech. Eng., Univ. of Washington, Seattle, WA), Randall P. Williams (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Joo Ha Hwang (Dept. of Medicine, Stanford Univ., Palo Alto, CA), Yak-Nam Wang, and Oleg A. Sapozhnikov (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Inertial cavitation induced by pulses of nonlinearly distorted focused ultrasound (FUS) at moderate intensity can result in mild mechanical disruption of tissue, short of its complete mechanical disintegration—histotripsy. This effect can be used to enhance diffusion of subsequently systemically administered drugs or biologics. Previously, single-element passive cavitation detection (PCD) of broadband noise emissions was successfully used to quantify tissue disruption and enhancement in drug concentration. This metric, however, has limitations: minimal spatial resolution and challenges with calibration when used at large depths in acoustically variable tissue. To address this, we developed a combination of passive and active Doppler-based methods that relied on relative, rather than absolute signal metrics. Specifically, destructive cavitation behaviors were previously linked to substantial motion of the bubbles during the FUS pulse due to acoustic radiation force and shock scattering. This was reflected in the backscattered FUS harmonics as Doppler shift and measurable from PCD. The bubbles were observed to dissolve within milliseconds following each FUS pulse, thus their distribution could be visualized as an area of rapid change via fast plane wave Doppler ensemble following the FUS pulse. The Doppler power distribution was spatially correlated with the area of tissue disruption in the *in vivo* experiments. [Work supported by NIH R01CA154451, R01EB025187, and R01EB23910.]

**10:05**

**4aBAa4. Detection and quantification of bubble activity in therapeutic ultrasound: Magnetic resonance imaging for cavitation detection and quantification.** Steven P. Allen (Elec. and Comput. Eng., Brigham Young Univ., 155 McDonald Bldg., Provo, UT 84660, spallen@byu.edu)

For some applications of therapeutic ultrasound, magnetic resonance imaging (MRI) may serve as a useful tool for detecting, measuring, and quantifying cavitation activity—especially in cases where direct sampling of acoustic emissions are difficult. Doing so can be challenging because the physical phenomena that drive ultrasonic cavitation (e.g., pressure, surface tension, micro-second time scales) have little overlap with the phenomena that drive MRI (e.g., quantum spin, Faraday induction, milli-second time scales). However, this same principle also protects MRI-based cavitation detection from common confounds that plague traditional direct acoustic detection. The proposed paper will first review known methods for encoding cavitation behavior into MR images. These methods generally employ one of two strategies: (1) Encode transient cavitation activities into otherwise approximately static MR signal sources.; and (2) Leverage the time-average effects of many, many cycles of cavitation activity to alter MR signal sources. This paper will then discuss how these methods may be useful to therapeutic ultrasound applications. Finally, this paper will pose opportunities for future development.



**10:25–10:45 Break**

**10:45**

**4aBAa5. Real-time passive cavitation mapping with high spatial-temporal resolution.** Mucong Li, Daiwei Li (Duke Univ., Durham, NC), Yun Jing (Acoust., Penn State Univ., State College, PA), Pei Zhong (Duke Univ., Durham, NC), and Junjie Yao (Duke Univ., 100 Sci. Dr., Hudson Hall Annex 261, Durham, NC 27708, junjie.yao@duke.edu)

Shock wave lithotripsy (SWL) and laser lithotripsy (LL) have been widely used for clinical treatment of kidney stones. Cavitation plays an important role in stone fragmentation in both SWL and LL, yet it may also contribute to renal tissue injury. It is therefore crucial to determine the spatiotemporal distributions of cavitation activities to maximize stone fragmentation while minimizing tissue injury. Passive cavitation mapping (PCM) has most practical applications in deep biological tissues and is most promising for clinical translation. We have developed a set of technologies for 2D/3D PCM that can be seamlessly integrated with ultrasound imaging and photoacoustic imaging. Our 2D/3D PCM has achieved a spatial resolution of hundreds of micrometers and a temporal resolution of several microseconds. We also developed a transient angular spectrum approach for PCM reconstruction, which is ten times faster than the traditional delay-and-sum method. Using the 2D/3D PCM system, we imaged shockwave- and laser-induced single cavitation bubbles in

both free field and constricted space, as well as on large animal models. Collectively, our results have demonstrated the high reliability and spatial-temporal accuracy of the 2D/3D PCM approach, which paves the way for future *in vivo* applications and human studies during SWL and LL.

**11:05**

**4aBAa6. Detection, characterization, and spatial mapping of nucleated cavitation for drug delivery: From *in-vitro*-, through *in-vivo* to the clinic.** Paul Boulos, Edward Ellis, Maura Power, Edward Jackson, Anant Shah, Charles Pearce, Patrycja Szostek, Naomi Berry, Erika Vojtasova, Mohammed Alkattan, Massimo Masiero, Calum Crake, Cliff Rowe (OxSonics Therapeutics, Ltd., Oxford, United Kingdom), and Christian Coviello (OxSonics Therapeutics, Ltd., OxSonics Therapeutics, The Magdalen Ctr., Robert Robinson Ave., Oxford OX4 4GA, United Kingdom, christian.coviello@oxsonics.com)

Cavitation-nucleation agents that stabilize gas bubbles, be they micro or nano in scale, allow for consistent and reliable induction of cavitation compared to endogenous nuclei. When these agents are used in conjunction with an ultrasound system that can perform simultaneous image-guidance to target tumors, focused ultrasound delivery to cause cavitation, and imaging of cavitation emissions as a treatment monitor, a complete drug delivery system is possible that can enhance the penetration and distribution of oncology agents into solid tumors. Development of novel, clinical cavitation agents requires, from their early prototyping through to their ultimate product release, that the detection and characterization of their cavitation performance is considered. Additionally, the methods and equipment used to detect and image cavitation both in preclinical and clinical drug delivery systems must be similarly assessed to ensure that the cavitation "dose" estimated from the agents is understood in precision to ensure reliable drug delivery. To achieve these objectives, OxSonics continues to develop, alongside its new bubble formulations, a cavitation metrology program that investigates cavitation performance and quantification, cavitation test articles and protocols, validation of these methods, and equipment calibration and qualification. This talk explores all the aspects of this from the prototyping stage to clinical deployment.

**11:25–11:55**
**Panel Discussion**

## Session 4aBAb

# Biomedical Acoustics and Signal Processing in Acoustics: Novel Ultrasound Beamforming Techniques and Their Applications I

Jian-yu Lu, Cochair
*Bioengineering, The University of Toledo, 2801 West Bancroft Street, Toledo, OH 43606*

Hérve Liebgott, Cochair
*CREATIS, Univ. of Lyon 1, Lyon, France*

**Chair's Introduction—9:00**

### *Invited Papers*

**9:05**

**4aBAb1. Spatial-coherence-based beamforming for image quality enhancement in high frame-rate ultrasound imaging.** Giulia Matrone (Dept. of Elec., Comput. and Biomedical Eng., Univ. of Pavia, Via Ferrata 5, Pavia 27100, Italy, giulia.matrone@unipv.it) and Alessandro Ramalli (Dept. of Information Eng., Univ. of Florence, Florence, Italy)

Several high frame-rate ultrasound imaging techniques have been recently proposed which allow to increase image acquisition rates by transmitting multiple focused beams simultaneously or defocused waves, and reconstructing multiple scan lines in parallel. These methods, which include for example multi-line transmission, plane/diverging wave imaging, and parallel beamforming, have proved to be successful in increasing the frame-rate, but on the other hand they have also showed some limitations in terms of the achievable image quality, partly degrading image contrast and resolution. Different solutions have been proposed to address this problem so far, such as the use of advanced receive beamforming algorithms able to improve the beam shape and the quality of the obtained image. Among these, beamformers based on backscattered signals spatial coherence, as e.g. Filtered Delay Multiply and Sum beamforming, Coherence-Factor-based methods, and Short Lag Spatial Coherence Imaging, have gained increasing attention for their ability to enhance image contrast and suppress clutter. In this presentation, an overview of some recent results obtained by jointly exploiting high frame-rate ultrasound imaging and coherence-based beamforming methods will be presented, showing through experimental tests the performance improvement achievable with these techniques.

**9:25**

**4aBAb2. Adaptive beamformer with echo statistics.** Hideyuki Hasegawa (Univ. of Toyama, 3190 Gofuku, Toyama 9308555, Japan, hasegawa@eng.u-toyama.ac.jp), Takumi Akamatsu, Masaaki Omura, and Ryo Nagaoka (Univ. of Toyama, Toyama, Japan)

Delay-and-sum (DAS) beamforming is commonly used in commercial scanners and computationally efficient for real-time imaging. However, the ability to suppress off-axis signals is limited. A minimum variance (MV) beamformer realizes a superior performance in suppression of off-axis signals and improves lateral resolution significantly. On the other hand, MV degrades the contrast-to-noise ratio (CNR) compared to DAS because it alters speckle statistics. In this study, we developed a method for improvement of CNR in MV beamforming by evaluating envelope statistics of echo signals. It is well known that the envelope statistic of speckle echoes from a random medium obeys the Rayleigh distribution. The echo envelope statistics were evaluated using the shape parameter of the Nakagami distribution. The proposed beamformer is worked as DAS for speckle echoes and MV for non-speckle echoes by referring to the Nakagami shape parameter. In the phantom experiment, the lateral resolution of MV was 0.18 mm, which was significantly better than 0.53 mm obtained by DAS. However, CNR was degraded from 6.73 dB to 4.05 dB by MV. The proposed beamformer realized a lateral resolution of 0.25 mm, which was significantly better than DAS, with a CNR value of 6.03 dB, which was comparable to DAS.

**9:45**

**4aBAb3. High-frame-rate limited-diffraction beam imaging.** Jian-yu Lu (Bioengineering, The Univ. of Toledo, 2801 West Bancroft St., Toledo, OH 43606, jian-yu.lu@ieee.org)

Limited-diffraction beams such as Bessel beam, X wave, and array beam are a class of beams that in theory do not spread as they propagate to infinite distance. When realized with a finite aperture, these beams have a large depth of field. In this paper, limited-diffraction beams and their applications to high-frame rate imaging are reviewed and further studied. Using a limited-diffraction beams (in either transmission or reception, or both, or with spatial Fourier transform of received signals), images can be reconstructed in Fourier domain. This significantly reduces the amount of computation in 3D imaging since FFT can be used. As the bandwidth of the transducer is increased, the image reconstructed can cover a larger field of view since a frequency in limited-diffraction beam corresponds to one angular direction of the beam (similar to the "frequency encoding" in the magnetic resonance imaging or MRI). Using the angular

4a THU. AM

dependence with frequency, focused limited-diffraction beams can be used for high-frame-rate and high-resolution tissue property imaging. When limited-diffraction beams are used in transmissions, no time delay (only amplitude weighting) is needed, which simplifies imaging system hardware. Also, flow vector imaging can be obtained with limited-diffraction beams using sine and cosine (quadrature) aperture weighting.

**10:05**

**4aBAb4. Beamforming of 3D ultrasound signals with full or sparse probes: New perspectives for volumetric imaging.** François Varray (Univ. Lyon, INSA-Lyon, Université Claude Bernard Lyon 1, UJM-Saint Etienne, CNRS, Inserm, CREATIS UMR 5220, U1294, 21 Ave. Jean Capelle, Bat Leonard de Vinci, Villeurbanne 69621, France, francois.varray@creatis.insa-lyon.fr)

The advances of 2D arrays and 3D ultrasound systems open new perspectives for volumetric US imaging. Coupled with the ultrafast acquisitions, the quantity of acquired data to process expand and the beamforming could become a bottleneck in the processing, especially if volumes are reconstructed in real time. In this work, we propose to study the impact of the beamformer on 3D volumetric images, such as coherence factor, sign coherence factor, phase coherence factor or pDAS. Indeed, such beamformers, adaptive or not, can be easily transfer to 3D but their studies are limited in 3D volumetric situation. Moreover, the introduction of fully populated array required the used of advance and expensive systems. To this end, the utilisation of sparse probes is a promising solution to reduce the number of signals to manage and the complexity of the used hardware. However, the impact on the reconstructed volumes have to be evaluated and the presented beamforming strategies will be presented in both fully and sparse probes.

**10:25–10:40 Break**

**10:40**

**4aBAb5. On the use of denoing algorithms for ultrasound beamforming.** Sobhan Goudarzi (Dept. of Elec. and Comput. Eng., Concordia Univ., Montreal, QC, Canada), Adrian Basarab (CREATIS, Univ. of Lyon, Université Claude Bernard Lyon 1 CREATIS UMR 5220 Bât. Léonard de Vinci, 21 Ave. Jean Capelle, Villeurbanne Cedex 69621, France, adrian.basarab@creatis.insa-lyon.fr), and Hassan Rivaz (Dept. of Elec. and Comput. Eng., Concordia Univ., Montreal, QC, Canada)

Beamforming in receive, whose objective is to estimate an image from raw RF data acquired by the probe piezoelectric elements, plays a crucial role in ultrasound imaging. The standard method, called delay and sum (DAS) and implemented in most of the commercial scanners, consists in coherently summing the RF signals, providing the backprojection solution of the inverse problem of beamforming. Despite real-time properties, DAS results into images with limited spatial resolution and contrast. The literature of ultrasound beamforming is rich and mainly consists in alternatives to DAS based on non-adaptive or adaptive (e.g., minimum variance, coherence factor) methods or image reconstruction algorithms in the Fourier domain. Furthermore, inverse problem formulations have been shown to be well-adapted to ultrasound beamforming. They consist in minimizing a cost function formed by two terms: a data fidelity term modelling the acquisition setup, and a regularization term. The choice of the latter is not straightforward in ultrasound imaging, mainly because of the need to conserve statistical properties of the speckle. In this paper, denoising algorithms are shown to be good regularizers for ultrasound beamforming, providing a good performance in spatial resolution and contrast gain, without deteriorating the quality of the speckle texture.

**11:00**

**4aBAb6. Correlation-based ultrasound imaging: A diagnostic enabler.** Maxime Bilodeau, Tamara Krpic, Nicolas Quaegebeur (Mech. Eng., Universite de Sherbrooke, Sherbrooke, QC, Canada), and Patrice Masson (Mech. Eng. Dept., Universite de Sherbrooke, 2500 blvd Universite, Sherbrooke, QC J1K 2R1, Canada, Patrice.Masson@USherbrooke.ca)

Correlation-Based (CB) ultrasound imaging relies on the correlation of measured signals with signals from a simulated or measured database. The CB imaging algorithm *Excitelet* has demonstrated better ultrasound image quality in Non-Destructive Testing (NDT) applications. In medical applications, ultrasound imaging is characterized by weak image dynamics when compared with Magnetic Resonance Imaging (MRI) or Computed Tomography (CT). However, portable, low-cost, and reliable diagnostic tools based on ultrasound imaging could help reducing the delays between patient assessment and treatment. In order to see more ultrasound-based diagnostics, collective research efforts are still required.Recent work on the use of the CB framework in medical imaging has enabled new diagnostic modalities. Indeed, CB imaging extends the field of view due to the near perfect compensation of the intrinsic properties of the transducers (directivity, dynamics, imperfections, lenses). New automated tools allow using a measured signal database, as opposed to simulated database previously required in CB imaging. Due to its frequency formulation, a realistic signal database enables local acoustical impedance estimation through the reconstruction of the reflectors/diffusers local phase. It is herein proposed to overlay this color-coded phase information, as typically done in Doppler imaging, to enrich the ultrasound image, and facilitate image interpretation.

**11:20**

**4aBAb7. From ultrafast ultrasound imaging to "matrix imaging".** Mathias Fink (Langevin Inst., ESPCI Paris, 1 rue Jussieu, Paris 75005, France, mathias.fink@espci.fr)

Both ultrasound medical imaging and NDT requires reflection-mode detection. As tissues, as well as many metallic samples, are complex disordered media, containing random distribution of scatterers, these techniques suffer various limitations as distortion induced by aberrating layers. A multi-illumination strategy is the solution to solve these problems. It was first used in the context of ultrafast imaging with multiple plane wave illumination. I will show that, recording a "reflection matrix" provides enough information to greatly improve beam-forming techniques. I will describe how to extract from the coherence properties of this reflection matrix enough information both to compensate the effects of aberrating layers and to provide quantitative information. Various strategies to measure this matrix and to exploit it will be discussed. This is the domain of "matrix imaging" recently developed for ultrasound imaging.

11:40

**4aBAb8. Distributed aberration correction in diagnostic ultrasound by time-delay computation from sound speed estimates using common midpoint gathers.** Jeremy J. Dahl (Radiology, Stanford Univ., 136 Hudson Hall, Box 90281, Durham, NC 27708, jeremy.dahl@duke.edu), Thurston Brevett (Elec. Eng., Stanford Univ., Palo Alto, CA), and Sergio Sanabria (Radiology, Stanford Univ., Stanford, CA)

We describe a method of distributed aberration correction from local sound speed estimates of tissue. In this case, compensated beamforming delays are computed using the eikonal equation to account for refracted paths of the ultrasound waves based on an estimation of the local sound speed of the tissue. Signals acquired from a multistatic synthetic aperture sequence are used for image reconstruction in order to refocus the ultrasound beam at every image point without re-transmission. We estimate sound speed using the common midpoint gathers (CMGs) of the synthetic aperture signals, which have high signal coherence that can be used for accurate time measurements. These time measurements are used in overlapping ray paths traced onto a discrete grid to solve an inverse ultrasound computed tomography problem. We separately utilize absolute delay measurements at each depth (to capture axial variations) together with differential time-delays between CMG gathers (to capture lateral variations). For generalized sound speed reconstruction, we balance the contributions of both terms with regularization. In simulations, local sound speed was reconstructed with an rmse of 5.3 m/s in four-layered media and 6.0 m/s in focal lesions. In multilayer propanol/agar phantoms, local sound speed was reconstructed with an rmse of 5.0 m/s.

THURSDAY MORNING, 8 DECEMBER 2022 — RAIL HEAD, 9:00 A.M. TO 11:20 A.M.

## Session 4aMU

## Musical Acoustics: Modeling and Simulation of Physical Effects in Sound Reproduction

Preston S. Wilson, Cochair
*Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 East Dean Keeton Street, Mail Stop C2200, Austin, TX 78712-0292*

Mark Rau, Cochair
*Music, Stanford University, 660 Lomita Court, Stanford, CA 94305*

**Chair's Introduction—9:00**

### *Contributed Paper*

9:05

**4aMU1. Brief overview and demonstration of some commercial off-the-shelf modeling tools.** Preston S. Wilson (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 East Dean Keeton St., Mail Stop: C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

This paper is intended to lead the special session on "Modeling and Simulation of Physical Effects in Sound Reproduction." To begin, a brief overview of linear systems theory is presented to set the stage for demonstrating some tools that use impulse responses and convolution to generate various "effects." This will be followed by a brief overview and demonstration of some commercially available modeling tools, ranging from the emulation of amplifiers and microphones, to reverberation and microphone placement. Some examples of the modeling of electromechanical instruments will be shown as well, including the Hammond organ and the Leslie rotating speaker.

## *Invited Papers*

**9:20**

**4aMU2. Modeling and correction of piezoelectric string instrument pickups.** Champ C. Darabundit (Music Technol., McGill Univ., 555 Sherbrooke St. West, Montréal, QC H3A 1E3, Canada, champ@ccrma.stanford.edu) and Mark Rau (Music, Stanford Univ., Stanford, CA)

Under-string piezoelectric pickups are often used to amplify and record stringed instruments such as guitars and violins. The mounting configuration of pickups differ with the most common placements being under the saddle, bridge, or directly under the strings, where the pickup functions as the saddle. Each of these placements, as well as the design of the piezoelectric pickup imparts a particular characteristic on the sound of the instrument. We investigate the equalization imparted by the pickup and present methods to correct for the frequency domain differences to produce a neutral sound. Additionally, piezoelectric sensors can exhibit nonlinear behavior such as hysteresis. Measurements are made with an attempt to model and correct for any nonlinear behavior observed.

**9:40**

**4aMU3. Measurement and modeling of an electromechanical spring reverb device.** Kyle S. Spratt (Image-Line Software, Franklin Rooseveltlaan 348 D, Ghent B-9000, Belgium, sprattkyle@gmail.com) and Jonathan S. Abel (CCRMA, Stanford Univ., Stanford, CA)

A spring reverb is an electromechanical device used to artificially reverberate an audio signal, i.e. to impart a sense of spaciousness to the audio signal, as if the sound were being emitted into a reverberant acoustic space. The device consists of a number of helical springs set into motion by electrically driven magnetized beads, with the elastic wave motion along the springs being analogous to acoustic waves propagating along the various dimensions of a room. While originally invented by the Hammond organ company to add a sense of spaciousness to the sound of their electric organs, spring reverbs have been used in a number of electronic instruments and audio devices over the years, with probably the most iconic and enduring use being in classic electric guitar amplifiers. In this talk, a general overview of the history and technical aspects of spring reverb devices will be given, with particular emphasis placed on the highly dispersive nature of elastic wave propagation along helical springs, which it will be argued is primarily responsible for the distinctive sound of the device. Finally, a simplified computation model will be presented that captures the salient features of a spring reverb device.

**10:00–10:15 Break**

**10:15**

**4aMU4. Physical modelling for analog tape emulation.** Jatin Chowdhury (11 Red Tail Dr., Littleton, CO 80126, chowdsp@gmail.com)

Reel-to-reel magnetic tape recording has a unique and specific sound which is highly sought after by music recording engineers. This talk will discuss the use of physical modelling signal processing to create algorithms which can emulate the sound of any reel-to-reel tape recorder, including ones which may not actually exist in the real world. These algorithms are designed to be used in modern music production software, meaning that they must be able to process audio in real-time with minimal latency, and offer interactive controls to the end user.

## *Contributed Papers*

**10:35**

**4aMU5. Atom Tones: investigating waveforms and spectra of atomic elements in an audible periodic chart using techniques found in music production.** Jill A. Linz (Phys., Skidmore College, 815 N Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu) and Christian Howat (Phys., Skidmore College, Saratoga Springs, NY)

Atom Music was introduced in 2019 as a way to create unique audible tones for each atomic element that are direct translations of that element's spectral lines. Each atomic element produces a unique spectral line pattern that can be recognized as the fingerprint of that element. Sonification is the process of translating non-audible data into audible signals as a way to gain an understanding of the original data. In this paper, sonification is applied to atomic spectra, using technology primarily from music production. These were applied to atomic spectra using additive synthesis methods and analyzed using digital audio workstations. Interest in the audible tones has primarily been in element identification through each tone, as well as those interested in musical interpretation. We investigate what insights can be made by observing the digital waveforms and spectra of each element tone. We consider whether there are patterns within the different element waveforms; if there is any correlation between the elements producing similar beat patterns; and if there are any harmonic relations between electron states that are represented by the spectra itself. Results indicate that this method could be a useful tool in investigating atomic structure.

**10:50**

**4aMU6. Estimating steelpan note class from attack transients.** Colin Malloy (Music, Univ. of Victoria, 3330 Richmond Rd., Victoria, BC V8P 4P1, Canada, malloyc@uvic.ca)

The estimation of fundamental frequency of instruments is an important task in computational audio analysis. The current state of the art methods use neural networks for this task. This process is typically computed periodically over very short segments of a monophonic audio signal so that minute shifts in intonation can be detected. However, the steelpan has discretely tuned notes where the performer has no direct control over pitch once a note has been activated. The activation of a note has great influence over the acoustical properties of the resultant note. Much research has been devoted to the tonality, construction, and acoustical properties of steelpans, but relatively little focuses on the attack transient specifically.This paper evaluates the application of pitch detection methods to the attack transients of steelpan notes. A dataset containing labeled audio samples from multiple tenor steelpans is used for training and evaluation. The accuracy of this approach for pitch detection is compared with established methods applied to both entire notes and only attack transients. Determining a steelpan note's pitch from the attack transient is an important first step in building a robust low latency automatic transcription system that can be used for both analysis as well as live performance.

**11:05–11:20**
**Panel Discussion**

## Session 4aNS

# Noise, Computational Acoustics, Structural Acoustics and Vibration, Signal Processing in Acoustics, and Physical Acoustics: Jet and Launch Vehicle Noise I (Hybrid Session)

Alan T. Wall, Cochair
*Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433*

Kent L. Gee, Cochair
*Department of Physics and Astronomy, Brigham Young University, N281 ESC, Provo, UT 84602*

Caroline P. Lubert, Cochair
*Mathematics and Statistics, James Madison University, 301 Dixie Avenue, Harrisonburg, VA 22801*

**Chair's Introduction—8:30**

## *Invited Papers*

**8:35**

**4aNS1. Simplified prediction of near-field jet coherence using the cross-power spectral density acoustic analogy.** Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, PO BOX 116250, 939 Sweetwater Dr., Gainesville, FL 32611, saem@ufl.edu)

The cross-power spectral density (CPSD) acoustic analogy is a generalization of Lighthill's analogy that directly predicts near-field statistics at multiple spatial points. The methodology was previously applied and validated to predict noise from isotropic turbulence and off-design supersonic jet flows. We present a simplification of the CPSD model for near-field predictions of acoustic coherence from jet fine-scale mixing noise. Validation predictions are presented with a NASA Glenn Research Center experimental jet database. The simplified mathematical model is presented along with the physical significance of each term. The application of the model can be applied to predict loading and coherence on the fuselage of aircraft, rockets, and associated structures. This approach can help designers minimize sonic fatigue or failure in flight-vehicles and support structures.

**8:55**

**4aNS2. Optical-acoustics source analysis of supersonic jet noise reduction using micro vortex generators.** Mohammad Saleem (Univ. of Cincinnati, 745 Baldwin Hall, Cincinnati, OH 45221, saleemmd@mail.uc.edu), Omar L. Rodriguez, Aatresh Karnam, Ephraim Gutmark (Univ. of Cincinnati, Cincinnati, OH), and Junhui Liu (Naval Res. Lab., Washington, DC)

A new supersonic jet noise reduction technology has been developed using Micro Vortex Generators (MVGs) by the collaboration between the University of Cincinnati and the Naval Research Laboratory. MVGs are used on model scale nozzles that are representative of GE F404 engine nozzles. Noise reductions up to $-10$ dB have been observed in both laboratory measurements and LES simulations at conditions related to take off in the overexpanded regime. Analysis of the acoustic field and flow field using Schlieren visualization reveal the noise reduction mechanisms associated with MVGs. Direct visualization of the changes in shock cell spacing, Large Coherent Structures (LCS) formation, and their convective velocity are identified and those changes modify the downstream propagating hydrodynamic waves and the upstream propagating acoustics waves. Spectral Proper Orthogonal Decomposition (SPOD) is utilized to examine the flow sources at frequencies associated with the noise components observed in the acoustic spectra to explain the noise reduction mechanisms of MVGs.



**9:15**

**4aNS3. The full spectrum of broadband shock cell noise.** Christopher Tam (Mathematics, Florida State Univ., 1017 Academic Way, Tallahassee, FL 323064510, tam@math.fsu.edu)

Broadband shock cell noise has been studied by numerous investigators since the work of Harper-Bourne and Fisher (1973). This noise component radiates in the upstream direction together with another noise component, the fine scale mixing noise. In measured noise spectra, broadband shock cell noise shows up as a small peak protruded up from a broad background noise. The spectral shape of the peak has been studied by Tam (1990) and Kuo *et al.* (2015). Good agreements are found between predictions and measurements. This situation begs the question of the shape of the full noise spectrum. The answer lies in devising a way to separate the two noise components. Here, this is done at the raw data level. We know broadband shock cell noise is generated by the interaction between large turbulent structures and shock cells whereas mixing noise is generated by fine scale turbulence. One, therefore, expects a scale separation between the two noise components. Broadband shock cell noise sound pulses are expected to have a longer duration or larger pulses

while turbulent mixing noise much shorter pulses. This idea has now been implemented on a set of F-18E noise data at 28 degrees. Results and relevant details will be reported.

**9:35**

**4aNS4. Deflector shape impact on aero-acoustic noise generation and propagation.** Mara S. Escartí-Guillem (IUMPA, Universitat Politècnica de València, Camino de Vera s/n, Valencia 46022, Spain, mescarti@comet-ingenieria.es), Sergio Hoyas, and Luis M. Garcia-Raffi (IUMPA, Universitat Politècnica de València, Valencia, Spain)

The vibroacoustic loading generated during the launch of space vehicles can cause the failure of electronic and mechanical components. Therefore, the prediction and mitigation of these vibroacoustic levels are crucial to improve the reliability of launchers and payload comfort. Because a properly designed flame deflector has the power to significantly reduce the acoustic pressure level, the aeroacoustics characteristics of diverse types of flame deflectors must be understood. Three different deflector geometries have been analysed: a wedge-type deflector, which is currently used on the VEGA rocket launch pad, a 30-degree inclined deflector, since new studies highlight its noise reduction capacity, and a flat deflector, since the impact on a flat plate is the simplest case of reflection. The sound generation and propagation in the launch platform full domain for each case were studied using dedicated computational fluid dynamics in BSC MareNostrum. To assess noise generation, the main shock waves were identified, and the evolution of the generated sound pressure was assessed. Moreover, the sound pressure levels at the fairing surface have been studied. Further research is focused right now on the use of an efficient solver running on graphics processing units that is capable of computing large-scale turbulence.

**9:55–10:10 Break**

**10:10**

**4aNS5. A nonlinear distortion length of the statistics of high-amplitude propagating noise.** Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil) and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The nonlinear evolution of high-amplitude broadband noise is important to the psychoacoustic perception of high-speed jet noise. One method to characterize the nonlinear evolution of such noise is to consider a characteristic nonlinear distortion length for the signal. A common length scale for this analysis is the shock formation distance of an initially sinusoidal signal. However, application of the shock formation distance for an initially sinusoidal signal to broadband noise, even with the amplitude and source frequency replaced with characteristic values, may lead to inaccurate interpretations. Here we derive an alternative length scale from the evolution of the skewness of the pressure time-derivative of Gaussian noise that may be more appropriate when analyzing the nonlinear evolution of broadband noise signals. This length scale is then used to demonstrate how numerically propagated noise signals are fundamentally different than numerically propagated initially sinusoidal signals.

**10:30**

**4aNS6. Reanalysis of subjective crackle ratings using a logistic curve fit.** S. Hales Swift (Sandia National Labs., P.O. Box 5800, MS 1082, Albuquerque, NM 87123-1082, shswift@sandia.gov) and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Prior work has shown a robust relationship between crackle from high-performance jets and the skewness of the distribution of the first time derivative of the pressure waveform; however, prior efforts have characterized this relationship in terms of a linear relationship between the log of the derivative skewness and the category scaling responses. While the relationship is linear over important portions of its range, the use of a logistic curve fit more fully captures the characteristics of the relationship between log first time derivative skewness and category scaling relationship including implied non-negativity, and saturation. Accordingly, a logistic curve is fit to these data, and the residual magnitudes are compared between the two fitting approaches.

**10:50**

**4aNS7. Application of a crackle-based adjustment to military aircraft noise levels.** Blaine M. Harker (Blue Ridge Res. and Consulting, LLC, 29 N Market St., Ste. 700, Asheville, NC 28801, blaine.harker@blueridgeresearch.com), J. M. Downing, Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Crackle is a sound quality that can be identified in the jet noise of tactical aircraft and may contribute to perceived community noise levels in the vicinity of military airports. Crackle has been linked to the skewness of the waveform pressure derivative, or derivative skewness. Recently, listener trials quantified the perceptual relationships between derivative skewness, loudness, and annoyance. We present a method to calculate a crackle-based adjustment to aircraft noise levels derived from annoyance. This crackle adjustment is applied to a large sample set of community noise measurements in the vicinity of departing military aircraft. Finally, we develop an empirical model to apply crackle-based adjustments to community noise models of tactical aircraft departures. The empirical model allows airport planners to examine how aircraft flight paths, altitudes, and engine conditions affect the crackle adjustment, which is significant in the airport's immediate vicinity and along the aircraft departure path. [Work funded by an AFRL SBIR.]

## *Contributed Paper*

**11:10**

**4aNS8. The Federal Aviation Administration (FAA) allows Americans to be exposed to unsafe levels of aviation noise.** Daniel Fink (The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733, DJFink@thequietcoalition.org)

The American Public Health Association states, Noise is unwanted and/or harmful sound. The FAA considers noise an annoyance (cf. Schultz Curve), and adopted 65 dBA Day-Night Level (DNL) as the threshold for significant aviation noise, below which residenial land uses are compatible. This is not safe for humans. According to the Environmental Protection Agency, safe noise levels are 45 dB for indoor noise, 55 dB for outdoor noise, and DNL 55 dB. The World Health Organization recommends nighttime noise levesl of only 45 dB. Recent research has established conclusively that noise is a stressor, causing a cascade of physiological events involving the autonomic nervous system, stress hormones, and inflammatory pathways, in turn causing cardiovascular disease, including hypertension and heart attacks, and other adverse health effects. Epidemiological studies demonstrating these effects have been confirmed by human and animal research. The biological mechanisms are now understood at the cellular, subcellar, molecular, and genetic levels. Additionally, aviation noise disproportionately affects poor and minority communities and vulnerable populations. A summary of this research will be presented. No more research is needed to know that aviation noise is hazardous to health. The FAA must establish lower noise standards to protect Americans exposed to aviation noise.

THURSDAY MORNING, 8 DECEMBER 2022 LIONEL, 8:00 A.M. TO 12:00 NOON

# Session 4aPAa

## Physical Acoustics and Biomedical Acoustics: Interaction of Light and Sound I (Hybrid Session)

E. Carr Everbach, Cochair
*Engineering, Swarthmore College, 500 College Avenue, Swarthmore, PA 19081*

Jason L. Raymond, Cochair
*Dept. of Engineering Science, University of Oxford, 17 Parks Road, Oxford OX1 3PJ, United Kingdom*

**Chair's Introduction—8:00**

## *Invited Papers*

**8:05**

**4aPAa1. Light, sounds, and magnetism: Enhanced image-guided adoptive cell therapy using T cells tagged with multifunctional nanoparticles.** Kelsey Kubelick, Myeongsoo Kim, Jinhwan Kim, Jeungyoon Lee, Anamik Jhunjhunwala, Anthony Yu (Georgia Inst. of Technol., Atlanta, GA), and Stanislav Emelianov (Georgia Inst. of Technol., 777 Atlantic Dr., Atlanta, GA 30332, stanislav.emelianov@gmail.com)

Methods to noninvasively increase and assess T cell infiltration are critical for success of adoptive cell therapy (ACT) of solid tumors where only a small fraction of adoptive T cells typically accumulates at the target. We developed an approach based on photoacoustic (PA) and ultrasound (US) imaging to visualize magnetic field driven accumulation of T cells tagged with photo-magnetic nanoparticles (PMNPs). Specifically, Au@$Fe_3O_4$ shell-core PMNPs (200 nm diameter), absorbing at 1064 nm wavelength, were synthesized and characterized using a UV-Vis-NIR spectrophotometry, dynamic light scattering (DLS), and transmission electron microscopy (TEM). Ovalbumin (OVA)-targeted PMNP-tagged OT1 murine primary T cells were injected intravenously into OVA-expressing tumor-bearing C57BL/6 mice. US/PA imaging (20 MHz, 1064 nm and 680–970 nm, Vevo LAZR, Visualsonics Inc.) of the primary tumor and regional lymph nodes showed accumulation of PMNP-tagged T cells. Our results indicate feasibility of the T cell tagging with PMNPs and magnetic delivery of adoptive PMNP-tagged T cells using US/PA imaging thus providing critical imaging feedback to improve the adoptive T cell therapy of solid tumors. Overall, our studies show that a synergistic combination of US/PA imaging and magnetic delivery of nanoparticle (NP)-tagged adoptive T cells can expedite development, translation, and expansion of ACT.

**8:25**

**4aPAa2. Nanoparticle-enhanced photoacoustic diagnosis and photothermal treatment of small-animal early-stage liver cancer.**
Puxiang Lai (Dept. of Biomedical Eng., Hong Kong Polytechnic Univ., ST410, Hung Hom, Hong Kong, puxiang.lai@polyu.edu.hk), Xiazi Huang, Yingying Zhou, and Weiran Pang (Dept. of Biomedical Eng., Hong Kong Polytechnic Univ., Kowloon, Hong Kong)

Synergy of light and sound, such as photoacoustic imaging, has demonstrated promising potentials in advancing the state-of-the-art of biomedical imaging by ultrasonically detecting the absorption of light, no matter the light is diffusive or not. As a result, optical contrast can be revealed with ultrasound or sub-ultrasound spatial resolution, which can be exploited to map sensitively the early tissue changes associated with the onset and development of early-stage diseases. The sources of photoacoustic emissions, however, could be very complicated within living biological tissue, as many tissue constituents may absorb light and generate heat to emit the ultrasonic waves, which considerably reduces the contrast between the target and the background. Therefore, exogenous agents, such as nanoparticles, that have absorption spectrum distinctive from that of background tissues have been widely used in the field to enhance the photoacoustic detection contrast and sensitivity at selected optical wavelengths. This talk summarizes our recent efforts in this direction that have explored various nanoparticles for more robust and more sensitive diagnosis and treatment of early-stage liver cancer based on photoacoustic imaging and photothermal effect. While the studies are based on merely small animal models, they may inspire further explorations towards preclinical and clinical trials.

**8:45**

**4aPAa3. Realization of clinical translation of photoacoustic imaging.** Sophinese Iskander-Rizk (Precision and Microsystems Eng., Delft Univ. of Technol., Aronskelkstraat 109, Rotterdam 3053XB, the Netherlands, s.iskander-rizk@erasmusmc.nl) and Gijs van Soest (Biomedical Eng., Erasmus MC, Rotterdam, the Netherlands)

Photoacoustic (PA) imaging capitalizes on sound and light to probe tissue optical absorption at depth. Owing to acoustic propagation of absorbed diffuse photons it exceeds imaging depth of most pure optical methods. Moreover, since the detection of PA signals relies on ultrasound transducers, the modality inherently allows for acquisition of co-registered ultrasound images, enabling a structural and functional examination of tissue. Diagnosis of diseases such as atherosclerosis could benefit greatly from uncovering such dual information. Clinical translation of photoacoustic imaging, led by breast cancer imaging application is—pending a couple of hurdles—on the brink of happening. Systematic approaches in assessing contrast through spectroscopy, designing acquisition systems through optimization for both PA/US signal sensitivity and image metrics and finally designing instrumentation to account for tissue access limitations have been researched and show promising results. In this talk, we discuss the final challenges in translating photoacoustic imaging to the clinics with respect to contrast, instrumentation and system development.

**9:05**

**4aPAa4. Optically excited nanoparticle enhanced high intensity focused ultrasound therapy of *in vivo* cancer models.** Nicola Ingram, Teklu Egnuni (St. James' University Hospital, Leeds Inst. of Medical Res., Leeds, West Yorkshire, United Kingdom), Steven Freear (School of Electron. and Elec. Eng., Univ. of Leeds, Leeds, West Yorkshire, United Kingdom), P. Louise Coletta (St. James' University Hospital, Leeds Inst. of Medical Res., Leeds, United Kingdom), and James McLaughlan (School of Electron. and Elec. Eng., Univ. of Leeds, Leeds, West Yorkshire LS2 9JT, United Kingdom, j.r.mclaughlan@leeds.ac.uk)

The use of plasmonic gold nanoparticles (AuNP) that have been actively targeted to a specific tumour region is a highly active area of research for both drug delivery and hypothermia applications. One application is providing *in situ* cavitation nuclei for enhancing high intensity focused ultrasound (HIFU) ablation. When the nanoparticles are simultaneously exposed to HIFU and pulsed laser illumination, microscopic vapour bubbles form providing a controllable way to nucleate cavitation. Using an EGFR-expressing head and neck xenograft model, the biodistribution of targeted and untargeted AuNPs were examined using multispectral, whole body photoacoustic imaging, and inductively coupled plasma mass spectrometry (ICP-MS) analysis following systemic or intratumoral injections. Systemic delivery showed negligible concentration of AuNRs in the tumour; however, intratumoral injection increased its concentration, but remained in the tumour at high concentration for the first 48hrs. Evaluation of ablation performance was undertaken using the same tumour model. An intratumoural injection of untargeted particles, with single 10 s continuous wave HIFU exposures for peak positive/negative pressures of 2.4/2.0 MPa, with simultaneous illumination of 1064 nm pulsed laser light with a surface energy density of 70 mJ/cm$^2$. Post treatment tumour volume was monitored using high frequency ultrasound imaging and calliper measurements, to evaluate exposure efficacy.

**9:25**

**4aPAa5. Assessing the microarchitecture and microenvironment of cancer *in vivo* by photoacoustic imaging techniques.** Xueding Wang (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., 2125 Gerstacker, Ann Arbor, MI 48109, xdwang@umich.edu)

Majority of current studies on photoacoustic (PA) imaging are focused on the total signal magnitudes as the reflection of the macroscopic optical absorption by specific chemical contents at single or multiple optical wavelengths. Our recent research has demonstrated that the frequency domain power distribution of radio-frequency PA signals contains the microscopic information of the optically absorbing materials in biological samples. In this research, I will present our recent development of quantitative PA methods for *in vivo* evaluation of histological microarchitectures of biological tissues as well as potential clinical applications of these methods such as the assessment of prostate cancer aggressiveness. In addition, powered by our recently developed polyacrylamide hydrogel nanoparticles containing a variety of functional contrast agents, multi-wavelength PA imaging plus spectral unmixing technique can enable quantitative mapping of the chemical and metabolic properties in cancer microenvironments, such as tissue oxygenation, pH level, and ion concentrations. Imaging of these chemical makeups of an entire tumor *in vivo*, non-invasively, with high sensitivity and high spatial resolution can shed new light to the understanding of cancer onset, progress, and responses to therapies.

**9:45–10:00 Break**

10:00

**4aPAa6. Detection of HIFU lesions by optical coherence tomography.** Jason L. Raymond (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Manuel Marques (School of Physical Sci., Univ. of Kent, Centerbury, Kent, United Kingdom), E. Carr Everbach (Eng., Swarthmore College, 500 College Ave., Swarthmore, PA 19081, ceverba1@swarthmore.edu), Michael Hughes (School of Physical Sci., Univ. of Kent, Canterbury, Kent, United Kingdom), Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and Adrian Podoleanu (School of Physical Sci., Univ. of Kent, Canterbury, Kent, United Kingdom)

The use of high-intensity focused ultrasound (HIFU) to induce irreversible changes in tissue due to heating is well established. We have shown that changes in tissue optical properties (scattering and absorption coefficients) could be used as a proxy to improve sensing and imaging of HIFU lesion formation, as an alternative to conventional methods such as thermometry. Optical coherence tomography (OCT) is a non-invasive optical imaging method which relies on low-coherence interferometry to determine the depth of individual scattering centres within the tissue. Previous studies have demonstrated that OCT signals are sensitive to morphological changes in heated tissue, likely due to denaturation of proteins concomitant with formation of crosslinked structures. The goal of this study was to assess the use of OCT for sensing and imaging HIFU lesions. We demonstrate the feasibility of imaging near-surface lesions in *ex vivo* chicken breast tissue exposed to HIFU. This technique has potential for detecting changes in optical properties corresponding to the progression of surface lesion formation which are antecedents of skin burn during HIFU exposures, thereby increasing safety and reducing treatment times.

10:20

**4aPAa7. Opto-mechanical material characterization via dual-geometry Brillouin confocal microscopy.** Antonio Fiore (HHMI Janelia Res. Campus, 19700 Helix Dr., Ashburn, VA 20147, fiorea@janelia.hhmi.org) and Giuliano Scarcelli (Univ. of Maryland, College Park, MD)

Brillouin scattering is a light-matter interaction that induces a frequency shift in the scattered photons. Such shift directly depends on the optical and mechanical properties of the material itself. In the last decade, Brillouin frequency shift has become a reliable proxy for mechanical stiffness in biological samples such as cells and tissues, with the subsequent development of Brillouin confocal microscopy; however, such correlation does not apply to samples with high degree of inhomogeneity. Recently, we showed that a dual geometry Brillouin microscopy enables the direct measurement of refractive index and speed of sound. We now report an improved dual-geometry microscopy technique that allows measurement of optoacoustical properties within a confocal volume. This goal has been achieved through the measurement of frequency, linewidth, and intensity of Brillouin components of the scattered light spectrum. Finally, we proved our three-dimensional imaging capability by mapping, with micron-scale resolution, the refractive index, density, and viscoelastic properties of a liquid-liquid phase separation system. This technique can be applied to a multitude of biological heterogenous scenarios such as nucleolus characterization, high concentration lipid sites, dynamic protein aggregates and more.

10:40

**4aPAa8. Multispectral photoacoustic fluctuation imaging for full visibility $SO_2$ imaging.** Guillaume Godefroy (LiPhy, Université Grenoble Alpes, CNRS, Grenoble, France), Bastien Arnal (LiPhy, Université Grenoble Alpes, CNRS, 140 rue de la Physique, CS 47100, Grenoble Cedex 9 38058, France, bastien.arnal@univ-grenoble-alpes.fr), and Emmanuel Bossy (LiPhy, Université Grenoble Alpes, CNRS, Grenoble, France)

Photoacoustic (PA) imaging (PAI) images blood vessels through the hemoglobin contrast. However, conventional images are affected by visibility artefacts which prevents seeing all the blood vessels morphology. We introduced PA fluctuation imaging (PAFI) exploiting the absorption fluctuations due to blood flow. Here, we demonstrate how PAFI enhances the image quality and elaborate if it can be used for quantitative multispectral (MS) imaging for 3D blood oxygenation ($SO_2$) imaging. A spherical array (256 elements, 8 MHz, Imasonic) connected to a Verasonics Vantage scanner, was coupled to the laser (Innolas Spitlight) through a fiber bundle (Ceramoptek). The PAFI sequence consisted in acquiring 1000 frames at 100 Hz repetition rate while scanning continuously the wavelength from 680 nm to 880 nm. Image processing resulted in PAFI image per wavelength. The compensation of predictable effects of pertubations provided quantitative $SO_2$ images with visibility and contrast enhancement. The technique was validated in glass capillaries and in the chicken embryo model by comparing the $SO_2$ values with the one obtained in the visible structures using conventional imaging (resp. 3.5% and 9% errors). Thus, MS-PAFI can provide quantitative $SO_2$ full-visibility 3D imaging with an intrinsic specificity to blood which simplifies spectral unmixing.

11:00

**4aPAa9. Deep learning for fast photoacoustic wave simulations.** Ko-Tsung Hsu, Steven Guan (Bioengineering, George Mason Univ., Fairfax, VA), and Parag V. Chitnis (Bioengineering, George Mason Univ., 4400 University Dr., 1J7, Fairfax, VA 22030, pchitnis@gmu.edu)

Photoacoustic tomography involves absorption of pulsed light and subsequent generation of ultrasound, which when detected using an array of sensors can produce clinically useful images. Simulation tools for photoacoustic wave propagation have played a key role in advancing photoacoustic imaging by providing quantitative and qualitative insights into parameters affecting image quality. Classical methods for numerically solving the photoacoustic wave equation rely on a fine discretization of space and can become computationally expensive for large computational grids. In this work, we apply Fourier Neural Operator (FNO) networks as a fast data-driven deep learning method for solving the 2D photoacoustic wave equation in a homogeneous medium. Comparisons between the FNO network and pseudo-spectral time domain approach demonstrated that the FNO network generated comparable simulations with small errors and was an order of magnitude faster. Moreover, the FNO network was generalizable and could generate simulations not observed in the training data.

**11:20**

**4aPAa10. Fiber Bragg grating technology for biomedical ultrasound applications.** Himanshu Shekhar (Elec. Eng., Indian Inst. of Technol. Gandhinagar, AB6/207, IIT Gandhinagar Campus, Near Palaj, Gandhinagar, Gujarat 382355, India, himanshu.shekhar@iitgn.ac.in), Kuldeep Jajoria, Chandan K. Jha, and Arup L. Chakraborty (Elec. Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India)

Fiber Bragg gratings (FBGs) are a class of optical sensors that have been used widely in non destructive testing and monitoring. These sensors are compact, robust, inexpensive and immune to electromagnetic interference. FBGs serve as wavelength selective mirrors that reflect light at the Bragg wavelength. An applied strain results in a shift in the Bragg wavelength, which can be used to detect physical parameters such as temperature and pressure. Recent work suggests that FBGs could have potential for measurements in the biomedical ultrasound frequency range. In this presentation, I will provide an overview of ongoing work in our group on employing FBGs for applications in therapeutic ultrasound. Specifically, I will discuss calibration of these sensors along with their use in passive cavitation detection. I will also present recent results on detecting nonlinear waveforms relevant to focused ultrasound along with temperature measurement for thermal therapy. We envisage that these sensors will be useful in laboratory research environments, and for specialized therapy applications in the clinical setting.

**11:40**

**4aPAa11. Photoacoustic non-human primate brain imaging.** Xinmai Yang (Dept. of Mech. Eng., Univ. of Kansas, 1530 W 15th St., 3138 Learned Hall, Lawrence, KS 66045, xmyang@ku.edu) and Xueding Wang (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Non-human primates (NHPs) play significant roles in brain research because of the physiological similarities between humans and NHPs. Current functional brain imaging techniques for NHPs are difficult to meet the required high spatiotemporal resolution for behavior active NHPs. In this presentation, we review our initial study on evaluating the feasibility of photoacoustic imaging (PAI) for monitoring hemodynamic responses in the NHP brains due to various functional activities. PAI systems, including array-based photoacoustic computed tomography (PACT) system and photoacoustic microscopy (PAM), were used to detect hemodynamic responses in the NHP brains through a cranial window. The NHPs were subjected to different functional stimulations or actively performing behavioral tasks. Strong increases in PA signal amplitude during functional activities can be detected with single-blood-vessel spatial resolution in the cortex and subcortical regions of the NHP brains. Realtime-temporal-resolution hemodynamic response can also be obtained for the detected blood vessels in the brain. Our results demonstrate that PAI can reliably detect brain activations in NHPs with high spatiotemporal resolution.

# Session 4aPAb

## Physical Acoustics: General Topics in Physical Acoustics II

Joel Mobley, Chair
*Physics and Astronomy, University of Mississippi, PO Box 1848, 108 Lewis Hall, University, MS 38677*

### *Contributed Papers*

9:00

**4aPAb1. Errors using the spatially averaged free-field pressure approximation for description of the receiving properties of piezoelectric transducers.** Eivind N. Mosland (Dept. of Phys. and Technol., Univ. of Bergen, Allégaten 55, Bergen 5007, Norway, eivind.mosland@uib.no), Jan Kocbach (NORCE Norwegian Res. Ctr. AS, Bergen, Norway), and Per Lunde (Dept. of Phys. and Technol., Univ. of Bergen, Bergen, Norway)

In modelling of ultrasonic measurement systems and calculation of diffraction corrections for such systems, it is often assumed that the spatially averaged free-field pressure over the front surface of a piezoelectric transducer, $\langle p \rangle$, can be used to calculate its voltage output. It is of interest to investigate whether errors due to this $\langle p \rangle$-approximation are acceptable for piezoelectric transducers applied in high-precision ultrasonic measurements in fluids. 3D-axisymmetric finite element (FE) modelling is used to investigate such errors in a transmit-receive measurement setup employing two identical transducers. The approximate output voltage is found by $V_{out} = \langle p \rangle M_V$ and compared to the output voltage from FE modelling of the transmitter-medium-receiver system, $V_{out,ref}$, using the appropriate boundary conditions at the two transducers. $M_V$ is the open-circuit free-field receiving voltage sensitivity of the receiver for normally incident plane wavefronts, calculated assuming spherical reciprocity. Two examples of piezoelectric transducer pairs are studied, operated in the frequency range of their lower radial modes. It is shown that the $\langle p \rangle$-approximation may introduce notable deviations between $V_{out}$ and $V_{out,ref}$, of up to approximately 40° for the phase angle and 2 dB for the magnitude. The deviations depend on the fluid medium, separation distance, frequency, and the transducer's construction and vibrational characteristics.

9:15

**4aPAb2. A ray tracing approach to focusing ultrasonic beams in isotropic and anisotropic solids.** Lauren Katch (Eng. Sci. and Mech., Penn State Univ., 212 Earth and Eng. Sci. Bldg., State College, PA 16802, Luk50@psu.edu) and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

Focal depths in solid materials are generally calculated using analytical expressions that rely on the paraxial approximation (i.e., small angles related to shallow focusing). However, with increasing use of highly focused probes for acoustic microscopy and backscattering studies, these assumptions are often challenged. Additional challenges are posed by the presence of material anisotropy within the solid. In this presentation, we present a ray tracing based approach to characterize ultrasonic beam focusing when using highly focused ultrasonic transducers. The results are compared to traditional methods to calculate focal regions to evaluate the range of applicability for isotropic and anisotropic materials. For isotropic solids, a non-paraxial focusing equation is derived and compared to the conventional focusing equation for both normal incidence and oblique incidence immersion setups. Both focusing equations are compared through ray diagrams where the proximity to the true geometric focus is explored. The proposed focusing equation results in a closer approximation to the geometric focus, a smaller beam cross-section, and a greater time convergence compared to the conventional focusing equation. Lastly, the results are expanded to anisotropic media where potential for multiple foci and complex beam behavior arise.

9:30

**4aPAb3. Noninvasive tracking of solidification using ultrasound.** Caeden Smith (Eng. Sci. and Mech., Penn State, 212 Earth and Eng. Sci. Bld., University Park, PA 16802, cks5474@psu.edu) and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State, University Park, PA)

Noninvasive tracking of solidification is typically done by monitoring the sample's surface temperature. This approach is error-prone and not always appropriate for complex geometries and often does not provide complete information on the sample's transitioning phases. This feasibility study investigates using ultrasonic waves to noninvasively track solidification in wax. Ultrasonic sensors are used in through-transmission mode to monitor the response over the solidification period. The difference in speed of sound between the wax's liquid and solid phase allows for the solid fraction of the sample to be calculated over the solidification period and the time of complete solidification to be identified. Changes in attenuation during solidification are consistent across experiments. These changes are likely to provide additional information about the cast once they are fully understood. Temperature probes are used to verify the results obtained from the ultrasonic data. The ultimate goal of the project is to use a metal sample and connect information from the ultrasonic signal to the material properties and grain structure of the cast using noninvasive ultrasound technology.

9:45

**4aPAb4. Acoustic wave propagation in a toroid cavity carrying a steady flow.** Charles Thompson (Elec. and Comput. Eng., UMASS Lowell, 1 Univ. Ave., Lowell, MA 01854, charles_thompson@uml.edu), Sarah Kamal, Nishi Shah, and Kavitha Chandra (Elec. and Comput. Eng., UMASS Lowell, Lowell, MA)

This paper presents an analytical study of acoustic wave propagation in a toroid-shaped cavity. The interior wall of the cavity is rigid and encloses a fluid driven into steady motion along its azimuth. The analysis will focus on acoustic excitation near the principle resonance frequency of the system. The impact of steady primary and secondary flow on the acoustic velocity potential flow-reverse symmetry is evaluated.

10:00

**4aPAb5. Vibrational characteristics of a helical antenna for L-Band satellite communications.** Nathan Hill (Dept. of Phys., University of Mississippi, University, MS 38677, ndhill@go.olemiss.edu), Scott W. Chumley (Phys. and Astronomy, Univ. of Mississippi, Auburn, AL), Wayne Prather (National Ctr. for Physical Acoust., Univ. of Mississippi, Oxford, MS), and Joel Mobley (Phys. and Astronomy, Univ. of Mississippi, University, MS)

Helical coils serve many functions in mechanical systems and are often subjected to a variety of impulsive forces. In this talk we examine the dynamics of a helix modeled after an L-Band helical antenna used for satellite communications in low Earth orbit (LEO). Since the components of a deployed satellite are not accessible, it is critical to understand the dynamics

of the antenna given its position outside of the main body. The intent of this work is to identify the frequencies, shapes and Q-values of the vibrational modes of the helix subject to impulsive stimuli and evaluate changes due to environmental exposures and structural modifications. We report on coil-by-coil measurements of the spectra using eight distinct stimulus-response polarizations. Two specific studies are emphasized in this talk. To gauge the sensitivity of the element to thermal variations under (LEO) conditions, we report on the impact of thermal cycling on the vibrational spectra of the helix. We also report on changes in the dynamic response of the helix due to the addition of structural support elements.

**10:15–10:30 Break**

**10:30**

**4aPAb6. Mechanism of low-frequency spectral scattering by a side-branch electromagnetic device with switching shunt.** Lixi Huang (Dept of Mech. Eng., Univ. of Hong Kong, Hong Kong NA, Hong Kong, lixi@hku.hk)

Sound waves are reflected and absorbed by a passive side-branch device in a duct. The performance of such a configuration is limited for low frequencies and if the cavity is compact. In this study, an electro-magnetic mechanism to enhance such low-frequency performance is examined. A common loudspeaker diaphragm is used as a passive interface to the cavity, with its moving-coil immersed in the magnetic field, and a shunt analogue circuit is attached. When the diaphragm is driven by the incident wave, the reactive Lorentz force exerts extra acoustic impedance. A MOSFET switch is added to periodically toggle between the shunt-off state, which is made damping-free in this study, and the shunt-on state which can almost forbid the diaphragm to vibrate if the shunt is close to a short circuit. The repeated transitions scatter a significant portion of the incident sound energy to frequencies other than the incident, with its peak scattering efficiency found when the switching is twice the frequency of the incident. The sudden removal of the Lorentz force by MOSFET switch-off creates a boost in the diaphragm response which is otherwise suppressed by the cavity stiffness, leading to much enhanced sound reflection.

**10:45**

**4aPAb7. Nonlinear acoustical transmission through a weak shock wave.** Francois Coulouvrat (CNRS, Sorbonne Universite - 4 Pl. Jussieu, Institut Jean Le Rond d'Alembert, Paris 75005, France, francois.coulouvrat@upmc.fr)

Recent experiments (Ducousso *et al.*, Phys. Rev. Appl., L051002, 2021) demonstrated the possibility to image weak shock propagation in solids by an ultrasonic probe wave. Wave interaction with a steady, ideal step shock in air has been previously described (Burgers, Selected Papers, Springer, 478–486, 1995—McKenzie and Westphal, Phys. Fluids, 11, 2350, 1968), without consideration for the particular case of a weak shock nor for the influence of the medium. The present paper considers a weak shock interacting in any inviscid fluid with an incident probe wave. No reflected wave arises. The transmitted wave, vortex and entropy modes behind the shock, and the shock front disturbance, are determined by the linearisation of the Rankine-Hugoniot relations. For a weak shock, entropy mode and energy jump relation can be omitted. The shock motion induces a Doppler effect dependant on the medium, air and water giving opposite trends. The transmitted wave amplitude is either increased or reduced through energy exchanges with the shock. For an incidence beyond the critical angle, instead of a total reflexion, an inversion of the direction of the transmitted wave occurs, propagating in the same direction as the shock. This phenomenon seems specific to weak shocks.

**11:00**

**4aPAb8. Thin ultrasound non-contact airborne sensor using flexural vibration.** Natsumi Nakaoka (Sci. and Eng., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, ut6tuzn@gmail.com), Eimei Yamamoto, and Daisuke Koyama (Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan)

A non-contact acoustic sensing system using changes in the radiation impedance was proposed to detect the position of an object in front of an ultrasonic vibrator. The sensor is composed of a rectangular vibrating plate ($20 \times 100 \times 1$ mm$^3$) and a PZT transducer ($20 \times 20 \times 2$ mm$^3$). The sensor configuration was determined based on the results of a finite element analysis simulation. The flexural vibration modes were excited on the plate at two resonance frequencies, 29.7 and 46.2 kHz. The radiation angle of ultrasound from the sensor and the acoustic field between the object and the plate were compared at the two frequencies to investigate the detection characteristics. When an acoustic standing-wave field was generated in the air, the electrical impedance of the ultrasound transducers dramatically increased, indicating that the radiation impedance of the sensor was dependent on the object position. By measuring the amplitude of the input current to the transducers and the phase difference between the input current and the voltage applied to the sensor, the object position could be determined uniquely within a two-dimensional area.

**11:15**

**4aPAb9. Waveguide demultiplexer based on Helmholtz-resonator mediated extraordinary acoustic transmission.** William M. Robertson (Phys. & Astronomy, Middle Tennessee State Univ., MTSU Box X116, 1301 East Main St., Murfreesboro, TN 37132, wroberts@mtsu.edu), Robert Carlyon, and Kyle Sprague (Phys. & Astronomy, Middle Tennessee State Univ., Murfreesboro, TN)

The design of an acoustic demultiplexer based on in-line Helmholtz resonators is demonstrated analytically via a modified transfer matrix method and computationally through finite element simulations. The modeled system consists of a single input waveguide that splits in a Y-configuration into two output waveguides. Each output arm has a single tuned Helmholtz resonator embedded in-line along the length of the guide. The Helmholtz resonators in each arm consist of a single cavity with two necks—one directed towards the input and output sides of the guide. The phenomenon of extraordinary acoustic transmission results in near perfect transmission of sound along each output arm in a small frequency interval about the Helmholtz resonant frequency. The demultiplexed frequencies are determined by the physical dimensions of the Helmholtz resonator. Using a single Helmholtz resonator in each output arm means that the system is more compact compared to other proposed schemes using either side-loaded Helmholtz resonators or stubs.

# Session 4aPP

## Psychological and Physiological Acoustics: Psychological and Physiological Acoustics Poster Session II

Nathan C. Higgins, Chair
*Communication Sciences and Disorders, University of South Florida, 4202 E. Fowler Avenue, PCD 1017, Tampa, FL 33620*

All posters will be on display from 10:00 a.m. to 12:00 p.m. Authors of odd-numbered papers will be at their posters from 10:00 a.m. to 11:00 a.m. and authors of even-numbered papers will be at their posters from 11:00 a.m. to 12:00 noon.

### *Contributed Papers*

**4aPP1. Transfer effects of discrete tactile mapping of musical pitch on discrimination of vocoded stimuli.** Aaron Hodges (CCRMA, Stanford, 660 Lomita Ct, Stanford, CA 94305, athodges@stanford.edu), Raymond L. Goldsworthy (Otolaryngol., Univ. of Southern California, Los Angeles, CA), Matthew B. Fitzgerald (Otolaryngol., Stanford, Palo Alto, CA), and Takako Fujioka (CCRMA, Stanford, Stanford, CA)

Many studies have found benefits of using somatosensory modality to augment sound information for individuals with hearing loss. However, few studies have explored the use of multiple regions of the body sensitive to vibrotactile stimulation to convey discrete F0 information, important for music perception. This study explored whether mapping of multiple finger patterns associated with musical notes can be learned quickly and transferred to discriminate vocoded auditory stimuli. Each of eight musical diatonic scale notes were associated with one of unique finger digits 2-5 patterns in the dominant hand, where pneumatic tactile stimulation apparatus were attached. The study consisted of a pre and post-test with a learning phase in-between. During the learning phase, normal-hearing participants had to identify common nursery song melodies presented with simultaneous auditory-tactile stimulus for about 10 min, using non-vocoded (original) audio. Pre- and post-tests examined stimulus discrimination for 4 conditions: original audio + tactile, tactile only, vocoded audio only, and vocoded audio + tactile. The audio vocoder used cochlear implant 4 channel simulation. Our results demonstrated audio-tactile learning improved participant's performance on the vocoded audio + tactile tasks. The tactile only condition also significantly improved, indicating the rapid learning of the audio-tactile mapping and its effective transfer.

**4aPP2. Using frequency selectivity to examine category-informative dimension-selective attention.** Sahil Luthra (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, sahil.bamba.luthra@gmail.com), Chisom O. Obasih (Psych., Carnegie Mellon Univ., Pittsburgh, PA), Adam Tierney (Birkbeck, Univ. of London, London, United Kingdom), Frederic Dick (Univ. College London, London, United Kingdom), and Lori Holt (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

In everyday situations, listeners may selectively attend to acoustic dimensions (e.g., frequency) within complex sounds and ignore other simultaneous dimensions. Speech sound categories are defined over multiple dimensions that vary in informativeness; thus, speech perception may demand selective attention to diagnostic dimensions. Here, we examine how selective attention affects cortical representations of informative acoustic dimensions during category learning. Participants completed a five-day training regimen wherein they learned four novel non-speech categories to criterion (Obasih *et al.*, in prep). Crucially, stimuli were designed such that for two categories, categorization required reliance on – and perhaps selective attention to – acoustic patterns within high-frequency bands; the other two categories were differentiated by acoustic patterns in low-frequency bands. After training, listeners completed an fMRI session using the same non-speech categorization task. Best frequency for each auditory cortex voxel was obtained using tonotopic mapping. We hypothesized that when categorization depended on distinctions in certain frequency ranges (e.g., high frequencies), listeners would selectively attend to those frequencies, resulting in greater recruitment of voxels preferring those frequencies and possibly attenuation of signal in voxels preferring other frequencies. We present preliminary data from our study, with the goal of clarifying the cortical mechanisms supporting dimension-based auditory selective attention.

**4aPP3. Fusion and identification of vowels in the free field.** Briana N. Martinez (Commun. Disord., California State Univ., Los Angeles, 5151 State University Dr., Los Angeles, CA 90032, bmarti113@calstatela.edu), Michelle R. Molis (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, Portland, OR), and Lina A. Reiss (Dept. of Otolaryngol., Oregon Health & Sci. Univ., Portland, OR)

Binaural fusion occurs when signals arriving separately at the two ears are perceived as one sound. Previously, Reiss and Molis (2021) showed that listeners with normal or impaired hearing fused and perceptually averaged dichotic vowels presented through headphones, especially vowels with the same fundamental frequency. To extend these findings to a more real-world listening context, the current study investigated if vowel fusion and averaging would replicate in the free field. Stimuli were good exemplars of /æ/, /a/, /u/ or /i/ synthesized with fundamental frequencies of 106.9, 151.2, or 201.8 Hz. On each trial, vowels were presented simultaneously via speakers located on the left and the right at $\pm 60°$. On test trials, vowels presented from the left and right were different; fundamental frequencies could be the same or different. On catch trials, identical non-exemplar vowels were presented from both speakers; these trials reduced the likelihood listeners would notice that different categories were presented on test trials. Listeners indicated which single vowel or two vowels they heard. Preliminary findings from normal-hearing listeners suggest there are fewer single vowel responses—fusion is decreased—in free field compared with headphone presentation. As demonstrated with headphones, vowels with the same fundamental frequency are more often fused. Results indicate that vowel fusion patterns previously observed with headphones can also be observed in the free field. [Work supported by NIH R01DC013307 and CSULA SPROUTS program.]

**4aPP4. Explaining the deterioration of the relative timing accuracy for across-channel by a simple mathematical model.** Satoshi Okazaki (Kagawa Univ., 1750-1 Ikenobe, Miki-cho, Kita-gun, Kagawa 761-0793, Japan, okazaki.satoshi.u2@kagawa-u.ac.jp)

The perceptual simultaneity range, within which two asynchronous pure tones are perceived to start simultaneously, and the gap detection threshold are known to be wide when the frequency separation of the tones is large. It

is generally said that the accuracy of the relative timing for different frequency channels deteriorates. However, there is no clear explanation of why such deteriorations are necessary. This study aimed to show that a simple mathematical model leads to the deterioration of relative timing accuracy for two tones with different frequencies. As a result of the calculations, the model simulated the deterioration of the perceptual simultaneity range not only with the increase of the frequency separation but also with the decrease of the frequency region of two tones. The model also simulated the behavior of the gap detection threshold for two different frequencies (across-channel) with its asymmetric deterioration. Further, the model simulated the behavior of the gap detection threshold for two identical frequencies (within-channel). These results suggest that one simple mathematical model may explain the mechanisms underlying perceptual simultaneity, within-channel gap detection, and between-channel gap detection for two tones.

**4aPP5. Statistical regularities of task-irrelevant dimensions impact auditory decisions.** Austin Luor (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, aluor@andrew.cmu.edu), Sahil Luthra (Psych., Carnegie Mellon Univ., Pittsburgh, PA), Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA), Adam Tierney, Frederic Dick (Psychol. Sci., Birkbeck, Univ. of London, London, United Kingdom), and Lori Holt (Psych., Carnegie Mellon Univ., Pittsburgh, PA)

Listeners build up statistically driven expectations of what they will hear; however, there is no consensus on *how* these statistics influence perception, attention, and behavior. Here, we manipulate two statistical properties: global probability (the likelihood of single ‘sound events’) and predictiveness (how often does one sound precede another). We ask how the probability and predictiveness of different acoustic frequencies affect performance on two paradigms where frequency is task-irrelevant: suprathreshold duration identification and near-threshold tone-detection-in-noise. We found that duration decisions are faster and detection decisions are more accurate for high-probability tone frequencies, compared to low-probability tone frequencies. Moreover, when a preceding “cue” tone’s frequency predicts that of a subsequent “target” tone, listeners are faster at judging the duration of the target tone. This latter effect is not solely a result of temporal cueing, as target responses are not facilitated if a cue does not predict the target tone’s frequency. Blending these paradigms to examine the same global and transitional probabilities across duration and detection decisions suggests that statistical learning shapes attention to perceptual dimensions, even when the dimensions are irrelevant to optimal task performance.

**4aPP6. Estimating human neural tuning curves at low-level sound from human psychophysical masking data and the cochlea-frequency maps.** Xinhang Song (ECE, Univ. of Illinois at Urbana-Champaign, 1901 N Lincoln Ave., Apt. 203, Urbana, IL 61801, xinhang@illinois.edu) and Jont B. Allen (ECE, Univ. of IL, Urbana, IL)

We proposed a three-slope method to estimate human frequency tuning curves at low sound levels. As a starting point, cat neural excitation patterns (Allen and Fahey, 1993) are derived from Liberman’s cat cochlea neural tuning curves data (Liberman, 1978) in a mathematical way. A three-slope structure inspired by (Rhode,1978) is proposed to represent neural excitation patterns. The aim of this paper is to estimate the three-slope structure for human neural excitation patterns and then transform human neural excitation patterns into human tuning curves by using a human cochlea-frequency map (Greenwood, 1990). In 1993, Allen and Fahey introduced the concept of the second cochlea map, which was shown to be highly correlated with both neural excitation patterns and distortion product otoacoustic emissions (DPOAE). Thus, three slopes of human neural excitation patterns may be estimated via the second cochlea map and human masking data (Wegel and Lane, 1924). In the present study, slopes are estimated from limited existing psychophysical data, which can be improved with more accurate experimental data in the future.

**4aPP7. Mismatch negativity linked with sound lateralization in elderly individuals.** Kazumoto Morita (Chuo Univ., 1-13-27, Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan, mtkkojiro@gmail.com), Moeko Shiroki (Chuo Univ., Tokyo, Japan), Sunao Iwaki (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Ibaraki, Japan), and Takeshi Toi (Chuo Univ., Bunkyo-ku, Tokyo, Japan)

Elderly individuals tend to experience a decline in auditory functions. One of these is poor lateralization ability. This study aimed to investigate the relationship between sound lateralization and mismatch negativity (MMN) in elderly individuals using a lateralization test and event-related potential (ERP) measuring test. In the lateralization test, participants were presented with sound stimuli with an interaural time difference (ITD) in the right or left ear and were required to indicate the sound direction. The ERP measuring test estimated unconscious brain activity during participants’ exposure to sound stimuli using an ITD in the right or left ear. Two ITDs (i.e., 0.4 ms and 0.8 ms) of a 1-kHz pure tone were measured in elderly and young participants. For sounds with phase reversal between the onset and ongoing parts, young individuals performed lateralization based on the onset part. In contrast, at an ITD of 0.8 ms, elderly participants unable to perceive the onset part tended to demonstrate less negative MMN. In conditions where the preceding ear in onset and ongoing parts was reversed, elderly participants exhibited ambiguous lateralization, and their cortical evoked potentials did not significantly differ from activity associated with standard stimuli without an ITD.

**4aPP8. The influence of binaural cues on the promotion or inhibition of auditory stream segregation.** Nathan C. Higgins (higgins1@usf.edu) and Erol J. Ozmeral (Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Though much is known about detection thresholds for binaural cues (interaural time [ITD] and level differences [ILD], and interaural correlation [IAC]), less is known about how these cues influence the separation of auditory sources. The ABA auditory stream segregation paradigm (Bregman, 1990) in a bistable configuration, elicits roughly equivalent proportions of integrated and segregated percepts, where listeners spontaneously switch back and forth between the two. To determine the influence of binaural cues we periodically (25 s) manipulated the binaural cue carried by the A components of the ABA triplet (made of narrowband noise, 6-semitone separation) while maintaining the B component at a stable binaural cue. Participants continuously indicated their perception via button box. Analyzed as a function of binaural cue, lateral cues were more likely to be reported as segregated than midline cues. Binaural cue boundaries (perceptual transition point) were defined using logarithmic regression and were significantly correlated for ILD and ITD ($r^2 = 0.65$, $p < 0.05$). IAC modulation resulted in significantly more switches when both the A and B components were centered ($IAC = 1$) than when the IAC of the A component was more diffuse ($IAC < 0.9$). These results provide evidence of how listeners functionally use binaural cues to segregate auditory streams.

**4aPP9. Sound lateralization ability of elderly individuals in three onset conditions.** Kazumoto Morita (Chuo Univ., 1-13-27, Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan, mtkkojiro@gmail.com), Moeko Shiroki (Chuo Univ., Tokyo, Japan), and Takeshi Toi (Chuo Univ., Bunkyo-ku, Tokyo, Japan)

Authors had previously reported that elderly individuals could not sufficiently perceive the onset portion of the sound. To further this research, the current study varied the sound conditions in the onset portion to three—(1) condition in which the onset portion on the delayed ear side was silent, (2) condition in which sound was presented to the onset portion of the delayed ear side as well, (3) condition in which silence was inserted only during onset portion on the delayed ear, with no phase delay in the ongoing portion. Two sound frequencies of 1 kHz and 500 Hz were presented to 23 elderly

participants and 22 young adults. Interaural Time Differences (ITDs) of 0.2, 0.4, 0.6, and 0.8 ms were relayed separately to each ear, in addition to the condition of zero ITD to both ears. In the experiments, a test stimulus sound was presented after the reference sound, and participants answered the direction of the sound from among three responses of "'Right," "Left," or "Same" compared with the reference sound. The results of condition (3) revealed that the elderly participants could not perceive the onset portion of sound compared to the younger participants.

**4aPP10. Cochlear mechanics from correlation between multiple travelling waves from oval window.** Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Many models of cochlear mechanics have been commonly perceived to be plausible. These models primarily assume a resonance in the neuro-mechanical system of the cochlea. An inherent challenge thatremains to be addressed is the rapid changes in speech sounds during rapid utterances of syllables that could not be demonstrated to be precisely recognized by such a traditional resonance system that needs much ore response delay to work properly. Whereas, we know the human auditory system can easily identify slight changes in syllables in terms of rate of production, slight changes in intonation, prosody, and so forth. For example, an average listener can quickly identify whether the singer of a recorded song is a particular celebrity and whether another recording is an identical copy or slightly different version, even if the singer is the same person. Therefore, there is a definite need to explore other mechanism in the cochlear mechanics other than simple resonance that can not respond so rapidly and so precisely simultaneously. As we know, rapid response of traditional resonance system requires sacrificing high precision. And high precision of a traditional resonance system requires sacrificing high speed of response. This trade off behavior is analogous to the well known Heissenberg Uncertainty Principle. Therefore, some alternative mechanism must be present that has not been discovered yet. We will discuss a few plausible paradigms of radically distinct cochlear mechanics from correlation between multiple travelling waves.

**4aPP11. Relation between temporal fine structure processing and global processing speed.** Kelli Sugai (VA Loma Linda Healthcare System, 11201 Benton St., Loma Linda, CA 92357, kellisugai@gmail.com), Nicole Whittle, Christian Herrera Ortiz (VA Loma Linda Healthcare System, Loma Linda, CA), Marjorie R. Leek (Res., Loma Linda VA Healthcare System, Loma Linda, CA), Grace Lee, and Jonathan H. Venezia (VA Loma Linda Healthcare System, Loma Linda, CA)

Strelcyk *et al*. (2019) recently found that interaural phase discrimination in older hearing-impaired listeners was correlated with both visuospatial processing speed and interaural level discrimination. This suggests that temporal fine structure (TFS) processing relies on global processing speed and/or spatial cognition, though it is possible that, generally, complex auditory discrimination engages multiple cognitive domains. Here, 50 Veterans (mean age = 48.1, range = 30–60) with normal or near-normal hearing completed batteries of temporal processing and cognitive tests. Composite cognitive test scores reflecting processing speed/executive function (PS-EX) and working memory (WM) were obtained. Temporal processing tasks included measures of envelope (ENV; gap duration discrimination, forward masking) and TFS (frequency modulation detection, interaural phase modulation detection) processing. Bayesian hierarchical regression was used to fit psychometric functions simultaneously to all ENV and TFS tasks in all subjects. Fixed effects of PS-EX and WM on thresholds and slopes were estimated for the psychometric functions in each temporal task. In general, (i) PS-EX and WM influenced both TFS and ENV thresholds, but not slopes; and (ii) TFS thresholds were best explained by PS-EX scores while ENV thresholds were best explained by WM scores. These findings suggest a specific relation between global processing speed and TFS.

**4aPP12. Computer model of sensory-motor dilemma in stuttering of spoken sounds.** Amitava Biswas (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu), Steven Cloud, and Mary Schaub (Speech and Hearing Sci., Univ. of Southern MS, Hattiesburg, MS)

The reaction time of any typical person is usually noticeable by others but not perceived by the same person. Whenever such an individual performs any activity even if very fast such as eye blinking, the individual apparently perceives simultaneity between the instant of activating the muscles and the instant of receiving a sensory, visual, tactile, or auditory confirmation of accomplishing the task. Obviously, any motor response involves some unavoidable neural processing delay. Any sensory perception involves some unavoidable neural processing delay also. Therefore, the real instant of the sensory perception cannot be perceived to coincide with the perception of the instant of motor action without a very precise neural adjustment of perception in temporal position. This situation reminds of the merging of the two visual perspectives from two eyes merging together to provide a single visual perspective. It appears that there are two clocks maintained in the central nervous system: one clock for recording the time when a motor command was sent, and another clock for recording the time when a sensory confirmation was received. These two clocks must differ by a small delay to give the illusion of zero reaction time to the individual's own activities and sensory feedback. We will discuss implications of disruptions of that neural adjustment of timing perception with a simple computer model.

**4aPP13. Investigating the influence of auditory streaming cues on binaural pitch fusion.** Nicole Dean (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., 3181 SW Sam Jackson Park Rd., Portland, OR 97239, nikki.lavee.dean@gmail.com), Lina A. Reiss, and Frederick J. Gallun (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Binaural pitch fusion occurs when dichotic signals eliciting a different pitch are perceived as one sound. Previously, Oh *et al*. (ASA 2018) investigated the effects of auditory streaming cues on binaural fusion using a modification of the ABA streaming paradigm. A dichotic stimulus (fixed tone at frequency $f_{ref}$ in the reference ear and variable frequency in the contralateral ear) was presented at alternating intervals with a "capture" tone at $f_{ref}$. Binaural fusion frequency range decreased, suggesting competition of the capture with the variable frequency tone for fusion. The present study added more capture tones, allowing for a galloping percept, and varied the interstimulus interval (ISI), both hypothesized to decrease binaural fusion by increasing salience of the capture tone. The dichotic stimulus was presented alternating with two or three repetitions of the capture tone at $f_{ref}$ for triplet and quadruplet paradigms, respectively. ISIs of 40 or 80ms were used. Participants reported whether they perceived galloping or two streams. Preliminary data from 5 normal-hearing listeners suggest reductions in binaural fusion range with the triplet and quadruplet paradigms, and longer ISI, compared to the original paradigm. These findings indicate that auditory grouping cues may potentially reduce binaural fusion. [Research funded by NIH/NIDCD R01DC013307.]

4a THU. AM

## Session 4aSAa

## Structural Acoustics and Vibration: Fluid Structure Interaction

Benjamin M. Goldsberry, Cochair
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Micah Sheperd, Cochair
*Brigham Young University, Provo, UT 84602*

### *Invited Papers*

**9:00**

**4aSAa1. Modeling of airborne ultrasound reflection from water surface waves.** Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., University, MS 38655, zhang@olemiss.edu) and Zheguang Zou (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, University, MS)

Airborne ultrasound reflection from water surface waves is modelled to advance uses of acoustic signals to measure water surface waves and apply the measurements to explore interactions of water waves with rigid structures in a laboratory setting. When the ultrasound is incident on a moving periodic water surface wave, the reflected signal can be treated as diffraction from a moving corrugated reflection grating. Under the condition that the amplitude of the water surface waves is much less than the incident acoustic wavelength, diffraction theory leads to analytical formulas for the spectra of the acoustic signal relating to the water wave amplitudes and frequencies. Complementary modeling based on ray theory and wave superposition illustrates the diffraction and validates formulas of water wave reflection from a surface-piercing barrier structure, where two counter-propagating water waves are involved.

**9:20**

**4aSAa2. Monostatic and bistatic acoustic scattering from smooth and rough elastic cylinders insonified by directional sonars.** Miad Al Mursaline (Mech. Engineering/Appl. Ocean Phys. & Eng., Massachusetts Inst. of Technology/Woods Hole Oceanographic Inst., 70 pacific St., Cambridge, MA 02139, miad@mit.edu), Timothy K. Stanton, and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Acoustic scattering from cylinders has been previously studied extensively for idealized incident plane waves and point receivers. However, the more realistic case of scattering from cylinders insonified by directional sonars has received limited attention. Operational sonars are directional and transmit waves that spread spherically. Due to the spherical spreading, the overall scattered pressure levels are affected and the cylinder is insonified across a continuum of oblique incident angles, even at broadside incidence. The obliqueness in the incident field, in turn, influences the structure of the scattering spectrum by exciting guided wave natural modes. A recently derived theory, accounting for the above realistic effects, is tested against laboratory measurements involving both smooth and rough elastic cylinders, spanning a range of scattering geometries and roughness profiles. The effects of bistatic angle between the source and receiver on overall scattered pressure levels and resonances are investigated and compared with results from the monostatic geometry. The influence of correlation length and root mean square roughness on the scattered field is also studied.

9:40

**4aSAa3. Extending the vibration-based sound power method to determine the sound power radiated from acoustic sources.** Ian C. Bacon (Dept. of Phys. & Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, ianbacon24@gmail.com), John C. Ebeling (Mathematics, Brigham Young Univ., Sandy, UT), Scott D. Sommerfeldt (Dept. of Phys. & Astronomy, Brigham Young Univ., Provo, UT), and Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

A vibration-based sound power (VBSP) method has been developed as an alternative method for determining the acoustic energy radiated from structures. Many acoustic sources, such as a blender or motor, have sound power contributions that originate from within the structure and therefore cannot be scanned properly using a vibrometer. For these types of sources, a rectangular enclosure with five rigid sides and a single mylar side was fabricated to enclose the acoustic source, so that the VBSP method could be utilized to obtain the radiated sound power. A calibration curve was developed to account for the enclosure effects on these sources, and this curve is used to adjust the measured results to give the true free-field radiated sound power. Experimental results will be shown comparing the sound power obtained from the adapted VBSP method with the sound power of the source obtained in a reverberation chamber using the ISO 3741 standard. Furthermore, computational results will be shown comparing the sound power obtained through the boundary element method (BEM) with the sound power obtained by using the velocities calculated from that model processed with our VBSP method. Funding for this work was provided by the National Science Foundation (NSF).

9:55

**4aSAa4. Numerical investigation on the effects of wake-induced vibration fluid flow over airfoil geometries.** Eli Bloomfield (Mech. Eng., Kennesaw State Univ., Marietta, GA) and Muhammad Salman (Mech. Eng., Kennesaw State Univ., 1100 South Marietta Pkwy, Marietta, GA 30047, msalman1@kennesaw.edu)

Currently, many companies are looking to innovate and decrease economic and environmental impacts in the aerospace, energy, and marine sectors. As companies continue to innovate, airfoil designs used in the aerospace, wind turbine, commercial, and military marine industries, the lighter materials used are more susceptible to fatiguing failures due to vibrations. When these geometries encounter unsteady flow, the airfoil creates a vibration response. If the vibration response is near the natural frequency of the airfoil geometry, catastrophic failure can occur. This paper analyzes the correlation between geometry and frequency response for NACA symmetric airfoils in unsteady flow due to von Karman vortex shedding. The correlation between the shedding frequency of the cylinder and the frequency response of the airfoil is also considered. The paper also aims to determine if there is a correlation between geometry and frequency response amplitude across the frequency spectrum and at the natural frequency of each airfoil. These results were compared to the airfoil natural frequency, determined through Finite Element Analysis (FEA).

# Session 4aSAb

## Structural Acoustics and Vibration: Analysis of Vibratory Systems

Stephanie Konarski, Cochair
*Johns Hopkins University, Applied Physics Laboratory,*

Thomas Bowling, Cochair
*Naval Surface Warfare Center Carderock Division, Bethesda, MD 20817*

### *Contributed Papers*

**10:30**

**4aSAb1. Impedance tube method for characterization of one-dimensional electro-momentum coupled materials.** Matthew A. Casali (Appl. Res. Labs. & Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcasali@utexas.edu), Andrew R. McNeese, Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Appl. Res. Labs. & Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Electro-momentum coupling is a macroscopically observable material response resulting from heterogeneous piezoelectric media with microscale asymmetries that produce unique cross-coupling between the bulk momentum of the material and the generated electric field. Recently, Pernas-Salomon *et al*. used a one-dimensional transmission line model to demonstrate that the electro-momentum coupling effect must be considered in order to retrieve physically meaningful effective properties of heterogeneous media with subwavelength asymmetries [*Wave Motion* **106**, 102797, (2021)]. This work presents the specialization of their transmission line model to the classical impedance tube measurement technique in which the scattering coefficients and the spatial averages of the mechanical and electrical fields of a one-dimensional material can be measured in order to obtain estimates of the frequency-dependent effective properties of an electro-momentum coupled sample. We investigate an idealized analytical approximation and then a finite element model of the realistic impedance tube configurations in order to link the analytical model to realistic experimental conditions and geometry. We then provide preliminary test results extracted from an electro-momentum coupled unit cell in a water-filled impedance tube. [Research sponsored by the Defense Advance Research Project Agency and the Army Research Office and was accomplished under Grant No. W911NF-20-1-0349.]

**10:45**

**4aSAb2. Initial studies into camera-based modal force reconstruction.** Sean Collier (Graduate Program in Acoust., The Penn State Univ., 218 Appl. Sci. Bldg., University Park, PA 16804, smc604@psu.edu) and Tyler P. Dare (Graduate Program in Acoust., The Penn State Univ., State College, PA)

Much work has been done toward the reconstruction of an unknown force using measured response. Modal Force Reconstruction (MFR) attempts a solution by using measured modes and vibrations at some number of degrees of freedom. While developed by way of traditional sensors (e.g., accelerometers), the full-field nature of cameras may be exploited for such tasks. In particular, the spatial density of an image can re-establish the traditionally underdetermined problem into one that is overdetermined. Not only does this make the reconstruction well-posed, but the modal matrix being that which needs inversion establishes a niche solution in certain force reconstruction scenarios. This talk will explore the proposed theory behind camera-based MFR and present proofs of concept using analytical models. Limitations and points of error will also be discussed, helping to further establish credibility and understanding. Initial results suggest camera-based MFR to be a viable reconstruction technique with high spatial and temporal accuracy.

**11:00**

**4aSAb3. Microphone vibration sensitivity: What it is, why it is important, and how to measure it.** Charles B. King (R&D, Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60143, charles.king@knowles.com)

Microphones are designed to respond to acoustic pressure fields, but they can also be excited by external sources of mechanical vibration. In hearing aids, feedback is a difficult problem to resolve due to the high gain used to amplify sounds. The problem is worsened by the location of microphones and loudspeakers in close proximity, coupling both the acoustic and mechanical feedback paths. The first proposals of microphone vibration measurement techniques and definitional structure were published by Mead Killion in the 1970's. Significant improvements in measurement technique have been published over the last 10 years. This talk will focus on a new measurement technique that allows for direct measurement of the "intrinsic vibration sensitivity". The technique gives the ideal zero acoustic pressure condition at the microphone port and isolates room acoustic noise from the desired vibration signal. The noise floor of the new technique is 30 dB below previous methods, allowing for exceptionally clean measurement of primary-axis vibration sensitivity up to 10 kHz. Improvements are robust enough to enable the clean measurement of off-axis sensitivity of the microphone, which is often below 50 dBspl equivalent at 9.81 $m/s^2$ of acceleration.

**11:15**

**4aSAb4. Highly damped single-chamber pneumatic vibration isolator using effects of thermal conductivity in the air volume.** Vyacheslav M. Ryaboy (Photonics Solutions, MKS, 1791 Deere Ave., Irvine, CA 92606, vryaboy@newport.com)

State-of-the-art pneumatic isolators have a two-chamber design with metered orifices between the chambers. This arrangement produces energy dissipation necessary to avoid excessive resonance vibration [1]. However, this leads to reduced isolation performance at high frequencies compared to a single-chamber isolator of equivalent total volume. Analysis shows that, while the dynamic process of the air compression in the pneumatic chamber is adiabatic for geometric parameters usually employed in vibration isolators, substantial dissipation of mechanical energy can be achieved by introducing certain thermally conductive features into the chamber. That makes practical a single-chamber design with sufficient damping. The paper presents the optimization of the geometry of thermally conductive features aimed at maximizing the loss factor at the resonance frequency of the isolation system. Experimental results illustrate the concept [2] and demonstrate the increased isolation effect. [1] V. M. Ryaboy, *Vibration Control for*

*Optomechanical Systems* (World Scientific, 2022), p. 280. [2] Vibration isolation apparatus with thermally conductive pneumatic chamber, and method of manufacture, U.S. patent application No. 17/721834, USPTO, 2022.

**11:30**

**4aSAb5. Minimization of radiated noise.** Mladen Chargin (None, 985 Heavenly View Ct., Gardnerville, NV 89460, chargin@garlic.com)

Radiatednoise from various components is of increasing concern for today's automotive manufacturers. Traditionally, this analysis was usually performed with Boundary Element Method (BEM) software. Analysis results, although interesting in general, do not provide the design engineer with information on how to improve the design. Unfortunately, there is no sensitivity or optimization capability within any known BEM code, especially when coupled with a FEM code, which provides the vibration of the structure. This paper addresses the problem by describing the software developed by CDH, which combines the structural FEM analysis and optimization capability (MSC/Nastran, SOL200)with CDHBEM analysis code such that one can perform analysis *and*optimization very quickly and efficiently. This approach is applied to an automotive air cleaner with internal acoustic domain, which is excited by pulsation of intake valves, which then vibrates and radiates noise. The radiated noise is then minimized at five predefined microphone positions in the far field. The maximum pressure was reduced by approximately 16 dB.

**11:45**

**4aSAb6. Introduction to Ansys NVH solutions focusing on the Vibro Acoustics.** Hardik Shah (Product Management, Ansys Inc., 1039 Granite Dr., Mc Donald, PA 15057-2519, hardik.shah@ansys.com)

Product validation focusing on NVH heavily relies on physical testing which put NVH focused validation on right hand side of the Product Validation Process. Due to this, lot many times design changes are not possible or expensive or time consuming which reduces overall product quality. Industries like Automotives and ground transportation, Aero and Defense, High Tech, Industrial products making huge investments on NVH validation to win over customer perception battle. Ansys geared up in providing industry leading NVH Solution with Mechanical, LS-Dyna, Motion, Sound along with other products like Maxwell, Fluent, Optislang. NVH workflows driven by highly correlated simulation models, Integration of our key industry leading technologies and Guided, Automated, DOE friendly, Cloud native NVH Solutions are our TOP 3 goals. To introduce Ansys NVH Solutions focusing on the Vibro-Acoustics to Acoustical Society of America communities, we will cover key capabilities from different products briefly and demonstrate NVH Workflows in detail. Electric Motor NVH Simulation—Excitation generated due to Electro Magnetic forces Powertrain/Driveline/Gearbox NVH Simulation—Excitation generated due to multi body dynamics simulationFan NVH Simulation—Excitation generated due to structural vibration of fan driven by flow simulation.

# Session 4aSC

## Speech Communication and Psychological and Physiological Acoustics: Bilingualism and the Brain

McCall E. Sarrett, Cochair
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Joseph C. Toscano, Cochair
*Villanova University, 800 E Lancaster Ave., Villanova, PA 19085*

**Chair's Introduction—8:40**

### *Invited Papers*

**8:45**

**4aSC1. An "asset" rather than "deficit" view of second-language (L2) speech.** Ann Bradlow (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL, abradlow@northwestern.edu)

This talk will present the rationale and empirical foundation for an "asset-" rather than "deficit-based" framework for understanding bilingual speech communication. In this framework, first-language (L1) and second-language (L2) speech are viewed as distinct speech styles each of which is shaped by four interacting sources of influence: (1) the individual talker's vocal tract and ideolectal speech patterns ("trait" characteristics), (2) the sound structure of the language being spoken ("structural" characteristics), (3) the sound structure of other language(s), if any, in the talker's repertoire ("repertoire" characteristics), and (4) general patterns of dominant versus non-dominant language function ("mode" characteristics). This "Trait-Structure-Repertoire-Mode" perspective contrasts with "deficit" models of the L1-L2 relationship in which metaphors of vectors (L1 source to L2 target) or filters (L1 filter for L2 production and processing) set up a hierarchy with L1 and L2 as "standard" and "deviant," respectively. While still acknowledging L1 advantages in communication speed, accuracy, and ease, this perspective emphasizes the confluence of language-general properties (Trait and Mode) and language-specific properties (Structure and Repertoire) for speech production and perception by bilinguals in both/all of their languages.

**9:05**

**4aSC2. Bilingualism, brain and development: A neuroemergentist perspective.** Arturo Hernandez (Psych., Univ. of Houston, 3114 Red Ridge Court, Manvel, TX 77578, aehernandez@uh.edu)

Neuroemergentism, (NM) is a novel framework which has sought to consider language development as involving the organization and reorganization of cognition and its underlying neural substrate. Work to support this framework comes from studies of language and cognitive development. In this talk, I will focus on two separate levels, the sensorimotor plasticity needed to adjust to new input and the cognitive flexibility needed to select between these competing sources of information. This talk will discuss both these levels with regard to the neurocognitive adaptations seen in bilinguals. This will include structural brain differences in monolinguals and bilinguals that vary in the age of second language acquisition. In the second part, of the talk work that has focused on the cognitive flexibility will be presented. This will focus on the adaptations of the basal ganglia and frontostriatal tracts as a gating mechanism crucial for selecting the correct motor response. This includes newer work which links genes associated with dopamine to cognitive and language flexibility in bilinguals. The ways in which sensorimotor plasticity and cognitive flexibility represent accurate but incomplete conceptualizations of the competitive processes involved in language and cognitive processing will be discussed. The talk will conclude with potential future directions using an NM framework.

**9:25**

**4aSC3. Effects of multilingual and monolingual social networks on speech perception.** Ethan Kutlu (Psychol. and Brain Sci., Univ. of Iowa, 1253 Dodge St. Ct, Unit 623, Iowa City, IA 52245, ethankutlu@gmail.com), Alexandra Fell, Keith Apfelbaum, and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Speech perception is gradient—listeners track continuous acoustic differences within a category (McMurray *et al*., 2022; Kapnoula & McMurray, 2022). Listeners use this gradiency to adjust subphonetic details (McMurray & Jongman, 2011), recover from ambiguity (McMurray *et al*., 2009), and aid learning and adaptation (McMurray & Farris-Trimble, 2012; Clayards *et al*., 2008). However, it is unclear whether gradiency is a developmental product of linguistic experience, particularly the variability of speech that is experienced. This ongoing project (current n = 31, planned n = 60) is testing school-aged children (6–11 years old) using the visual analogue scaling task (Kong & Edwards, 2011). Children hear tokens from a speech continuum (e.g., beach/peach) and make continuous ratings about how /b/- or /p/- like the sound is. This is related to social network information regarding children's language and social background. Preliminary results suggest that linguistic diversity impacts speech perception gradiency. The implications of bilingual education and linguistic environment in development will be discussed.

9:45

**4aSC4. Non-native speech sound learning as a window to individual differences in the neural encoding of speech.** Emily Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Storrs Mansfield, CT 06269, emilybmyers@gmail.com)

Non-native speech sound learning in adulthood is a skill marked by enormous individual differences, especially when compared to acquisition of speech sounds in childhood. This has led to the proposal that adult speech sound acquisition might be achieved through different mechanisms (or relying on fundamentally altered neural circuitry) compared to sound learning in infancy and childhood. One possibility we explore in a recent line of work is that individual differences in non-native speech learning are attributable to differences in the structure of brain areas responsible for processing speech, especially the transverse temporal gyrus. Of interest, the morphology of these same regions also predicts individual differences in native language speech perception, suggesting that sound acquisition (whether in childhood or adulthood) is limited by (likely innate) differences in neural architecture supporting sound processing. However, evidence that the best non-native perceivers are also the most finely-tuned native-sound perceivers is limited. Taken together, these results suggest that differences in brain structure are a soft constraint on the precision of acoustic-phonetic encoding, but that native language speech category structure is not the explanation for the difficulty of non-native sound learning.

**10:05–10:20 Break**

10:20

**4aSC5. Examining the links between subphonemic sensitivity and nonnative speech perception.** Efthymia Kapnoula (Basque Ctr. on Cognition, Brain and Lang.; Ikerbasque, Mikeletegi Pasealekua, 69, 2nd Fl., San Sebastian, Gipuzkoa 20009, Spain, kapnoula@gmail.com)

In general, listeners are sensitive to acoustic differences within their L1 phonemic categories (Andruski *et al*., 1994; McMurray *et al*., 2002; Toscano *et al*., 2010). However, some listeners appear to be more sensitive to subphonemic information than others (Kapnoula *et al*., 2017; Kong & Edwards, 2016). Recent electrophysiological evidence points to an early perceptual locus behind these individual differences; listeners with higher subphonemic sensitivity exhibit more linear (less warped) encoding of acoustic cues (Kapnoula & McMurray, 2021). It is unclear how these individual differences come to be and how they may affect listeners' ability to learn a new language. Here, I will discuss the potential role of bi/multilingual exposure in shaping speech perception and I will present data that offer valuable insights into how subphonemic sensitivity may be linked to nonnative speech perception.

10:40

**4aSC6. Neural and cognitive mechanisms of accented speech processing: the role of speaker identity and listener experience.** Janet van Hell (Psych., Penn State Univ., University Park, PA), Sarah Grey (Fordham Univ., Fordham University, New York, NY, sgrey4@fordham.edu), and Fatemeh Abdollahi (Boston Univ., Boston, MA)

Current everyday communication is a linguistic melting pot. Many learners of English as a second language have a nonnative accent when speaking English. We are also likely to interact with people from different language backgrounds, whose accent may be similar or different from one's own accent. Although behavioral measures indicate that listeners adapt quickly to nonnative-accented speech, neurocognitive studies have shown distinct neural mechanisms in processing nonnative-accented sentences relative to native-accented sentences. I will present recent behavioral and EEG/ERP experiments that studied how speaker identity and listener experience affect the comprehension of nonnative- and native-accented sentences. Specifically, we studied how listeners' experience with nonnative-accented speech modulates accented speech comprehension by testing different listeners (young and older adults with little experience with nonnative-accented speech, listeners immersed in nonnative-accented speech, and bilingual (nonnative-accented) listeners). We also examined how faces cuing the speaker's ethnicity create language expectations, and how these biases impact the neural and cognitive mechanisms associated with the comprehension of nonnative- and native-accented sentences. Implications of the findings will be discussed by integrating neurocognitive theories of language comprehension with linguistic theories on the role of socio-indexical cues in speech comprehension.



11:00

**4aSC7. How do non-native phonemes impact learning words in a second language? Evidence from eyetracking and EEG in a laboratory word learning study.** Marc Joanisse (Psych., The Univ. of Western ON, Western Interdisciplinary Res. Bldg., Rm. 3190, London, ON N6A 5B7, Canada, marcj@uwo.ca) and Félix Desmeules Trudel (Psych., The Univ. of Western ON, London, ON, Canada)

A classic finding holds that listeners have significant difficulty categorizing and discriminating unfamiliar/nonnative phonemes. In the present study we examined how this influences learning new words in a second language (L2). Adult monolingual English speakers were trained on a pseudo-French vocabulary, by matching images of cartoon "aliens" to auditory CVCV words incorporating French vowels and consonants. Of interest was comparing words incorporating vowels similar to English to those containing highly unfamiliar vowels (here, the French high front rounded vowel [y]). Accuracy, eyetracking and event-related potentials (ERPs, measured with EEG) were then used to assess word recognition post-training. The neurocognitive measures indicated weakened recognition of words containing the novel [y] vowel, compared to words with vowels that more closely resembled those in English. Furthermore, we found that a training regime that emphasized discriminating easily confused vowels (i.e., [u] vs. [y]) during learning yielded somewhat improved recognition, both immediately after training and in a follow-up session. Interestingly, learning words containing the unfamiliar [y] vowel was not accompanied by improved AX discrimination of this vowel. The results have key implications for how we understand the role of phonology in L2 word representations, and for how we approach L2 teaching.

**11:20–11:40**
**Panel Discussion**

# Session 4aSP

## Signal Processing in Acoustics: Dispersive Wave Signal Processing I

Julien Bonnel, Cochair
*Woods Hole Oceanographic Institution, 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050*

Zoi-Heleni Michalopoulou, Cochair
*Department of Mathematical Sciences, New Jersey Institute of Technology, Newark, NJ 07102*

**Chair's Introduction—8:30**

### *Invited Papers*

**8:35**

**4aSP1. The effects of higher higher-order dispersion on pulse propagation.** Leon Cohen (Phys., City Univ. of New York, Hunter College, 695 Park Ave., New York, NY 10065, leon.cohen@hunter.cuny.edu)

We give explicit formulas for the propagation of a pulse in a dispersive medium governed by a general dispersion relation. In particular, we consider the mean and standard deviation of a propagating pulse and relate them to the parameters of the initial pulse and the dispersion relation. If the dispersion relation is expanded in a Taylor series, higher-order dispersion is when there are terms that are higher than quadratic. We show the effects of the higher-order dispersion on the propagation of the pulse's mean, standard deviation, and contraction/expansion time. Explicit examples will be given.

**8:55**

**4aSP2. Instantaneous frequency: Definitions, interpretations and misconceptions.** Patrick Loughlin (Dept. of Bioengineering, and Elec. & Comput. Eng., Univ. of Pittsburgh, 302 Benedum Hall, Pittsburgh, PA 15261, loughlin@pitt.edu)

Due to the waveguide nature of the ocean surface and bottom, sounds propagating in shallow water can undergo frequency-dependent distortion (specifically, dispersion). This distortion manifests as changes in the instantaneous frequency of the sound, among other attributes. The most common approach for calculating the instantaneous frequency is via the analytic signal, developed by Gabor in the 1940s. Yet, many other definitions of instantaneous frequency have been given, before and since, resulting in a multitude of different answers for a given signal. What, then, is "the instantaneous frequency" of a signal, and what exactly is it a measure of? How are we to choose a particular method? A related question is, given a method, how do we determine whether or not it is a "good" method? We discuss these challenges and the history of the problem; physical constraints that have been proposed in attempts to uniquely define the instantaneous frequency; some common misconceptions; and the intriguing interpretation of instantaneous frequency as the "average frequency at each time" that arises from time-frequency distribution theory.

**9:15**

**4aSP3. Inference of both source and seabed characteristics from broadband signals on the New England continental slope.** David P. Knobles (KSA Corp, 5416 Tortuga Trail, Austin, TX 78731, dpknobles@kphysics.org), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Andrew R. McNeese (ARL:UT, Austin, TX), William Hodgkiss (MPL Scripps Inst. of Oceanogr., San Diego, CA), and Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

In this study the characteristics of the modal dispersion of acoustic waves in littoral ocean environments are utilized to infer both source properties (range, depth, speed, and source level) and geoacoustic properties of the seabed. An analysis is made of broadband acoustic data (10–4500 Hz) taken during 2021–2022 on the New England continental slope over water depths from 200 to 500 m. The acoustic data are generated by both moving broadband sources and explosive sources over range scales of 1–30 km. Such range scales can reveal the nature of the modal dispersion. Conductivity, Temperature, and Depth (CTD) and Expendable Bathythermographs (XBTs) measurements provide a sparse sampling of the temporal and spatial variability of the ocean during the acoustic measurements. Along with bathymetry measurements the sound speed profiles derived from the CTDs and XBTs provide prior information for an inverse problem from which insight can be obtained on the effects of the seabed on the dispersion of broadband propagation.

9:35

**4aSP4. Estimating dispersion relations of ultrasonic guided waves in bone using a modified matrix pencil algorithm.** Tho Tran, Boyi Li (Inst. of Biomedical Eng. and Technol., Acad. for Eng. and Technol., Fudan Univ., Shanghai, China), Ying Li (Ctr. for Biomedical Eng., Fudan Univ., Shanghai, China), Lawrence H. Le (Dept. of Radiology and Diagnostic Imaging, Univ. of Alberta, Edmonton, AB, Canada), and Dean Ta (Ctr. for Biomedical Eng., Fudan Univ.Fudan University, Shanghai, China, tda@fudan.edu.cn)

Guided wave ultrasound technology is well recognized for non-destructive testing. The technology is increasingly applied in bone characterization and imaging to evaluate bone strength and fracture risk. Cortical bone with porous microstructure induces substantial dispersion and attenuation effects on ultrasonic guided waves (UGW). Estimating frequency-dependent propagation characteristics of co-excited wave modes is significant to studying UGW propagation and developing wave-based approaches. This work implements a modified matrix pencil method to simultaneously compute modal wavenumber and attenuation coefficient from dispersive bone UGW signals with improved convergence rate and noise reduction ability. The dispersion estimation is formulated as a matrix pencil or generalized eigenvalue problem with Loewner matrices. The extracted eigenvalues are estimated complex wavevectors, in which the wavenumber and attenuation can be deduced from the real and imaginary components respectively. The performance of the proposed algorithm is demonstrated with low signal-to-noise-ratio synthetic and experimental datasets acquired using axial-transmission measurement settings. The computed dispersive features are validated via comparison with the theoretically calculated dispersion curves by semi-analytical finite-element simulation. The dispersive wave properties are accurately reconstructed in a computationally efficient manner and can be further utilized for bone parametric analyses and subsequently clinical bone health assessment.

9:55

**4aSP5. Quantifying information content of modal dispersion data in geoacoustic inversion.** Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, University of Victoria, Victoria, BC V8W 2Y2, Canada, sdosso@uvic.ca) and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

This paper illustrates the information content of ocean acoustic modal-dispersion data to constrain parameters and uncertainties of seabed geoacoustic models. In particular, time-frequency warping analysis is applied to extract dispersion data consisting of arrival times as a function of frequency for 18 of the first 21 propagating modes from recordings of an impulsive sound source on a vertical hydrophone array. To quantify the information content of these dispersion data to resolve seabed structure, a Bayesian inversion formulation is applied that includes rigorous approaches to model selection and data error modeling. Model selection considers both layered and gradient representations of seabed profiles using trans-dimensional inversion and Bernstein-polynomial basis functions, respectively. In both cases, model parameterizations are determined probabilistically from the data as part of the inversion. The error model assumes a multi-variate Gaussian distribution with unknown variance and covariance for each mode; covariance estimation is formulated in terms of trans-dimensional sampling of zeroth- and first-order autoregressive processes. The applicability of these assumptions/approaches is validated with qualitative (graphical) and quantitative residual analyses. Results are considered as marginal probability profiles for geoacoustic properties, which quantify the resolution of seabed structure versus sub-bottom depth. [Work supported by the Office of Naval Research.]

**10:15–10:25 Break**

10:25

**4aSP6. A physics-informed machine learning based dispersion curve estimation for non-homogeneous media.** Harsha Vardhan Tetali (Elec. and Comput. Eng., Univ. of Florida, 968 Ctr. Dr., Gainesville, FL 32603, vardhanh71@ufl.edu) and Joel Harley (Elec. and Comput. Eng., Univ. of Florida, Gainesville, FL)

Modern machine learning has been on the rise in many scientific domains, such as acoustics. Many scientific problems face challenges with limited data, which prevent the use of the many powerful machine learning strategies. In response, the physics of wave-propagation can be exploited to reduce the amount of data necessary and improve performance of machine learning techniques. Based on this need, we present a physics-informed machine learning framework, known as wave-informed regression, to extract dispersion curves from a guided wave wavefield data from non-homogeneous media. Wave-informed regression blends matrix factorization with known wave-physics by borrowing results from optimization theory. We briefly derive the algorithm and discuss a signal processing-based interpretability aspect of it, which aids in extracting dispersion curves for non-homogenous media. We show our results on a non-homogeneous media, where the dispersion curves change as a function of space. We demonstrate our ability to use wave-informed regression to extract spatially local dispersion curves.



10:45

**4aSP7. Dispersion of broadband signals in the Seabed Characterization Experiment-2022.** Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, gpotty@uri.edu), James H. Miller, Lindsey J. Moss (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Andrew R. McNeese (ARL:UT, Austin, TX), Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), and David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, Seattle, WA)

Broadband sources such as SUS charges and Rupture Induced, Underwater Sound Sources (RIUSS) were deployed in the New England Mud Patch in support of the 2022 Seabed Characterization Experiment. Data measured on the Ocean Bottom Recorders (OBX) and hydrophones will be discussed. Five OBXs were deployed on the seabed and four hydrophones were configured as a tetrahedral array on the bottom mounted Geosled. The OBXs measured three components of particle velocities and acoustic pressure. The spatial variability in the dispersion of broadband signals observed in the pressure and particle velocity data will be discussed. Presence of Scholte waves on the OBX data will be explored using the random decrement technique. Results will be compared with modeling using the seismo-acoustic propagation code (OASES). Outputs of preliminary inversions will be compared with core data collected as part of the experiment. [Work Supported by Office of Naval Research.]

## *Contributed Papers*

**11:05**

**4aSP8. Polarization spectrograms: Unraveling the polarization of ocean acoustic normal modes using the Stokes framework.** Julien Bonnel (Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), Peter H. Dahl, David Dall'Osto (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Nicolas Le Bihan (CNRS GIPSA-Lab, Université de Grenoble Alpes, Grenoble, France), and Julien Flamant (CNRS CRAN, Université de Lorraine, Nancy, France)

Coastal oceanic environments act as dispersive waveguides for acoustic propagation, which is conveniently described using normal mode theory. Here we propose a framework to describe and estimate the polarization of normal modes, as measured using a single vector sensor in the water column. We introduce the Stokes parameters, four real-valued parameters widely used to describe polarization properties in wave physics, notably optics, but largely ignored in ocean acoustics. Analytic expressions for modal Stokes parameters are derived, and a signal processing framework to estimate them is introduced. The concept of polarization spectrograms, which enables visualization of the Stokes parameters in the time-frequency domain, is notably introduced. The whole framework is illustrated for dispersed impulsive signals using both simulations and experimental marine data collected during the Seabed Characterization Experiment. [Work supported by the Direction Générale de l'Armement and by the Office of Naval Research.]

**11:20**

**4aSP9. A computational Bayesian processor for inference regarding range, depth, and velocity of a mobile scatterer under limited aperture.** Abner C. Barros (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, abarros1@umassd.edu) and Paul J. Gendron (ECE, Univ. of Massachusetts Dartmouth, Dartmouth, MA)

A computational Bayesian framework for inference regarding the range, depth and velocity of a submerged mobile localized body in an ocean waveguide is constructed. The approach incorporates the various mixed mode Doppler-frequency dispersion effects associated with the reverberant body's vertical angle scattering. Such coupled eigen returns present angle-frequency modes of intermediate Doppler residing between that of the two coupled specular eigen path Doppler frequencies. It is exactly these modes which present a severe limitation to exploiting closely spaced arrivals in coherence and aperture limited environments. Conditional densities of arrival angle-Doppler are solved via a fast inverse quantile sampler. All other conditionals offer closed form representations in the Gaussian-inverse gamma family. The joint posterior probability density (PPD) of the arrivals are numerically solved and the implied PPD of the object's range, depth, and speed is inferred through acoustic ray interpolation. Case studies are presented with various refractive ocean waveguide environments as well as isovelocity cases. The framework offers a means of incorporating spatio-temporal arrival structure for recursive tracking in an active sonar system. [Work is funded by the Office of Naval Research.]

# Session 4aUW

## Underwater Acoustics and Acoustical Oceanography: Memorial Session for Lisa Zurk I (Hybrid Session)

Kathleen E. Wage, Cochair
*George Mason University, 4400 University Drive, Fairfax, VA 22151*

Martin Siderius, Cochair
*Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201*

Daniel Rouseff, Cochair
*University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:10**

**4aUW1. Lisa Zurk and Matched Field Processing—Her early career at MIT Lincoln Laboratory (1996–2005).** Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

After completing her doctoral program in electromagnetics in 1995 Lisa decided to go to MIT Lincoln Laboratory to work on Matched Field Processing (MFP) in a nascent group formed for sonar and underwater acousitcs efforts. Then MFP introduced new methods of beamforming which were found to be sensitive to assumptions made for the sonar system, the environment and array processing. The so called adaptive one were especially sensitive. Lisa examined the importance of source motions and array are important for any sonar. In this context she introduced (i) mode based rank reduction, or modal MFP, (ii) array tile corrections, (iii) a "dynamic" snapshot correction method a focused MFP while the source is moving similar to beam steered MVDR, and (iv) "instantaneous" data and model based interference filtering. All these were subsequently discovered by later MFP researchers. Lisa and her coauthors were well ahead of there time.

**8:30**

**4aUW2. Lisa Zurk's contributions to MIT Lincoln Laboratory.** Jennifer A. Watson (ISR & Tactical Systems Div., MIT Lincoln Lab., 244 Wood St., Lexington, MA 02421, jwatson@ll.mit.edu)

Lisa Zurk made numerous technical contributions making a substantive impact on the ISR community at a national level. This talk will highlight some of her many technical contributions, ranging from early work in UHF and VHF propagation in rough, overland, terrain, to her pioneering work in adaptive signal processing applied to underwater acoustics. This talk will also recognize her outstanding leadership, always pushing the boundaries as a researcher and as a leader, laying the groundwork for future ISR decision support and architectures. The talk will close with a summary of her role as a mentor and colleague who always challenged those around her to bring out their best. DISTRIBUTION STATEMENT A. Approved for public release. Distribution is unlimited. [This material is based upon work supported by the Department of the Air Force under Air Force Contract No. FA8702-15-D-0001. Any opinions, findings, conclusions or recommendations expressed in this material are those of the author(s) and do not necessarily reflect the views of the Department of the Air Force.]

**8:50**

**4aUW3. A survey of Lisa Zurk's contributions to physics-based signal processing in underwater acoustics.** Douglas Abraham (Ellicott City, MD, abrahad@ieee.org)

The incorporation and exploitation of the physics of underwater acoustic propagation (including reflection and scattering) in signal and information processing can improve the performance of detection, classification, localization and tracking (DCLT) in sonar systems. This presentation will survey several areas of Prof. Lisa Zurk's research from this perspective of physics-based signal processing. Contributions are found in the areas of (i) assessing the impact of propagation through a non-stationary underwater environment on the performance of signal processing algorithms, (ii) adapting signal processing algorithms to be more robust to these environmental effects, and (iii) the exploitation of specific characteristics of the propagation channel in novel approaches to solving the above inferential objectives. Of these areas, identifying consistently observable propagation phenomena that can be exploited to improve DCLT performance is the most essential and yet the most difficult because it requires balancing just enough accuracy in the physical model with a simple

4a THU. AM

enough representation to develop robust signal and information processing algorithms. Impressively, Prof. Zurk was quite successful in this crucial area of physics-based signal processing.

**9:10**

**4aUW4. Focusing on motion and averaging the diagonal: Lisa Zurk's many contributions to passive sonar array processing.** John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, jbuck@umassd.edu), Savas Erdim (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, North Dartmouth, MA), and Yang Liu (Bose Corp., Framingham, MA)

Lisa Zurk's numerous contributions to passive sonar signal processing include motion compensation and regularizing sample covariance matrices (SCM) in snapshot deficient environments. While working at MIT Lincoln Labs, Lisa and colleagues compared the performance of several motion compensation methods including mode-based rank reduction and target motion compensation, or focusing, on the Santa Barbara Channel Experiment dataset [Zurk, Lee & Ward, JASA, 2003]. Later at Portland State, Jorge Quijano and Lisa developed methods for regularizing the SCM through Toeplitz averaging, followed by maximum entropy extrapolation of the SCM to additional lags, [Quijano & Zurk, JASA, 2017]. Toeplitz regularization found significant application in developing "augmented covariance matrices" for DOA estimators on sparse arrays. This talk reviews some of Lisa's contributions to adaptive beamforming and highlights how some of her contributions inspired research in the UMass Dartmouth Signal Processing Group. [Work supported by ONR Code 321US.]

## *Contributed Paper*

**9:30**

**4aUW5. Non-stationary covariance estimation for adaptive beamforming.** Bruce K. Newhall (7504 Broadcloth Way, Columbia, MD 21046, bknewhall@outlook.com)

Standard adaptive beamforming (ABF) assumes a quasi-stationary covariance. However, in low frequency shipping noise the covariance is not stationary, due to the relative motion of ships and receiver. A physical model was previously developed for the behavior of the covariance due to ship motion, and a technique was developed to provide improved estimation. Relatively wide filters were used previously in order to simultaneously estimate the covariance from all moving ships. It has recently been recognized the it is most important in ABF to handle the loudest ship, which is usually the closest and therefore the most non-stationary. Here we will show simulations of the results of using more narrow filters to estimate the covariance due to the loudest ship. The technique will be contrasted to a method of matched field processing with moving ships developed by Lisa Zurk, as well as with derivative-based updating, another non-stationary estimation technique. The potential to improve the estimate by using waveguide invariance across a band of frequencies will also be discussed.

**9:45–10:00 Break**

## *Invited Papers*

**10:00**

**4aUW6. Lisa Zurk's contributions to striation-based signal processing for active sonar.** Daniel Rouseff (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, rouseff@apl.washington.edu), Scott Schecklman (Metron, Portland, OR), and Jorge Quijano (JASCO, Victoria, BC, Canada)

Strong multipath propagation adds considerable complication to acoustic signal processing in the ocean. When mapped in range-frequency space, the acoustic field exhibits striations, alternating bands of high and low intensity due to constructive and destructive interference between the paths. Nearly forty years ago, Russian scientists showed how this striation pattern could be described by a single scalar parameter, the so-called waveguide invariant that subsequently became a staple of their passive sonar signal processing methods. Lisa Zurk's contribution was to show how a striation-based approach could be adapted to active sonar processing. Together with her students, she did tank experiments and analysis for monostatic and bistatic configurations. For a horizontal array, she developed a beamformer featuring a linear frequency shift across the array designed to align with high-intensity striations. The result was improved performance in noisy environments. She showed how the waveguide invariant can improve tracking accuracy by providing a constraint on possible tracks. She and her students demonstrated the improved tracking with continuous active sonar data. Lisa Zurk's innovative work on these topics continues to inspire present day research.

**10:20**

**4aUW7. Phase space approximations for dispersive propagation.** Patrick Loughlin (Dept. of Bioengineering, and Elec. & Comput. Eng., Univ. of Pittsburgh, 302 Benedum Hall, Pittsburgh, PA 15261, loughlin@pitt.edu)

Sounds propagating in shallow water can undergo frequency-dependent changes, specifically damping and dispersion. Phase space analysis and approximations have yielded a clear physical picture of these frequency-dependent effects. In particular, in position-wavenumber phase space, each mode propagates, approximately, at constant velocity given by its group velocity. In time-frequency phase space, each mode undergoes a frequency-dependent time shift given, approximately, by its group slowness. The effects of damping (i.e., a complex dispersion relation) are also clearly manifest. We review and extend various phase space approximations for dispersive propagation, and explore their accuracy. We show that one can recover the exact spectral magnitude and group delay of the propagating pulse from the phase space approximations. One reason the approximations are generally very good is that one does not have to assume that the initial spectrum is slowly varying, as is done, for example, in the stationary phase approximation.

## *Contributed Paper*

**10:40**

**4aUW8. Joint range-velocity estimation with continuous active sonar.** Jeffrey Krolik (Dept. of Elec. and Comp. Eng., Duke Univ., 100 Sci. Dr. Rm. 130, Durham, NC 27708, jlk@duke.edu) and Granger Hickman (ECE, Duke Univ., Durham, NC)

Active sonars using linear frequency-modulated (LFM) continuous waveforms are typically processed in sub-bands to facilitate high target range update rates. Target range-rate is then estimated using incoherently processed matched-filter outputs from each sub-band as input to a tracker. Target returns can easily be confused with clutter because sub-band width is typically chosen to avoid decoherence caused by target and/or platform motion. In this paper, we present methods for joint range-velocity estimation which allows for coherent combination of sub-band outputs and accounts for target/platform motion. The approach involves pre-filtering beamformed sub-band outputs using the transmitted LFM waveform, match-filtering each frequency-domain pre-filtered sub-band under a range-velocity hypothesis, and coherently combining the results. A maximum likelihood estimate (MLE) of range-velocity is formed which involves adaptively suppressing clutter and noise using an estimate of sub-band clutter covariance matrices. Simulation results are presented which indicate that joint range-velocity estimation using coherent processing across sub-bands offers a significant improvement over conventional methods and allows for more flexible selection of sub-band processing bandwidth. Moreover, the approach lends itself to handling the case of partial coherence across sub-bands and offers the potential for improved discrimination of slowly moving targets versus clutter discretes. Work supported by ONR.

## *Invited Papers*

**10:55**

**4aUW9. Tracking in ocean acoustics: Insights from the work of Lisa Zurk.** Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., 323 M. L. King Boulevard, Newark, NJ 07102, michalop@njit.edu)

Tracking is of paramount importance in ocean acoustics. One of the goals is the continuous location estimation of moving sources. Tracking needs also arise in identifying the structure of dispersion curves for long-range sound propagation and multipath arrival time identification across vertically separated hydrophones. Zurk's work has shown that tracking can play a significant role in invariance estimation from striation patterns in spectrograms. We investigated ideas from tracking across the spectrum of ocean acoustics and looked into this latter problem. We found novel approaches for the estimation of the passive and active waveguide invariants, building on a seminal contribution of Lisa Zurk in the field of sonar signal processing.

**11:15**

**4aUW10. Jammer robust model-aided deep learning-based target detection for cognitive sonar.** Touseef Ali (ECEE, Arizona State Univ., Tempe, AZ) and Christ D. Richmond (Dept. of Elec. and Comput. Eng., Duke Univ., Rm. 327, Gross Hall, Box 90984, Durham, NC 27708, christ.richmond@duke.edu)

This work extends the development of an adaptive deep learning-based target detection algorithm for a channel matrix-based cognitive sonar framework that makes use of both a primary and secondary dataset. The channel matrix-based framework models all waveform dependent components, i.e. the sonar echo and clutter, via transfer function matrices, whereas all other waveform independent noise is characterized via the data covariance. By leveraging lessons learned from a derived model-based generalized likelihood ratio test (GLRT), a data driven deep learning algorithm was made adaptive to waveform dependent clutter returns in our previous work. The algorithm, however, remains fragile to waveform independent jamming and interference, and the extensive training required to obtain robust performance is practically infeasible. Thus, in this work further modifications are included to robustify the algorithm against these waveform independent components by leveraging knowledge from the model-based GLRT. The performance of the proposed algorithm is tested using various scenarios with jamming and interference signals.

# Session 4pBAa

## Biomedical Acoustics and Signal Processing in Acoustics: Novel Ultrasound Beamforming Techniques and Their Applications II

Jian-yu Lu, Cochair
*Bioengineering, The University of Toledo, 2801 West Bancroft Street, Toledo, OH 43606*

Hérve Liebgott, Cochair
*CREATIS, Univ. of Lyon 1, Lyon, France*

### *Invited Papers*

**1:30**

**4pBAa1. Monitoring cavitation activity through bone: Passive acoustic mapping for monitoring ultrasound therapy in the central nervous system.** Andrew Frizado, Grace Farbin (Physical Sci. Platform, Sunnybrook Res. Inst., Brampton, ON, Canada), and Meaghan O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Rm C736a, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

The blood-brain barrier (BBB) and blood-spinal cord barrier (BSCB) are major impediments to the pharmacological treatment of central nervous system disorders. Focused ultrasound in combination with intravenously administered microbubbles can transiently increase the permeability of the BBB and BSCB to enable passage of therapeutics from the blood pool to the brain and spinal cord tissues. Safely controlling these treatments relies on information such as the cavitation state (i.e., stable vs. inertial), and location. Passive acoustic mapping (PAM) is a beamforming approach that does not require absolute time of flight knowledge and is therefore well suited for monitoring ultrasound therapy where therapy pulses are long, ranging from several cycles to milliseconds in length. PAM applied with large receiver arrays can enable three-dimensional monitoring of cavitation activity through the bone for non-invasive treatment monitoring. This talk will discuss the application of PAM in brain and spinal cord and the challenges posed by the skull and vertebral bony geometry, with a focus on recent in silico and experimental work in transvertebral PAM.

**1:50**

**4pBAa2. Ultrasound image formation in the deep learning age.** Muyinatu Bell (Johns Hopkins Univ., 3400 N. Charles St., Barton 208, Baltimore, MD 21218, mledijubell@jhu.edu)

The success of diagnostic and interventional medical procedures is deeply rooted in the ability of modern imaging systems to deliver clear and interpretable information. After raw sensor data is received by ultrasound and photoacoustic imaging systems in particular, the beamforming process is often the first line of software defense against poor quality images. Yet, with today's state-of-the-art beamformers, ultrasound and photoacoustic images remain challenged by channel noise, reflection artifacts, and acoustic clutter, which combine to complicate segmentation tasks and confuse overall image interpretation. These challenges exist because traditional beamforming and image formation steps are based on flawed assumptions in the presence of significant inter- and intrapatient variations.In this talk, I will introduce the PULSE Lab's novel alternative to beamforming, which improves ultrasound and photoacoustic image quality by learning from the physics of sound wave propagation. We replace traditional beamforming steps with deep neural networks that only display segmented details, structures, and physical properties of interest. I will then transition to describing a new resource for the entire community to standardize and accelerate research at the intersection of ultrasound beamforming and deep learning. This resource includes the first internationally crowd-sourced database of raw ultrasound channel data and integrated beamforming and evaluation code (see https://cubdl.jhu.edu/ & https://pulselab.jhu.edu/ for more details).

## Contributed Papers

**2:10**

**4pBAa3. Acoustic Signature: A focused ultrasound guidance technique with sub-millimeter accuracy.** Thomas J. Manuel (BME, Vanderbilt Univ., 1161 21st Ave., South Medical Cent, Nashville, TN 37232, thomas.j.manuel@vanderbilt.edu), Aparna Singh (BME, Vanderbilt Univ., Nashville, TN), Jiro Kusunose, and Charles F. Caskey (Radiology, Vanderbilt Univ. Medical Ctr., Nashville, TN)

Accurate targeting is paramount for focused ultrasound (FUS) procedures. Here we describe a method for targeting FUS dubbed acoustic signature, which uses acoustic feedback from a target's unique reflection patterns to guide the transducer to a previously defined orientation. We demonstrate convergence to the desired target in a 3 degree of freedom (DOF) water tank scenario and a 5 DOF robotic arm scenario. We also tested the acoustic signature technique for targeting with a phantom skull on a 3 DOF motor stage and a 5 DOF manual stereotactic frame designed for transcranial FUS procedures in an MRI environment. In both the water tank and the robotic arm convergence tests, the method converges to the target. The convergence was sufficiently smooth for automated gradient descent. In the 3-DOF test, the net targeting error was $0.30 \pm 0.27$ mm ($n = 10$). In the stereotactic frame 5-DOF test, the targeting error was $2.11 \pm 0.99$ mm and $3.2 \pm 2.2°$ ($n = 10$) summed rotational error. The accuracy of this technique is limited by the positioning apparatus. This method may enable precise and repeatable FUS targeting for therapies which require multiple FUS sessions at the same target.

**2:25**

**4pBAa4. A phase-shifting method for computation reduction for high-frame-rate imaging.** Jian-yu Lu (Bioengineering, The Univ. of Toledo, 2801 West Bancroft St., Toledo, OH 43606, jian-yu.lu@ieee.org)

High-frame rate (HFR) imaging using steered plane wave (SPW) or limited-diffraction beam has a high-temporal resolution and thus has found many applications. To further study the HFR imaging methods, computer simulations were performed. However, the simulations require a large number of computations, especially for 3D imaging with 2D array transducers for a large imaging volume. In this paper, a phase-shifting method was developed to reduce the number of computations. In the method, the grid points of the transmit and receive beams were calculated at 1-mm interval in the depth direction that is perpendicular to the transducer surface. The interval is much larger than the 1/4 of the 0.58-mm wavelength required for an accurate interpolation for millions of random scatterers in pulse-echo response without aliasing. Since the HFR imaging uses either SPW or LDB, the wave vectors of these beams are fixed at each frequency. Due to the fact that the amplitude of ultrasound beams changes very little over a couple of wavelengths, the interpolation in the depth direction was replaced with a phase shift. Results show that images reconstructed with the phase-shifting method removed the artifacts caused by aliasing when conventional tri-linear interpolations were used for 3D imaging.

**2:40–2:55 Break**

**2:55**

**4pBAa5. 3D-printed gradient-index phononic crystal lens for transcranial focused ultrasound.** Eetu Kohtanen (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr NW, Atlanta, GA 30332, ekohtanen3@gatech.edu), Ahmed Allam, and Alper Erturk (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Transcranial focused ultrasound (tFUS) shows great promise as a noninvasive tool to treat neurological conditions such as essential tremor. The existing clinical phased array systems are mostly intended for ultrasound delivery to the center of the brain (as in thalamotomy for essential tremor), in addition to being complex and expensive. To seek an alternative focusing approach especially for the brain periphery, we explore a 3D-printed gradient-index (GRIN) lens as a simple and an orders of magnitude more cost-effective approach. The lens is constructed using a phononic crystal (PC) architecture with varying lattice geometry and hence refractive index distribution. Specifically, the lens uses an axisymmetric hyperbolic secant refractive index profile to focus ultrasonic waves generated by a 1 MHz single-element ultrasonic transducer. Finite element simulations are performed to design and analyze the GRIN-PC lens, and to explore the effects of various parameters such as the distance from the skull and the incidence angle. The numerical results are validated experimentally for a 3D-printed lens by scanning the 3D pressure field generated through a temporal bone. This cost-effective approach to tFUS can open new possibilities to 3D print lenses based on patient computed tomography scans for various applications from tissue ablation to neurostimulation.

**3:10**

**4pBAa6. Reconstruction of thermoacoustic emission sources from proton irradiation using numerical time reversal.** T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu), David A. Johnstone (Medical Phys., Univ. of Cincinnati, Cincinnati, OH), Charles L. Dumoulin (Radiology, Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), Michael A. Lamba (Radiation Oncology, Univ. of Cincinnati, Cincinnati, OH), and Sarah K. Patch (Acoust. Range Estimates, Chicago, IL)

Dose delivery in proton beam therapy for cancer treatment can be mapped by analyzing thermoacoustic emissions measured by ultrasound arrays. Here, a method is presented for spatial mapping of thermoacoustic sources using numerical time reversal, simulating physical re-transmission of measured emissions into the medium. The spatial distribution of acoustic sources is shown to be approximated by the amplitude envelope of the time-reversed field, evaluated at the time of emission. Given calibration of the array sensitivity and knowledge of tissue properties, this approach approximately reconstructs the induced acoustic pressure, equal to the product of radiation dose, density, and Grueneisen parameter. Numerical time reversal is implemented using two models for array elements, as either ideal line sources or diffracting rectangular radiators. Demonstrated reconstructions employ previously reported measurements of thermoacoustic emissions from proton energy deposition in tissue-mimicking phantoms. For a phantom incorporating a bone layer, reconstructions account for the higher sound speed in bone. Spatial resolution of reconstructions, assessed by widths of reconstructed Bragg peaks, is improved in the array direction by incorporation of diffraction effects. In comparisons with corresponding Monte Carlo simulations, source distributions correspond well with simulated proton dose, while source localization with respect to room coordinates is improved by incorporating sound speed inhomogeneities.

**3:25**

**4pBAa7. A straightforward method for simulating phase aberration with high fidelity using PDMS phantoms.** Ying-Chun Pan (Biomedical Eng., Vanderbilt Univ., 5824 Stevenson, Nashville, TN 37232, ying-chun.pan@vanderbilt.edu), Christopher Khan, Katelyn Craft, Braden Huneycutt, Rachel Hecht, Eric Tang, and Brett Byram (Biomedical Eng., Vanderbilt Univ., Nashville, TN)

Phase aberration arises from the speed of sound heterogeneity in the imaging environment and degrades image quality. Accurate aberration simulation is essential for developing aberration correction methods. Existing works often apply an aberration value at each channel in simulation software, but this approach introduces aberration integration error, assumes a fixed profile across all beams, and does not account for harmonic generation. We propose to address these limitations by making a PDMS aberration phantom. The manufacturing process involves (1) integrating a software-generated profile into a 3D-printed mold, (2) treating the mold with acrylic lacquer to prevent cure inhibition, (3) casting the mold with PDMS and degassing for an hour, and (4) baking at 75 °C for 4 h before demolding. The phantom is smooth and retains software-specified root mean square and full width half max of the autocorrelation function, and can be placed at the transducer to simulate aberration with higher fidelity than software. OCT data suggests that 3D-printed molds are accurate to within 21 $\mu$m of the

software profiles and beamformed aberrated data exhibits a profile-dependent increase in sidelobes relative to clean data.

**3:40**

**4pBAa8. Passive cavitation imaging in fuel injector nozzles.** Ankush Gupta (Dept. of Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, ag1@bu.edu), Grace K. McDonough, Jennifer Moreira, Jeremy Lau, Victoria V. Doheny, Emily Ryan, Sheryl M. Grace (Mech. Eng., Boston Univ., Boston, MA), and R. Glynn Holt (Phys. & Astronomy Dept., Hampden-Sydney College, Hampden-Sydney, VA)

Modern fuel injection systems operate at high pressures and flow velocities. Injectors can be pressurized in excess of 10 MPa, resulting in fuel velocities on the order of hundreds of meters/second in the mm and sub-mm internal confines of a fuel injector. Subsequent ejection velocities at the nozzle yield characteristic atomization droplet size distributions and spread angles. Studies have shown or inferred the presence of cavitation in such fuel injectors, typically beneficially decreasing ejection droplet sizes while increasing the spray spreading angle. While beneficial for fuel atomization, it is known that bubble collapse near a solid surface produces a strong jet which impinges on the surface and causes erosion. In this study, Fourier based image reconstruction is used to perform Passive Cavitation Imaging (PCI) in laboratory nozzles to detect, characterize, and most importantly localize inertial cavitation. [Work funded by DOE.]

**3:55**

**4pBAa9. Superresolution using synthetic shaped acoustic vortex waves in water.** Amber M. Groopman (Acoust. Div., Code 7160, U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Bldg. 2, Washington, DC 20375, amber.groopman@nrl.navy.mil) and Matthew D. Guild (Acoust. Div., Code 7160, U.S. Naval Res. Lab., Washington, DC)

Acoustic vortex waves have previously been shown to be able to overcome the diffraction limit, through the use of subdiffraction-limited pressure nulls that propagate well into the farfield. Acoustic vortex waves, therefore, present a novel means to achieve farfield superresolution imaging. However, generating acoustic vortex waves in an experimental setting typically requires a complicated phased array consisting of multiple active elements in a fixed geometrical configuration. In this work, we describe how an acoustic vortex wave can be generated using a synthetized vortex wave array applied during post-processing to in-water 2D plane measurements obtained with a single moving acoustic source and a fixed receiver. The geometric versatility of the synthetic vortex array enables different shaped acoustic vortex patterns to be achieved using the same in-water measurements. Experimental and theoretical results of the magnitude and phase of the nearfield and farfield pressure fields will be presented for a variety of geometric configurations and vortex integer wave modes of the synthesized vortex wave array. The data show excellent agreement with expected results and demonstrates shaped acoustic vortices with superresolved features in both the nearfield and farfield. [Work supported by the Office of Naval Research.]

## Session 4pBAb

# Biomedical Acoustics: Detection and Quantification of Bubble Activity in Therapeutic Ultrasound II

Adam Maxwell, Cochair
*Department of Urology, University of Washington, Seattle, WA 98195*

Eli Vlaisavljevich, Cochair
*Dept. of Engineering Mechanics, Virginia Tech, Blacksburg, VA 24061*

### *Contributed Papers*

**1:00**

**4pBAb1. Thrombolytic efficacy of histotripsy combined with thrombolytic-loaded echogenic liposomes.** Connor Centner (Dept. of Radiology, Univ. of Chicago, 5812 S. Ellis, Chicago, KY 60637, ccentner@uchicago.edu), Aarushi Bhargava (Dept. of Radiology, Univ. of Chicago, Chicago, IL), Shaoling Huang, David D. McPherson (UT Health Houston, Houston, TX), and Kenneth B. Bader (Dept. of Radiology, Univ. of Chicago, Chicago, IL)

Deep vein thrombosis (DVT) is a substantial burden to the American public. Thrombolytics drugs are the front-line approach to treat the most egregious forms of DVT. Histotripsy is a bubble-based focused ultrasound therapy that enhances thrombolytic activity while simultaneously breaking down erythrocytes within the thrombus mechanically. Loading thrombolytics inside echogenic liposomes may accentuate the action of histotripsy via site-specific thrombolytic release and prompt bubble nuclei. The objective of this study was to assess clot degradation under the action of histotripsy and echogenic liposomes in an *in vitro* venous flow model. Histotripsy pulses were applied along the length of the clot at pulse peak negative pressures of 0 (control) or 42 MPa. Echogenic liposomes were diluted into the perfusate such that the thrombolytic concentration was 0 or 2.68 $\mu$g/ml. The combination of histotripsy and echogenic liposomes displayed increased clot mass loss, thrombolytic activity, hemolysis, and intact erythrocytes in the perfusate relative to echogenic liposomes alone (ANOVA $p < 0.05$, $n = 6$–8/group). Expect for hemolysis, markers of clot degradation were increased for histotripsy and echogenic liposome arms relative to histotripsy exposure alone. Together, these data indicate histotripsy combined with thrombolytics is a promising approach for the treatment of DVT.

**1:15**

**4pBAb2. The role of histotripsy pulsing rate on clot degradation when combined with thrombolytic drug.** Joel Toledo-Urena (Univ. of Chicago, 5841 S. Maryland Ave., MC2026, Chicago, IL 60637, jtoledo@uchicago.edu), Connor Centner (Univ. of Chicago, Chicago, KY), Aarushi Bhargava, and Kenneth B. Bader (Univ. of Chicago, Chicago, IL)

Histotripsy is a bubble-based focused ultrasound therapy under development for tissue ablation, including the treatment of venous thrombosis. Histotripsy is effective for lysing erythrocytes within the clot, but not its fibrin structures. Our prior work has demonstrated when combined with a standard-of-care thrombolytic drug, histotripsy can degrade both cellular and extracellular clot components. To date, only histotripsy pulsing rates of 20–40 Hz have been tested for this combination approach. Previous studies have shown the pulsing rate influences the bubble dynamics, and therefore treatment outcomes. The goal of this study was to determine the effect of pulsing rate on clot degradation. Human whole blood clots were exposed to thrombolytic drug and histotripsy pulses. The pulsing rate ranged from 5 to 500 Hz. Following treatment, metrics of clot degradation (change in clot mass, and markers of erythrocyte and fibrin degradation) were tabulated. An improvement in treatment outcomes was observed for the combination approach relative to histotripsy alone for the 50 Hz pulsing rate. At 500 Hz, the benefit of combining thrombolytic drug with histotripsy was not well defined. Overall, these results indicate the pulsing rate has a role in the ablative and drug delivery capacity of histotripsy.

**1:30**

**4pBAb3. Effects of pulse repetition frequency on bubble cloud characteristics and ablation for single-cycle histotripsy.** Alex Simon (Biomedical Eng. and Mech., Virginia Tech, 325 Stanger St., Blacksburg, VA 24061, alex34@vt.edu), Connor Edsall, and Eli Vlaisavljevich (Biomedical Eng. and Mech., Virginia Tech, Blacksburg, VA)

Histotripsy is a cavitation-based focused ultrasound ablation method in development for multiple clinical applications. This work investigates the effects of pulse repetition frequency (PRF) on histotripsy bubble cloud characteristics and ablative capabilities for single cycle histotripsy. Bubble clouds produced by a 500 kHz histotripsy system at PRF's from 0.1 to 1000 Hz were visualized using high speed imaging in 1% agarose phantoms. Cloud images were analyzed to determine bubble cloud density (bubbles/$mm^2$) and pulse-to-pulse bubble correlation. Ablation was assessed through lesion analysis in red blood cell (RBC) phantoms. Results showed cavitation clouds generated at low PRF were characterized by consistently dense bubble clouds ($49.6 \pm 7.7$ bubbles/$mm^2$, 1 Hz) that closely matched regions above the histotripsy intrinsic threshold. Bubble clouds formed at high PRF had significantly lower cloud density ($20.0 \pm 8.3$ bubbles/$mm^2$, 1000 Hz). In addition, bubbles in higher PRF clouds had significantly increased pulse-to-pulse correlation, characteristic of what has been reported as the cavitation memory effect. Results from RBC ablation showed that higher PRF generated lesions had lower adherence to the focal region and less repeatability compared to low PRF ablations. This study demonstrates essential differences in bubble cloud characteristics that will help guide future histotripsy pulsing strategies.

**1:45**

**4pBAb4. Spatio-temporal evaluation of anti-biofilm cavitation activity by passive acoustic mapping.** Sara Keller (Inst. of Biomedical Eng., Univ. of Oxford, Botnar Res. Ctr., Windmill Rd., Headington, Oxford OX3 7LD, United Kingdom, sara.keller@eng.ox.ac.uk), Gareth LuTheryn (School of Pharmacy, Univ. College London, London, United Kingdom), Michael Gray, Eleanor P. Stride, Robin O. Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Headington, Oxford, Oxfordshire, United Kingdom)

Bacterial biofilms present a major challenge to achieving effective antibiotic therapy, as these sessile communities of microbes confer protection to bacteria by decreasing antibiotic efficacy. Focused ultrasound can

mechanically disrupt biofilms, offering a new 'drug-free' antibiotic paradigm. The goal of this work is to validate, quantify, and optimize the role of acoustic cavitation in the biofilm disruption process through spatiotemporal monitoring of cavitation activity. A clinical isolate strain of *Staphylococcus aureus* from native valve endocarditis was cultured for 72 hours within a flow channel to form a biofilm. A 1.1 MHz spherically focused transducer was used to expose the biofilm from below at a 45° angle. The *in situ* acoustic field was characterized with a fibre-optic hydrophone. A calibrated 5–11 MHz linear array was placed 25 mm above the biofilm in order to record acoustic emissions during biofilm disruption from which passive acoustic maps of cavitation could be derived. Biofilms were exposed to 4.5 MPa peak rarefactional pressure (derated), 10,000 cycles, at a 1 Hz PRF. Qualitative reduction of biofilms was assessed by live/dead staining with Syto 9/propidium iodide, which was correlated with cavitation activity observed in the passive acoustic maps.

**2:00**

**4pBAb5. Characterization and monitoring of cavitation behavior induced by a dual-mode pulsed high-intensity focused ultrasound array.** Randall P. Williams (Div. of Gastroenterology & Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th Street CIMU - Portage Bay Buiilding, Seattle, WA 98105, rpw1@uw.edu), Vera A. Khokhlova (Univ. of Washington and M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Univ. of Washington and M.V. Lomonosov Moscow State Univ., Seattle, WA), and Tatiana D. Khokhlova (Div. of Gastroenterology & Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

Pulsed high-intensity focused ultrasound (pHIFU) is capable of inducing cavitation in tissue without the need for contrast agents, thereby enhancing permeability and passive drug penetration in tissue with poor perfusion. We have developed and characterized the performance of a 64-element linear ultrasound array for image-guided therapy, capable of generating sufficient levels of inertial cavitation at 1 MHz, along with B-mode and Doppler imaging for targeting and cavitation assessment, respectively. Cavitation behaviors induced by different pHIFU exposures were investigated using high-speed imaging in agarose and polyacrylamide phantoms in conjunction with passive cavitation detection (PCD) and plane wave Doppler imaging. Image quality assessment was performed on *in vivo* porcine models, and cavitation monitoring was demonstrated on *ex vivo* tissues. High-speed images of the induced cavitation behavior while steering the beam focus both axially and laterally (within $\pm$1.2 cm) were in agreement with Doppler power distributions and broadband noise emissions from PCD. Destructive cavitation behaviors in both gels and tissue were induced at relatively low peak-negative pressures (3–4 MPa), provided that the pHIFU focal waveform was nonlinearly distorted. B-mode image quality is sufficient for recognizing the relative position of anatomic landmarks and targeting cm-sized regions. [Work supported by NIH grants R01EB023910 and R01EB025187.]

**2:15**

**4pBAb6. Extension of boiling histotripsy lesions by axial focus steering during pulse delivery.** Gilles P. Thomas (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, gillespierre.thomas@gmail.com), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Vera A. Khokhlova (Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation)

Boiling histotripsy (BH) is a pulsed high intensity focused ultrasound (HIFU) method relying on the generation of high amplitude shocks and bubble activity to induce tissues liquefaction. A sequence of pulses, 1–20 ms long, generates boiling bubbles at the focus of the HIFU transducer within each pulse, and the remainder of the pulse then interacts with those bubbles. One effect is the creation of a prefocal bubble cloud due to shock scattering: the shock is inverted when reflected from the bubble wall resulting in sufficient negative pressure to reach intrinsic cavitation threshold immediately proximally to these bubbles. Here, a methodology is proposed to extend the length of this prefocal bubble cloud by steering the focus toward the transducer during the BH pulse and thus accelerate treatment. A BH system comprising a 1.5 MHz 256-element phased array connected to a Verasonics V1 system was used. High-speed imaging in transparent gels was performed to observe the extension of the bubble cloud resulting from shock scattering. Volumetric BH lesions were generated in *ex vivo* tissue. Results showed a threefold increase of the volumetric ablation rate with focus steering compared to standard BH. [Work supported by NIH R01EB007643, R01GM122859, and R01EB25187.]

**2:30**

**4pBAb7. Histotripsy significantly decreases tumor viability in neuroblastoma xenograft model.** Isabella J. Iwanicki (Dept. of Pediatric Surgery, Univ. of Chicago, 5841 S. Maryland, Ste. AB524, Chicago, IL 60637, iji@uchicago.edu), Lydia Wu, Fernando Flores-Guzman (Dept. of Pediatric Surgery, Univ. of Chicago, Chicago, IL), Connor Centner (Univ. of Chicago, Chicago, KY), Kenneth B. Bader (Univ. of Chicago, Chicago, IL), and Sonia Hernandez (Dept. of Pediatric Surgery, Univ. of Chicago, Chicago, IL)

Children diagnosed with high-stage neuroblastoma (NB) have a $<$50% 5-year survival rate, resisting intensive treatment. Histotripsy, a focused ultrasound method, can ablate subcutaneous tumors. Here, we characterize histotripsy in an abdominal NB-xenograft model.Intrarenal injection of NGP-Luciferase cells in female NCr Nude mice generated 1–2 g NB tumors after 5–6 weeks. We assessed tumor viability with bioluminescence before, after, and 24h after Histotripsy. A 1-MHz focused source under ultrasound image guidance delivered 4–6 pulses per tumor with individual targets separated by $\sim$1 mm. Immunostains of the apoptosis marker TUNEL, endothelial marker Isolectin-B4, and hypoxia-marker pimonidazole were imaged or scanned. Statistics were performed using Graphpad. Histotripsy decreased bioluminescence by $\sim$50% ($p = 0.02$, $n = 7$), suggesting a decrease in viability. Untreated tumors did not change ($n = 4$). TUNEL staining increased in Histotripsy-treated tumors compared to controls (56 $\pm$ 6 versus 9 $\pm$ 3 %area, $n = 3$ to 8, $p < 0.0001$). Histotripsy increased pimonidazole positivity adjacent to targeted areas, suggesting hypoxia. Finally, Histotripsy increased red blood cells compared to controls. Histotripsy dramatically reduces tumor viability by inducing apoptosis of targeted areas. Hypoxia patterns suggest histotripsy alters perfusion and/or permeability within the tumor, indicating potential for synergy with chemotherapy.

**2:45–3:00 Break**

**3:00**

**4pBAb8. Extending the Iterative Nonlinear Contrast Source method to simulate mutual interaction in large populations of microbubbles.** Martin D. Verweij (Section Medical Imaging, Dept. Imaging Phys., Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CD, Netherlands, M.D.Verweij@tudelft.nl) and Agisilaos Matalliotakis (Section Medical Imaging, Dept. Imaging Phys., Delft Univ. of Technol., Delft, Netherlands)

There is an ongoing development of ultrasound contrast agents (UCAs) for specific medical diagnostic and therapeutic applications. Recent researches involve all kinds of targeted and loaded microbubbles, monodisperse microbubbles, and phase-change nanodroplets. To design and improve applications that use UCAs, it is necessary to have a thorough understanding of the mutual interaction of ultrasound fields and UCAs. The individual response of microbubbles and nanodroplets is extensively studied and, in the case of bubbles, multiple variants of the Rayleigh-Plesset equation are available to describe their behavior. However, novel applications, such as therapeutic proton beam localization, may strongly depend on the mutual interaction in a population of UCAs. In this presentation, a numerical approach for the analysis of ultrasound-bubble interaction in a population with many (order one million) microbubbles will be discussed. This approach is based on the earlier developed Iterative Nonlinear Contrast Source (INCS) method for simulating nonlinear ultrasound waves. In the current case, the population of microbubbles will be represented by a set of pressure-dependent contrast point sources that are iteratively updated by the INCS method, where each iteration adds an order of multiple scattering. Results will be presented for the linear scattering in populations with different concentrations of microbubbles.

**3:15**

**4pBAb9. Simulating multiple scattering inside a population of nonlinearly oscillating microbubbles using the Iterative Nonlinear Contrast Source method.** Agisilaos Matalliotakis (Section Medical Imaging, Dept. Imaging Phys., Delft Univ. of Technol., Delfgauseweg 35, Delft 2628EG, the Netherlands, a.matalliotakis@tudelft.nl) and Martin D. Verweij (Section Medical Imaging, Dept. Imaging Phys., Delft Univ. of Technol., Delft, the Netherlands)

For several decades, microbubbles have been the primary choice for ultrasound contrast agents both for efficiency and safety reasons. To optimize their performance in various applications e.g. proton therapy, it is important to understand the dynamics of populations of nonlinearly oscillating microbubbles. Especially for high concentration clouds, multiple scattering can significantly influence the propagation of medical ultrasound. To numerically study the higher-order interaction between nonlinear microbubbles, we present a modified version of the Iterative Nonlinear Contrast Source (INCS) method. Based on a Neumann iterative scheme, the acoustic pressure is obtained by treating the nonlinearly scattering bubbles as individual contrast sources in a "linearized" background medium. In each iteration, the full nonlinear pressure field is updated by applying the 4D spatiotemporal convolution between the background Green's function and the contrast sources obtained from the previous iteration. In this study, all microbubbles may exhibit an individual behavior. To accommodate this, in each iteration the solution of the Marmottant equation for each microbubble is obtained and used as the temporal signature of the respective contrast source. Using this scheme, each iteration adds an order of multiple scattering. The accuracy and the efficiency of INCS results will be demonstrated for monodisperse and polydisperse concentrations.

**3:30**

**4pBAb10. Intrusive temperature measurement of the cavitation bubbles using cold wire.** Roshan Kumar Subramanian (Aerosp. and Ocean Eng., Virginia Tech, 460 Old Turner St., 215 Randolph Hall, Blacksburg, VA 24060, roshan@vt.edu) and Olivier Coutier-Delgosha (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA)

Cavitation is a phenomenon of rupturing a liquid when the liquid is subjected to a decrease in pressure at roughly constant temperature. Cavitation is the fundamental reason for the material erosion when the bubbles are collapsing closer to the surface. These bubbles also produce a very high temperature (thousands of degrees Celsius) for a very short time (in the order of nano or microseconds). Cavitation is also being used in the non-invasive cancer tumor ablation treatment using high intensity focused ultrasound (HIFU) called histotripsy. So, our research in measuring the temperature of the bubble collapse will help the histotripsy treatment to understand its thermal behavior. We have designed a cold wire sensor of the thickness of $\sim$10 to 50 micrometers. And, the sensing region is 2 millimeter. These cold wires are made with kevlar or nylon fibers, titanium, nickel, silver, and silicon dioxide. The single bubble or histotripsy cloud, is allowed to collapse on the sensor, and we measure the temperature based on its change in resistance. These changes in resistance are corresponded to the temperature using the mathematical function which is calibrated prior to the experiment. We measure it in three different experimental settings like Tube Arrest Method (TAM), LASER induced bubbles, and HIFU induced bubbles. The preliminary results have shown that the temperature of the single spherical bubble collapse (TAM) can rise to thousands of degrees celsius as mentioned in the literature based on the theoretical analysis.

**3:45**

**4pBAb11. Bubble-cloud characteristics and ablation efficiency in dual-frequency intrinsic threshold histotripsy.** Connor W. Edsall (Biomedical Eng. and Mech., Virginia Polytechnic Inst. and State Univ., 325 Stanger St., 340 Kelly Hall, Blacksburg, VA 24061, edsallcw@gmail.com), Laura Huynh (Mater. Sci. and Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA), Timothy L. Hall (Univ. of Michigan, Ann Arbor, MI), and Eli Vlaisavljevich (Biomedical Eng. and Mech., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

Intrinsic threshold histotripsy has been shown to nucleate bubble clouds and generate ablation closely matching the dimensions of the focal region above the intrinsic threshold ($\sim$25–28MPa) with the ablation efficiency dependent upon the size and density of bubbles within the cloud. This work investigates the effects of dual-frequency histotripsy pulsing on bubble-cloud characteristics and ablation efficiency. Histotripsy was applied to agarose tissue phantoms (1%) using a dual-frequency 500kHz–3MHz array transducer with equal pressure delivered by each frequency. The arrival time of the 3 MHz pulse was varied relative to the 500 kHz pulse, and high-speed imaging was used to characterize the bubble-cloud dimensions, bubble density, and individual bubble size. Ablation was determined using red blood cell phantoms. Results indicated that dual-frequency pulsing generated bubble clouds with intermediate cloud size, bubble size, and bubble density compared to single-frequency (500 kHz and 3 MHz) pulsing, with these characteristics further modulated by the respective pulse arrival times. Additionally, dual-frequency pulsing increased ablation efficiency compared to previously published single frequencies (500 kHz and 3 MHz) with the most efficient ablation observed for cases where the 3 MHz pulse arrived at the leading edge of the 500 kHz pulse. Overall, this investigation demonstrates possible benefits of dual-frequency histotripsy pulsing, warranting further investigation.

**4:00**

**4pBAb12. An *ex vivo* experimental demonstration of passive acoustic mapping through the human spinal column.** Grace Farbin (Physical Sci., Sunnybrook Res. Inst., 38 Spotted Owl Crescent, Brampton, ON L7A0J9, Canada, gracefarbin@gmail.com), Andrew Frizado, and Meaghan O'Reilly (Physical Sci., Sunnybrook Res. Inst., Toronto, ON, Canada)

The delivery of drugs to the central nervous system is grossly limited by the presence of the blood-brain and blood-spinal cord barriers. These barriers can be transiently permeabilized using low-intensity focused ultrasound combined with circulating microbubbles. The spectral content of acoustic emissions from cavitating microbubbles can be analyzed to control for and mitigate potential bioeffects. However, there is a large degree of uncertainty as to where these cavitation signals are originating. Passive beamforming of microbubble emissions recorded using multi-element arrays can enable spatial mapping of cavitation activity. Passive acoustic mapping (PAM) is challenging to implement in the presence of an intervening bone layer but has been successfully demonstrated through the skull. Here we present the first experimental demonstration of PAM through human vertebral bone. A tube containing flowing microbubbles was placed in the canal of *ex vivo* human thoracic vertebrae (stack of 3 vertebrae) and excited (250 kHz) through the right laminae. A 64-element large aperture 2D array was used to receive the harmonic emissions through the left laminae. Reconstructed maps successfully localized the cavitation activity to the spinal canal. Future work will examine effects of phase/amplitude correction, receiver number, and location of the cavitation relative to the vertebral anatomy.

**4:15**

**4pBAb13. Passive and Doppler-based assessment of cavitation activity induced by pulsed focused ultrasound.** Minho Song (Mech. Eng., Univ. of Washington, 1715 NE Columbia Rd., Seattle, WA 98195, songmh@uw.edu), Oleg A. Sapozhnikov, Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Joo Ha Hwang (Dept. of Medicine, Stanford Univ., Palo Alto, CA), and Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA)

Pulsed focused ultrasound (pFUS) exposures utilizing short, nonlinearly distorted pulses at low duty cycle have been shown to enhance drug and gene delivery to targeted tissue through inertial cavitation activity. Passive cavitation detection (PCD) and mapping of broadband emissions are current conventional methods to monitor and quantify cavitation but provide limited spatial resolution. Here, plane-wave Doppler imaging was used with PCD to quantify pFUS-induced cavitation in *ex vivo* bovine tissues and *in vivo* surgically exposed porcine liver, kidney, and pancreas. A 1.5 MHz FUS transducer (aperture 75 mm, F-number 0.75) was used to deliver 60 pulses (duration 1 ms, 0.1% duty cycle, focal pressure $p^+=70$i–110 MPa, $p^-=13$–20 MPa). A coaxially mounted ATL P7-4 ultrasound imaging probe was used for PCD during the FUS pulse, and Doppler and B-mode sequences. Disrupted tissue areas were collected for histology and compared to Doppler power images. Maximum Doppler power was found to correlate to broadband noise level for each

FUS pulse. The Doppler power map integrated over the exposure was observed to correlate spatially with tissue disruption area from histology, which thus represents a promising real-time metric for quantifying cavitation activity induced by pFUS exposures. [Work supported by NIH R01CA154451, R01EB025187, and R01EB23910.]

**4:30**

**4pBAb14. Power Doppler-based ultrasound detection of cavitation for burst wave lithotripsy.** Adam Maxwell (Urology, Univ. of Washington, Dept. of Urology, University of Washington, Seattle, WA 98195, amax38@u.washington.edu), Bryan W. Cunitz, Yak-Nam Wang, Christopher Hunter, Ga Won Kim, Stephanie Totten, Michael R. Bailey, and Wayne Kreider (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Burst wave lithotripsy (BWL) is a noninvasive method to fragment urinary stones with pulsed focused ultrasound. Cavitation activity produced by ultrasound pulses in BWL can cause multiple effects including stone fragmentation, acoustic shielding, and tissue injury. Monitoring cavitation activity and spatial extent is therefore important to evaluating these effects and determining parameters that maximize effectiveness and safety. *In vivo* experiments in a porcine model were conducted using a Verasonics ultrasound system to detect the onset and spatial extent of cavitation through synchronized power Doppler ultrasound imaging during 350-kHz BWL exposures to stones and kidney tissue. Kidneys ($n = 15$) were exposed *in vivo* to transcutaneous BWL at multiple pressure levels, and cavitation in tissue was found to be detected as an increase in the measured Doppler power by an average of 28.6 dB (range 11–51 dB) over the background. Cavitation occurred at average levels between 5.8 and 8.0 MPa peak negative pressure, depending on targeting location, stone presence, and exposure parameters. Cavitation presence further correlated with gross observation of injury. These studies demonstrate suggest filtered power Doppler imaging can sensitively detect presence of sustained cavitation during burst wave lithotripsy exposures. [Work supported by NIDDK K01 DK104854 and P01 DK043881.]

THURSDAY AFTERNOON, 8 DECEMBER 2022 SUMMIT C, 1:00 P.M. TO 4:25 P.M.

## Session 4pCA

## Computational Acoustics and Biomedical Acoustics: Finite Difference Time Domain Methods Across Acoustics

Jennifer Cooper, Cochair
*Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, MD 20723*

Michelle E. Swearingen, Cochair
*Construction Engineering Research Laboratory, US Army ERDC, PO Box 9005, Champaign, IL 61826*

Subha Maruvada, Cochair
*U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993*

### *Invited Papers*

**1:00**

**4pCA1. Why use the finite difference time domain method for studying echolocation bats?** Yu Teshima (Doshisha Univ., Ishinkan IN505N, Tataramiyakodani 1-3, Kyotanabe 610-0394, Japan, yuteshima18@gmail.com), Takao Tsuchiya, and Shizuko Hiryu (Doshisha Univ., Kyotanabe, Japan)

To study bat echolocation, it is essential to investigate echoes containing environmental information that bats use to determine their behavior. However, even with the recent advanced technique, it is difficult to measure all echoes reaching the bats in flight. Therefore, we proposed an approach that combines behavioral measurements (flight path measurements using high-speed video cameras and emitted ultrasound measurements using a telemetry microphone attached to the bat's back) with the FDTD method to calculate echoes from the surroundings, resulting in new behavioral findings. Analysis of binaural echoes reaching the bats in flight suggested the space composed of echo information is different from the visually perceived space, and the Doppler shift compensation behavior of bats revealed the transition of attention during flight by extracting echoes that bats pay attention to. In addition, by introducing a 3D digital model of a bat's head into the simulation space, we were able to study the echo information required for bat source localization based on the head transfer function. The proposed method enables the acquisition of echoes, which has been a longstanding problem in bat echolocation research and is a promising new method that could lead to a better understanding of bat behavior decisions.

1:20

**4pCA2. Piezoelectric finite-difference time-domain simulations of piezoelectric signal generated in cancellous bone by ultrasound irradiation.** Atsushi Hosokawa (Dept. of Elec. and Comput. Eng., National Inst. of Technol., Akashi College, 679-3 Nishioka, Uozumi, Akashi 674-8501, Japan, hosokawa@akashi.ac.jp)

Bone formation can be driven by mechanical loads applied to the bone. By taking advantage of this mechanism, the accelerated healing of bone fracture using low-intensity pulsed ultrasound (LIPUS) has been medically practiced. Bone can behave as a piezoelectric material, and the piezoelectric effects are considered to accompany the bone formation. However, the piezoelectric properties in bone, particularly in cancellous bone with a porous structure, at ultrasound frequencies are too complex to easily clarify. In such a case, numerical simulations can be helpful because they enable visualization in the black box. Numerical simulations by an elastic finite-difference time-domain (FDTD) method have been widely performed to investigate ultrasound behaviors in bone. In this study, the elastic FDTD method with piezoelectric constitutive equations (PE-FDTD method) was used to simulate the piezoelectric signals generated in water-saturated cancellous bone by ultrasound irradiation. The cubic cancellous bone model was reconstructed from the x-ray microtomographic image. The piezoelectric signal waveforms when an ultrasound burst wave was irradiated in three orthogonal directions were calculated, together with the ultrasound signal waveforms propagated through cancellous bone. From the calculated results, the effect of the trabecular orientation was investigated.

1:40

**4pCA3. Viscoelastic transcranial wave propagation modeling for neuronavigated focused ultrasound procedures in humans.** Samuel Pichardo (Radiology, Univ. of Calgary, 1403 29th St. NW, Foothills Medical Ctr. - MRG 013, Calgary, AB T2N 2T9, Canada, samuel.pichardo@ucalgary.ca)

Transcranial focused ultrasound (FUS) has gained significant attention as a non-invasive therapeutic technique that can reach both cortical and subcortical targets in the human brain. However, the skull barrier distorts the ultrasound beam and introduces significant losses that need to be accounted for. In this talk, I will present our efforts to develop high precise transcranial ultrasound modeling based on the open-source viscoelastic wave propagation solver BabelViscoFDTD and its application in neuronavigated procedures for FUS-based neuromodulation. The viscoelastic wave equation is expressed in isotropic stress tensors and displacement vectors, and their nodes are placed in a staggered grid arrangement. The solver uses a Cartesian O(4)-space O(2)-time FDTD scheme. A narrow-band quality factor models attenuation. A reduction of staircase artifacts can be enabled using a superposition operator. BabelViscoFDTD is optimized for multiple computing GPU and CPU backends (CUDA, OpenCL, Metal, and X86-64) and includes GPU-accelerated solvers of the Rayleigh-Sommerfeld equation and Bio-Heat thermal equations. We adapted the viscoelastic and thermal solvers to neuronavigate two FUS devices: A 128-element concaved array operable at 250 kHz and 700 kHz and a 4-ring concaved array operable at 500 kHz. The high degree of optimization of the solvers allow that numerical procedures (neuronavigation and modeling) can be performed in an Apple M1 Max (64 GB RAM) laptop computer.

2:00

**4pCA4. Infrasound propagation with realistic terrain and atmospheres using a finite-difference time-domain method.** Jordan W. Bishop (Wilson Alaska Tech. Ctr., Geophysical Inst., Univ. of Alaska Fairbanks, 2156 Koyukuk Dr., P.O. Box 757320, Fairbanks, AK 99709, jwbishop2@alaska.edu), Philip S. Blom (Earth & Environ. Sci., Los Alamos National Lab., Los Alamos, NM), and David Fee (Wilson Alaska Tech. Ctr., Geophysical Inst., Univ. of Alaska Fairbanks, Fairbanks, AK)

Numerous infrasound observations and complementary numerical simulations have shown that infrasound propagation is strongly influenced by terrain within approximately 10 km from the source. Recent computational efforts using ray theory have shown that terrain influence extends over hundreds of km and is especially strong for waves ducted in the troposphere. Wind and temperature gradients also have a strong influence on propagation at these distances, which suggests that both terrain and atmospheric structure need to be accounted for in waveform modeling at a wide range of distances. Here we show preliminary results from numerical simulations of linear acoustic propagationthrough a moving, inhomogeneous atmosphere using a finite-difference time domain propagation code. We compare our synthetic waveforms in two and three dimensions with existing community infrasound propagation codes and discuss future developments, including open source licensing. Finally, we present preliminary comparisons between modeling and observations from a large-scale explosion at the Utah Testing and Training Range. This scenario highlights the importance of using full-wave modeling compared to ray methods to explain the observed waveforms. [Work funded by the Defense Threat Reduction Agency. Cleared for release.]

2:20

**4pCA5. An implementation of a finite difference time domain method for second-order nonlinear acoustic equations.** Kyle G. Dunn (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, kyle.g.dunn@erdc.dren.mil) and Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

The framework of the numerical scheme presented in this talk is that of the finite-difference time-domain (FDTD) method published by Sparrow and Raspet [J. Acoust. Soc. Am. 90, 1991]. The method is fourth-order in space and second-order in time when propagating in homogeneous media. Our implementation resides in a Cartesian coordinate system and is meant to be used in inhomogeneous media. In the neighborhood of inhomogeneities the FDTD method is reduced to second-order in space. Of interest to our applications is the simulation of infinite domains, which is achieved using the perfectly matched layer (PML). The particular PML implementation we consider is one that was designed to absorb linear acoustic waves neglecting thermoviscous effects. In this talk, we will give an overview of our FDTD implementation and show one- and two-dimensional examples to illustrate performance in heterogeneous media that include changes in material properties and scattering objects.

**2:40–2:55 Break**

## *Contributed Papers*

**2:55**

**4pCA6. Simple insertion loss calculations executed using a 13 point 3d FDTD scheme.** Arthur W. van der Harten (Acoust., Acoust. Distinctions/ Open Res. in Acoust. Sci. and Education, 400 Main St. Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com)

Simple applications of a 13 point finite difference time domain method have been developed to address difficult problems concerning the attenuation of particularly complicated noise attenuation problems. When in doubt concerning engineering equations for sound attenuation, a wave-based technique can be a practical tool for determining the attenuation due to obstructions and physical structures. Several instances ranging from mechanical applications, site noise problems, and architectural features are presented.

**3:10**

**4pCA7. Finite-difference diffusion-equation modeling of reverberation chambers in time-domain.** Jiahua Zhang (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Graduate Program in Architectural Acoust., Troy, NY 12180, zhangj45@rpi.edu), Juan Navarro (Polytechnic Sci. Dept., San Antonio's Catholic Univ., Murcia, Spain), Ning Xiang, and Mélanie Nolan (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Conventional chamber-based methods of random incidence absorbing coefficients overlook the non-diffused sound field in the room acoustics, which decreases accuracy and may even lead to conflicting results. This work applies the diffusion equation model to room-acoustic simulations of standardized reverberation chambers. The simulations can more efficiently capture the chamber's non-diffuse sound field and energy flows than wave-based simulation models. The diffusion equation represents the governing equation of reverberant sound energy densities within the reverberation chamber which is solved using the finite-difference time-domain method. Its computational efficiency of diffusion equation-based modeling lies in a highly sparse domain meshing condition dictated by the mean-free path length rather than wavelengths and still derives a wideband simulation result. This work also dedicates the effort to reexamine the meshing condition, particularly for standardized finite sizes of sound absorbers for measurements of random incidence absorption coefficients. By comparing the outcomes of simulations with measurement data, an *a posteriori* absorption coefficient is inversely estimated involving Bayesian parameter estimation.

**3:25**

**4pCA8. An open-source implementation of GPU-accelerated Finite-Difference Time Domain (FDTD) simulations in the Unity game engine to visualize elastic wave propagation and scattering for pedagogical or training scenarios.** Seth Golembeski (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, sgole1@unh.newhaven.edu) and Eric A. Dieckman (Mech. Eng., Univ. of New Haven, West Haven, CT)

Simulations of elastic wave propagation through solid materials can be realized through industry-standard software or custom implementations in a variety of programming languages. These codes are generally optimized for numerical accuracy and can be difficulty to interact with, slow to run, and often require external processing of results to create 3D visualizations. This is especially true for parallelized or Graphical Processing Unit (GPU)-based simulations which require additional libraries. These factors tend to limit the use of numerical simulations of wave propagation in pedagogical applications to simple 2D implementations of acoustic propagation through fluids. Here we present a fully-3D open-source Finite-Difference Time Domain (FDTD) simulation of elastic wave propagation that is implemented in the Unity game development engine. Due to the nature of the engine the simulation code runs entirely on the GPU without additional coding. Additionally the solutions are rendered as the simulation runs which removes the need for saving and post-processing large data files, as well as allowing the user to rotate the viewpoint in real-time to better visualize the wave propagation and scattering in complex geometries.

**3:40**

**4pCA9. FDTD simulation of induced potentials in bone by ultrasound irradiation.** Hidehisa Suzuyama (Doshisha Univ., Tataratudani1-3, Kyotanabe City, Kyoto Prefecture 610-0394, Japan, ctwg0348@mail4.doshisha.ac.jp), Taisei Tsubata, Keigo Maehara (Doshisha Univ., Kyotanabe, Kyoto, Japan), Atsushi Hosokawa (Dept. of Elec. and Comput. Eng., National Inst. of Technol., Akashi College, Akashi, Japan), Takao Tsuchiya, and Mami Matsukawa (Doshisha Univ., Kyotanabe, Kyoto, Japan)

Low Intensity Pulsed Ultrasound (LIPUS) can shorten the healing time of fractures by irradiating low-power ultrasound. However, the full mechanism of bone fracture healing is still unknown. One possible mechanism is the ultrasonically induced electrical potential due to the weak piezoelectricity of bone. There are several studies reporting the acceleration of fracture healing due to electrical stimulation. In this study, the piezoelectric finite-difference time-domain (PE-FDTD) method, which is an elastic FDTD method with piezoelectric constitutive equations in the stress-charge form, was used to consider the induced electrical potentials. Simulations were performed with a heterogeneous digital model of the human radius created from the 3 D CT data. Assuming a fracture at the distal part of the radius, ultrasound in the MHz range entered bone perpendicular to the bone axis. The guided longitudinal waves first propagated along the pseudo-piped shape bone, followed by the weak shear and surface waves. However, the intensity of the electric field in the bone increased owing to the shear wave. Shear wave seems a key factor to understand the ultrasonically induced electrical potentials in bone.

**3:55**

**4pCA10. Finite difference time domain ray-based modelling of acoustic scattering for target identification and tracking.** Grant Eastland (Test and Evaluation, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

The Finite Difference Time Domain (FDTD) method has provided a powerful technique for modelling and simulation of solutions of a variety of acoustics problems. The purpose of this investigation is to present work on the development of time-domain models of acoustic scattering from targets near a flat pressure-release boundary for use in identification and tracking from a moving receiving platform. The timing from acoustic source to reception are dependent on the location dependent sound speed profile and the specular scattering points on the target. There can be multiple specular points revealed on the target because of interactions with the flat boundary; each of which are modelled utilizing the FDTD method providing path corrections due to variation in propagation brought on by the sound speed profile of the environment.

**4:10**

**4pCA11. Acoustic scattering from dynamic rough ocean surfaces using finite-difference time-domain modeling techniques.** Alex Higgins (Elec. & Comput. Eng., Portland State Univ., 1900 SW 14th Ave., Ste. 25-01, Portland, OR 97201, higginja@ece.pdx.edu) and Martin Siderius (Portland State Univ., Portland, OR)

Established models for underwater acoustic propagation and scattering typically assume the sea-surface to be either perfectly smooth or rough but static in time. When the sea-surface is rough and moving, received signals show anomalies such as additional transmission losses (due to scattering) and Doppler effects (due to surface motion). Understanding the mechanisms behind these anomalies leads to better sonar system designs without having to perform expensive at-sea experiments. The Finite-Difference Time-Domain (FDTD) method is ideal to predict these physical phenomena. This work presents a FDTD method implemented to model the impact of both

sea-surface roughness and motion on underwater acoustic propagation. The FDTD method allows an arbitrary function to define the rough sea-surface and its time evolution. The surface characteristics are modeled using a Pierson-Moskowitz (PM) frequency spectrum. The PM surface model is simple to implement and fully defined by wind speed and direction. Time-domain results from FDTD simulations of static rough sea-surfaces are compared to an established Helmholtz integral equation (HIE) method to establish the validity of the approach. Broadband signals are used as the source waveform. Results demonstrate the anomalous transmission loss and Doppler in received signals. [Work supported by the Office of Naval Research.]

THURSDAY AFTERNOON, 8 DECEMBER 2022 SUMMIT A, 1:00 P.M. TO 4:25 P.M.

## Session 4pED

## Education in Acoustics: Connecting Industry and Education (Hybrid Session)

Jim DeGrandis, Cochair
*Acoustics First Corporation, 2247 Tomlyn Street, Richmond, VA 23230*

Daniel A. Russell, Cochair
*Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802*

### *Invited Papers*

**1:00**

**4pED1. Fostering relationships between industry and academia is good business.** Jim DeGrandis (Acoust. First Corp., 2247 Tomlyn St., Richmond, VA 23230, Jim@acousticsfirst.com)

From the perspective of an industry manufacturer, it is sometimes difficult to promote activities such as academic outreach and research. While business-to-business investments are commonplace and provide the most direct investment-to-return ratio calculation, some companies find that allocating resources for academia provides long-term benefits and opportunities. The symbiotic relationship between business and education helps manifest research resources, knowledge exchange, mentoring, and future opportunities for the students, staff, facilities, and industry as a whole. Here are some cases where investing in the intangible relationships with academia proved to be good business.

**1:20**

**4pED2. Impact of industrial partnerships in the acoustics program at Kettering University.** Timothy A. Stiles (Natural Sci., Kettering Univ., 1700 University Ave., Flint, MI 48504, tstiles@kettering.edu), Ronald E. Kumon, and Daniel Ludwigsen (Natural Sci., Kettering Univ., Flint, MI)

Kettering University has an experiential learning model in which students typically alternate between academic and cooperative work terms every quarter during their entire undergraduate degree program. In this approach, industrial partners play a particularly critical role. In this talk, we will discuss how partnerships with industry have impacted our classroom experiences and inspired curricular elements that incorporate theoretical, computational, and experimental approaches to solve problems in acoustics. We will highlight some of the work related to acoustics that students have done in their co-op projects and how students have served as a bridge between the academic and corporate environments. The course "Acoustics in the Human Environment" has recently benefited from a grant from the Head Acoustics Foundation. The course was revamped in Spring 2022 to include more detailed analysis and a jury study that make use of this donation. The course "Acoustic Testing and Modeling" uses Siemens Simcenter Testlab to acquire experimental data and Comsol for computer simulations. Recently, this course has added a module on Laser Doppler Vibrometry that makes use of equipment from Polytec. The efforts in these courses have enabled students to be better prepared for acoustic testing projects for a thesis or co-operative employment terms.

**1:40**

**4pED3. Education to practice, the stories of two early-career acousticians.** Zachery O. L'Italien (McKay Conant Hoover, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, zlitalien@mchinc.com) and Henry Ashburn (McKay Conant Hoover, Inc., Westlake Village, CA)

This presentation will highlight several key educational and extracurricular endeavors that have enriched the careers of two recent Michigan Technological University graduates. Examples include experiences at Michigan Technological University, such as diffuser modelling and construction projects with professionals in acoustics, broad course material, and excellent faculty mentorship. Additionally discussed topics include internship opportunities, professional development, and firsthand technical experience in performing arts and studio environments.

**2:00**

**4pED4. Internships in the acoustical disciplines: How can we attract a more diverse student population?** Felicia M. Doggett (Metropolitan Acoust., LLC, 1628 JFK Blvd, Ste. 1902, Philadelphia, PA 19103, f.doggett@metro-acoustics.com)

A problem that many professional scientific organizations are tackling now, including the Acoustical Society of America (ASA) and the Institute of Noise Control Engineering (INCE), is the lack of diversity in our current membership. As the chair of INCE's Diversity, Equity, and Inclusion committee, we are exploring ways to engage with a more diverse student population for both internships as well as fulltime employees after graduation. In 2021, the National Science Foundation (NSF) published *Women, Minorities, and Persons with Disabilities in Science and Engineering*, which provides statistical information about the participation of these three groups in science and engineering education and employment. In order to have a plan as to how to proceed engaging with students, this document provides a good benchmark as to the current enrollment in the United States. This presentation explores methods that INCE's DEI committee is implementing to reach out to students and connect them with companies and industries in acoustics and noise control engineering who can make internships possible.

**2:20**

**4pED5. Creating effective assessment to prepare students for the workforce.** Kimberly A. Riegel (Phys., Farmingdale State College, 652 Timpson St., Pelham, NY 10803, kriegel@qcc.cuny.edu)

Effective assessment is one of the most difficult aspects of instruction. Assessment not only needs to accurately capture the student knowledge and abilities for the instructor but should prepare students for the kind of evaluation that they will face once they enter the field. These connections can help the student better connect with the material and understand the relevance of the assessment. Many common forms of assessment (e.g. tests) require the student to perfume timed and memorized tasks with few available resources for reference. As acoustics students head out into the professional world, the assessment that they encounter during their academic careers should reflect the kind of skills that they will be evaluated on in the workforce. Several examples of assessment that more closely align with industry will be presented. This presentation will include a plea to the industrial partners to help faculty effectively create more of these types of assessments.

**2:40–2:55 Break**

## *Contributed Papers*

**2:55**

**4pED6. RAMP summer bridge: Room acoustics: Connecting research, academics and mentoring.** Grace Remillard, Aidan Keefe, Nguyen Le (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA), Charles Thompson (Elec. and Comput. eng, UMASS Lowell, Lowell, MA), and Kavitha Chandra (Elec. and Comput. Eng., Univ. of Massachusetts Lowell, Lowell, MA 01854, kavitha_chandra@uml.edu)

The Research, Academics and Mentoring Pathways (RAMP) to Success summer bridge program at the University of Massachusetts Lowell prepares new engineering students from all majors for an early transition from high school to college. Over the last five years, RAMP has engaged 107 students, the majority being women and underrepresented minorities, in a six-week research, academics and mentoring experience. During RAMP, students are enrolled in Calculus and Introduction to Engineering courses, earning six credits towards their degree pathway. Groups of 5–6 students take on research projects with engineering faculty while also meeting weekly with professionals from industry. The structuring of a room acoustics research project is presented. A project-based learning model helps build skills in collaborative problem solving while also supporting each student to find authentic interest in the research. The project integrates experiments on measuring room impulse responses and spatial variation of source pressure amplitude in rooms of different sizes with a model-driven analysis of the data collected. Concepts of reverberation time, backward integration and room constants reinforce calculus concepts being studied. Coding experiments on rendering speech and music through measured impulse responses and filters created interest across a range of acoustics topics from concert halls to virtual audio.

**3:10**

**4pED7. Experiments on the nonlinear interaction of crossed turbulent streaming jets: A case study in mentorship.** Jenna M. Cartron (Phys., U.S. Naval Acad., Laurel, MD) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu)

During (2017–2018) co-author JMC began a mentorship at USNA as a HS senior. Projects led to developing an apparatus to study the interaction between crossed turbulent streaming jets in water. The ISNA 21 (Santa Fe) was an opportunity to present this research – which included a live demonstration. See [Proc. Mtgs. Acoust. **34**, 045022 (2018)]. Majoring in material science at JHU, JMC worked as a college intern at USNA (during 4 summers) to further develop the turbulent streaming jets experiment which now included a LaVision 2-D PIV (particle imaging velocimeter). Homemade MOSFET power amplifiers, Langevin 28 kHz ultrasonic transducers and aluminum stepped horns, were replaced by kits developed by Sonics & Materials. The kits including 40-kHz oscillators, 20 Watt amplifiers, impedance matching to the 40 kHz transducers CV245 and titanium 1/2, 1/4 and 1/8 inch diameter stepped horns Part # 630-0594 – to generate stronger downward East and West crossed turbulent jets (at a 60 deg angle) in an open 10 cc acrylic tank. As a by-product, cavitation bubbles allow flow visualization. Lighthill's nonlinear theory of acoustic streaming (1978) for a single source will be discussed including "Stuart streaming," Reichardt and

Schlicting's (turbulent jets) and Squire's theory on nonlinear laminar jet flow.

**3:25**

**4pED8. Penn State ASA student chapter educational workshops in engineering design tools for loudspeaker and amplifier system design.** Peter J. Riccardi (Graduate Program in Acoust., The Penn State Univ., University Park, PA), Zane T. Rusk (Dept. of Architectural Eng., The Penn State Univ., University Park, PA), John A. Case (Graduate Program in Acoust., The Penn State Univ., University Park, PA), Heui Young Park (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, hkp5188@psu.edu), and Eric Rokni (Graduate Program in Acoust., The Penn State Univ., State College, PA)

Hands-on workshops and projects are an effective way to provide meaningful and memorable STEM education. To promote practical acoustics and engineering knowledge, the Pennsylvania State University Acoustical Society of America student chapter hosted a series of workshops wherein students learned the basics of electroacoustic transduction, mechanical modeling, electronics design, and PCB routing. This educational effort was accomplished by sponsoring a three-way loudspeaker design project. Transduction topics were covered by optimizing the model of a ported woofer in MathWorks SimScape, using the recently developed *Acoustical Domain for Simscape* available from MATLAB central. Mechanical modeling of the three-way speaker cabinets to produce manufacturing drawings was covered in SolidWorks workshops. Finally, the basics of circuit design, schematic capture, and PCB routing were covered in DipTrace through the design of a stereo integrated amplifier with an active crossover that was built to power the loudspeaker. Details about the workshops and performance of the designed components will be presented.

**3:40**

**4pED9. Open-source acoustics simulation in the education environment.** Arthur W. van der Harten (Acoust., Acoust. Distinctions/Open Res. in Acoust. Sci. and Education, 400 Main St. Ste. 600, Stamford, CT 06901, Arthur.vanderharten@gmail.com)

Pachyderm Acoustic Simulation is a professional tool for geometrical and numerical simulation of acoustics in the built environment that happens to be open source under the General Public License. Over the course of the 14 year life of the open-source acoustics simulation project, Pachyderm Acoustic has had the privilege of partnering with numerous academic institutions worldwide to educate architecture and engineering students in acoustics. This has been in the form of support for student projects, lectures and workshops on acoustics simulation and other topics, and, in some cases, PHD advice. This session will explore several notable examples of successful pedagogy using the open source software as a tool for higher education. There will also be some discussion of approaches to pedagogy that have not been as successful, in an effort to further discussion regarding an appropriate approach to higher learning in acoustics.

**3:55–4:25**
**Panel Discussion**

## Session 4pNS

# Noise, Computational Acoustics, Structural Acoustics and Vibration, Signal Processing in Acoustics, and Physical Acoustics: Jet and Launch Vehicle Noise II (Hybrid Session)

Alan T. Wall, Cochair
*Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, OH 45433*

Kent L. Gee, Cochair
*Department of Physics and Astronomy, Brigham Young University, N281 ESC, Provo, UT 84602*

Caroline P. Lubert, Cochair
*Mathematics and Statistics, James Madison University, 301 Dixie Avenue, Harrisonburg, VA 22801*

### *Invited Paper*

**1:00**

**4pNS1. Microphone location investigation for standard aircraft ground run-up noise measurements.** Alan T. Wall (711 Human Performance Wing, Air Force Research Lab., Bldg. 441, Wright-Patterson AFB, OH 45433, alantwall@gmail.com), Steven C. Campbell (Ball Aerosp., Beavercreek, OH), and Frank S. Mobley (711 Human Performance Wing, Air Force Reseach Lab., Dayton, OH)

The correct placement (height, orientation, and mounting techniques) for microphones used in standard data collection practices for aircraft environmental noise models is a topic of debate. Previous studies have demonstrated and quantified the dependence of spectral data recorded as a function of microphone placement variations and relative source origin. In this work, an in-depth investigation of microphone mounting effects is performed with an emphasis on (1) microphone height and (2) ground-mounting configuration for the purpose of quantifying ground run-up aircraft noise emissions for environmental noise model generation. Data for this investigation were collected on the T-7A Red Hawk in both standard (see ANSI S12.75-2012) and off-standard configurations for comparison.

### *Contributed Papers*

**1:20**

**4pNS2. Assessing the impact of ground reflection on measurement of the overall sound power level and acoustic radiation efficiency of rocket launches.** Grant W. Hart (Dept. of Phys. and Astronomy, Brigham Young Univ., N357 ESC, Provo, UT 84602, grant_hart@byu.edu) and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Determining the sound power levels of a rocket is the first step in characterizing its radiation efficiency. Because the sound radiation from a launch vehicle is anisotropic, well-calibrated ground measurement stations are used, along with trajectory data, to obtain sound power. Historically, the effect of ground reflections appears to largely have been neglected in the literature, despite the potential to inflate OAPWLs and therefore radiation efficiency. This study investigates the likely effect of a finite-impedance ground on spectra, overall sound pressure levels, and power levels. The single-parameter ground reflection model by Embleton *et al*. [J. Acoust. Soc. Am. 74, 1239–1244 (1983)] is used to obtain an estimate for change in OASPL for a model spectrum based on space vehicle launches. When the correction is applied to the OAPWL it produces in a nearly 3dB reduction in level and reducing the efficiency by a factor of 2. This indicates the possibility that the 0.5% radiation power efficiency assumed for rockets in the past is too high.

**1:35**

**4pNS3. Implementing a heuristic method to correct ground reflection effects observed in full-scale tactical aircraft noise measurements.** Matthew A. Christian (Dept. of Mech. Eng., Brigham Young Univ., ESC N284, Provo, UT 84602, matthew.christian@byu.edu), Kent L. Gee, Jacob B. Streeter (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and Steven C. Campbell (Ball Aerosp., Dayton, OH)

In a recent study of noise from a T-7A-installed GE F404 engine, microphones along 38 and 76 m (125 and 250 ft) arcs were mounted 1.5 m (5 ft) above the ground to quantify human impact. While helpful for this purpose, the resulting multipath effects pose challenges for other acoustical analyses. For jet noise runup measurements, these effects are complicated by the fact that the noise source is extended and partially correlated and its spatial properties are frequency dependent. Furthermore, a finite-impedance ground surface and atmospheric turbulence affect interference nulls. This study applies a ground-reflection method developed previously [Gee *et al*., Proc. Mtgs. Acoust. 22, 040001 (2014)] for rocket noise measurements. The model accounts for finite ground impedance, atmospheric turbulence, and extended source models that are treated as coherent and incoherent arrays of monopoles. Application to the ground runup data to correct the 38 and 76 m spectra

at a range of angles suggests the incoherent line source model is more appropriate at sideline angles whereas the coherent source model is more appropriate for upstream and downstream propagation. Comparisons with near-field data and similarity spectra show that, while imperfect, this method represents an advancement in correcting jet noise spectra for ground reflection effects. [Work supported by ONR Grant No. N00014-21-1-2069.]

**1:50**

**4pNS4. Assessing impact of near-ground meteorology on spectral variability in static jet aircraft noise measurements.** Jacob B. Streeter (Brigham Young Univ., Provo, UT, jacobbstreeter@gmail.com), Matthew A. Christian (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and Steven C. Campbell (Ball Aerosp., Beavercreek, OH)

ANSI/ASA standard S12.75 (2012) provides guidance on allowable meteorological conditions for acoustical measurements of installed high-performance jet engines. This paper investigates meteorological effects on acoustic data acquisition by analyzing recent measurements of a T-7A-installed GE F404 engine. During this measurement, the aircraft was run up six times at engine powers from idle to full afterburner, with test conditions following those prescribed by S12.75. However, far-field spectra show variability between runs, despite relatively uniform test conditions. Measurements of the vertical temperature gradient show a correlation between the gradient and spectral characteristics. This analysis suggests that local temperature profiles must be considered more carefully in future full-scale measurements. [Work supported by ONR Grant No. N00014-21-1-2069.]

**2:05**

**4pNS5. Source decomposition and reduced-order modeling of an installed F404-GE-103 engine using the optimized-location virtual reference method.** Logan T. Mathews (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmathew3@byu.edu), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Principal component analysis-related methods (i.e., proper orthogonal decomposition or partial field decomposition) are used widely in aeroacoustic applications for source characterization and reduced-order modeling. A previous paper [A. T. Wall *et al*., J. Acoust. Soc. Am. 144, 1356 (2018)] developed a coherence-based, virtual-reference method for performing physically meaningful PFD known as the optimized-location virtual references (OLVR) method. This paper performs a source decomposition using the OLVR method on acoustical holography data from an installed GE F404 engine. The OLVR-derived partial fields are compared with far-field radiation characteristics and partial equivalent sources are fitted to analytical wavepackets. Characteristics of this reduced-order model are discussed as a function of frequency and engine condition. [Work supported ONR Grant No. N00014-21-1-2069.]

**2:20**

**4pNS6. Joint time-frequency domain analysis of F404 engine noise sources using event-based beamforming methods.** Tyce Olaveson (Dept. of Phys. and Astronomy, Brigham Young Univ., N284 Eyring Science Ctr. BYU, Provo, UT 84602, tyceolaveson@gmail.com), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and Jon P. Johnson (Dept. of Phys., Brigham Young Univ. - Idaho, Rexburg, ID)

Spatiospectral lobes are features identified in the noise fields surrounding full-scale tactical aircraft that are unseen in most lab-scale experiments. Prior studies have explored their characteristics in the frequency domain, but a joint time-frequency domain (JTFD) analysis has potential to further explore these phenomena and connect them to source-related events. This paper applies the event-based beamforming technique developed by Vaughn *et al*. [AIAA J. 2021] to acoustical data collected at a 120-microphone array near a T-7A-installed F404 engine. The algorithm correlates time-domain events between pairs of adjacent microphones to find an event propagation direction and then ray traces to the jet nozzle lipline to identify an apparent source location. In addition to the large-derivative events used previously to identify crackle-related phenomena, this paper uses a JTFD wavelet analysis to expand the possible triggers to extract physical insights into the lobes. The lobe directivities are explored and compared to frequency-domain studies. [Work supported by Grant No. N00014-21-1-2069.]

**2:35–2:50 Break**

**2:50**

**4pNS7. Near-field coherence analysis of noise from an installed F404 engine.** Kristi A. Epps (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, kristi.epps5@gmail.com), Tyce Olaveson (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH), and S. Hales Swift (Sandia National Labs., Albuquerque, NM)

Because phenomena observed in noise radiated from high-performance military aircraft are not fully understood, different analysis techniques can yield complementary insights into their characteristics. Here, a coherence analysis from a 120-microphone near-field array is used to identify and characterize the noise radiation and source-related properties of a T-7A-installed F-404-GE-103 engine over a broad range of frequencies. Some of the noteworthy findings include: a) coherence trends for mixing noise and spatiospectral lobes generally match those observed for previously studied aircraft; and b) bands of coherence between upstream and downstream locations are observed at higher engine powers, suggesting a coherent interaction between upstream broadband shock-associated noise radiation and downstream Mach wave radiation. [Work supported by ONR Grant No. N00014-21-1-2069.]

**3:05**

**4pNS8. Analysis of spark-generated waveforms coalescing in air using schlieren imaging.** William A. Willis (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029, william.willis@utexas.edu), John A. Valdez, Charles E. Tinney, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Mach waves generated by turbulent structures in supersonic jet flows have the potential to intersect and coalesce, a process proposed as a significant contributor to acoustic waveform steepening in the near field of a jet and to the noise referred to as crackle [Baars *et al*., AIAA (2013); Fiévet *et al*., AIAA (2016)]. Numerical simulations of intersecting waveforms have demonstrated that coalescence can lead to increased steepening that is dependent on intersection angle, waveform duration, and geometrical spreading [Willis *et al*., AIAA (2022)]. In this study, two simplified experiments involving a spark source in air are examined. The first involves grazing incidence on a rigid plane to model symmetric intersection between waveforms. The second involves intersecting waveforms emanating from side-by-side openings in a 3D-printed enclosure. Measurements of nonlinear evolution were made for the intersecting waveforms produced with each experimental setup and compared with measurements made for one waveform alone. Schlieren images of the coalescing waves allow for comparison of waveform steepening between the two cases without the limitations imposed by microphones. The measurements permit assessment of the extent to which coalescence enhances waveform steepening. [WAW is supported by the ARL:UT Chester M. McKinney Graduate Fellowship in Acoustics.]

3:20

**4pNS9. Aeroacoustic and aerodynamic interaction effects between electric vertical takeoff and landing rotors.** Gustavo Resende Coelho (Mech. and Aerosp. Eng., Univ. of Florida, 939 Ctr. Dr., Gainesville, FL 32611, gresendecoelho@ufl.edu), James D Goldschmidt, Henry J. Tingle, Peter G. Ifju, Ukeiley Lawrence (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL), Ben Goldman, Maicon Secchi (Archer Aviation Inc., San Jose, CA), and Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

Electric vertical take-off and landing (eVTOL) aircraft are characterized by their unconventional wing and electric rotor configurations, which involve both side-by-side and tandem rotor configurations. These configurations create unique aerodynamic and acoustic flow-fields. We numerically investigate the interaction effects between rotor pairs as well as their individual and combined acoustic radiation. We examine horizontal and vertical spacing, rotor tilt angles, and forward flight effects. Performance is characterized by thrust coefficient, blade passage frequency (BPF) sound pressure level (SPL), and overall sound pressure level (OASPL). This study is performed with a mid-fidelity aerodynamic solver, Dust, which is used to predict the aerodynamic flow-field. The tonal acoustic pressure at observer positions is predicted via the Farassat F-1A solution of the Ffowcs Williams and Hawkings equation utilizing the aerodynamic flow-field. The configurations studied show strong aerodynamic interaction effects in thrust, as well as out-of-plane acoustic radiation from the aft rotor. Base predictions of thrust and noise are validated via experimental measurement. As rotor separation decreases, we observe that aft rotor thrust decreases and BPF SPL increases. The most forward rotor, however, is marginally impacted by the interactions. [This research is supported by Archer Aviation Inc.]

THURSDAY AFTERNOON, 8 DECEMBER 2022 — SUMMIT E, 1:00 P.M. TO 3:00 P.M.

## Session 4pPAa

## Physical Acoustics: Physical Acoustics Best Student Paper Competition

The ASA Technical Committee on Physical 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 Physical Acoustics. Additionally, each student will give an oral presentation in a regular/special session.

Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions 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.

**1aPA6. Interactions between epithelial cells after their exposure to ultrasounds**
Student presenter: Iciar Gonzalez

**1aPA7. Using motile cells to characterize surface acoustic wave-based acoustofluidic devices**
Student presenter: Advaith Narayan

**1aPA8. Cell-like microparticles with tunable acoustic properties for calibrating devices**
Student presenter: Clara Elizabeth Tandar

**1pPA2. Particle manipulation with acoustic waves based on Hertz-Mindlin mechanics**
Student presenter: Marina Terzi

**1pPA3. Noncontact rotation of a small object using ultrasound standing wave and traveling wave**
Student presenter: Eimei Yamamoto

**2aPAb3. Generalized inverse problems in resonant ultrasound spectroscopy**
Student presenter: Juraj Olejnak

**2pPA7. Resonant ultrasound spectroscopy studies of high-entropy fluorites**
Student presenter: Rubayet Tanveer

**3aPAa6. Automated detection of dust-devil-induced pressure signatures**
Student presenter: Louis Urtecho

**3pPAb3. Methods for accurate acoustic characterization with ultra-low noise and minimal effect from reflection wave**
Student presenter: Junpeng Lai

**4aPAa5. Assessing the microarchitecture and microenvironment of cancer in vivo by photoacoustic imaging techniques**
Student presenter: Xueding Wang

**4aPAb2. A ray tracing approach to focusing ultrasonic beams in isotropic and anisotropic solids**
Student presenter: Lauren Katch

**4aPAb3. Noninvasive tracking of solidification using ultrasound**
Student presenter: Caeden Smith

**4aPAb5. Vibrational characteristics of a helical antenna for L-band satellite communications**
Student presenter: Nathan Hill

**4aPAb8. Thin ultrasound non-contact airborne sensor using flexural vibration**
Student presenter: Natsumi Nakaoka

**4pPAb4. Measurement of airborne ultrasound using laser Doppler vibrometry**
Student presenter: Zihuan Liu

**4pPAb5. Low frequency photoacoustic optical absorption measurement of aerosols**
Student presenter: Arthur Case

**4pPAb6. Evaluation of acousto-optic effect in liquid using high-frequency and high-intensity ultrasound**
Student presenter: Yuki Harada

**5aPA2. Effect of bone degradation on axially transmitted low frequency (<500 kHz) ultrasonic guided waves**
Student presenter: Aubin Antoine Chaboty

**5aPA3. Assessment of flaws in cold sintered ZnO via longitudinal ultrasonic wave speed and attenuation measurements**
Student presenter: Haley Nicole Jones

**5aPA5. Ultrasonic characterization of the microstructure for cold sintered sodium molybdate**
Student presenter: Christopher Wheatley

**5aPA7. Ultrasonic damage detection in Lithium-ion batteries subjected to localized heating**
Student presenter: Tyler McGee

**5aPA8. A model-driven approach to ultrasonic detection of state of charge in Lithium-ion batteries**
Student presenter: Tyler McGee

**5aPA9. Investigating the influence of smoothing approximations on analytical models of elastic waves in strongly scattering polycrystals**
Student presenter: Anubhav Roy

**5pPA2. Using random matrix theory to quantify pulmonary fibrosis: investigating the effect of time window duration**
Student presenter: Azadeh Dashti Cole

**5pPA3. Mechanical and acoustical characterization of elastic properties for additively manufactured polymers**
Student presenter: Celeste A. Brown

**5pPA5. The Doppler ultrasound twinkling artifact on crevices etched in silicon wafers**
Student presenter: Eusila C Kitur

**5pPA7. Investigation of the planar reflector substitution method using the finite element method**
Student presenter: George West

**5pPA9. Time-varying elastic wave mode conversion in vibrating elastic beams with subwavelength nonlinearity**
Student presenter: Samuel D Parker

**5pPA10. A technique for measurement of ultrasonic waves propagating in time-varying media**
Student presenter: Samuel D Parker

## Session 4pPAb

# Physical Acoustics and Biomedical Acoustics: Interaction of Light and Sound II (Hybrid Session)

E. Carr Everbach, Cochair
*Engineering, Swarthmore College, 500 College Avenue, Swarthmore, PA 19081*

Jason L. Raymond, Cochair
*Dept. of Engineering Science, University of Oxford, 17 Parks Road, Oxford, OX1 3PJ, United Kingdom*

### *Invited Papers*

**1:30**

**4pPAb1. Brillouin light scattering spectroscopy as a versatile probe of hypersound in diverse materials systems.** Dillon F. Hanlon, Bradley D. McNiven, Stephen J. Spencer (Phys. and Physical Oceanogr., Memorial Univ., St. John's, NF, Canada), and G. T. Andrews (Dept. of Phys. and Physical Oceanogr., Memorial University, St. John's, NF A1B 3X7, Canada, tandrews@mun.ca)

An overview of the inelastic laser light scattering technique of Brillouin spectroscopy and its application to the study of hypersound in a diverse set of materials systems will be presented. In particular, results obtained from recent Brillouin scattering experiments on natural gastropod mucus, layered high temperature superconductor $Bi_2Sr_2CaCu_2O_{8+\delta}$, biotite micas, and satellite tobacco mosaic virus crystals will be highlighted. Collectively, these results demonstrate the utility of Brillouin light scattering spectroscopy as a sensitive non-destructive probe of hypersound velocity and attenuation in the vicinity of structural phase transitions, the influence of chemical composition and incommensurate structure on material elasticity, and of phonon dynamics in challenging materials classes.

**1:50**

**4pPAb2. Three-dimensional characterization of spatial sound fields via acousto-optic sensing.** Samuel A. Verburg (Acoust. Technol., Dept. of Elec. and Photonics Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, saveri@dtu.dk), Earl G. Williams (U.S. Naval Res. Lab., Code 7106, Washington, DC), and Efren Fernandez-Grande (Acoust. Technol., Dept. of Elec. and Photonics Eng., Tech. Univ. of Denmark, Kgs Lyngby, Hovedstaden, Denmark)

Characterizing audible sound fields over space is at the core of many applications in acoustics and audio technology, including active sound field control, spatial audio, and the experimental analysis of sound radiation. For this purpose, arrays of microphones are commonly deployed across the sound field, with an inter-microphone spacing that depends on the highest frequency studied. Capturing sound fields at mid and high frequencies is nonetheless challenging, as the number of transducers required becomes impractically large and the scattering of sound introduced by the array is often significant. Acousto-optic sensing enables the remote and non-invasive characterization of acoustic fields, and thus it represents an attractive alternative to conventional electro-mechanical transducers in several applications. The phase-shift that laser beams experience as they travel through a pressure field are measured via optical interferometry. The optical measurements are then used to reconstruct the field that originated such phase variations. In this study, we project the measured data into a set of functions suitable to represent sound fields over space. The projection makes it possible to improve the reconstruction accuracy and extrapolate the reconstruction outside the measured region. We demonstrate the method by reconstructing the three-dimensional sound field inside a room using optical data.

**2:10**

**4pPAb3. Distributed acoustic sensing with optical fibres: Fundamentals and applications.** Ali Masoudi (Univ. of Southampton, Southampton, United Kingdom) and Gilberto Brambilla (Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, gb2@orc.soton.ac.uk)

Because of their intrinsic geometrical features and extraordinary transparency, optical fibers allow for a continuous and dynamic monitoring of strain along all their length. In state of the art systems, 100 000 points can be resolved over a range of the order of 50 km, providing an extremely cost effective solution per sensing unit. In its basic implementation, a light pulse is injected into the sensing optical fibre and the Rayleigh backscattered light is continuously monitored, in a process similar to a radar system. While the phase relation between different point along the fiber is used to reconstruct the strain (thus the acoustic information), the time of flight (e.g., delay between injected pulse and received backscattered light) provides information about the physical location along the fiber. In the last decade, research on novel detection schemes and fibers has allowed to extend the detection range to over 170km, to decrease the spatial resolution below 10 cm and significantly increase the signal-to-noise-ratio. Because of this, distributed acoustic optical fiber sensing (DAS) has found applications in the oil and gas industry for oil well monitoring and geoseismics and more widely for the monitoring of border and perimeter security, resulting in market of the order of $1 billion. More recent applications include monitoring of railways and road traffic, structural health, earthquakes and other geophysical events.

**2:30**

**4pPAb4. Measurement of airborne ultrasound using laser Doppler vibrometry.** Zihuan Liu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, 2501 Speedway, EER 4.822 Hall Lab, Austin, TX 78712, zihuanliu@utexas.edu), Ehsan Vatankhah, Yuqi Meng, Xiaoyu Niu, and Neal A. Hall (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

The speed of light in air is dependent on the air's instantaneous density. Since air density is modulated by sound, sound in the air can be observed and measured using optical methods. One such optical method is Laser Doppler Vibrometry (LDV). Most commonly, LDVs measure the mechanical velocity of a surface. However, by placing a rigid reflector beneath a sound beam in air, it is possible to measure the time rate-of-change of optical refractive index and thus to measure dynamic changes in air density, or sound. In prior demonstrations, this method has been used to visualize sound fields in the audible frequency range and ultrasonic range underwater. Here, we present the first measurements of high-intensity airborne ultrasound beams in the frequency range spanning 100 kHz–300 kHz. We observe accumulated distortion, wave steepening, and weak shock formation as high intensity sound beams propagate. LDV measurements are compared against numerical simulations of the sound field. Advantages of the LDV technique are discussed, and we also attempt to quantify limitations of the technique which include spatial averaging of the measurand along and normal to the optical beam path.

**2:45–3:00 Break**

**3:00**

**4pPAb5. Low frequency photoacoustic optical absorption measurement of aerosols.** John A. Case (Appl. Res. Lab., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, jac7175@psu.edu) and Robert W. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA)

Atmospheric optical absorption, including the contribution from aerosols, is important in modeling optical propagation and for climate research. An open-celled photoacoustic aerosol sensing system was developed to operate at 150 Hz for measurement of absorption aerosols up to 3 microns in diameter with the design goal to achieve a sensitivity adequate to measure an absorption corresponding to a 1/e optical absorption length of 1 Mm. The advantage of a photoacoustic system is that it measures optical absorption only and is not sensitive to optical extinction due to scattering at low absorption levels. The system was calibrated against an existing black carbon absorption measurement unit made by Droplet Measurement Technologies, which operates at a shorter optical wavelength and higher acoustic frequency. With previous work focusing on measuring optical absorption by black carbon aerosols [J.A. Case and R.W. Smith,182nd meeting of the Acoustical Society of America in Denver, CO (5aPA)], this work covers measuring absorption by wet salt solutions of similar salt concentration to that oceanic aerosols. Experiments with the current system showed an anomalous photoacoustic signal appearing in the absence of any aerosols. Several solutions to eliminating the unwanted signal from the desired measurement are proposed and tested.

**3:15**

**4pPAb6. Evaluation of acousto-optic effect in liquid using high-frequency and high-intensity ultrasound.** Yuki Harada (Faculty of Sci. and Eng., Doshisha, 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, cyjh1301@mail4.doshisha.ac.jp), Mutsuo Ishikawa (Faculty of Biomedical Eng., Toin Univ. of Yokohama, Yokohama, Kanagawa, Japan), Shuto Inamoto, Hidehisa Suzuyama, Mami Matsukawa, and Daisuke Koyama (Faculty of Sci. and Eng., Doshisha, Kyotanabe, Kyoto, Japan)

Piezoelectric thin film transducers that can receive and transmit high-frequency and high-intensity ultrasound are powerful tools for future medical and industrial applications. Our group has developed acousto-optic devices where the refractive index can be controlled at high speed and high precision using ultrasound, but there are few reports on the interaction between high-frequency (tens to hundreds MHz) and high-intensity (MPa) ultrasound and visible light in liquids. In this study, we evaluated the acousto-optic interaction using polarized CW laser (532 nm) and a piezoelectric $KNbO_3$ thin film transducer formed by hydrothermal method. The transducer was driven continuously at the fundamental or higher resonance frequencies (32, 96, 160, or 210 MHz) in a water cell, and the projection light of a laser beam penetrating perpendicularly to the sound axis was observed. The length of the projection light was elongated in the sound axis on the screen as the driving frequency increased. These results implies that the relationship between the wavelength of ultrasound and the laser beam width is the critical parameter for the acousto-optic phenomenon; the profile of the projection light is dependent on a ratio of the wavelength of ultrasound and the beam width of the laser beam.

**3:30**

**4pPAb7. Optical fiber tip damage by sparks in holmium:YAG laser lithotripsy.** Yuri A. Pishchalnikov (Applaud Medical, Inc., 953 Indiana St., San Francisco, CA 94107, yuri.pishchalnikov@applaudmedical.com), William M. Behnke-Parks (Applaud Medical, Inc., San Francisco, CA), and Marshall L. Stoller (Dept. of Urology, Univ. of California San Francisco, San Francisco, CA)

Laser lithotripsy disintegrates urinary stones using laser pulses delivered to the stone via an optical fiber. The fiber tip is frequently damaged during the procedure, but the mechanism of damage is not understood. Here we show that the free-running long-pulse ($>$150 $\mu$s) infrared holmium:YAG lasers that currently are most often used for laser lithotripsy can produce sparks and shock waves, which is an unexpected result as the optical breakdown is typically observed with more powerful Q-switched lasers. We used an ultrahigh-speed camera Shimadzu HPV-X2 at frame rate up to 10 million frames per second to record single laser pulses with 242-$\mu$m glass-core-diameter fibers in contact with whole surgically retrieved urinary stones, BegoStones, and hydroxyapatite-coated glass slides in air and water. Fiber-tip damage occurred even without cavitation and was greater at 1.0 J than at 0.6 J ($p = 0.010$). No fiber-tip damage occurred without sparks (1.0 J, 500 pulses, water). Shock waves produced by sparks and subsequent collapses of vapor bubbles could break optical fiber tip in a single laser pulse. These observations suggest that the previously unappreciated optical breakdown with the free-running holmium:YAG lasers can be responsible for fiber-tip damage in laser procedures. [Work supported by NIDDK of NIH under award R43DK129104.]

**3:45**

**4pPAb8. Imaging acoustic phenomena with transmission electronic speckle pattern interferometry.** Thomas R. Moore (Dept. of Phys., Rollins College, Box 2743, Winter Park, FL 32789, tmoore@rollins.edu)

Optically imaging acoustic phenomena requires sensing the pressure-induced change in index of refraction of the medium, and imaging this change in index has been accomplished in the past using a variety of methods. However, these methods typically require a static ambient atmosphere, the sensing area is often limited by the size of the optics, and in some cases the phenomena of interest must be either reproducible or stable for time periods exceeding tens of minutes. We demonstrate that transmission electronic speckle pattern interferometry (TESPI) is a fast and effective method for imaging gas flow, acoustic standing waves, and standing waves in the presence of flow, with sensing times of less than one second and a viewing area of a square meter or more. [Work supported by NSF grant No. PHY–2109932.]

**4:00**

**4pPAb9. Effective Grüneisen constant in sodium fluoride crystals.** Farkhad Akhmedzhanov (Lab. of Thermophysics of Multiphase Systems, Inst. of Ion-plasma and Laser Technologies of the Acad. of Sci. of Uzbekistan, 33 Durmon yuli St., Tashkent, Tashkent 100125, Uzbekistan, akhmedzhanov.f@gmail.com)

In this paper, we propose a new approach to determining the components of the Grüneisen tensor and establishing unambiguous relationships between the effective Grüneisen constant, which characterizes the nonlinearity of interatomic interaction forces, and the attenuation anisotropy of acoustic waves. The cubic crystals of sodium fluoride were taken as an object of study. Measurements were carried out by the method of Bragg diffraction of light by sound in the frequency range 0.4–1.6 GHz.The obtained values of velocity and attenuation of acoustic waves were used to determine the real and imaginary components of the elasticity tensor. Next, the acoustic damping coefficients and the independent components of the Grüneisen tensor in sodium fluoride crystals were determined. It is shown that the calculations performed trough the attenuation coefficients make it possible to determine the effective Grüneisen constants in two ways, and these results are in good agreement with each other. Thus, the components of the acoustical Grüneisen tensor for sodium fluoride crystals are determined for the first time and the dependence of the effective Grüneisen constant on the direction of propagation and polarization of acoustic waves is calculated.

THURSDAY AFTERNOON, 8 DECEMBER 2022 SUMMIT E, 1:00 P.M. TO 4:00 P.M.

# Session 4pSC

## Speech Communication: Bilingualism and Second Language Acquisition (Poster Session)

Laura Spinu, Chair
*Communications & Performing Arts, CUNY Kingsborough, 2001 Oriental Blvd. Room E329, Room E329, Brooklyn, NY 11235*

All posters will be on display from 1:00 p.m. to 4:00 p.m. Authors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

### *Contributed Papers*

**4pSC1. The third language imitation of Spanish rhotic-lateral contrast by Akan-English bilingual children and adults: The effect of first language transfer, orthography, and age of acquisition.** Linda Ahima (Lang. and Cultures, Western Univ., London, ON, Canada), Gabriela Martinez Loyola (Lang. and Cultures, Western Univ., 1151 Richmond St., London, ON N6A 3K7, Canada, gmarti49@uwo.ca), and Yasaman Rafat (Lang. and Cultures, Western Univ., London, ON, Canada)

This study explores the third language imitation of Spanish rhotics and laterals by Akan-English bilingual children and adults. Specifically, it examines this phenomenon, with respect to (i) first language transfer in the imitation of the Spanish rhotic contrast, (ii) the effect of orthographic input on the imitation of the rhotic-lateral contrast in Spanish as a third language, and (iii) the effect of age of acquisition via the inclusion of a comparative analysis between children and adults. Whereas rhotics and laterals are contrastive in most varieties of Spanish and English, they are in free variation in Akan. The participants in this study consisted of 20 Akan-English bilinguals between the ages of 8–10 years and 18–55. The conditions included an imitation task with auditory input only and another with orthographic input. The results demonstrated evidence of L1 transfer, a positive effect of orthography, and a better performance of children. This study has important implications as it provides data from an understudied population in the field of L2/L3 speech learning.

**4pSC2. An acoustic examination of English fricative production by Korean- and Farsi-English bilinguals: The role of language- and orthographic-specific effects.** Yasaman Rafat (Lang. and Cultures, Western Univ., London, ON, Canada), Laura Spinu (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, 2001 Oriental Blvd. Rm. E329, Brooklyn, Brooklyn, NY 11235, laura.spinu@kbcc.cuny.edu), Veronica Whitford (Univ. of NB, St. John, NB, Canada), Marc Joanisse (Psych., The Univ. of Western ON, London, ON, Canada), Vidhi Trivedi (Western Univ., London, ON, Canada), Natasha Swiderski (McMaster Univ., Hamilton, ON, Canada), Celina Valdivia (Education, Western Univ., London, ON, Canada), Sarah Cornwell (Western Univ., London, ON, Canada), and Mercedeh Mohaghegh (Univ. of Toronto, Toronto, ON, Canada)

Recent work has shown that exposure to orthographic effects can promote first-language (L1) phonological transfer. However, it is relatively unknown whether orthographic effects persist in highly proficient bilinguals. Here, we examined how L1 orthographic depth (regularity in grapheme-phoneme correspondences) modulated Korean-English and Farsi-English bilinguals' ($n = 25$ each) production of fricatives in English words (e.g., <mellow> /ˈmeloʊ/ vs. <melon> /ˈmelən/). Native English speakers ($n = 25$) served as a control group. Because fricatives are produced as geminate sounds in both Korean and Farsi, we expected L1-based transfer.

Participants completed four tasks: an eye-movement reading task, word-naming task, cloze test, and language background questionnaire. The stimuli were controlled for word frequency, word length, number of syllables, and stress. We expect language-specific differences, corroborating previous neuro-linguistic evidence that shallow and deep orthographies differentially rely more heavily on phonological and lexical pathways, depending on language-specific demands. To explore this aspect, we employed an acoustic classification method for fricatives extracting cepstral coefficients and using HMMs to divide the sounds in regions based on their internal variance, aimed at determining whether fricatives produced by different groups can be classified correctly. This work has implications for both second-language (L2) speech learning models and classroom instruction.

**4pSC3. Exploring the connection between articulatory skill and phonetic and phonological learning: A magnetic resonance imaging study.** Laura Spinu (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, 2001 Oriental Blvd. Rm. E329, Brooklyn, Brooklyn, NY 11235, laura.spinu@kbcc.cuny.edu), Duke Shereen (Adv. Sci. Res. Ctr., CUNY Graduate Ctr., New York, NY), Anastasiia Myslyk (Commun. Arts, Sci., and Disord., CUNY Brooklyn College, Brooklyn, NY), and Mariany Rojas (School of Health and Natural Sci., Mercy College, Dobbs Ferry, NY)

Research has revealed positive consequences of bilingualism on cognition, some of which have been collectively referred to as the 'bilingual advantage', although the term remains controversial. The lack of replicability of such studies is often ascribed to methodological issues, such as the difficulty of quantifying the bilingual experience. Emerging areas of research where a consistent bilingual advantage *has* been identified include studies on phonetic and phonological learning (PPL)—the ability to learn the features of a novel accent effectively after initial exposure. Our main goal is to explore articulatory skill as a potential mechanism underlying the differences in PPL between mono- and bilingual populations. We used MRI instrumentation to visualize the movements of the tongue, lips and velum as our English monolingual and English-Spanish bilingual participants (n = 24) were trained to produce unfamiliar speech sounds, specifically high front rounded vowels, low back nasalized vowels, and secondarily palatalized labiodental fricatives. We also administered a PPL task and a performance-based test of language proficiency. With data analysis underway, we hypothesize that PPL correlates with articulatory skill and proficiency modulates the results across the board. Our study thus adds to the body of work on the coupling between sensory and cognitive functions.

**4pSC4. Decoding the neural dynamics of within- and between-language lexical competition: A pediatric neurosurgical case report of a Spanish-English bilingual.** McCall E. Sarrett (Psychol. and Brain Sci., Villanova Univ., Tolentine Hall 334, 800 E Lancaster Ave., Villanova, PA 19085, mccall.sarrett@villanova.edu), Ariane E. Rhone (Neurosurgery, Univ. of Iowa Hospitals & Clinics, Iowa City, PA), Christine Shea (Spanish and Portuguese, Univ. of Iowa, Iowa City, IA), Kristi Hendrickson (Commun. Sci. and Disord., Univ. of Iowa Hospitals & Clinics, Iowa City, IA), John B. Muegge (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA), Christopher Kovach (Neurosurgery, Univ. of Iowa Hospitals & Clinics, Iowa City, IA), Brian J. Dlouhy (Neurosurgery, Univ. of Iowa Hospitals & Clinics, Iowa City, IA), and Bob McMurray (Psychol. and Brain Sci., Univ. of Iowa, Iowa City, IA)

Spoken word recognition proceeds by immediately activating items in the mental lexicon which match the incoming speech signal. These items compete for recognition over time. In multilinguals, the array of possible lexical competitors includes items across all their languages. Eye-tracking work indicates that the strength of early competitor activation relates to listeners' proficiency: words from the dominant language outcompete those from the other(s). However, the neural mechanisms that give rise to this effect are not well known. We present a case study of a pediatric neurosurgical patient who was bilingual in English (nondominant) and Spanish (dominant). The participant passively listened to English and Spanish cohort competitors. We used machine learning techniques to decode within- and between-language lexical competition dynamics from the pattern of activity on the superior temporal plane. Spanish words were more robustly decodable than English words, regardless of their role as a target word (the actual word heard on a given trial) or as a competitor (the phonologically related word competing for activation). This is consistent with a Spanish-dominant system—corroborating evidence from eye-tracking—and suggests that activation of cohorts may be in part due to perceptual tuning of early auditory areas for the sounds of a language.

**4pSC5. An examination of the perceptions of Cuban and Peninsular Spanish varieties by native, second language learners of Spanish, and monolingual English speakers.** Gabriela Martinez Loyola (Lang. and Cultures, Western Univ., 1151 Richmond St., London, ON N6A 3K7, Canada, gmarti49@uwo.ca) and Yasaman Rafat (Lang. and Cultures, Western Univ., London, ON, Canada)

Individuals have been found to make judgments on a person's personality, income, education, and employment in as little as 30 seconds after listening to their voice. This study investigates the perceptions of Cuban and Peninsular Spanish varieties by native Cuban and Peninsular Spanish speakers, second language (L2) Spanish learners, and monolingual English speakers, to analyze whether (i) these groups differ in their ability to recognize these varieties, and (ii) whether there is stigma attributed to either accent. The study consisted of 5 Cuban (Havana) and 5 Peninsular (Madrid) voices which were disguised and rated by 20 adult listeners. The methodology included the administration of a Bilingual Language Profile (BLP) questionnaire and a survey intended to gauge at listeners' perceptions. Results revealed that listeners do make unconscious assumptions on an individual's voice, as the Peninsular variety was often attributed to a higher educational level (84% SP, 48% CU), income (60% SP, 40% CU), and was more closely associated with a CEO position (32%), while the Cuban voice was associated with being funnier and slightly more intelligible. Furthermore, native Cuban listeners were found to outperform all other groups in their ability to correctly categorize the accents with an accuracy rate of 92%.

**4pSC6. Voicing and manner of articulation in L2 perception: The role of markedness.** Faisal Alkahtani (Univ. of Delaware, 125 E Main St., Newark, DE 19711, kahtani@udel.edu)

Arabic labial obstruents exhibit gaps in voicing: both the voiceless stop /p/ and the voiced fricative /v/ are absent. Thus, Arabic speakers learning English must acquire the missing labial segments along with segments that are already in their phonemic inventory. In this study, I examine markedness in L2 acquisition of English by Arabic speakers, focusing on its role in perception. Specifically, I test two general markedness patterns: (a) stops are unmarked relative to fricatives and (b) voicing contrast in word-initial position is unmarked relative to word-final position. If unmarked sounds are easier to acquire than marked ones, we may expect Arabic speakers to perceive the new segment /p/ better than /v/. Moreover, their perception of the voicing contrast in word-initial position should be better than in word-final position. 31 Saudi Arabic speakers heard 144 nonce words produced by a native speaker of English. The words included: /p/, /b/, /f/ and /v/ in word-initial or word-final position. Results showed Arabic speakers were significantly better at perceiving the fricative /v/ than the stop /p/ ($p < .01$), contrary to what was expected, suggesting that the acoustic cues of fricatives are perceptually more prominent. The markedness relation was, however, observed with better perception word-initially than word-finally.

**4pSC7. Perception of English vowel length modulation due to vowel identity and the identity of the following consonant by Korean learners of English.** Juyeon Chung (Linguist., Indiana Univ., Memorial Hall 322, Bloomington, IN 47405-7005, chungjulia29@gmail.com)

In English, consonant voicing has large effects on the previous vowel duration, as does the status of the vowel as tense or lax. Our aim is to examine the relationship between discrimination and identification of these contrasts by Korean L2 listeners. An ABX discrimination task shows little correlation between the two contrasts, so that listeners who distinguish English consonant voicing contrasts do not tend to do better with the English tense/lax contrasts. An identification task, however, indicates a strong correlation across the two different contrasts such that listeners who identify the correct underlying final consonant are able to identify the correct underlying

vowel identity in their English perception. Further, comparing the participants' performance on the two tasks shows a different relationship between discrimination and identification. For voicing contrasts, discrimination performance needs to reach a level of accuracy before the listeners improve in identification, suggesting that auditory discrimination is a foundation for identification. However, for the tense-lax contrast, the listeners' performance in the two tasks is entirely uncorrelated, suggesting that part of the difficulty for the listeners acquiring the tense-lax contrast is that they auditorily attend to aspects of the signal which are inappropriate for identification.

**4pSC8. Perception of English lexical stress in different intonational contexts by Mandarin listeners.** Tzu-Hsuan Yang (Linguist., Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045-3129, thyang@ku.edu) and Annie C. Tremblay (Lang. and Linguist., Univ. of Texas at El Paso, El Paso, TX)

The present study seeks to elucidate the nature of first language transfer effects on the perception of English lexical stress in different intonational contexts. Pitch (F0) is one of the acoustic cues to lexical stress in English, but crucially, pitch can only serve as a cue to stress when interpreted within the intonation system of English. In contrast, Mandarin is a tonal language where pitch carries greater functional weight for signaling lexical contrasts. Associating pitch to lexical contrasts, Mandarin listeners might therefore assume a one-to-one relationship between pitch and lexical stress in English. To examine whether Mandarin learners of English make this assumption, the current study tested their stress perception in four different intonations: H*L-L%, L*H-H%, H*H-H%, and L*L-L%, where F0 cues to stress were realized differently in each intonation. The results of the stress identification task showed that Mandarin listeners falsely associated higher F0 with stress and lower F0 with the absence of it, and their performance did not improve as they became increasingly proficient in English. The findings provide corroborating evidence for the Cue-Weighting Transfer Hypothesis by showing that the use of a suprasegmental cue can transfer from one phonological category (tones) to another (stress).

**4pSC9. Evaluating how language and masked-speech recognition change over time for bilingual children in the Midwest.** Stefani Garcia (Case Western Reserve Univ., 10900 Euclid Ave., Cleveland, OH 44106, stefani.garcia@case.edu), Tiana Cowan (Boys Town National Res. Hospital, Omaha, NE), Emily Buss (The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Lauren Calandruccio (Case Western Reserve Univ., Cleveland, OH), Ryan McCreery, Margaret Miller (Boys Town National Res. Hospital, Omaha, NE), Barbara Rodriguez (The Univ. of New Mexico, Albuquerque, NM), and Lori Leibold (Boys Town National Res. Hospital, Omaha, NE)

Spanish-English bilingual children living in the US often become more proficient in English compared to Spanish over time, in part due to education models. For example, most bilingual children in the Midwest receive educational instruction in English or rapidly transition to English-only instruction. If a bilingual child's relative proficiency shifts, how does that shift affect their speech recognition? To address this question, we invited participants from the Children's English and Spanish Speech Recognition Test (ChEgSS) normative study back three years later to retest their speech recognition. The initial study evaluated word recognition in 83 Spanish-English bilingual children with normal hearing. Children completed Spanish and English word recognition tests in speech-shaped noise and two-talker speech using a 4AFC procedure. Language proficiency and language use were assessed using standardized tests and a questionnaire. Results indicated that age and receptive vocabulary influenced word recognition performance. In this study we retested a subgroup of children who were between the ages of 4-8 years in the initial study using the same procedures. Results will advance our understanding of the association between language proficiency and word recognition in this demographic of bilingual children.

**4pSC10. Abstract withdrawn.**

**4pSC11. The perception of English vowels by native Korean and Mandarin speakers.** Alexis N. Zhou (Linguist., Purdue Univ., 100 University St., B290, West Lafayette, IN 47907, atews@purdue.edu) and Olga Dmitrieva (Linguist., Purdue Univ., West Lafayette, IN)

Second language learners often experience difficulties discriminating L2 contrasts absent in their L1. Theoretical frameworks, (e.g., SLM, PAM, L2LP), make predictions about the acquisition of L2 phonemes based on the existence of equivalent or similar L1 phonemes, but do not explicitly consider the extent of L1 allophonic variability as a contributing factor. The present study compared the perceptual discrimination of English contrasts /ɪ-i/ and /ɛ-æ/ by learners with Mandarin and Korean L1 backgrounds. Both languages have only one L1 phoneme for the two relevant L2 contrasts. Furthermore, Mandarin [ɛ] is an allophone of the highly variable mid vowel phoneme /E/, while Korean /ɛ/ lacks this variability. Allophonic variability may condition learners to accept diverse vowel qualities as acceptable renditions of the corresponding phoneme, therefore lowering their discriminability. In an AX perceptual discrimination task, 16 Standard Mandarin and 14 Korean speakers were similar in their discrimination of the /ɪ-i/ vowels, but, as predicted, the Mandarin group was less accurate for the /ɛ-æ/ pairs. These findings suggest that allophonic variability in the L1 needs to be considered when making theory-driven predictions concerning the acquisition of specific phonological categories.

**4pSC12. "New" sounds detract attention from "similar" sounds: "New" and "similar" L2 sounds for English speakers learning French.** Danilo A. Lombardo (Program in Speech-Language-Hearing Sci., The Graduate Ctr., City Univ. of New York, Speech Production, Acoust., and Percept. Lab., The Graduate Ctr., CUNY, 365 Fifth Ave., New York, NY 10016-4309, dlombardo1@gradcenter.cuny.edu) and D. H. Whalen (CUNY Graduate Ctr., New York, NY)

The pronunciation of a foreign language (L2) is difficult for adult learners, which has been attributed to two classes of L2 sounds: "New" sounds, with unfamiliar articulatory gestures, and "similar" sounds, which are likely to be replaced by native (L1) sounds. For American English (AE) speakers learning French, /ʁ/ is a "new" sound, while the voicing of stops is a "similar" feature. We hypothesized that both "similar" and "new" sounds require attention for correct pronunciation, and that "new" sounds attract more attention than "similar" ones. We predicted that "similar" sounds will be more L1-like when "new" sounds are in the same utterance. Preliminary results for AE speakers producing French utterances with or without /ʁ/ suggest that the presence of /ʁ/ impacted surrounding sounds. In particular, the VOT of voiceless stops was sometimes longer (i.e., more English-like) in sentences with than without /ʁ/. The "new" sound /ʁ/ appears to require attention that is then less available for controlling details of the L2 stops. This result points, seemingly for the first time in the L2 speech literature, to competition between "new" and "similar" sounds for the learner's attention, which allows positing a previously unexplored cognitive mechanism in L2 accent.

**4pSC13. Vowel categorization accuracy and its relationship to phonemic awareness.** Charles C. Ball (School of Psych., Western Sydney Univ., Bldg. 24, Bankstown Campus, Locked Bag 1797, Penrith, New South Wales 2751, Australia, charles.ball21@gmail.com), Tamara Watson, and Michael D. Tyler (School of Psych., Western Sydney Univ., Sydney, New South Wales, Australia)

Cross-language speech perception studies regularly employ categorization tasks to determine listeners' repertoire of phonological categories. In these tasks, participants select a phoneme from a list of letters or keywords to indicate which speech category they heard. However, recent studies have

shown poor categorization accuracy when native (L1) English speakers are asked to categorize English vowels. Categorization tasks require *phonemic awareness* (PA), the explicit knowledge that words consist of segments, which is acquired alongside alphabetic literacy. Individual differences in PA may account for variability in L1 categorization accuracy. That is, participants with poor phonemic awareness may have difficulty explicitly identifying phonemes in a categorization task, particularly for English vowels, where grapheme-phoneme correspondences are relatively opaque. To test whether PA is related to categorization accuracy, 115 L1 English listeners completed an L1 vowel categorization task and three PA tasks: phoneme counting, deletion, and reversal. Strong correlations were observed between categorization accuracy and all three PA tasks, suggesting PA may account for a substantial proportion of individual variability in vowel categorization tasks. This finding questions the utility of the categorization task for determining listeners' phonemic categories and has further implications for assessing second language speech categories.

**4pSC14. Characterization and normalization of second language speech intelligibility through lexical stress, speech rate, rhythm, and pauses.** Okim Kang (English, Northern Arizona Univ., 705 S Beaver St., Flagstaff, AZ 86011, okim.kang@nau.edu), Kevin Hirschi (English, Northern Arizona Univ., Flagstaff, AZ), John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX), and Stephen Looney (Penn State Univ., State College, PA)

While a range of measures based on speech production, language, and perception are possible (Manun *et al*., 2020) for the prediction and estimation of speech intelligibility, what constitutes second language (L2) intelligibility remains under-defined. Prosodic and temporal features (i.e., stress, speech rate, rhythm, and pause placement) have been shown to impact listener perception (Kang *et al*., 2020). Still, their relationship with highly intelligible speech is yet unclear. This study aimed to characterize L2 speech intelligibility. Acoustic analyses, including PRAAT and Python scripts, were conducted on 405 speech samples (30 s) from 102 L2 English speakers with a wide variety of backgrounds, proficiency levels, and intelligibility levels. The results indicate that highly intelligible speakers of English employ between 2 and 4 syllables per second and that higher or lower speeds are less intelligible. Silent pauses between 0.3 and 0.8 s were associated with the highest levels of intelligibility. Rhythm, measured by $\Delta$ syllable length of all content syllables, was marginally associated with intelligibility. Finally, lexical stress accuracy did not interfere substantially with intelligibility until less than 70% of the polysyllabic words were incorrect. These findings inform the fields of first and second language research as well as language education and pathology.

**4pSC15. Measuring second language acquisition of spanish lenition.** Rachel Meyer (Linguist., Univ. of Florida, 4131 Turlington Hall, P.O. Box 115454, Gainesville, FL 32611, rmeyer2@ufl.edu), Ratree Wayland, and Fenqi Wang (Linguist., Univ. of Florida, Gainesville, FL)

In intervocalic position, Spanish /b/, /d/, and /g/ are lenited to fricatives or approximants [$\beta$], [ð], and [ɣ] (Hualde 2005).The goal of this study is to examine the ability to learn allophones of these Spanish voiced stops among L2 learners. Eighteen native English-speaking Spanish learners were recorded reading word lists at the beginning and end of a semester-long course on Spanish pronunciation, which included explicit instruction on voiced stop allophones. Ennever, Meakins, and Round (2017)'s automatic segmentation method was used to analyze degree of lenition. This method uses the intensity contour to determine the on- and offset of intervocalic consonants. It yields three measurements: difference between maximum and minimum intensity ($\Delta$i), maximum change in intensity over time, i.e. velocity, ($v_{peak}$) and duration. A smaller absolute value of $\Delta$i and $v_{peak}$, and a shorter duration correspond with a greater degree of lenition. Between the two recording times, $\Delta$i, $v_{peak}$, and duration all decreased, demonstrating some learning of Spanish lenition. Results on the impact of preceding and following vowels, and consonant place of articulation were mixed. Syllable stress and age of onset of Spanish instruction had no effect on lenition.

**4pSC16. The role of sensorimotor mechanisms in phonetic and phonological learning.** Nicole Nieves (City Univ. of New York - Kingsborough Community College, 2001 Oriental Blvd, Brooklyn, NY 11235, nicolernieves@gmail.com), Laura Spinu (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, Brooklyn, NY), Anastasiia Myslyk, Llewellyn Duncan (City Univ. of New York - Brooklyn College, Brooklyn, NY), Christie Oreste, and Kevin Roodnauth (City Univ. of New York - Kingsborough Community College, Brooklyn, NY)

Phonetic and phonological learning was reported to be enhanced in bilinguals; however, the underlying mechanisms remain understudied. Prior work suggests a bilingual advantage in articulatory skill and auditory sensory memory, raising the question of the involvement of sensorimotor functions in phonetic and phonological learning. To explore this relationship, we compare the performance of English monolinguals and English-Spanish bilinguals on three tasks:(1) Articulatory skill, using a novel sound learning task with articulatory complex sounds not present in our participants' inventory.(2) Auditory sensory memory, measured via an adaptive digit span task. (3) Phonetic and phonological learning, using an artificial accent of English displaying diphthongization of mid-front lax vowels and tapping of intervocalic liquids. All tasks were conducted online 1-on-1 via Zoom. Tasks 1 and 3 were scored based on four listeners' perceptual judgments. For Task 2, we obtained response accuracy and input-output similarity scores. While data analysis is currently underway, preliminary findings replicate the previously observed bilingual advantage in phonological learning and auditory sensory memory and reveal a correlation between them. Bringing together motor, sensory, and acquisitional components and investigating the ways in which they are related adds to lesser explored mechanisms potentially underlying the bilingual advantage.

**4pSC17. Linguistic transfer of voice onset time: Aspiration and voicing.** Jarry C. Chuang (Dept. of English, National Chengchi Univ., No. 64, Sec. 2, ZhiNan Rd., Wenshan District, Taipei City 11605, Taiwan R.O.C., Taipei 11605, Taiwan, cwchuang.academia@gmail.com)

Crosslinguistic comparison of VOT has indicated linguistic transfer of voicing and aspiration contrasts in many languages. Mandarin has clear aspiration contrasts for voiceless stops, while Min presents another complicated VOT pattern, where voicing and aspiration contrasts are involved. The present study makes a crosslinguistic comparison between languages with voicing and aspiration contrasts as well as the potential linguistic transfer of VOT in English contexts from Mandarin and Min. There are three subject groups, including American English natives and Mandarin-Min bilinguals with different levels of Min-fluency. Mandarin-Min bilinguals have more aspirations for aspirated voiceless stops than English natives. They also present two surfaces for English underlying voiced stops, voiced and unaspirated voiceless. Different levels of Min fluency are found to influence the tendencies towards voiced or unaspirated voiceless representations of English voiced stops. The overall finding presents a clear crosslinguistic influence on VOT patterns.

**4pSC18. Bilingual effects on lexical retrieval and performance on a word guessing game.** Marta Pyrih (Commun., Sci. and Disord., City Univ. of New York- Brooklyn College, 2900 Bedford Ave., Brooklyn, NY 11210, martapyrih16@gmail.com), Mariana Vasilita, Julia Wallace, Ketevan Shavdia, and Revette Hinkson (Commun. Dept., City Univ. of NY, Kingsborouh Community College, Brooklyn, NY)

Bilingualism has been associated with cognitive disadvantages (compared to monolinguals) in terms of lexical retrieval, even in bilinguals' native language. According to Gollan *et al*. (2002), a possible source of the disadvantages is cross-language interference. Another potential explanation is that bilinguals process even their first language differently from monolinguals. Higby *et al*. (2013) suggest that, to understand the advantages and disadvantages of multilingualism, one must consider the linguistic system as a whole and how effective management of more than one language affects

all aspects of cognition, both linguistic and nonlinguistic. Our experimental study includes two tasks: a lexical decision task in which participants are shown a single word and have to decide whether it is a real word, and the web-based word game Wordle, which provides six attempts at guessing a 5-letter word. Each participant was tested individually and recorded via Zoom under 1-on-1 supervision, sharing their screen throughout the process. Our measures include reaction times for the lexical decision task, and overall duration, number of trials required to arrive at a correct answer, and accuracy for Wordle. Based on previous findings, we hypothesize slower lexical decision reaction times in bilinguals but higher performance on the word game.

**4pSC19. Phonological learning of American English: An acoustic study of Flapping in Mandarin-speaking EFL Contexts.** Jarry C. Chuang (Dept. of English, National Chengchi Univ., No. 64, Sec. 2, ZhiNan Rd., Wenshan District, Taipei City 11605, Taiwan R.O.C., Taipei 11605, Taiwan, cwchuang.academia@gmail.com)

The study explores the nature of non-native flapping under Mandarin-speaking EFL contexts. Flapping as a dominant feature in American English (AE) offers an indicative insight into phonological learning. Forty college students were asked to read words and sentences with intervocalic flapping potentially produced. (Non-)English major students (ENGs/non-ENGs) were respectively invited to make distinctions on the impact of language exposure on crosslinguistic flapping production. Acoustics results have examined and showed that ENGs had a higher frequency to produce [è*f*] than non-ENGs might. While more flapping was produced, ENGs possibly make erroneous phones [d]. It has been interpreted as the results of phonetic upward convergence and hypercorrection, where [ɾ] for /t/ can be produced as [d] for the similar quality and the same voicing they share. It is also found that female subjects had more flapped sound [ɾ] in production than males do. Generally, acoustic findings demonstrate that flapping is achievable phonological processing in non-English native contexts. More exposure to English and linguistic training can potentially acoustically trigger a higher occurrence of flapping in production (, while It remains uncertain whether flapping could be articulatorily transferred).

**4pSC20. Prosodic modulations on the formant excursions of diphthongs.** Miao Zhang (Linguist., Univ. at Buffalo, SUNY, 6 Affinity Ln., Apt. 152C, Buffalo, NY 14215, miaozhan@buffalo.edu)

The excursions of the first two formants of diphthongs in Chinese, English, and Japanese were investigated to see how diphthongs are affected by the hierarchical prosodic structure. The production of /ai, au, ou/ was examined in Chinese and English, and /ai, au, ae/ in Japanese. Three prosodic contexts were examined: word-final, list-final, and IP (intonational phrase)-final positions. The results of GAM analysis of F1 and F2 show that (1) F2 was more affected than F1; (2) diphthong offset was more affected than the onset; (3) only the diphthong onset in Japanese was affected by the prosody. The by-language analysis further shows that the offset of diphthongs at word-final positions in Chinese was less resistant to the coarticulatory effect: F2 raises earlier and higher toward the end of the diphthong due to the influence of the following coronal consonant. Meanwhile, in English and Japanese /ai, ae/ with lower F2 and higher F1 and /au, ou/ with higher F1 and F2 at word-final positions were found, indicating hypoarticulation at lower prosodic boundaries. Some of the diphthong onsets in Japanese were influenced as well. The result suggests hyperarticulation as a prosodic strengthening effect in English and Japanese.

**4pSC21. Vowel nasalization and Nasal Coda in Mandarin Chinese.** Jarry C. Chuang (Dept. of English, National Chengchi Univ., No. 64, Sec. 2, ZhiNan Rd., Wenshan District, Taipei 11605, Taiwan, cwchuang.academia@gmail.com)

Vowel nasalization for onset or coda nasal is a common phenomenon in many languages. As coda nasals belong to the rime and involve syllable timing, the interaction between nucleus vowel and nasal coda might reflect onto duration patterns. In Mandarin, coda nasal [n, ŋ] and coda glide [j,w] are in complementary distribution, while acoustic results have found a coda nasal takes a longer duration than a coda glide. Besides, with the increase in the nasality of vowels (triggered by the nasal coda), the duration of the nasal coda is correspondently longer, accompanied by the durative lenition of vowels. This demonstrates that the nasal coda starts sharing the mora (originally belonging to the vowel), with the help of vowel nasalization.

## Session 4pSP

# Signal Processing in Acoustics: Dispersive Wave Signal Processing II

Zoi-Heleni Michalopoulou, Cochair
*Department of Mathematical Sciences, New Jersey Institute of Technology, Newark, NJ 07102*

Julien Bonnel, Cochair
*Woods Hole Oceanographic Institution, 266 Woods Hole Rd, MS# 11, Woods Hole, MA 02543-1050*

**Chair's Introduction—1:00**

## *Invited Papers*

**1:05**

**4pSP1. Normal mode extraction using sparse Bayesian learning in shallow water.** Haiqiang Niu (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China) and Peter Gerstoft (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, gerstoft@ucsd.edu)

The low-frequency signals propagating in the shallow-water waveguides are dispersive. They are composed of several normal modes according to the normal mode theory. For the vertical array data, the horizontal wavenumbers and the associated multi-frequency modal depth functions were estimated using block sparse Bayesian learning (*Niu et al., JASA 2020*), while *a priori* knowledge of sea bottom, moving source, and source locations is not needed. For the impulsive or known-form signals received by one hydrophone, the sparse Bayesian learning (SBL) approach can be also used to extract the modes (*Niu et al., JASA 2021*). It uses the approximate modal dispersion relation, connecting the horizontal wavenumbers (phase velocities) for multiple frequencies, to build the dictionary matrix for SBL. Different from warping transforms based on the group slowness (or group speed) dispersion curves, mode separation using SBL is performed in frequency domain based on the phase speed (or equivalently horizontal wavenumbers) dispersion relation. The simulation results demonstrate that the proposed approach is adapted to the environment where both the reflected and refracted modes coexist, whereas the performance of the time warping transformation degrades significantly in this scenario.

**1:25**

**4pSP2. In-sensor signal processing techniques for Lamb waves inspections.** Luca De Marchi (Univ. of Bologna, Viale del Risorgimento 2, Bologna, BO 40136, Italy, l.demarchi@unibo.it) and Masoud Mohammadgholiha (Univ. of Bologna, Bologna, Italy)

Structural health monitoring based on guided waves (GW) is typically performed by controlling a large number of piezoelectric transducers placed on the component to be inspected. The permanent installation of this technology allows to acquire information about the structural integrity on demand, but it has several limiting factors: bulky hardware instrumentation and large number of connecting cables, complex signal processing, high power consumption and, consequently, high integration costs. Such limitations hamper the adoption of the GW technology in application domains with stringent weight requirements (e.g., aerospace and automotive).These limitations can be addressed by adopting transducers with inherent beam steering capabilities. Such capabilities can be achieved by properly shaping the electrodes of the piezotransducers according to the dispersion characteristics of the waveguide to produce anisotropic wavenumber filtering effects. This "In-Sensor" signal processing can be used to selectively generate ultrasonic waves along arbitrary directions depending on the frequency content of a single excitation signal, or to automatically detect the direction of arrival of mechanical waves propagating within the structure with minimal hardware/software requirements. [In this work (supported by the Horizon2020 project GW4SHM), numerical and experimental validations of this concept will be presented, together with possible future developments.]



**1:45**

**4pSP3. Geo-acoustic inversion using ground waves of explosive sounds.** Lin Wan (Elec. and Comput. Eng., Univ. of Delaware, 139 the green, 301 Evans Hall, Newark, DE 19716, wan@udel.edu) and Mohsen Badiey (Elec. and Comput. Eng., Univ. of Delaware, Newark, DE)

The ground waves of explosive sounds exhibit strong dispersion in shallow water waveguides. Their travel time and amplitude are related to the geo-acoustic properties in the seabed and could be utilized to construct cost functions for geo-acoustic inversion. Following the analysis of the Airy phase [Wan *et al.*, JASA, 143, EL199-205, (2018)], this research focuses on the extraction and utilization of ground waves, which propagate at frequencies below the corresponding Airy frequency. First, the ground waves of low order modes are identified in the received signals from explosive SUS charges deployed at various depths/ranges and recorded by vertical line arrays during the series of Seabed Characterization Experiments (SBCEX) conducted in the New England continental shelf and slope region. Then, the extracted ground waves, showing strong dispersive characteristics in a bandwidth of a few Hertz (i.e., between cut-off frequency and Airy frequency), are fed to a proposed inversion algorithm to obtain the estimated geo-acoustic parameters. Finally, the advantage of using ground waves rather than water waves is discussed and a physical explanation is provided. [Work supported by ONR Ocean Acoustics.]

2:05

**4pSP4. Recent advances in phase-array based wideband multimodal dispersion curves extraction and plate-waveguide parameter inversion.** Kailiang Xu (Ctr. for Biomedical Eng., School of Information Sci. and Technol., Fudan Univ., No. 2005, Songhu Rd., Yangpu District, Shanghai, Shanghai, China, xukl@fudan.edu.cn), Feiyao Ling (Ctr. for Biomedical Eng., School of Information Sci. and Technol., Fudan Univ., Shanghai, China), Honglei Chen (Acad. for Eng. & Technol., Fudan Univ., Shanghai, China), Yifang Li (Ctr. for Biomedical Eng., School of Information Sci. and Technol., Fudan Univ., Shanghai, China), Tho N. H. T. Tran (Acad. for Eng. & Technol., Fudan Univ., Shanghai, China), Pascal Laugier (Laboratoire d'Imagerie Biomédicale (LIB) CNRS 7371 - INSERM 1146, Sorbonne Université, Paris, France), Jean-Gabriel Minonzio (Escuela de Ingeniería en Informática, Universidad de Valparaíso, Valparaíso, Chile), and Dean Ta (Ctr. for Biomedical Eng., School of Information Sci. and Technol., Fudan Univ., Shanghai, China)

Multimodal ultrasonic guided waves with rich dispersion information have been widely applied for waveguide evaluation. However, there are still some challenges to extract multimode dispersion curves using the traditional pitch-catch method, including lack of spatial information and poor signal-to-noise ratio. Phase array has been recently developed for measuring ultrasonic guided waves, which brings many advantages for wideband multimodal dispersion curves extraction and parameter inversion. In the last years, basing on a multi-emitter and multi-receiver measurement, we proposed some array dispersive signal processing strategies for analyzing guided waves in long cortical bone, including the sparse singular vector decomposition (sparse-SVD) method for enhancing wavenumber estimation resolution and the low-amplitude mode extraction, Radon transform and dispersive Radon transform (DRT) with the capability to project temporal array dispersive signals on the space of parameters of interest for solving the inverse problem, and our recent work of deep neural networks for solving the intractable multiparameter inverse problem from the array signals to the waveguide elasticity. In the talk, we present (1) recent advances in array signal processing for extracting wideband multimodal dispersion curves; (2) some new perspectives to retrieve the waveguide parameters; (3) some pilot clinical results of long cortical bone evaluation using ultrasonic guided waves.

**2:25–2:35 Break**

2:35

**4pSP5. Inversion of ocean acoustic modal-dispersion and amplitude data for seabed geoacoustic models including attenuation.** Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, University of Victoria, Victoria, BC V8W2Y2, Canada, minj@uvic.ca), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

The long-range acoustic field due to an initial sound source in a shallow-water waveguide can be expressed as the sum of a number of dispersive, propagating modes. Modal-dispersion data (arrival time as a function of frequency), as extracted with warping time-frequency analysis, have been inverted to estimate seabed sediment sound-speed and density profiles. In this work, measurements of modal amplitudes as a function of frequency are combined with modal-dispersion data in a joint (simultaneous) inversion which also provides sensitivity to frequency-dependent sound attenuation coefficients in the sediment layers. Since the source signature (spectrum) is often unknown in practice, the relative amplitudes between modes can be considered as data, or the source spectrum can itself be included as additional unknowns in the inversion. Here, trans-dimensional Bayesian inversion is applied to modal-dispersion and amplitude data to estimate marginal probability profiles for sediment geoacoustic properties, including attenuation. The signal processing and inversion methods are demonstrated using modal-dispersion and amplitude data extracted from acoustic measurements made by a hydrophone-equipped underwater glider during the 2017 Seabed Characterization Experiment (Work supported by the Office of Naval Research).

## *Contributed Papers*

2:55

**4pSP6. An end-to-end deep learning approach for joint detection, source localization and environmental characterization using a single hydrophone in shallow water.** Ariel Vardi-Chouchana (MIT/WHOI, 29 Wiltshire Rd., Newton, MA 02458, arielv@mit.edu) and Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA)

Environmental characterization of underwater environments is crucial for modeling acoustic propagation in the ocean, which in turn enables underwater applications such as performing environmental impact studies on the effect of ocean noise on marine life, evaluating sonar performance and more. In this work, the aim is to automate and improve run-time performance of environmental characterization, compared to classical methods which employ high-cost computational schemes and sometimes require manual inputs from a skilled operator. To do so, we use 1D Convolutional Neural Networks (1D-CNN), a type of Deep Learning (DL) model, to acoustically characterize the underwater environment using unprocessed pressure time-series recorded on a single hydrophone. We notably propose an end-to-end approach to detect, classify and localize acoustic sources along with estimating environmental parameters using a single DL model. The experimental data used for testing the resulting 1D-CNN models are signals that were generated by navy explosives (SUS charges) deployed during the Seabed Characterization Experiment performed in the New England Mud-patch off the coast of Massachusetts.

3:10

**4pSP7. A Bayes factor based high frequency broadband active acoustic detection scheme for vertical line arrays in ocean waveguides.** Daniel Lopes (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, dlopes5@umassd.edu), Ryan Ferreira, Paul J. Gendron, and Christopher Norton (ECE, Univ. of Massachusetts Dartmouth, Dartmouth, MA)

From the log posterior odds we develop a Bayes Factor (BF) inference processor for high frequency broadband active monostatic vertical receiver arrays operating in shallow water ocean waveguides. The relevant information regarding the refractive media, rough surface and volume reverberation are incorporated to build the marginal likelihoods under each of the composite hypotheses of null and alternative. Acoustic scattering from a mobile object of interest under considerable depth uncertainty characterizes the compound alternative hypothesis. Approximations are presented and inferences regarding the presence of the mobile body of interest are determined against a composite null hypothesis of reverberation and ambient acoustic noise. Reverberation is modeled via Lambert rough surface scattering. The BF processor is shown to be a time-varying quadratic form of array observations over the beam-delay space. We discuss the structure of the quadratic form relative to the expected angle-delay spectra of target and reverberation. We illustrate the BF inferential approach by considering various refractive waveguides as well as the iso-velocity case. The BF inference structure can

be employed for decision theoretic loss functions in order to establish processors that optimally incorporate vertical arrival structure for decision making. [This work supported by the Office of Naval Research.]

**3:25**

**4pSP8. Performance of a Bayes factor (BF) based broadband active acoustic system.** Ryan Ferreira (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747, rferreira6@umassd.edu), Paul J. Gendron, and Daniel Lopes (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Dartmouth, MA 02747)

We explore the performance of a Bayes Factor (BF) active acoustic inference approach under various refraction and rough surface scattering regimes that typify diverse ocean waveguides. The BF processor, a time varying quadratic form over beam-angle observations, enables various user risk minimization strategies. The nature of the BF quadratic form is shaped by the expected ambient noise, reverberation and target angle-delay spectra. The structure of the BF processor provides intuition regarding the role of prior environmental and bathymetric information on detection in shallow water ocean waveguides with rough boundary scattering. The distribution of the BF under the composite null hypothesis when target and reverberation are well approximated as uncorrelated is shown to be a superposition of heteroskedastic gamma densities with the degree of freedom being the dimension of the target hypothesis scattering sub space. The distribution of the BF under the composite alternative is a superposition of heteroskedastic scaled non-central chi-square distributions. These distributions do not admit simple closed forms but can be well approximated. We consider a number of refractive and scattering regimes and relate the probability of detection as a function of probability of false alarm under conventional fixed threshold and cost rules. [This work supported by the Office of Naval Research.]

THURSDAY AFTERNOON, 8 DECEMBER 2022 NORTH COAST B, 1:00 P.M. TO 3:50 P.M.

## Session 4pUW

## Underwater Acoustics and Acoustical Oceanography: Memorial Session for Lisa Zurk II (Hybrid Session)

Kathleen E. Wage, Cochair
*George Mason University, 4400 University Drive, Fairfax, VA 22151*

Martin Siderius, Cochair
*Portland State Univ., 1600 SW 4th Avenue, Suite 260, Portland, OR 97201*

Daniel Rouseff, Cochair
*University of Washington, 1013 NE 40th St., Seattle, WA 98105*

**Chair's Introduction—1:00**

### *Invited Paper*

**1:10**

**4pUW1. Office of Naval Research Undersea Signal Processing, Contributions of Dr. Lisa Zurk.** Keith L. Davidson (Ocean Battlespace and Expeditionary Access, Office of Naval Res., 875 N Randolph St., Code 32, Arlington, VA 22203, keith.l.davidson4.civ@us.navy.mil)

The Undersea Signal Processing team at the Office of Naval Research (ONR 321US) is tasked with developing advanced active/passive sonar signal and information processing technologies that improve the anti-submarine warfare (ASW) performance of US Navy sonar systems. Pursuant to this goal ONR supports academics whose research, being on the cutting edge, lead to high-risk technologies with potential for the greatest advancement of fleet capability. Dr. Lisa Zurk was such an academic, first supported by ONR 321US in 2005. Dr. Zurk's research focused on developing signal and information processing algorithms which exploited the physics of the underwater operating environment. She believed the path to significant gains in sonar system performance would be obtained through exploitation of environmental knowledge. This belief is now a realization within the Naval Research Enterprise (NRE) and was a primary motivation for initiation of the ONR Task Force Ocean (TFO) Program in 2017. From establishment of the NEAR-Lab at Portland State University in 2005 until her transition to a Program Manager at the Defense Advanced Research Projects Agency in 2016, Dr. Zurk made significant contributions to the NRE. This presentation looks to highlight these past contributions and to consider their present impact.

## *Contributed Paper*

**1:30**

**4pUW2. Extensions to Lisa Zurk's depth-based signal separation method.** Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375, altan.turgut@nrl.navy.mil) and Warren T. Wood (Naval Researh Lab., Hancock County, MS)

Lisa Zurk and her colleagues introduced a depth-based signal separation method that uses interference between the direct and surface-reflected CW signals which are received on a bottom-moored vertical line array in deep water [Asilomar Conference on Signals, Systems and Computers, IEEE, 2130–2312, 2013, *J. Acoust. Soc. Am.* **133**(4), EL320-EL325, 2013]. In this work, her method was extended to broadband signals for instantaneous depth-based signal separation and range estimation. The broadband method was successfully tested by using a single phone or a pair of phones, deployed in the Northern Gulf of Mexico. A broadband (250–800 Hz) source was towed at 20 m, 50 m, and 100 m depths and received at four bottom-moored Environmental Acoustic Recording Systems (EARS). Source depth estimation and localization by trilateration was successfully demonstrated. Realizing the lack of comprehensive CW and broadband data sets, two experiments were planned for late 2022 and mid 2023 in collaboration with Lisa Zurk. These experiments will still be conducted to demonstrate her depth-based signal separation method experimentally. [Work supported by the ONR.]

## *Invited Papers*

**1:45**

**4pUW3. Ambient acoustic enviroment—Diurnal soundscapes.** John S. Allen (Mech. Eng., Univ. of Hawaii Manoa, Holmes 302, 2540 Dole St., Honolulu, HI 96822, alleniii@hawaii.edu)

Soundscape ecology has emerged as an important technique for monitoring complex coastal environments. Spatial and temporal acoustic variability exists between different regions and habitats. The sound characteristics have been hypothesized to be related to the associated community's diversity and health. The prominent sounds may originate from physical, biological, and anthropogenic sources. Biological sounds especially those at coral reefs locations may have seasonal, monthly and diurnal variations. Snapping shrimp noise can dominate the ambient noise background in these sub-tropical coastal regions. While seasonal variations of snapping shrimp sounds have been studied extensively, the diurnal variations across habitats are less understood. Ambient noise data from two sites on Oahu, Hawaii are investigated with respect to diurnal variations using both spectral and time series analysis. Motivations and inspirations from Dr. Lisa Zurk's extensive work on passive acoustics are highlighted. In addition, Dr. Zurk made notable theoretical and numerical contributions to acoustic scattering research. Some recent studies on the scattering from gas bubbles in complex media are also outlined in tribute.

**2:05**

**4pUW4. Wind-driven underwater ambient noise for seabed remote sensing: A review of methods and results.** Jorge Quijano (JASCO Appl. Sci., Victoria, BC, Canada, Jorge.Quijano@jasco.com), Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Martin Siderius (Portland State Univ., Portland, OR)

Estimation of seabed geoacoustic parameters in shallow water by acoustic remote sensing remains a challenging task due to constraints on hardware, data collection and analysis, and cost of maritime surveys. The work of Lisa Zurk and Martin Siderius, among many others, inspired investigation of signal processing techniques to process wind-driven underwater noise for the estimation of seabed geoacoustics. This paper presents a summary of the work on Bayesian geoacoustic inversion of ambient noise carried out at University of Victoria and Portland State University, using fixed and drifting vertical line arrays (VLA). The Bayesian inversion framework is based on Markov-chain Monte Carlo sampling. It estimates a joint posterior probability density function, from which marginal density functions, moments and covariances between geoacoustic parameters of interest can be quantified. The approach was applied to simulated and experimental ambient noise data for the estimation of seabed layer thicknesses, sound speeds, densities, and sediment attenuations. The resolution of the method was explored as a function of experiment factors such as array design and wind speed, and a method for extending the aperture of a VLA by extrapolating the noise coherence was proposed.

**2:25–2:40 Break**

**2:40**

**4pUW5. Reflections on Lisa Zurk as an Educator.** Gabriel P. Kniffin (The Johns Hopkins Univ. Appl. Phys. Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099, gabriel.kniffin@jhuapl.edu)

Lisa Zurk was my advisor and mentor through both my masters and doctoral degree programs. I had the pleasure of working with her beginning in 2008; first as her graduate student, then in my subsequent professional career while she was a program manager in the Strategic Technology Office at DARPA. I consider myself extremely fortunate to have been part of the Northwest Electromagnetics and Acoustics Research Laboratory (NEAR-Lab), which she founded at Portland State University in 2005. Under her guidance and leadership, the group became a diverse community of tight-knit students, faculty, and other researchers; united in our passion for research in terahertz imaging and spectroscopy, underwater acoustics, and physics-based signal processing. My experience there, and that of my fellow students, paved the way for our future careers in science and engineering in industry, academia, and everywhere in between. Today, NEAR-Lab alumni can be found at top tech companies such as Apple, Boeing, Intel, Tektronix, and Metron; nationally-renowned research organizations like The Johns Hopkins University Applied Physics Laboratory, MIT Lincoln Laboratory, and the Pacific Northwest National Laboratory; and even internationally at the University of Victoria, British Columbia, and JASCO Applied Sciences. The success of her students speaks volumes about her impact on the generation of innovators that followed her. This talk will reflect on Dr. Zurk as an educator and will present some of the work she coauthored and published with her students.

**3:00**

**4pUW6. Video Tributes to Lisa Zurk.** Kathleen E. Wage (George Mason Univ., 4400 University Dr., Fairfax, VA 22151, kwage@gmu.edu), Daniel Rouseff (Univ. of Washington, Seattle, WA), and Martin Siderius (Portland State Univ., Portland, OR)

This multimedia presentation is a compilation of reflections on Lisa Zurk's career and legacy.

**3:20–3:50**
**Panel Discussion**

THURSDAY AFTERNOON, 8 DECEMBER 2022 NORTH COAST B, 1:00 P.M. TO 3:50 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. All meetings will begin at 7:30 p.m., except for Signal Processing in Acoustics (4:30 p.m.) and Engineering Acoustics (4:45 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:

| | |
|---|---|
| Signal Processing in Acoustics (4:30) | Rail Yard |
| Engineering Acoustics (4:45) | Lionel |
| Acoustical Oceanography | North Coast A |
| Animal Bioacoustics | Grand Hall A |
| Architectural Acoustics | Summit A |
| Physical Acoustics | Golden Pass |
| Psychological and Physiological Acoustics | Grand Hall C |
| Structural Acoustics and Vibration | Golden Eagle B |

### Committees meeting on Wednesday:

| | |
|---|---|
| Biomedical Acoustics | Mill Yard A |

### Committees meeting on Thursday:

| | |
|---|---|
| Computational Acoustics | Summit C |
| Musical Acoustics | Rail Head |
| Noise | Summit B |
| Speech Communication | Grand Hall B |
| Underwater Acoustics | North Coast B |

# Session 5aAA

## Architectural Acoustics and Noise: Sound with Context: Cultural Heritage Acoustics

Miriam A. Kolar, Cochair
*Center for Computer Research in Music and Acoustics (CCRMA), Stanford University, School for Advanced Research (SAR), PO Box 2188, Santa Fe, NM 87504-2188*

David Lubman, Cochair
*DL Acoustics, Westminster, CA 92683-4514*

**Chair's Introduction—9:00**

### *Invited Papers*

**9:05**

**5aAA1. From room acoustics to paleoacoustics: A preliminary acoustical study in Chauvet Cave.** Miriam A. Kolar (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., School for Adv. Res. (SAR), PO Box 2188, Santa Fe, NM 87504-2188, miriamakolar@gmail.com), Luna Valentin (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA), Peter Svensson, Sara Martin (Norwegian Univ. of Sci. and Technol. (NTNU), Trondheim, Norway), Jonathan S. Abel (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA), Romain Michon (INRIA & INSA, Lyon, France), Carole Fritz, Gilles Tosello (Ctr. de Recherche et d'Etudes de l'Art Préhistorique (CREAP) — Emile Cartailhac & Ctr. National de la Recherche Scientifique (CNRS), Toulouse, France), John Chowning, and Matt Wright (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA)

Due to geological closures between 21 000 and 29 000 years ago, the acoustics of the UNESCO World Heritage site, Chauvet Cave (Ardèche, France) have been in slow flux via mineral deposition processes that continue to alter its interior. Since Upper Paleolithic humans created extensive and elaborate artworks throughout this grand limestone cavern more than 30 000 years ago, the cave's interior has changed with calcite and other minerals forming a diversity of features including the best-known stalactites, stalagmites, and flowstone floor coverings. Here, we report on archaeoacoustics fieldwork in 2022 that initiated acoustical mapping and reconstructive modeling to enable archaeological acoustics research and the creation of auralizations and multimodal experiences for virtual public access to this conservation-restricted place. We present here a comparative room acoustics study of two substantively enclosed cave areas (Salle du Fond and Galerie du Cactus) whose volumes differ significantly, but whose extant reverberation times are similar across most center bands, providing important information about the dynamical contributions of surface materials and structural features distinct to each impulse-response-measured location. Our study exemplifies an archaeological application of room acoustics methods with site-responsive techniques that offer a human-centered approach for understanding and translating cultural heritage acoustics across time.

**9:25**

**5aAA2. Ringing tone and drumming sages in the Crevice Cave of Pirunkirkko, Koli, Finland.** Riitta Rainio (Dept. of Cultures, Univ. of Helsinki, Fabianinkatu 24, h. 132 (PL 4), Helsingin yliopisto 00014, Finland, riitta.rainio@helsinki.fi) and Elina Hytönen-Ng (Univ. of Eastern Finland, Joensuu, Finland)

Pirunkirkko ("Devil's Church") is one of the famous caves in Finland. Tradition says that this Crevice Cave leading inside the Koli mountain was a meeting place for sages, who typically used sound to contact the spirit world. Today, the place is visited by practitioners of shamanism, who organise drumming sessions at the back of the cave. This paper examines Pirunkirkko and the related traditions from the perspective of acoustics, hypothesising that the acoustic characteristics of the narrow crevice might have played a role in the ritualization of the place and the power of its sonic rituals. Methods employed include impulse response recording, spectrum analysis, archival research, and an interview of a shamanic practitioner analysed with discursive psychology. The results indicate that the corridor-like back of the cave houses a distinct resonance phenomenon. A standing wave between the smooth parallel walls generates a ringing tone at 230 Hz that stays audible after sharp impulses or tones vocalised at the same frequency. Surprisingly, the local folklore or the interviewed practitioner do not mention this ringing tone at all. Instead, they speak about the "spirit of the cave," "special energy," or "new horizons" opened up by drumming. This leads to reflection on cultural frameworks of thought that guide sensory perceptions leading to differing experiences and interpretations.

## *Contributed Paper*

9:45

**5aAA3. Case study of a Brothertown Indian Nation cultural heritage siteToward a framework for acoustics heritage research in simulation, analysis, and auralization.** Timothy Hsu (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, 535 W. Michigan St., IT 371, Indianapolis, IN 46202, hsut@iu.edu), Seth Wenger (New York, NY), and Hope Leonard (Music and Arts Technol., Indiana Univ. - Purdue Univ., Indianapolis, Indianapolis, IN)

The Brothertown Indian Nation and their ancestors have a centuries old heritage of group singing beginning along the Northeast Atlantic coast of what is now the United States and following them along their migration to the Midwest. The 18th century Curricomp cabin still stands in Connecticut and was an important location in the singing history and political movement that built Brothertown (Eeyamquittoowauconnuck). In this paper, a collaboration with Brothertown Indian Nation and an ongoing public humanities project regarding the Tribe's aural traditions, a case study will be presented that investigates (1) the results of acoustic modeling and simulation of the Curricomp cabin and (2) auralizations for both binaural listening and a spatial audio installation using those models. Significantly, this paper employs a novel theoretical framework for acoustic heritage research that allows for (3) an analysis of how the scientific process and technological practices mediate intangible heritage. The acoustic models and auralization techniques created in this case study provide acoustic access to a heritage location that would otherwise be inaccessible to the Brothertown Community. The auralizations used in the spatial audio installation serve as a public humanities tool to amplify the contemporary voice of the Brothertown Indian Nation.

**10:00–10:15 Break**

## *Invited Paper*

10:15

**5aAA4. Virtual acoustic reconstruction of the Roman amphitheater of Avella in Italy.** Umberto Berardi (Toronto Metropolitan Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca), Gino Iannace, Amelia Trematerra (Università della Campania Luigi Vanvitelli, Aversa, Caserta, Italy), and Antonella Bevilacqua (Università della Campania Luigi Vanvitelli, Parma, Italy)

In ancient Rome, gladiator fights were very popular. The places where these shows took place were called amphitheaters, due to the particular shape of the elliptical building. Amphitheaters were widespread, and in each city, one or more of these buildings was present. This paper describes the virtual reconstruction of the acoustics of the amphitheater Avella, close to Naples, Italy. This amphitheater was discovered a few decades ago and was only partially rebuilt. Today, it is used for musical performances during the summer season mainly. The plan of the building is elliptical, and the dimensions of the arena are 35 m for the minor axis and 65 m for the major axis. Acoustic measurements were performed in the current state with an impulsive sound source and the acoustic parameters were obtained according to the ISO 3382 standard. Subsequently, from the dimensions of the current state and in analogy with the architecture of other amphitheaters existing today, a virtual reconstruction of the original shape was performed. The virtual model was developed with the architectural acoustics software Ramsete in order to obtain the acoustic characteristics of the amphitheater as it was in Roman times. In particular, the spatial distribution of the acoustic characteristics on the steps where the audience was seated is described and discussed.

## *Contributed Paper*

**10:35**

**5aAA5. Developing a virtual reality application for cultural heritage and room acoustics education.** Sang Bum Park (School of Architecture and Eng. Technol., Florida Agricultural and Mech. Univ., 1938 South Martin Luther King Jr. Blvd., Tallahassee, FL 32307, sang.park@famu.edu)

We tend to use visual factors, such as architectural styles, features, specific decorations, historical contexts, etc., to characterize cultural heritage. Sound is a transient, ethereal phenomenon that tends to be neglected in historical records. While photographs and drawings can preserve the visual aspect of a building or scene, documenting the sonic impact of the spaces is more complicated. In particular, the historic places used for sonic activities, such as music halls, performance halls, and worship spaces, are essential to document and preserve the acoustic qualities. An immersive experience using virtual reality (VR) technology effectively promotes public awareness about cultural heritage's importance. It simulates the room acoustics using spatial audio technology. It also can make the VR environment interactive to manipulate architectural features that change the room acoustics, such as room volume, finish materials, reverberation time, sound barrier, etc. The main goal of this paper is to develop a VR application that can be used as a template to create a VR environment where 3rd to 8th-grade students navigate and learn about the history and architectural features of cultural heritage and the basics of room acoustics using a Quest headset.

## *Invited Paper*

**10:50**

**5aAA6. Aural heritage preservation and access: Methodological explorations from data collection to immersive multimodal virtual reality.** Sungyoung Kim (RIT, ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623, sxkiee@rit.edu), Doyuen Ko (Belmont Univ., Nashville, TN), Miriam A. Kolar (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Santa Fe, NM), and Xuan Lu (RIT, Rochester, NY)

From prehistorical archaeological sites to middle-American banks, classic Nashville recording studios, and present-day concert halls, spatial acoustics influence human experience, behavior, and interactions. We previously proposed a definition of *aural heritage*: "spatial acoustics as physically experienced by humans in cultural contexts," and have tested equipment and methods to document aural heritage in diverse sites. In fieldwork and in the studio/lab, we have evaluated and extended standard room acoustics techniques to address ecological validity (realism) in aural heritage preservation including specific human-centered methods. For the translation of aural heritage data into auralizations, we have focused on multichannel audio with and without congruent visual immersion, anticipating multimodal virtual reality presentations of cultural heritage reconstructions. Here, we overview our multi-year project to develop, test, and share aural heritage preservation, translation, and access technologies via three detailed case study applications and associated outreach projects. We (1) summarize the translatable aural heritage data collection protocol for long-term preservation; (2) discuss preferred methods for processing aural heritage data into scalable auralizations for different audio platforms; and (3) preview extensibility pathways for other researchers to implement these methodologies that enable researchers and public audiences to explore and experience this aspect of intangible culture.

**11:10–11:40**
**Panel Discussion**

# Session 5aAB

## Animal Bioacoustics, Speech Communication, Signal Processing in Acoustics, and Psychological and Physiological Acoustics: Mapping Acoustic Features to Production Mechanisms in Speech and Animal Communication

Benjamin N. Taft, Chair
*Landmark Acoustics LLC, 1301 Cleveland Avenue, Racine, WI 53405*

### *Invited Papers*

**8:30**

**5aAB1. Neural and cognitive mechanisms for vocal communication.** Gregg A. Castellucci (Neurosci. Inst., NYU School of Medicine, 435 E30th St., NYULMC Sci. Bldg., New York, NY 10016, gregg.castellucci@nyulangone.org)

Vocal communication is a central feature of social behavior across numerous species. While the neural systems underlying vocalization in humans and animals is well described, less is known about how these circuits enable naturalistic vocal interactions. To investigate how the human brain gives rise to ethologically relevant vocal interactions—conversational turn-taking—a series of intracranial recording and perturbation experiments are performed to precisely assay neural activity while neurosurgical patients engage in both task-based and unconstrained turn-taking. In these social contexts, spatially and functionally distinct networks are uncovered, which are critical for a speaker's ability to comprehend their partner's turns, plan their own turns, and articulate the speech comprising those turns. To better understand the neural mechanisms underlying specific computations relevant to vocal communication, a theoretical framework is constructed, which consists of the cognitive modules required for generating communicative action during interaction (e.g., vocalization, co-speech gesture). This model is designed to account for the behavioral and neurobiological features of both naturalistic human language and animal communication; therefore, this species-general framework is intended to facilitate the identification of cognitive analogues between human and non-human interaction—which may rely on similar neural mechanisms. [Work supported by the NIH & Simons Collaboration on the Global Brain.]

**8:50**

**5aAB2. Vocal learning, chorusing seal pups, and the evolution of rhythm.** Andrea Ravignani (Comparative Bioacoustics Group, Max Planck Inst. for Psycholinguistics, Wundtlaan 1, Nijmegen 6525 XD, Netherlands, Andrea.Ravignani@mpi.nl), Marianna Anichini (Comparative Bioacoustics Group, Max Planck Inst. for Psycholinguistics, Nijmegen, Netherlands), Marlene Sroka (Sealcentre Pieterburen, Muenster, Germany), Mila Varola (Sealcentre Pieterburen, Italy, Italy), Anna Salazar Casals (Sealcentre Pieterburen, Pieterburen, Netherlands), Koen de Reus, and Laura Verga (Comparative Bioacoustics Group, Max Planck Inst. for Psycholinguistics, Nijmegen, Netherlands)

The presence of vocal learning and rhythm capacities in only a few mammals drives a cross-species hypothesis in evolutionary neuroscience: vocal learning and rhythm perception and synchronization may be causally related. Harbor seals (*Phoca vitulina*) are among these mammals, and their puppyhood is the most active vocal period in their lives. Building on both comparative psychology and animal behavior, this research integrates methods from human cognition with attention to species' ecological sensory niches. We present data from seal pups in four different setups: recordings of semi-natural vocal interactions in (1) individual and (2) group settings, (3) playback experiments to elicit vocal responses, and (4) perceptual listening experiments to measure behavioral responses. We complement empirical data with agent-based computational modeling aimed at reverse-engineering the putative mechanisms underlying rhythmic vocal interactions. Our data suggest that seal pups have developed capacities to produce and perceive rhythmic patterns. Seals' vocal exchanges show rhythmic interactivity and antisynchronous coordination. Evidence for antisynchrony, rather than synchrony, in seals suggests that the rhythm-vocal learning link across species is more complex than previously surmised.

**9:10**

**5aAB3. Understanding production variability in multimodal communication.** Yoonjeong Lee (Linguist, Univ. of Michigan, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeol@umich.edu)

A comprehensive view of speech communication requires a consideration of dynamic characteristics in language users' multimodal behaviors. This talk discusses dynamic properties of both verbal and non-verbal communication behaviors, focusing on variability in voice, speech, and co-speech gesture production. I present a series of experiments that leverage quantitative approaches to examine how the multimodal production units of the cognitive system vary in their dynamic control and realization in executing communication goals. The overarching hypothesis is that surface variations in multimodal behaviors are structured to reflect communicative intentions, exhibiting a tight relation between the linguistic system and recruited modalities. Supporting evidence emerges from two types of data: dynamic voice signals drawn from speech corpora of various speakers, speaking styles, and languages, and time-aligned multidimensional signals of speech and co-speech gestures (simultaneously recorded audio, kinematic, and visual signals). Voice variations are acoustically structured by both biologically relevant factors and factors that vary with language-specific phonology. Furthermore, the

spatiotemporal patterning of vocal tract actions and cooccurring body movements systematically varies, reflecting the prosodic structure of language. The findings and the utility of the analytical tools employed in these studies have implications for comparative investigations of communicative behaviors in animals. [Work supported by the NSF.]

## *Contributed Paper*

**9:30**

**5aAB4. Advances in aero-acoustics of flying beetles.** John S. Allen (Mech. Eng., Univ. of Hawaii Manoa, Holmes 302, 2540 Dole St., Honolulu, HI 96822, alleniii@hawaii.edu) and Kevin 0'Rourke (Adaptive Res., Las Vegas, NV)

An understanding of the acoustics of invasive species of beetles is needed for potential detection and tracking methods in agricultural monitoring. However, the underlying mechanisms of sound generation are not well understood, especially with respect to the higher harmonic sounds. The Coconut Rhinoceros Beetle (Oryctes rhinoceros) and the Oriental Flower Beetle (Protaetia orientalis) have been studied during tethered flight with synchronized microphone array measurements and high speed video (1000–10,000 fps). The larger Coconut Rhinoceros Beetles have fundamental ~50 Hz with distinctive torsional wing rotation compared to Oriental Flower Beetle (~100 Hz). Computational fluid dynamics simulations were performed using the unsteady compressible flow solver (CAESIM, Adaptive Research, Inc.) using a highresolution (TVD) methodology. Models of the wing flapping motion were accomplished using mesh deformation techniques with the flapping following from rotation with prescribed bending and coupled rotation and translation from the wing's hinge position. Fluid structure interactions with respect to the wing's flexibility are investigated in terms of the wing bending and the leading edge vortex formation.

**9:45–10:00 Break**

## *Invited Papers*

**10:00**

**5aAB5. Urban habitats and noise impact avian communities and shape vocal mimicry in a songbird.** Dana Moseley (Biology, James Madison Univ., 951 Carrier Dr MSC 7801, Bioscience James Madison U, Harrisonburg, VA 28807, moseledl@jmu.edu) and Jaclyn Tolchin (Biology, James Madison Univ., Harrisonburg, VA)

Urban landscapes present problems for wildlife including highly modified habitat, anthropogenic noise pollution, and competition for suitable habitat. These novel selection pressures filter species present in urban habitats or result in changes to behavior. Recent studies show that some bird species sing differently in noisier, urban areas compared to rural areas, and that anthropogenic noise alone can exclude species. We investigated how the level of urbanization affects (1) bird species composition detected acoustically, and (2) the assemblage of species that a songbird mimics from the local bird community. We studied gray catbirds, *Dumetella carolinensis*, across an urban gradient (Washington, DC to Virginia) and predicted catbirds in urban habitats would mimic fewer species or mimic more urban-adapted species. The results showed avian community composition differed along the urban gradient and was best predicted by impervious surface and noise level. The species composition of catbird mimicry was predicted by site soundscape, indicating catbird mimicry reflects the local avian community. Since urbanization level was found to significantly predict avian community composition, these findings support the hypothesis that urbanization impacts catbird song by influencing the community of sounds available for catbirds to mimic. Catbirds living in habitats of varying levels of urbanization have significantly different songs from catbirds in other habitats, due to their surrounding bird communities and noise levels differing.

**10:20**

**5aAB6. Individual variations in vocal and cognitive responses to simulated acoustic environments.** Keiko Ishikawa (Commun. Sci. and Disord., Univ. of Kentucky, Lexington, KY, ishikawa@uky.edu), Silvia Murgia, Hannah Li, Elisabeth Coster (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL), Elle Harris, Calley Moore (Commun. Sci. and Disord., Univ. of Kentucky, Lexington, KY), and Pasquale Bottalico (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Humans "automatically" make vocal and speech adjustments in response to room acoustics; however, it is unclear whether these adjustments require additional cognitive load. Beyond this automatic adjustment, speakers may have to employ a speech production technique to enhance their intelligibility in situations such as public speaking. How does room acoustics challenge speakers on such occasions? Understanding the relationship between room acoustics, speech production, and cognitive load has practical implications for various areas, including speech training and room design. This study describes the associated change in cognitive load when speaking with and without a speech production technique at three levels of reverberation time (i.e., 0.05 s, 1.2 s, and 1.83 s at 125 Hz). Ten adult native speakers of American English were asked to read sentences in simulated acoustic environments using two types of speech production: habitual speech and clear speech, an intelligibility-enhancing technique. Cognitive load during the speech tasks was monitored with pupillometry and a subjective self-rating scale. Participants' speech production behaviors were acoustically assessed via speech rate and alpha ratio. Preliminary results indicate that reverberation time significantly affects speech production behaviors and cognitive load and that the pattern of cognitive response varies substantially among the speakers.

**10:40**

**5aAB7. Should more effortful utterances map to more elaborate signals?** Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

Many songbirds tune the frequency of their whistled vocalizations by varying the length of their vocal tract. To do this, they must open their bills wider to produce higher tones and then close them again for lower pitches. The time course of frequency variation in such a bird's song, thus, maps directly to the time course of motor effort by its bill. In 2016, Podos *et al*. described frequency excursion as "for any given song or song segment, the sum of frequency modulations both within and between notes on a per-time basis." As predicted from the previous work, birds with larger bills are constrained to produce less frequency excursion than birds with smaller bills. After reviewing this result, I will elaborate on ways to extend the idea of a map between motor effort and signal characteristic from one-to-one to many-to-many.

## *Contributed Papers*

**11:00**

**5aAB8. Observation of a golden cheeked warbler (*Setophaga chrysoparia*), utilizing a typical B-song but an atypical A-song.** David P. Knobles (Phys., Knobles Sci. and Anal., 5416 Tortuga Trail, Austin, TX 78731, dpknobles@kphysics.org), Darrell Hutchinson (Austin Water Utility, Wildland Conservation Div., City of Austin, Austin, TX), Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Lisa O'Donnell (Austin Water Utility, Wildland Conservation Div., City of Austin, Austin, TX)

From 27 March through 2 June 2017, a golden cheeked warbler (*Setophaga chrysoparia*), code named "Piccolo," claimed a territory within the Balcones Canyonland Preserve near Austin, Texas and sang a combination of atypical A-songs combined with typical B-songs. To the authors' knowledge, this A-song variant has not been previously reported. *In situ* observations of Piccolo's movements were used to determine the extent of Piccolo's defended territory. In addition, between 14 April and 7 June, an acoustic recorder was deployed in the territory to record the variant song along with the soundscape. On April 28, a second-year competitor male entered Piccolo's territory and intense and prolonged counter-singing was recorded through May 24. The competitor produced typical A- and B-songs while Piccolo counter-sang in the aforementioned variant A-songs and typical B-songs. During the recording period, Piccolo did not vocalize any typical A-songs. Two additional competitor males that used typical B-songs were also observed and recorded. The *in situ* observations and acoustic analysis of Piccolo's atypical countersigning were compared to typical interactions between other nearby birds to ascertain whether the variant song provided an advantage or disadvantage for Piccolo. [Work supported by the City of Austin.]

**11:15**

**5aAB9. Investigating the individual difference in distance calls of Bengalese finches using acoustic analysis.** Wei Chen (Dept. of Life Sci., Graduate School of Arts and Sci., the Univ. of Tokyo, Kaga2-21-1, Itabashi-Ku, Tokyo 173-0003, Japan, chen-wei0601@g.ecc.u-tokyo.ac.jp) and Kazuo Okanoya (Adv. Comprehensive Res. Organization, Teikyo Univ., Tokyo, Japan)

Most songbirds are monogamous and exhibit biparental care. They socially interact with other individuals, including those outside the family, through auditory modality. The capacity to explicitly identify conspecific individuals via acoustic cues enables fast turn-taking communication in a crowd, especially when visual signals are not available. Bengalese finches (*Lonchura striata* var. *domestica*) are domesticated strains of the white-rumped munia imported from China to Japan about 260 years ago. Previous studies have reported distance calls, which are emitted when a bird is visually separated from its conspecifics, are sexually dimorphic (both acoustically and perceptually) in Bengalese finches. However, the individuality in distance calls has not been thoroughly discussed yet. In this research, we recorded vocalizations from four pairs of Bengalese finches in kinship. To extract distance calls from those recordings, we use the Harmonics-to-Noise Ratio (HNR) to detect and remove noise, which mainly consists of aperiodic components. By analyzing acoustic features such as length, bandwidth, and peak frequency, we found that calls from the same individuals form clear clusters, and the difference between other individuals was more pronounced in contrast to kins. [Work supported by MEXT/JSPS grant #4903, 17H06380.]

5a FRI. AM

# Session 5aBA

## Biomedical Acoustics: General Topics in Biomedical Acoustics II: Imaging and Therapeutics

Randall P. Williams, Cochair
*Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105*

Vera A. Khokhlova, Cochair
*University of Washington/Moscow State University, Physics Faculty, Moscow State University, Moscow 119991, Russian Federation*

### *Contributed Papers*

**9:15**

**5aBA1. High resolution focused-ultrasound-induced thermoacoustic imaging.** Tianqi Shan (Biomedical Eng., Chongqing Medical Univ., No.1 Xueyuan Rd., Chongqing 40016, China, tianqi@cqmu.edu.cn)

We report a new imaging modality, focused-ultrasound-induced thermoacoustic imaging (FUTAI). FUTAI provides high imaging resolution and unique contrast of acoustic absorption and thermodynamic properties of tissue. When a short pulse of the focused ultrasound beam is applied to biological tissue, the tissue will absorb the energy of the incident ultrasound causing a rapid change in local temperature, which leads to thermoelastic expansion and generates acoustic waves. In this work, we investigated this focused-ultrasound-induced thermoacoustic effect through theoretical simulation and experimental validation. A novel imaging modality FUTAI is proposed and demonstrated for the first time. By scanning the focus of the incident ultrasound beam, a 2D or 3D image of tissue properties can be formed. FUTAI provides better resolution compared to traditional B-mode ultrasound imaging, and deeper imaging depth compared to photoacoustic imaging. Additionally, since FUTAI is sensitive to acoustic absorption, it has unique advantages in guiding therapies like high intensity focused ultrasound (HIFU) in which treatment efficiency is determined by the acoustic absorption property. In this work, we also proposed a new method to localize the focus of HIFU with high accuracy using FUTAI. The results indicate that FUTAI has great potential in applications such as noninvasive imaging and therapy guidance.

**9:30**

**5aBA2. *In vivo* thermal ablation control using three-dimensional echo decorrelation imaging in swine liver.** Elmira Ghahramani Z. (Biomedical Eng., Univ. of Cincinnati, 3960 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, ghahraea@mail.uc.edu), Peter D. Grimm (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), R. Cutler Quillin, Sameer H. Patel, Syed A. Ahmad, Shimul A. Shah (Surgery, Univ. of Cincinnati, Cincinnati, OH), Nicholas S. Schoenleb, Ben Connolly, Bahar Saremi, and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

In experiments modeling clinical thermal ablation of liver tumors, radiofrequency ablation (RFA) of swine liver was controlled using a three-dimensional (3D) ultrasound echo decorrelation imaging. Up to six ablations with target diameter 2 cm were done in each animal's liver using a clinical RFA system (RITA/Angiodynamics, 50 W, 6–10 min). Paired sequential volumes of beamformed pulse-echo data (inter-frame time <50 ms) were acquired from a Siemens Acuson SC2000 scanner with a Z6Ms transesophageal matrix array and transferred via an Ethernet to a computer running a custom MATLAB program to compute 3D echo decorrelation images. When the average cumulative, motion-compensated echo decorrelation within the planned ablation zone exceeded a prespecified threshold determined from preliminary trials, ablation was ceased automatically. After each procedure, the animal's liver was excised, uniformly sectioned, and optically scanned to reconstruct 3D ablation zones. Local ablation prediction was assessed using receiver operating characteristic (ROC) curve analysis comparing echo decorrelation images to co-registered ablation zones. To assess differences in outcomes, ablation zone volumes, ablation rates, Dice coefficients for measured versus targeted ablation zones, and ROC curves were statistically compared for controlled versus uncontrolled trials. The results indicate promise for control of *in vivo* thermal ablation using motion-corrected 3D echo decorrelation imaging.

**9:45**

**5aBA3. The impact of the central opening on nonlinear effects in ultrasound fields generated by Sonalleve V1 and V2 MR-HIFU systems.** Maria M. Karzova (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Wayne Kreider (CIMU/APL, Univ. of Washington, Seattle, WA), Ari Partanen (Profound Medical Inc., Mississauga, ON, Canada), Tatiana D. Khokhlova (Dept. of Medicine, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (CIMU/APL, Univ. of Washington, Seattle, WA), Petr V. Yuldashev (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Vera A. Khokhlova (CIMU/APL, Univ. of Washington, Seattle, WA 98105, verak2@uw.edu)

Previous studies have shown that the Sonalleve MR-HIFU clinical system can generate shocks of 80–105 MPa at the focus, which is sufficient for boiling histotripsy. Two versions of the system (V1 and V2) have therapeutic transducers that differ in focusing angle, arrangement of elements, and the size of a central opening. The goal here was to reveal the impact of different array geometries on shock amplitudes at the focus. Nonlinear modeling of the array's field in water using boundary conditions reconstructed from holography shows that at the same power output, the V2 array generates 10–15 MPa lower shock amplitudes at the focus. Although this difference is mainly caused by the smaller focusing angle of the V2 array, the larger central opening of the V2 array has a nontrivial impact. By suppressing a direct wave, the central opening produces a somewhat higher shock amplitude. Axisymmetric equivalent source models were constructed for both arrays and the importance of including the central opening was demonstrated. These models can be used in the "HIFU beam" software for simulating nonlinear fields of the Sonalleve V1 and V2 systems in water and flat-layered biological tissues. [Work supported by RSF-22-72-00047 and NIH R01EB025187.]

10:00

**5aBA4. Palpating particles using the acoustic radiation force: A new approach to magnetic particle imaging.** Kristen Zarcone (Biomedical Eng., Vanderbilt Univ., 1161 21st Ave. S AA-1105, Nashville, TN 37232, kristen.m.zarcone@vanderbilt.edu), Charles F. Caskey (Radiology, Vanderbilt Univ. Medical Ctr., Nashville, TN), and Will Grissom (Biomedical Eng., Vanderbilt Univ., Nashville, TN)

Magnetic particle imaging (MPI) is a tracer-based imaging technique that does not require radiation or a cyclotron. However, it does require bulky MRI-like hardware and does not produce a background tissue image. We propose a new approach to MPI that uses the acoustic radiation force to dynamically move iron nanoparticles through a magnetic field gradient, thereby changing their magnetic moment and producing a detectable signal in a receiver coil. An image could be formed by electronically steering the ultrasound focus to push particles at different spatial locations, like an ultrasonic palpation. This would enable a device that attaches to an ultrasound imaging transducer for simultaneous MPI and inherently registered ultrasound imaging. To test the feasibility of the technique, a cylindrical agar and graphite phantom was placed between a Maxwell coil pair which generated a linear field gradient across it, and particle signals were detected using a receiver solenoid. 2-ms ultrasound pulses were applied using a 6 MHz transducer to generate displacements along the magnetic field gradient. Signal measurements confirmed a lack of signal when iron nanoparticles were absent from the phantom, but the signal was detected when particles were present, with amplitudes that linearly increased with the gradient coil current.

10:15

**5aBA5. Characterizing the steering performance of a diagnostic-therapeutic ultrasound array using measured and synthesized holograms.** Randall P. Williams (Div. of Gastroenterology & Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., CIMU - Portage Bay Buiilding, Seattle, WA 98105, rpw1@uw.edu), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Fedor A. Nartov (Faculty of Phys., M.V.Lomonosov Moscow State Univ., Moscow, Russian Federation), Tatiana D. Khokhlova (Div. of Gastroenterology & Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), Maria M. Karzova, Petr V. Yuldashev (Faculty of Phys., M.V.Lomonosov Moscow State Univ., Moscow, Russian Federation), Vera A. Khokhlova (Univ. of Washington and M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Oleg A. Sapozhnikov (Univ. of Washington and M.V. Lomonosov Moscow State Univ., Seattle, WA)

Acoustic holography is frequently used for the characterization of fields generated by ultrasound transducers. For arrays with electronic focusing and beam steering, separate holograms are needed to represent the 3D field for each combination of array parameters. In this paper, we show that holograms for the field generated by each individual element in an ultrasound array can be recorded in a single hydrophone scan, by interleaving a series of excitations at each measurement location in which each element is driven separately. Such holograms were recorded for a 64-element diagnostic-therapeutic linear array with a nominal focal length of 50 mm and pitch of 0.8 mm, which is smaller than the wavelength at the operating frequency of 1.05 MHz. The resulting holograms reveal aspects of crosstalk and variations between array elements. By summing the elemental holograms with the appropriate phase shifts, holograms are synthesized representing source conditions for cases where the entire array is used to generate different beam configurations, with focal lengths between 35 mm and 60 mm, and steering angles of up to 20°. These synthetic holograms are shown to accurately characterize non-ideal steering behavior of the transducer, confirmed through direct hydrophone measurements. [Work supported by NIH grants R01EB023910 and R01EB025187.]

10:30–10:45 Break

10:45

**5aBA6. The use of acoustic holography for simultaneous characterization of various focus steering configurations in ultrasound fields generated by multi-element phased arrays.** Vera A. Khokhlova (Faculty of Phys., M.V. Lomonosov Moscow State University, Moscow 119991, Russian Federation, va.khokhlova@gmail.com), Dmitry A. Nikolaev, Sergey A. Tsysar, Azamat Z. Kaloev (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Wayne Kreider (CIMU/APL, Univ. of Washington, Seattle, WA), and Oleg A. Sapozhnikov (Faculty of Phys., M.V. Lomonosov Moscow State University, Moscow, Russia)

Acoustic holography has been successfully applied for accurate characterization of ultrasound sources. However, its direct use for multi-element phased arrays with many possible field configurations is not feasible. Here, two methods based on transient acoustic holography are proposed allowing reconstruction of all field configurations over a range of frequencies. The methods were tested using a 2 MHz piezocomposite 12-element annular array. First, the holography measurement was performed with all array elements excited in phase. After field reconstruction on the array surface, each element was separated spatially. Second, the array elements were excited individually in sequence for each measurement location. After separating the measured signals in time, 12 separate holograms were obtained to representing each array element individually. For both methods, the array performance was predicted by numerically defining the phase of each element. To evaluate these predictions, a hologram for a specific case of focus steering toward the array was measured independently. Although both characterization methods captured the array performance with focus steering, the approach that extracts elements from time sequences provided higher accuracy and thus can be used for sources with stronger electrical and mechanical crosstalk between the elements. [Work supported by RSF-20-12-00145 and NIH R01EB025187.]

11:00

**5aBA7. A pipeline to enable large-scale generation of diverse 2D cardiac synthetic ultrasound recordings corresponding to healthy and heart failure virtual patients.** Nitin Burman (Dept. of Cardiovascular Sci., Katholieke Universiteit Leuven, UZ Herestraat 49 - box 7003, Leuven 3000, Belgium, nitin.burman@kuleuven.be), Claudia Manetti (Faculty of Health, Medicine and Life Sci., Maastricht Univ., Maastricht, Netherlands), Paulo Tostes (Dept. of Cardiovascular Sci., Katholieke Universiteit Leuven, Leuven, Belgium), Joost Lumens (Faculty of Health, Medicine and Life Sci., Maastricht Univ., Maastricht, Netherlands), and Jan D'hooge (Dept. of Cardiovascular Sci., Katholieke Universiteit Leuven, Leuven, Belgium)

Simulated ultrasound (US) data are widely used in echocardiography to develop and validate rapidly growing convolutional neural networks (CNNs) based learning algorithms for image processing and analysis. In this context, a large and diverse database of synthetic US scans is considered vital for CNN training purposes, as clinical US data are scarce and difficult to access. Major hurdles in creating an extensive database are the long US simulation time and unstable heart models for extreme parameter settings. Here, we developed and implemented a cardiac US simulation pipeline that kinematically connects two state-of-the-art solutions in the field of US simulation (COLE) and cardiac modelling (CircAdapt), benefiting from the fast simulation time of the convolution-based ultrasound simulator and stability of the mechanical heart model to produce 2D synthetic cardiac US recordings. Furthermore, using our pipeline, we generated diverse set of 600 2D synthetic cardiac US recordings of healthy and heart failure virtual patients with variations in the shapes, motion patterns, and functions of the heart, along with their ground truth 2D myocardial velocity profiles and deformation curves. The resulting database is a potential tool for augmenting training databases of machine learning based US image processing algorithms. [Work funded by European Union's Horizon 2020 research and innovation programme under the Marie Sklodowska-Curie grant agreement No. 860745.]

**11:15**

**5aBA8. Contrast-enhanced ultrasound to detect active bleeding.** Scott J. Schoen (Radiology, Harvard Med. School and Massachusetts General Hospital, 101 Merrimac St., Boston, MA 02114, scottschoenjr@gatech.edu), Viksit Kumar (Radiology, Harvard Med. School and Massachusetts General Hospital, Boston, MA), Brian Telfer, Laura Brattain (MIT Lincoln Lab., Lexington, MA), and Anthony E. Samir (Radiology, Harvard Med. School and Massachusetts General Hospital, Boston, MA)

Non-compressible internal hemorrhage (NCIH) is the most common cause of death in acute non-penetrating trauma. NCIH management requires accurate hematoma localization and evaluation for ongoing bleeding for risk stratification. The current standard point-of-care diagnostic tool, the focused assessment with sonography for trauma (FAST), detects free fluid in body cavities with conventional B-mode imaging. The FAST does not assess whether bleeding is ongoing, at which location(s), and to what extent. Here, we propose contrast-enhanced ultrasound (CEUS) techniques to better identify, localize, and quantify hemorrhage. We designed and fabricated a custom hemorrhage-mimicking phantom, comprising a perforated vessel and cavity to simulate active bleeding. Lumason contrast agents (UCAs) were introduced at clinically relevant concentrations ($3.5\times10^8$ bubbles/ml). Conventional and contrast pulse sequence images were captured, and post-processed with bubble localization techniques (SVD clutter filter and bubble localization). The results showed contrast pulse sequences enabled a 2.2-fold increase in the number of microbubbles detected compared with conventional CEUS imaging, over a range of flow rates, concentrations, and localization processing parameters. Additionally, particle velocimetry enabled mapping of dynamic flow within the simulated bleeding site. Our findings indicate that CEUS combined with advanced image processing may enhance visualization of hemodynamics and improve non-invasive, real-time detection of active bleeding.

**11:30**

**5aBA9. Effect of acoustic output on fetal ultrasound color Doppler performance.** Matthew Huber (Biomedical Eng., Duke Univ., 101 Sci. Dr., Campus Box 90281, Durham, NC 27705, matthew.huber@duke.edu), David Bradway, Katelyn Flint, and Gregg Trahey (Biomedical Eng., Duke Univ., Durham, NC)

Increasing transmit acoustic output in color Doppler ultrasound imaging improves the received data temporal signal-to-noise ratio (SNR), but such elevated acoustic pressures increase the potential for inertial cavitation and tissue heating. Safety regulations constrain maximum acoustic output to mitigate bioeffects, but lower output is encouraged following the ALARA (As Low As Reasonably Achievable) principle. In this study, color Doppler imaging with a Siemens ACUSON Sequoia and 10L4 transducer was performed on the placenta umbilical cord insertion site of 10 pregnant volunteers. Flow data were collected at five acoustic output levels and four frequency settings to assess each parameter's effect. The output Thermal Index (TI) ranged from 0.001 to 0.7 and frequency ranged from 4.0 to 7.3 MHz. Increased temporal SNR in the receive signals at high output levels was linked to lower pulse-to-pulse phase variance and velocity bias. These metrics changed most dramatically across low output levels, before converging towards constant phase variance and velocity bias at high temporal SNR. The output levels approaching stable phase variance and velocity were specific to each volunteer and frequency. This result provides a framework for future automated Doppler ALARA implementations that calculate performance metrics and adjust acoustic power in accordance with ALARA principles.

**11:45**

**5aBA10. Exploring the benefits of spatial and temporal block-wise filtering architectures.** Abbie Weeks (Biomedical Eng., Vanderbilt Univ., 2201 West End Ave., Nashville, TN 37235, abbie.e.weeks@vanderbilt.edu) and Brett Byram (Biomedical Eng., Vanderbilt Univ., Nashville, TN)

Spatial block-wise filtering architectures improve noise stationarity locally, making SVD-based filtering more robust (Song *et al.*, 2017). Temporal block-wise filtering has been proposed to augment spatial block-wise methods for 3D Doppler data acquired via mechanical translation (Chen *et al.*, 2021) and compensate for bulk motion in cardiac power Doppler imaging (Zhang et al., 2021). We combine spatial and temporal block-wise architectures to suppress noise through depth and better capture time-varying signals with power Doppler. We show that a combination of spatial and temporal block-wise methods provides improved CNR (3.9 dB gain), SNR (0.9 dB gain), and CR (11.7 dB gain) over spatial block-wise alone and propose two methods for selecting temporal block sizes. First, we select temporal ensembles from systole, early-, and late-diastole from matched ECG data. Second, we derive a temporal SVD cutoff between the blood and noise subspaces for the entire ensemble. We use that cutoff as a proxy for correlation length and select ensembles accordingly. We collected 2 s of liver data from a healthy volunteer (nine angled plane waves at 600 Hz PRF and 4.1667 MHz center frequency) and applied spatial and temporal block-wise SVD filtering with 100-sample ensembles and 50-sample temporal block sizes overlapped at 20%.

## Session 5aPA

# Physical Acoustics and Biomedical Acoustics: Ultrasonic Assessment of Properties in Complex Materials I

Andrea P. Arguelles, Cochair
*Engineering Science and Mechanics, Penn State University, 212 Earth-Engineering Sciences Building, University Park, PA 16802*

Marie Muller, Cochair
*MAE, North Carolina State University, 911 Oval Drive, Engineering Building III, Raleigh, NC 27606*

**Chair's Introduction—8:00**

### *Invited Paper*

**8:05**

**5aPA1. Ultrasonic characterization of distributed cracking damage in concrete using multiply scattered wave fields.** John Popovics (Civil and Environ. Eng., Univ. of Illinois, 205 N. Mathews St. MC-250, Urbana, IL 61801, johnpop@illinois.edu), Homin Song (Gachon Univ., Gachon, Korea (the Republic of)), and Steven Feldman (Imerys Performance Minerals, Johns Creek, GA)

This paper reports work to detect, visualize, and characterize alkali–silica reactivity cracking damage in concrete using multiply scattered ultrasonic wave fields. Numerical simulations and laboratory-scale experiments are performed to understand ultrasonic surface wave scattering caused by distributed cracks in concrete, distinguishing that caused by the internal aggregate network. The simulations and experimental results reveal that incident ultrasonic surface waves undergo complicated multiple scattering set up by distributed cracks. To extract the crack-induced multiply scattered ultrasonic wave fields, a frequency-wavenumber (f-k) domain signal filtering approach is proposed. The feasibility of the signal analysis approach is then established by a series of experiments on large steel reinforced concrete blocks with known and controlled cracking damage. The experimental results demonstrate that distributed cracks can be detected and visualized using the proposed ultrasonic signal analysis approach, even at very early stages of the damage process. In addition, we demonstrate that the progress of damage over time can be monitored, where the cracking damage extent is closely related to the extracted multiply scattered wave field energy.

### *Contributed Papers*

**8:30**

**5aPA2. Effect of bone degradation on axially transmitted low frequency ($<$500 kHz) ultrasonic guided waves.** Aubin A. Chaboty (PULÉTS, École de Technologie Supérieure, 1100 Rue Notre Dame O, Montréal, QC H3C1K3, Canada, aubin.chaboty.1@ens.etsmtl.ca), Vu-Hieu Nguyen, Salah Naili (Laboratoire Modélisation et Simulation Multiechelle, Université Paris-Est Créteil, Créteil, Val de Marne, France), Guillaume Haiat (Laboratoire Modélisation et Simulation Multiechelle, Ctr. National de la Recherche Scientifique, Créteil, Val de Marne, France), and Pierre Bélanger (PULÉTS, École de Technologie Supérieure, Montréal, QC, Canada)

The early diagnosis of osteoporosis through bone quality assessment has been extensively studied in the past few years. Research in axial transmission in long cortical bone using ultrasonic guided waves was shown to be sensitive to variations of the bone properties such as the geometry and the mechanical properties. The aim of this presentation is, therefore, to simulate a complex bone-like geometry using 3D finite elements (FE) in order to investigate the influence of the cortical thickness mechanical properties degradation. Three parameters were investigated: (1) the thickness of the periosteal region, (2) the degradation of the endosteal region, and (3) the position of the transition zone from periosteal to endosteal regions. Moreover, the influence of soft tissue was investigated on a plate-like structure using semi-analytical iso-geometric analysis method (SAIGA). Two cortical bone phantom plates with different mechanical properties covered with a soft tissue mimicking material were used to perform axial transmission measurement. The dispersion curves of the propagating modes were experimentally measured and then were compared to the solutions obtained with the SAIGA method. The results showed the possibility to retrieve the properties of the bone phantom plates through an inversion of the dispersion curves.

**8:45**

**5aPA3. Assessment of flaws in cold sintered ZnO via longitudinal ultrasonic wave speed and attenuation measurements.** Haley N. Jones (Mater. Sci. and Eng., Penn State Univ., N-225 Millenium Sci. Complex, State College, PA 16802, hnj5051@psu.edu), Susan Trolier-McKinstry (Mater. Sci. and Eng., Penn State Univ., University Park, PA), and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

The cold sintering process (CSP) is a low temperature processing technique that utilizes a transient phase to synthesize dense ceramics. However, some CSP parts contain microflaws that arise due to pressure gradients and inhomogeneities in temperature and distribution of the transient phase. This work uses high frequency ultrasound (20 MHz) to verify the presence of defects larger than 15 $\mu$m in effective radius in CSP ZnO samples of varying densities (84–97%). The acoustic data were compared to X-ray Computed Tomography (XCT) images to validate the findings. Acoustic metrics used in this work include wave speed, which is affected by differences in the effective elastic properties of the medium, and attenuation, which is wave energy loss due to scattering from defects, including those smaller than the

wavelength. Wave speed maps were inhomogeneous suggesting density gradients which were verified with SEM. Areas of high attenuation (> 300 Np/m) are present in all samples independent of relative density and correspond to defects identified in XCT ranging from 15 $\mu$m to 50 $\mu$m in effective radius. This suggests the presence of microflaws possibly due to the inhomogeneous removal of the transient phase. However, some high attenuation spots do not correspond to visible defects in XCT which suggests the presence of features such as texturing which are undetectable with XCT. These results show the viability of high frequency ultrasound for defect detection in cold sintered ZnO.

**9:00**

**5aPA4. Laser shock wave propagation in a 3D woven composite material.** Eduardo Cuenca, Mathieu Ducousso (Safran Tech, Magny les Hameaux, France), Laurent Berthe (PIMM, CNRS, Paris, France), and Francois Coulouvrat (CNRS, Sorbonne Universite - 4 Pl. Jussieu, Institut Jean Le Rond d'Alembert, Paris 75005, France, francois.coulouvrat@upmc.fr)

We investigate the propagation of a laser shock wave within a composite material used in aeronautics. This one is made of carbon fibers woven according to a 3D pattern within an epoxy matrix. The shock wave is generated by an impulsive (of order 10 ns) laser beam of power a few GW/cm$^2$, focused on the surface of a metallic layer attached to the composite. The metallic surface ablation results in an expanding plasma inducing a shock wave in the material with a peak stress of a few GPa. Its propagation can be measured by the velocity of the opposite surface of the sample detected by laser interferometry. The high frequency content of the signal and the heterogeneous and anisotropic behavior of the material induce a complex propagation. A measurement campaign has been performed for different samples. A 2D numerical model has been developed based on a finite difference discretisation of equations of linear elastodynamics for the propagation part, combined with a nonlinear model of the source. Sample geometry and fiber orientations are obtained from analysis of the images of transversal cuts of the samples. Experimental data are statistically compared to the model and discussed.

**9:15**

**5aPA5. Ultrasonic characterization of the microstructure for cold sintered sodium molybdate.** Christopher Wheatley (Mater. Sci. and Eng., Penn State Univ., 123 S. Osmond St., State College, PA 16801, cs.wheatleyjr@psu.edu), Javier Mena-Garcia, Clive Randall (Mater. Sci. and Eng., Penn State Univ., State College, PA), and Andrea P. Arguelles (Eng. Sci. and Mech., Penn State Univ., University Park, PA)

The Cold Sintering Process (CSP) is a novel manufacturing process that can create high-density metals, ceramics, and composite materials at significantly lower temperatures than conventional sintering. However, the microstructure of cold sintered materials can be more flawed and inhomogeneous than observed in conventionally sintered counterparts. Therefore, more information about the underlying densification mechanisms is required to reduce these flaws and promote CSP as an alternative industrial process. Ultrasonic testing provides a non-destructive approach to analyze the microstructure of these cold-sintered materials. In this presentation, longitudinal attenuation and wave speed are used to characterize sodium molybdate ($Na_2Mo_2O_7$), a dielectric material. The concentration of inhomogeneities in the microstructure is found to increase proportionally to the heating rate, and with a larger thickness relative to diameter ratio. The densification is ultimately dependent on balancing the kinetics of the transient liquid phase as it undergoes pressure-solution precipitation and evaporation. Finally, the processing parameters are adjusted using information from ultrasonic characterization to optimize for a homogeneous microstructure, showing CSP as a promising lower energy alternative for sintered materials.

**9:30–9:45 Break**

## *Invited Paper*

**9:45**

**5aPA6. Numerical and experimental analysis of ultrasonic scattering in two-phase polycrystalline materials.** Showmic Islam (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE), Michael Uchic ((AFRL/RXCA), Air Force Res. Lab., Wright-Patterson AFB, OH), and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE 68588, jaturner@unl.edu)

Many polycrystalline materials found in nature and used in industry consist of multiple phases. Careful nondestructive characterization of these materials is of fundamental importance for quality assurance because microstructure is strongly correlated with mechanical performance. A fundamental understanding of ultrasonic scattering from idealized two-phase materials would provide essential information regarding the predictive nature of scattering measurements to quantify such microstructures. For this purpose, samples were created using spark plasma sintering (SPS) from mixtures with different ratios (with respect to mass) of copper and iron powders. These samples were used for measurements of wave velocity, attenuation, and backscatter. Prior ultrasonic scattering models for continuous media were modified for discrete synthetic two-phase three-dimensional microstructures. In this way, the ultrasonic scattering responses were determined numerically using DREAM. 3D based on 30 realizations of synthetic microstructures representative of each volume fraction. In this presentation, the computational approach is described, and the results are then compared with the microstructures and ultrasonic data from the samples made from the binary mixtures. The results show some agreement as well as highlight some limitations of the computational model. Finally, prospects for more complex materials are discussed. [Research supported by the Air Force Research Laboratory (AFRL) under prime contract FA8650-15-D-5231.]

**10:10**

**5aPA7. Ultrasonic damage detection in lithium-ion batteries subjected to localized heating.** Tyler McGee (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712, tyler.m.mcgee@gmail.com), Barrett Neath (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Washington DC, DC), Samuel B. Matthews, Ofodike A. Ezekoye, and Michael R. Haberman (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

The transition from internal combustion to electrically powered vehicles (EVs) is accelerating, with some estimates predicting that 45% of new cars sold in the US will be fully electric by 2035. EVs will be powered by lithium-ion batteries (LIBs). This high-power application can subject LIBs to significant electrical, mechanical, and thermal abuse. Predicting and preventing thermal runaway (TR) is therefore of utmost importance to save lives and ameliorate the transition to renewable energy. Currently, battery management systems monitor cell voltage, current, temperature, and presence of gases, which does not provide sufficient advanced warning of a catastrophic event [J. Acoust. Soc. Am., **150**, A66 (2021)]. This work explores the viability of evaluating cell safety by monitoring and interpreting ultrasonic signals propagating through a battery as it undergoes localized heating. We monitor different portions of received waveforms, linking them to specific propagating modes or multi-path arrivals. A finite element model is then used to understand the mode of propagation of the chosen frequencies, and the methodology is applied to both 10Ah and 60Ah cells. The time-domain features of signal amplitude and time-of-flight are used to create safety metrics which warn of TR as much as 25 min in advance of failure.

**10:25**

**5aPA8. A model-driven approach to ultrasonic detection of state of charge in lithium-ion batteries.** Tyler McGee (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712, tyler.m.mcgee@gmail.com), Barrett Neath (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Washington DC, DC), Samuel B. Matthews, Ofodike A. Ezekoye (W), and Michael R. Haberman (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Ultrasonic inspection is one non-destructive method to monitor the internal state of lithium-ion battery (LIB) cells. During charging, lithium ions intercalate into the graphite anode, causing moderate volumetric expansion (approximately 10%) and as much as a three-fold increase in Young's modulus of the anode. Many researchers have observed changes in the time-of-flight (TOF) of through-thickness ultrasonic waves that correlate with changes in the state of charge (SOC). We introduce a transfer matrix method and Bloch-wave formalism for periodically layered media to show that the observed changes in TOF are partially due to the associated changes in anode stiffness. However, more detailed models that consider battery heterogeneity in structure and properties are needed to predict changes in TOF and signal amplitude (SA) for sophisticated ultrasonic measurement configurations and different cell chemistries. We therefore employ a quasi-static homogenization scheme for layered media to estimate the effective anisotropic stiffness of LIB pouch cells as a function of SOC as an input to a time-domain finite element model to determine the influence of changes in the anode, cathode, and separator on the direction-dependent propagation modes. This model is then employed to interpret experimentally obtained TOF and SA data for more robust SOC detection.

**10:40**

**5aPA9. Investigating the influence of smoothing approximations on analytical models of elastic waves in strongly scattering polycrystals.** Anubhav Roy (Eng. Sci. and Mech., The Penn State Univ., 212 Earth and Eng. Sci. Bldg, University Park, PA 16802, aroy_esm@psu.edu) and Christopher M. Kube (Eng. Sci. and Mech., The Penn State Univ., University Park, PA)

Ultrasonic scattering-based measurements are commonly used as sensitive tools for non-destructive evaluation (NDE) of metals. Existing analytical models can predict ultrasonic wavespeed and attenuation for statistically homogeneous metals exhibiting spatially dependent elastic constants. These models rely on two-point statistics that correspond to first-order smoothing approximation (FOSA) of the mass operator series present in the governing Dyson equation. The FOSA-based prediction agrees well with finite element (FE) results for a wide range of polycrystalline metals at ultrasonic frequencies in the stochastic regime. However, discrepancies become more pronounced with increasing single-crystal elastic anisotropy even at low frequencies in the Rayleigh limit. For example, a difference of 70% is present for Lithium. In this presentation, the next order or third-order smoothing approximation (TOSA) is introduced, and Rayleigh-limit attenuations and wavespeeds are derived analytically. In this context, the TOSA accounts for four-point correlation functions and, thus, a higher degree of multiple scattering effects are included. This presentation will focus on key points and assumptions pertaining to the derivation for results including the TOSA. The TOSA-based model improves predictions of longitudinal attenuation by more than 11% and 4% for polycrystalline Lithium or Nickel, respectively. Future extensions to higher frequencies and multiphase polycrystals will be described.

**10:55**

**5aPA10. Simulation study of the sensitivity of mechanical parameters on the reflected signal at the second interface of human cancellous bone—Application of Biot theory.** Mustapha Sadouki (Acoust. and Civil Eng. Lab., Khemis-Miliana Univ., Rte. Thenia el Had, Khemis-Miliana 44225, Algeria, mustapha.sadouki@univ-dbkm.dz)

A simulation study of mechanical parameters sensitivity to reflected ultrasound waves at the second interface of a hypothetical human cancellous bone sample is proposed. The sample is considered as a biphasic porous medium saturated by a fluid. The visco-inertial exchanges between the structure and the saturating fluid are described by the Biot theory modified by the Johnson model. An analytical expression of the reflection coefficient at the second interface is obtained in the frequency domain. This expression depends on three physical parameters, which are the porosity, the tortuosity, and the viscous characteristic length, as well as on mechanical parameters, Young's modulus and Poisson's ratio of the solid and the skeletal frame of the medium. The reflected signal is calculated in the frequency domain by the product of the spectrum of the incident signal by the reflection coefficient. In the time domain, the reflected wave is obtained by taking the inverse Fourier transform of the reflected frequency signal. The effect of mechanical parameters on the reflected signal inside the medium is studied. The results obtained are discussed and compared to those given in the literature.

# Session 5aSC

## Speech Communication: Speech Production (Poster Session)

Daniel Aalto, Chair
*Communication Sciences and Disorders, University of Alberta, 8205 114 St NW, Edmonton, AB T6G2G4, Canada*

All posters will be on display from 8:00 a.m. to 12:00 noon. Authors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

### *Contributed Papers*

**5aSC1. Evaluation of glottal inverse filtering in the presence of source-filter interaction.** Zhaoyan Zhang (Dept. of Head and Neck Surgery, Univ. of California, Los Angeles, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu) and Joyce Lin (Dept. of Head and Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA)

Glottal inverse filtering is often used to recover the glottal flow waveform from the produced voice outcomes. While there have been many studies aiming to evaluate its validity, few studies investigated its validity in voice conditions with strong source-filter interaction. The goal of this study is to evaluate the performance of glottal inverse filtering in conditions of epilaryngeal tube narrowing and/or reduced lip opening, both of which are expected to enhance source-filter interaction. Computational simulation of voice production was performed using a three-dimensional vocal fold model coupled to a vocal tract with varying degrees of constriction at the epilarynx and lips. Different algorithms of inverse filtering were then used to estimate the glottal airflow waveform from either the produced voice or airflow at the lips, and compared to the true glottal airflow from the simulation. The estimation errors in the glottal flow waveform and selected glottal flow-based measures will be evaluated as a function of the epilaryngeal and lip openings.

**5aSC2. Palatalization of /d/ across word boundaries in UK English.** Sara E. Miller (Linguist, Univ. of Georgia, 145 Gilbert Hall, 210 Herty Dr., Athens, GA 30602, betsy.miller@uga.edu), Austin Brailey-Jones, and Margaret E. Renwick (Linguist, Univ. of Georgia, Athens, GA)

In many spoken varieties of English, the coronal stop /d/ may palatalize or affricate before /j/, including at word boundaries; e.g., “did you” [ˈdɪdʒu], “would you” [ˈwʊdʒu]. We study this process in the large, naturalistic Audio BNC (http://www.phon.ox.ac.uk/AudioBNC). Previous studies have found that coronal fricatives /s, z/ palatalize preceding /j/ in predictable contexts and at increased speech rates, implying that palatalization results from coarticulatory gestural overlap. However, analysis of /t/ shows that palatalization to [tʃ] is a categorical stylistic variant, which is instead selected in formal contexts at slower speech rates. To determine whether [dʒ] results from gestural overlap or as a stylistic variant, we analyze /d/’s rate of palatalization using approximately 8000 tokens of /d#j/ gathered from the force-aligned Audio BNC. Tokens were impressionistically coded as palatalized to [dʒ], released as [d], or unreleased/deleted. With multinomial logistic regressioni, we test whether effects of lexical frequency, speech rate and duration, phonological context, discourse, or speaker characteristics (e.g., region, gender) predict the realization of /d#j/ in naturalistic speech. In line with previous findings, we hypothesize that [dʒ] is more likely in frequent collocations and at high speech rates, but style-driven effects of region, formality, or gender may also predict this variant.

**5aSC3. Vowel trajectories of African Americans in Georgia, USA.** Margaret E. Renwick (Linguist, Univ. of Georgia, 240 Gilbert Hall, 210 Herty Dr., Athens, GA 30602, mrenwick@uga.edu), Jon Forrest (Linguist, Univ. of Georgia, Athens, GA), Lelia Glass (School of Modern Lang., Georgia Inst. of Technol., Atlanta, GA), and Joey Stanley (Linguist, Brigham Young Univ., Provo, UT)

Within the United States, dialectal variation is often characterized by vowel shifts: systematic differences in vowels’ relative qualities and vowel-inherent dynamics. The African American Vowel Shift (AAVS), in particular, includes raising and fronting of front lax /ɪ ɛ æ/, among other features. The more recent pan-regional Low Back Merger Shift (LBMS), by contrast, includes lowering and backing of the same vowels. We evaluate these two shifts in an audio corpus of over 40 Black speakers from the Southern state of Georgia. Speakers, born between the 1930s and 2000, represent five demographic generations. Normalized formant values (F1,F2) from five temporal points per token are input to Generalized Additive Mixed Models (GAMMs). We test for significant changes in vowels’ trajectories across time by fitting Year of Birth as a continuous smooth term. Additionally, we use linear mixed-effects modeling to test for raising versus lowering on the (F2–F1) front-vowel diagonal, across generations. Evidence from GAMMs and linear modeling indicates raised positions of /ɪ ɛ æ/ among older generations (1950s–1980s), followed by significant retraction from 1990–2000. These acoustic results are consistent with strengthening of the AAVS in the third quarter of the 20th Century, followed by a rapid transition to the pan-regional LBMS.

**5aSC4. Aerodynamic and acoustic characterisitics of glottal closure insufficiency.** Stefan Kniesburges (Phoniatrics & Pediatric Audiol., Univ. Hospital Erlangen, Waldstrasse 1, Erlangen, Bavaria 91058, Germany, stefan.kniesburges@uk-erlangen.de), Sebastian Falk, Bernhard Jakubaß (Phoniatrics & Pediatric Audiol., Univ. Hospital Erlangen, Erlangen, Bavaria, Germany), Paul Maurerlehner, Stefan Schoder, Manfred Kaltenbacher (Inst. of Fundamentals and Theory in Elec. Eng., Tech. Univ. Graz, Graz, Austria), Matthias Echternach (Dept. of Otorhinolaryngology, Div. of Phoniatrics & Pediatric Audiol., Hospital of the Ludwig-Maximilian Univ. München, München, Bavaria, Germany), Anne Schützenberger, and Michael Döllinger (Phoniatrics & Pediatric Audiol., Univ. Hospital Erlangen, Erlangen, Bavaria, Germany)

Glottal closure insufficiency (GCI) and asymmetric oscillations of the vocal folds are often responsible for low voice quality and increased phonation effort for concerned patients. In this study, four clinically observed types of GCI with increasing glottal closure insufficiency were reproduced in the aero-acoustic computer model *simVoice*. Each type was simulated with symmetric and asymmetric oscillations extracted from high-speed

recordings obtained in *ex-vivo* porcine larynges. The results show a permanent glottal flow throughout the oscillation cycle and an increasing mean flow rate for increasing minimum glottal gap. Consequently, the aerodynamic work exerted onto the vocal folds decreases for an increasing gap corresponding to the reported patient's effort increase. Moreover, the quality of the computed sound decreases with an increasing gap based on the cepstral-peak-prominence. Even worse, asymmetric oscillations further degrade both, the aerodynamic energy transfer and the acoustic quality. The temporal evolution of energy transfer shows that the main decay occurs during the opening of the glottis. This shows that the blockage of the flow during glottis closure and the resulting large temporal gradients in pressure and velocity in the larynx are the key features for an efficient phonation process which is also visible in the aero-acoustic sound sources.

**5aSC5. Syllable complexity in 24-month-old infants resulting from spontaneous mother–child vocal interactions.** Farheen A. Ahmed (Hearing and Speech Sci., Univ. of Maryland, College Park, 7251 Preinkert Dr., College Park, MD 20742, farheen6055@gmail.com), Margaret Cychosz, and Rochelle S. Newman (Hearing & Speech Sci., Univ. of Maryland, College Park, MD)

Infant Directed Speech (IDS) influences language development via caregiver–child interactions and can promote early speech and language outcomes in some cultures (Golinkoff *et al*., 2015). IDS is characterized by a higher, more dynamic fundamental frequency (f0) as well as shorter caregiver utterances and simplified phono-lexical structures (Ramirez-Esparza *et al*., 2014). In line with social feedback theories of early vocal development (Warlaumont *et al*. 2014), this study proposes that certain acoustic aspects of IDS, when produced contingently with infant speech, will promote infants' early vocal maturity. N = 84 infant-caregiver dyads were recorded during semi-naturalistic play sessions at 7, 10, 18, and 24 months. Infant vocal maturity was quantified as syllable complexity at 24-months where syllables with consonant-vowel (CV or VC) transitions were more complex than syllables with only a nucleus (V). The complexity of children's syllables produced within 5 sof contingent caregiver speech will be compared to those produced outside of 5-swindows of caregiver speech. Infants' syllable production will also be related to acoustic characteristics of IDS including f0 height, modulation, and caregiver speaking rate. Positive correlations between IDS characteristics and infant syllable complexity will provide additional evidence that socially contingent IDS spurs early speech development.

**5aSC6. The role of voice fundamental frequency in the perception of anger in clear speech.** 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), Sierra N. Bennion, Tara E. Smalley, and Elizabeth D. Young (Commun. Sci. and Disord., Univ. of Utah, Salt Lake City, UT)

In previous work, listeners rated clearly spoken neutral speech materials as sounding angry significantly more often than identical materials spoken conversationally. This effect varies widely among talkers, and a recent study compared acoustic characteristics of clear and conversational speech produced by talkers whose clear speech sounds angry to those of talkers whose clear speech does not sound angry. Of several well-known clear speech acoustic changes, only raised voice fundamental frequency ($f_o$) differed for talkers who do and do not sound angry when speaking clearly. To test whether raised $f_o$ causes clear speech to sound angry, the present study used six talkers whose clear speech was rated as sounding angry and who also made sizeable $f_o$ shifts when speaking clearly. For each talker, nine pairs of identical sentences (one from each speaking style) were identified. Praat was then used to lower the pitch of the clear sentence to match the pitch of the conversational sentence and to raise the pitch of the conversational sentence to match the pitch of the clear sentence. Listeners with normal hearing will hear these sentences and rate the emotion they heard in each item from a set of six emotions.

**5aSC7. Vowel duration before voiced and voiceless stops in the Indian English of Assamese speakers.** Caroline Wiltshire (Linguist, Univ. of Florida, Box 115454, Gainesville, FL 32611-5454, wiltshir@ufl.edu) and Priyankoo Sarmah (Indian Inst. of Technol. Guwahati, Guwahati, Assam, India)

Speakers of many English varieties produce longer vowels before voiced obstruent codas than before voiceless (e.g., House and Fairbanks, 1953; Chen, 1970; Umeda, 1975; Van Santen, 1992), though the extent can vary (Jacewicz *et al*., 2007; Tanner *et al*., 2019; Deterding, 2005). While phonetic motivations have been proposed to explain this pattern (Ohala, 1983; Kluender *et al*., 1988), the effect in some varieties of English is observed to be larger than phonetically motivated (Ohala, 1983; Keating, 1985); furthermore, some languages have less of a voicing effect (e.g., Korean, Russian, and French in Chen, 1970) and others may have none at all (e.g., Polish and Czech in Keating, 1985). This study investigates the phenomenon in the second largest variety of English by population, Indian English. Twenty Indian English speakers, all with L1 Assamese, were recorded reading a word list including 40 monosyllables that place vowels before voiced and voiceless versions of the same stop. We measure vowel duration using a Praat script to determine whether and to what extent this variety shares in the lengthening pattern, and in addition to providing the first measures of the phenomenon in Indian English, our results will be compared with other varieties of English and other languages.

**5aSC8. Competing prosodic influences on VOT length in Hawaiian.** Lisa Davidson (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, lisa.davidson@nyu.edu) and 'Ōiwi Parker Jones (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Previous research on voice onset time shows that VOT in unaspirated stops is at best only weakly affected by prosodic strengthening (Hodgson, 2021; Simonet *et al*., 2014). VOT duration is analyzed in Hawaiian, which has only /p k/, in conjunction with a prosodic analysis, to determine its stop type, and investigate whether prosodic boundaries condition VOT length. Data comprise speech from seven speakers interviewed on the radio program Ka Leo Hawai'i in the 1970–80s (Kettig, 2021). Using the computational prosodic grammar in Parker Jones (2010), stops were coded for whether they occurred in lexical word-initial/medial position, in a stressed/unstressed syllable, and in prosodic word-initial/medial position. The results indicate that the average duration of /p/ is 24ms and /k/ is 41ms, the latter being on the higher end of unaspirated stop ranges (Cho and Ladefoged, 1999). VOT is significantly longer before stressed word-medial syllables, but shorter before stressed word-initial syllables. For word-medial stressed syllables, VOT is also significantly longer when the syllable is in prosodic-word initial position. VOT length may have separate functions in Hawaiian unaspirated stops: strengthen a lexical word boundary especially before an unstressed syllable via shortening, but lengthen word-medially to demarcate a prosodic word boundary.

**5aSC9. Formant frequency variability in English schwa-initial minimal pairs.** Emily R. Napoli (Linguist, The Ohio State Univ., 1712 Niel Ave. Columbus, OH 43210, napoli.44@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

The label "schwa" has traditionally been used to refer to a mid-central vowel produced with a neutral vocal tract. However, "schwa" has also been used by linguists to represent a wide range of sounds cross-linguistically, which show considerable variation depending on their context. Previous research on schwa quality has focused primarily on consonantal factors that impact the phonetic realization of schwa. One understudied factor is whether or not the schwa is a function word (as in *a company*) or the initial syllable of a content word (as in *accompany*). The current production study aims to look at both ambiguous (e.g., accompany and a company) and unambiguous (e.g., *accomplish* and *a comic*) schwa-initial pairs in neutral sentences, which, up to a critical point, can be read as having either schwa variant, and biasing sentences, which are contextually biased towards one of the schwa variants. Preliminary analyses reveal that F1 at the midpoint of the schwa is not affected by schwa type, ambiguity, or sentence bias but is determined primarily by schwa duration. This result suggests that variability in schwa quality may be a result of its short duration relative to other vowels, reflecting articulatory undershoot.

**5aSC10. The tonal system of Hengyang Xiang.** Yaqian Huang (Univ. of California San Diego, Dept. of Linguist, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, yah101@ucsd.edu)

Hengyang belongs to the Hengzhou branch of Xiang Chinese; it contrasts six tones: high rising (45), falling-rising (213), low-rising (24), low level (11), mid level (33), and low-mid (22). Liu (2010) found that the falling-rising (213) and the low-rising (24) tones have merged to become a distinct low-dipping tone (214), as in Mandarin. This merger is attributed to language contact and influence from Mandarin, because it is mostly observed in frequent words and has a stronger tendency among younger populations (Liu, 2010). However, it is unclear whether this tone merger is complete, because previous reports conflict as to the number of tones in the tonal system of Hengyang (Zhong, 2011; Li, 1986; Huang, 2013). This study investigates the tonal system in Hengyang using acoustic analyses to capture the tonal differences in pitch and voice quality characteristics including spectral tilt and periodicity measures. We recorded ten participants reading monosyllabic words. The participants were from four age groups ranging from 21 to 60 years old (21–30, 31–40, 41–50, and 51–60 years old). We hypothesize that the tone merger occurs more often in frequently used words and for younger speakers. The findings will depict the acoustic spaces of Hengyang tones and show whether the merged tonal patterns are associated with particular words and/or age groups. The discussion will emphasize how language contact gives rise to tonal neutralization and sound change.

**5aSC11. Voice quality as an indicator of speaker's social information and as a cue for sound change.** Xin Gao (Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104, kauhsin@sas.upenn.edu) and Rui Wen (Shanghai Univ., Shanghai, China)

Voice quality reflects a variety of information in speech. The present study examines how voice quality may indicate a speaker's social information. We examine the ongoing /yø∼y/ merger in Shanghainese Wu. The first vowel formant (F1) and Harmonics-to-noise ratio (HNR) are extracted from 80 speakers. We also collect social information including age, gender, and immigrant background, family language, and working language. Mixed-effects linear regression models are built with the F1/HNR difference between /yø∼y/ as the dependent variable. Significant age effects are found on both F1 ($\beta = 0.043$, $p = 0.006$) and HNR ($\beta = -0.055$, $p = 0.024$); F1 is also more merged for female speakers than males ($\beta = -0.521$, $p = 0.001$); HNR is more distinct for speakers from non-immigrant families than speakers from immigrant families ($\beta = 1.858$, $p = 0.008$). The study shows a tendency for /yø∼y/ to merge in terms of both vowel quality and voice quality. It is noteworthy that speakers of native origin participate in the vowel-quality merging (cued by F1) but not in the voice-quality merging (cued by HNR). Taken together, voice quality can function as an independent indicator of a speaker's social attributes.

**5aSC12. Tense voice without the high f0: the case of glottalized vowels in Zongozotla Totonac.** Hayley Dawson (UC San Diego, 9500 Gilman Dr. #0108, La Jolla, CA, hdawson@ucsd.edu), Marc Garellek (UC San Diego, La Jolla, CA), Osbel López-Francisco (Universidad Nacional Autónoma de México, Iztacala, Zongozotla, Mexico), and Jonathan Amith (Gettysburg College, Gettysburg, PA)

Prototypical creaky voice involves increased glottal constriction, lower periodicity, and lower f0. Keating *et al.* (2015, Proc. ICPhS) argue that other kinds of creaky voice manifest only some of these properties. For example, "tense voice" is constricted but has neither a low nor irregular f0. Although tense voice is usually produced with a high f0, the aforementioned authors suggest that tense voice can also occur with non-high f0. In this study, we argue that glottalized (also known as "laryngealized";) vowels of Zongozotla Totonac qualify as such: they are produced with tense voice but with non-high f0. Zongozotla Totonac is a Totonac-Tepehua language spoken in the municipality of Zongozotla, Puebla, Mexico. We recorded eight speakers producing nine word pairs that contrast in terms of modal versus glottalized vowels. Compared to modal vowels, glottalized vowels show increased constriction, as indexed by lower spectral tilt and weaker voicing. However, glottalized and modal vowels do not differ in terms of f0 or periodicity. Taken together, the results suggest that glottalized vowels are produced with tense voice, for which the increase in constriction is criterial. This analysis has implications for understanding sound change in Totonac as well as for the taxonomy of creaky voice subtypes.

**5aSC13. Perceiving the intersection of gender identity, sexual orientation, and gender expression in American English.** Benjamin Lang (Linguist, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, blang@ucsd.edu)

Listeners make assumptions about speaker gender identity (GI), sexual orientation (SO), or gender expression (GE) from speech (Tripp and Munson, 2021). Researchers have identified correlations between acoustic cues such as center of gravity (COG) of /s/ and perceived speaker's identity. In this study, I investigate how a multitude of acoustic features combine to influence nuanced perception of GI, SO, and GE in speech. In an online study, 197 listeners rated the speech of 66 speakers of American English for perceived GI, SO, and GE. Acoustic measures for f0, vowel formants and dispersion, segment durations, fricative spectra, and creaky voice were correlated with these judgments using analyses such as random forests and PCA. The results indicate that ensembles of acoustic features, including well-known features such as f0 and COG of /s/ but also creaky voice and diphthong formants, contribute to the perception of GI, SO, and GE such that common perceived social categories emerge. I discuss the complex interactions of acoustic cues that describe the ways in which listeners construct a speaker's identity along traditional binaries such masculine or feminine as well as outside those binaries for queer identities.

**5aSC14. Development of vowel acoustics and subglottal resonances in American English-speaking children: A longitudinal Study.** Vishwas Shetty (Elec. Eng., Univ. of California at Los Angeles, 66-147E Engr. IV, 420 Westwood Plaza, Los Angeles, CA 90095, shettyvishwas@ucla.edu), Steven M. Lulich (S), Pertti Palo (Speech, Lang. and Hearing Sci., Indiana Univ., Bloomington, IN), and Abeer Alwan (Elec. Eng., Univ. of California at Los Angeles, Los Angeles, CA)

Acoustic analysis of typically developing elementary school-aged (prepubertal) children's speech has been primarily performed on cross-sectional data in the past. Few studies have examined longitudinal data in this age group. For this presentation, we analyze the developmental changes in the acoustic properties of children's speech using data collected longitudinally over four years (from first grade to fourth grade). Four male and four female children participated in this study. Data were collected once every year for each child. Using these data, we measured the four-year development of subglottal acoustics (first two subglottal resonances) and vowel acoustics (first four formants and fundamental frequency). Subglottal acoustic measurements are relatively independent of context, and average values were obtained for each child in each year. Vowel acoustics measurements were made for seven vowels (i, ɪ, ɛ, æ, ʌ, ɑ, u), each occurring in two different words in the stressed syllable. We investigated the correlations between the children's subglottal acoustics, vowel acoustics, and growth-related variables such as standing height, sitting height, and chronological age. Gender-, vowel-, and child-specific analyses were carried out in order to shed light on how typically developing speech acoustics depend on such variables. [Work supported, in part, by the NSF.]

**5aSC15. Coarticulation is reduced in clear speech produced with protective face masks.** Zhe-chen Guo (Linguist, The Univ. of Texas at Austin, 307 E 31st ST APT 105, Austin, TX 78705, y9024131@gmail.com) and Rajka Smiljanic (Univ. of Texas at Austin, Austin, TX)

Talkers dynamically modify their coarticulatory patterns when producing listener-oriented hyperarticulated clear speeches. This study examined

how the use of protective face masks interacts with the production of intelligibility-enhancing clear speech to impact coarticulation. A native and a non-native speaker of English read sentences in a clear and conversational speaking style with and without a surgical mask. Coarticulation between word-internal adjacent segments was analyzed with a whole-spectrum analysis including spectral distance and segment overlap duration. Both speakers coarticulated less in clear than in conversational speaking style as indicated by the larger spectral distance and shorter overlap duration between adjacent segments. Coarticulation was further reduced when clear speech was produced with a mask by the native speaker but not by the non-native speaker. The findings showed that producing hyperarticulated intelligibility-enhancing clear speech also involves reducing coarticulatory overlap across adjacent segments. Coarticulatory resistance was adaptively reinforced in the presence of the additional communicative barrier, face mask, particularly for the speaker with extensive experience with the target language. Such cumulative reduction of coarticulation may in part underlie the larger perception-in-noise benefit for clear speech produced with a mask for the native compared to the non-native talker.

**5aSC16. Acoustic-based automatic speech intelligibility scoring using deep neural networks.** Nikita B. Emberi (Information Technol., Vidyalankar Inst. of Technol., Mumbai, Maharashtra, India), Tyler T. Schnoor (Linguist, Univ. of Alberta, 150 Assiniboia Hall, Edmonton, AB T6G2E7, Canada, tschnoor@ualberta.ca), Richard A. Wright (Linguist, Univ. of Washington, Seattle, WA), and Benjamin V. Tucker (Linguist, Univ. of Alberta, Edmonton, AB, Canada)

Human-generated measures of speech intelligibility are time-intensive methods for assessing the intelligibility of speech. The purpose of the present study is to automate the assessment of speech intelligibility by developing a deep neural network that estimates a standardized intelligibility score based on acoustic input. We extracted Mel-frequency cepstral coefficients from the UW/NU IEEE sentence corpus which had been manipulated with three signal-to-noise ratios (−2, 0, 2 dB). We obtained listener transcriptions from the UAW speech intelligibility dataset and calculated the Levenshtein distance between the transcriptions and the speaker's prompt. The neural network was trained to predict the Levenshtein distance given MFCC representations of sentences. We use tenfold cross-validation to verify the accuracy of the model and investigate the correlation of the model predictions with the average human responses. We also compare our model's accuracy with the Levenshtein distance generated by transcriptions produced by the DeepSpeech ASR model. This study investigates the reliability of deep neural networks as an alternative to human-based inference in quantifying the intelligibility of speech. We discuss the advantages and disadvantages of the different approaches to assessing speech intelligibility.

**5aSC17. Timing of smile suppression during the articulation of labials in smiled speech.** Kyra Hung (Linguist, The Univ. of British Columbia, 2329 West Mall, Vancouver, BC V6T1Z4, Canada, kmhung@mail.ubc.ca), Yadong Liu, Charissa Purnomo, and Bryan Gick (Linguist, The Univ. of British ColumbiaBC, Vancouver, BC, Canada)

Past studies have shown that opposing forces on the lips can be reconciled by suppressing a smile during speech-related lip closure movements [Liu *et al*., 2020, ISSP]. However, the exact timing of this suppression on the smiling posture remains unknown. The purpose of this study was to determine the onset and duration of smile suppression during labial production in smiled speech. We collected video footage of participants reading sentences that contain labial sounds (/m, f, v, b, p, w/) under three facial posture conditions (neutral, smile, laugh), and extracted short video clips of the labial sound productions. We then analyzed the extent of lip spreading in each facial posture condition using OpenFace 2.0 [Baltrušaitis *et al*., 2018, IEEE]. Our preliminary results show that smile is suppressed in the smiling and laughing conditions, with attenuation beginning approximately 300ms prior to labial sound production, and continuing for about 600ms regardless of conditions. Further analysis will investigate the duration of suppression after the offset of labials and will be conducted on more participants.

**5aSC18. Understanding and characterizing speaker roles within naturalistic task-based communications: The fearless steps APOLLO-11 corpus.** Meena Chandra Shekar (Elec., Univ. of Texas at Dallas, 7220 McCallum Blvd, Apt 1704, Dallas, TX 75252, Meena.ChandraShekar@utdallas.edu) and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

Naturalistic team based speech communications requires specific protocols/procedures to be followed to allow for effective task completion for distributed team members. NASA Apollo-11 was the first manned space mission to successfully bring astronauts to the moon and return them safely. Mission specialists roles within NASA Mission Control (MOCR) are complex and reflected in their communications. In this study, we perform speaker clustering to identify speech segments uttered by the same speaker from recently recovered Fearless Steps APOLLO corpus (CRSS-UTDallas). We propose a pretrained network to obtain speaker embeddings and use a framework that builds on these learned embeddings which achieves a clustering accuracy of 73.4%. We also track/tag key speakers-of-interest across three critical mission phases and analyze speaker roles based on speech duration. NASA communication protocols dictate that information be communicated in a concise manner. In automated communication analysis, individuals higher in trait dominance generally speak more and gain more control over group processes. Hence, speaker duration of primary- versus -secondary speaker and speaker turns are metrics used to determine speaker role. This analysis provides greater understanding of communications protocol and serves as a lasting tribute to the «Heroes Behind the Heroes of Apollo» as well as preserve "words spoken in space."

**5aSC19. Retention of devoiced vowels in Tokyo Japanese: Evidence from lip articulation.** Rion Iwasaki (Speech-Language-Hearing Sci., City Univ. of New York, 365 Fifth Ave., Rm. 7304, New York, NY 10016, riwasaki@gradcenter.cuny.edu), Kevin D. Roon (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY), Jason A. Shaw (Dept. of Linguist, Yale Univ., New Haven, CT), Mark Tiede (Haskins Labs., New Haven, CT), and D. H. Whalen (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY)

In Tokyo Japanese, high vowels /i/ and /u/ are frequently devoiced when they are surrounded by voiceless obstruents. Controversy remains over whether vowel gestures in devoiced vowels are retained or instead deleted. Both static (Iwasaki *et al*., 2020) and dynamic (Iwasaki *et al*., 2022) ultrasound data have indicated that vowel-specific lingual gestures can persist even when vowels are devoiced. This study focuses on the lip articulation of devoiced vowels by examining lateral lip aperture, where lower values index a greater degree of the rounding of vowels. Native speakers of Tokyo Japanese produced nonce word pairs with the form of /$C_1VC_2V_2$toko/. $V_1$ was either /i/ or /u/. $C_1$ and $C_2$ were either voiced or voiceless, which determined the voicing of $V_1$. Lateral lip aperture during the first mora was calculated by identifying facial landmarks using OpenFace 2.0 (Baltrusaitis *et al*., 2018), and compared across vowel quality (/i/ *vs*./u/) and vowel voicing (devoiced *vs*. voiced). Preliminary results show that lateral lip aperture is larger for /i/ than for /u/ in both devoiced and voiced environments, indicating that these vowels maintain their labial specifications even when devoiced, providing additional evidence that devoiced vowels can retain their articulatory gestures.

**5aSC20. Links between language, fluid dynamics, and the airflows that transport pathogens.** Junshi Wang (Princeton Univ., 41 Olden St., Princeton, NJ 08540, junshi.wang@princeton.edu), Simon Mendez (Univ. of Montpellier, Montpellier, France), Haibo Dong (Univ. of Virginia, Charlottesville, VA), and Howard A. Stone (Princeton Univ., Princeton, NJ)

Growing evidence shows that the airflows that accompany speech during social interactions contribute to the transport of pathogens, such as the SARS-CoV-2 virus. However, it is still elusive how the manners of articulation during speaking, such as time-varying airflow rate, and instantaneous teeth and lip movements, may affect the transport features of airflow exiting the mouth. We combine experimental and numerical approaches to

investigate the flow patterns produced by representative vowels, such as /a/, /o/, and /i/, and consonants, such as /p/, /k/, /s/, and /h/, that have distinct articulatory characteristics. A 3D vocal tract is modeled with a temporally varying exit that captures key morphologic and kinematic features of the human vocal tract, including teeth and lips, during speaking. An incompressible flow solver based on a sharp-interface immersed-boundary-method (IBM) is employed to compute the resultant airflow. By comparing representative utterance pairs, like /apa/ and /aha/, we are able to isolate the effect of articulatory features and show significant differences in spatial-temporal patterns of airflow under the influences of a time-varying orifice, different orifice aspect ratios, and different relative lip-teeth positions. This work helps bring insight into the understanding of articulatory phonetics, and the links to different languages, from a fluid dynamics perspective.

**5aSC21. Curating feedback for parents based on interactions during parent–child book reading activities.** Sarah A. Tao (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, 800 West Campbell Rd., Richardson, TX 75080, sarah.tao@utdallas.edu), Satwik Dutta (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX), Rebecca Hacker, Dwight Irvin, Jay Buzhardt (Univ. of Kansas, Kansas City, KS), and John H. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas - Dallas, Richardson, TX)

Assessing child language and communication skills in naturalistic settings requires a motivating framework for adults and children to interact. Book reading is widely used by parents to interact with children, providing parents opportunities to engage children in conversation and practice communication skills. Recordings of interactions between parents and children can be automatically analyzed to provide feedback to parents to maximize developmental benefits of book reading. Encouraging children to participate actively in book reading yields gains in terms of language acquisition and improved literacy. In this study, we focus on extracting fruitful feedback from parent–child book reading activities for parents. Audio evaluated were collected by consenting families in home settings during parent–child science book reading activities. A range of parameters are analyzed to track reading/learning behaviors that include number of questions asked, number of conversational turns, number of words spoken, repetition, and vocabulary diversity. These parameters can help track engagement of story reading, including changes over time. Discussions with various stakeholders including parents, teachers and researchers will be presented. Speech technology automation of this analysis process can help parents track language development, improve communication skills, and increase school preparedness of their children. [Work sponsored by NSF Grants #1918032 & #1918012.]

**5aSC22. Second language acoustic–prosodic entrainment in conversation and storytelling.** Grace Kuo (Foreign Lang. and Literatures, National Taiwan Univ., 1 Section 4, Roosevelt Rd., Taipei, Taipei City 106, Taiwan, graciakuo@ntu.edu.tw)

Prosodic entrainment is the tendency for individuals to modify their acoustic-prosodic speech behaviors to converge with the behaviors of their interlocutors. Evidence of entrainment in the native language (L1) is robust, yet research regarding its development in the second language (L2) is sparse. To examine prosodic entrainment in spoken dialogues, we developed a natural speech corpus in which spontaneous conversations of 30 young adults were collected. Data from 15 dyadic groups of participants in the conversation and the story-telling sessions in both their native language (Mandarin Chinese) and L2 (English) were collected and analyzed. Prosodic entrainment has been examined with the low-level prosodic features, a bundle of the acoustic measure (i.e., f0, speaking rate), and voice measures (i.e., jitter and NHR). Individual difference measures (i.e., attention to details, communication skills) are also examined, and their correlations with the prosodic features are reported. The research is expected to contribute to the study of L2 prosody acquisition and human–machine interaction as well as future research in turn-taking, conversational analysis, and individual differences in L1 and L2 speech behaviors.

**5aSC23. Growth of the hard palate between the ages of 6 and 10 years: A longitudinal study.** Steven M. Lulich (Speech, Lang. and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu)

Several studies have found that the shape of the palate can affect articulation of speech sounds, but not much information is available characterizing the three-dimensional shape of the palate. This is especially important in children, since their speech is maturing at the same time that the palate is still growing. This study examines the longitudinal growth of the palate from 8 children between the ages of 6 and 10 years, based on high-resolution three-dimensional digitized palate impressions. Measures of palate width, height, depth, and symmetry were derived from three anatomically defined points: the marginal gingiva between the front teeth, and between the second premolar and the first molar on both sides. While palate height was most difficult to measure accurately, it was also most highly correlated with age ($R^2 = .42$, p<.001), while palate depth, width, and symmetry were poorly correlated with age ($R^2$<.07, p>.17). Age and standing height were strongly correlated ($R^2 = .77$, p<.001). At the individual level, palate height appeared to grow linearly with age, while the growth of palate width and depth was nonlinear and even non-monotonic, indicating a complex interaction between growth and the ongoing structural rearrangement of dentition in this age group. [Work supported, in part, by the NSF.]

**5aSC24. Frequency stability of articulatory and acoustic modulation functions.** Jessica Campbell (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, jac95339@usc.edu), Dani Byrd, and Louis Goldstein (Linguist, Univ. of Southern California, Los Angeles, CA)

During speech perception, neural activity entrains with moments of high acoustic change. For example, periods of high change in speech amplitude envelope magnitude are tracked by neurons in the human superior temporal gyrus. However, it is unknown whether neural entrainment may also be driven by modulation in the articulatory domain. To locate periods of high articulatory change, a spatiotemporal modulation function (Goldstein, 2019) that quantifies change over time in global vocal tract posture can be used to investigate the potential for such entrainment. Here, the frequency patterning and stability of modulation maxima, called "pulses," are assessed using articulatory point-tracking data. The median frequency of both articulatory and acoustic pulses is found to be only slightly higher than theta band frequencies (6i–8 Hz), at which neural entrainment with speech has been reported. Within- and between-speaker variability of inter-pulse intervals is also compared to the variability of acoustic syllable and acoustic stress foot durations. The results show that intervals between pulses are more stable than syllable and foot durations. In sum, the spatiotemporal modulation function exhibits a stable frequency profile in the articulatory and acoustic domains that could be leveraged in the neurocognitive functions at work in speech perception. [Work supported by the NIH.]

**5aSC25. Southern dialects of United States English and automatic speech recognition.** Yolanda F. Holt (Commun. Sci. and Disord., East Carolina Univ., 300 Moye Bv 3310-X HSB, MS 668, Greenville, NC 27834, holty@ecu.edu) and Matthew Walenski (Commun. Sci. and Disord., East Carolina Univ., Greenville, NC)

Automatic Speech Recognition (ASR) technology is designed to provide more human to machine communication options. Emergent research has observed variant performance of ASR for African American English (AAE) and other minority dialects. Word prediction was observed to be better for AAE compared to majority dialects. The opposite was true for word identification accuracy. The researchers hypothesize the higher word error rate for AAE is related to phonetic factors of vowel duration, consonant production, rhythm, pitch, and syllable accent. This work evaluates that hypothesis. Recordings of two AAE and two White AE speaking women from North Carolina reading Comma Gets a Cure were submitted for transcription to the Microsoft ASR program. Several common transcription errors were noted across speaker groups with a greater variety of errors noted for the AAE women. Acoustic analyses of vowel duration, and the spectral acoustics of vowel and consonant production by talker and group will be completed. The results of this analysis will be useful to describing the specific aspects of AAE speech that may perturb the tested ASR. These data provide insight into the speech tasks and the sub and supra-segmental acoustic data that should be included in future ASR programming iterations.

**5aSC26. Effects of semantic predictability and speaking style on phonetic variation.** Yu Jin Song (Ohio State Univ., 1712 Neil Ave. Columbus, OH 43210, song.1693@osu.edu) and Cynthia G. Clopper (Ohio State Univ., Columbus, OH)

Many linguistic factors contribute to phonetic reduction in «easy» linguistic contexts and phonetic enhancement in «hard» linguistic contexts. However, previous findings are mixed as to whether interactions among these factors are simply additive and as to whether their effects are differentially exhibited across acoustic domains. To better understand phonetic variation processes, we explored the interaction between two factors, semantic predictability and listener-driven speaking style, on phonetic variation in duration, f0, and vowel formant frequencies. Talkers read aloud sentences differing in the semantic predictability of the sentence-final word (high versus low predictability) to purported listeners with different linguistic competence (a native listener versus a non-native listener with high proficiency versus a non-native listener with low proficiency). Talkers produced phonetic enhancement for less predictable words and for the low-proficiency non-native listener relative to the native and high-proficiency non-native listeners across measures, but the effects of predictability and speaking style did not interact. Together, these results suggest that phonetic variation due to semantic predictability and speaking style may be additive, independent processes, rather than the result of a single listener-oriented process.

**5aSC27. Vocalic contrasts in Hnaring Lutuv.** Amanda Bohnert (Linguist, Indiana Univ., Bloomington, Bloomington, IN), Grayson Ziegler, and Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 852, Bloomington, IN 47405-2201, kberkson@indiana.edu)

Lutuv, sometimes called Lautu, is an under-documented Chin language from the Tibeto-Burman language family spoken in Chin State in western Burma by 15 000 to 18 000 people (Eberhard *et al*., 2022, citing 2005 data that do not account for current military violence and displacement). Lutuv is also spoken in diaspora communities worldwide, including by about 1000 people in the Chin refugee community of Indianapolis (community estimate). Ongoing fieldwork with the Hnaring variety has revealed that, compared to related languages, Lutuv has undergone radical syllable structure simplification and attendant vowel inventory expansion. In addition to the low vowel /a/ and mid vowels /e, ə, ɔ/, Lutuv contains six to ten high vowels. These tentatively include four diphthongal vowels (/i$^{e}$, y$^{ə}$, u$^{ɔ}$, ɯ$^{ə}$/) as well as six monophthongs /i, y, ɨ, ʉ, ɯ, u/. We explore the distribution of the vowels via a combination of acoustic and comparative data in an effort to better understand the articulatory and featural specifications thereof, with special attention paid to the high vowels. If all of these monophthongs truly are high, it would represent a startlingly large and typologically uncommon high vowel inventory.

**5aSC28. Consonantal contrasts in Hnaring Lutuv.** Grayson Ziegler, Amanda Bohnert (Linguist, Indiana Univ., Bloomington, Bloomington, IN), and Kelly H. Berkson (Linguist, Indiana Univ., Bloomington, 1020 E. Kirkwood Ave., Ballantine Hall 852, Bloomington, IN 47405-2201, kberkson@indiana.edu)

Hnaring is a dialect of Lutuv, a member of the South Central branch of Tibeto-Burman spoken by 15 000 to 18 000 people in Chin State in western Burma and in refugee communities worldwide (community estimates, Eberhard *et al*., 2022; Salaz and Raymer, 2020). Like many other Tibeto-Burman languages, Lutuv features an extensive consonantal inventory, with three-way contrasts for labial and coronal stops and two-way contrasts for dorsals, laryngeal contrasts for sibilant and lateral affricates (/ts, ts$^{h}$/, /tl, tl$^{h}$/ ), and voicing contrasts for sonorants /r, r̥, l, l̥, n, n̥, m, m̥/. Of special note is the high degree of variability observed in the instantiation of voiceless sonorants, which can include alternately ordered phases of aspiration, frication, and/or voicing. For example, voiceless laterals can be realized in at least four ways ([l$^{h}$, $^{h}$l, ɬ, l̥]); rhotics alternate between rhotic and sibilant productions ([$^{h}$r, r$^{h}$, ʃ]); and nasals show phasing and voicing differences [$^{h}$m, m$^{h}$, m̥, m͡m̥]. Using both wordlist data and naturalistic speech from a conversational corpus collected during Summer 2022, this study presents a suite of temporal and spectral measures for Hnaring Lutuv consonants and combines qualitative and quantitative data to characterize the observed variation.

**5aSC29. Vocal and articulatory performance fluctuates across the menstrual cycle: Acoustic analysis of daily single participant speech data over three months.** Daniel Aalto (Commun. Sci. and Disord., Univ. of Alberta, 8205 114 St NW, Edmonton, AB T6G2G4, Canada, aalto@ualberta.ca), Courtney Cathcart, Riley Brennan, and Jacqueline Cummine (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada)

The impact of sex hormones on voice and speech has been observed across the menstrual cycle; however, the evidence remains mixed. Here, we investigated voice and articulatory performance as a function of the menstrual cycle over three months via an intensive longterm single case experimental design. A 17-yearold naturally cycling female who did not use oral/hormonal contraceptives gave a written consent to participate in the study over a period of three cycles (i.e., 90 days). Daily urine sample test strips were utilized to identify the onset of ovulation (via luteinizing hormone, LH). Daily voice and articulation tasks including a Maximum Phonation Time (MPT) task, a diadochokinesis (DDK) task were completed. A harmonic regression analysis showed a linear learning effect for DDK rate (p<.00001) with a fluctuation of values as a function of the menstrual cycle ($p = 0.024$). MPT showed a cyclic fluctuation ($p = 0.028$) but no linear learning effect. Linear and harmonic terms for fundamental frequency (measured from MPT recordings) showed no significant patterns. Understanding hormone driven variability in speech and voice helps to inform remediation strategies that aim to minimize the impact of speech motor and voice disorders across the lifespan.

**5aSC30. The effect of turn-taking on the kinematic properties of prosodic boundaries.** Ruaridh Purse (Linguist, Univ. of Michigan, Ann Arbor, MI, rupurse@umich.edu) and Jelena Krivokapić (Linguist, Univ. of Michigan, Ann Arbor, MI)

Gestures at prosodic boundaries are larger, longer, and further apart than gestures phrase-medially, and this effect is stronger for hierarchically higher boundaries. However, most studies examining kinematic properties of boundaries limit their investigation to read speech of individual utterances, so the kinematic properties of turn-taking are not known. In this study, six pairs of participants competed in a game eliciting semi-controlled interactive speech with target words at phrase-medial, phrase-final but utterance-medial, and turn-final prosodic boundaries. One member of each participant dyad was recorded in an articulatory magnetometer, and the other acted as a conversation partner. We test two competing hypotheses about the behavior of gestures at turn-final boundaries: Hypothesis 1 states that turn-final gestures show more lengthening than turn-medial gestures. This hypothesis comes from Conversation Analysis research suggesting final lengthening is used to indicate the end of a turn, and the fact that hierarchically higher boundaries generally show more lengthening. Conversely, hypothesis 2 states that turn-final gestures show less lengthening than turn-medial gestures. This alternative hypothesis is based on the idea that, since there is no imminent speech to be planned following the end of a turn, no additional lengthening is required in order to accommodate planning time.

# Session 5aUW

## Underwater Acoustics: General Topics in Underwater Acoustics I

Shima Abadi, Chair
*University of Washington, 185 Stevens Way, Paul Allen Center – Room AE100R, Seattle, WA 98195*

### *Contributed Papers*

**9:00**

**5aUW1. Overview of ocean ambient noise interferometry – Theory and simulation.** John Ragland (Elec. and Comput. Eng., Univ. of Washington, 185 W Stevens Way NE, Seattle, WA 98195, jhrag@uw.edu) and Shima Abadi (Univ. of Washington, Seattle, WA)

Ambient noise interferometry is a passive acoustic technique for environment characterization. The technique uses coherent ambient sound to approximate the Green's function between two sensors. It has previously been used in ocean acoustics to passively estimate water temperature, sound-speed structure, and mode shapes as well as for sensor localization. Since this technique utilizes the ambient noise field, whose characteristics are often unknown, understanding the effects of non-isotropic ambient sound is important for ambient noise interferometry. In this talk, an overview of the theoretical literature for noise interferometry will be presented with a specific emphasis on the effects of non-isotropic ocean noise source distributions. Additionally, simulations that explore the emergence of the Green's function will be presented. Specifically, sound source distributions and environmental parameters such as sound speed profile will be explored. Lastly, the implications of these works on the possibilities and limitations of ambient noise interferometry will be discussed. [Work supported by the ONR.]

**9:15**

**5aUW2. Detection and monitoring of seafloor methane bubbles using hydrophones.** Shima Abadi (School of Oceanogr., Univ. of Washington, 185 Stevens Way, Paul Allen Ctr. – Rm. AE100R, Seattle, WA 98195, abadi@uw.edu), Tor A. Bjorklund (School of Oceanogr., Univ. of Washington, Seattle, WA), Junzhe Liu (School of Oceanogr., Univ. of Washington, Seattle, WA), and H. P. Johnson (School of Oceanogr., Univ. of Washington, Seattle, WA)

Natural marine seeps of methane are important sources of greenhouse gas emissions that enter the global environment. Monitoring marine methane seeps will reveal important information about how they form, their source regions, and how much of the inventory is microbially consumed within the water column before the gas is released to the atmosphere. While active acoustics methods have been extensively used to detect and monitor methane emissions from the seafloor, there are only a few studies showing the use of passive acoustics for bubble sound detection and monitoring. In this presentation, we use passive acoustics data recorded in a methane seep field in Puget Sound to characterize the bubble sound and estimate the bubble radii from their sound frequency. We use the ratio of a short-term average from a bubble burst over the long-term average to automatically detect the bubble sounds. We also show the relationship between the number of bubble sound detections and the hydrostatics seafloor pressure fluctuations due to tidal changes in water height.

**9:30**

**5aUW3. Characterization of underwater soundscape variations pre- and post-ship shock trial underwater detonations.** Shyam Madhusudhana (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, skm246@cornell.edu), Holger Klinck (K. Lisa Yang Ctr. for Conservation Bioacoustics, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY), and Kerri D. Seger (Appl. Ocean Sci., Fairfax Station, VA)

During Full Ship Shock Trials (FSSTs), a ship is subjected to a series of underwater detonations conducted at various distances to assess its ability to withstand shock waves that simulate near misses during combat. In the summer of 2021, the US Navy successfully conducted a FSST of the USS Gerald R. Ford, in the Atlantic Ocean off the coast of Jacksonville, FL. We collected passive acoustic recordings during each of the three underwater detonations, using an array of bottom-moored autonomous underwater recorders—Rockhoppers and SoundTraps—that were deployed at 15 vantage positions around the three sites. While the acoustic recordings of the underwater detonations were aimed at providing the Naval Sea Systems Command valuable data for updating their underwater acoustic propagation models, it also provided opportunistic data to assess possible impacts on prevailing biota. Our analyses of the recordings included an assessment of the recorded soundscape pre- and post-trial and a characterization of relative vocal activity by large faunal groups (baleen whales, delphinids, and fish). The results provide an insight into behavioral changes in response to the underwater detonations at different distances from the detonation sites as well as valuable information for improving impact mitigation considerations during future FSSTs.

**9:45**

**5aUW4. Ambient sound characterization over decadal time scales in the Atlantic Ocean.** Mahdi H. Al-Badrawi (Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, mahdi.albadrawi@unh.edu) and Kevin D. Heaney (Appl. Ocean Sci., Fairfax Station, VA)

Gaining more knowledge about the variations in the ambient ocean soundscape plays a critical role in assessing the human impact (anthropogenic noise pollution) on marine species' behavior and habitat choices. Accessing historical recordings is important to understand the long-term changes in the ambient sound and to identify mechanistic drivers influencing the soundscape regionally and globally. Data recorded in the Northwest Indian Ocean (Bearing Stake exercise) in 1977 were compared with more recent (2003 and 2013) data from the north side of Diego Garcia Island. This comparison between datasets 40 years apart showed that the ambient sound levels in that region were not higher in the 2000s than they were in 1977. The spectral levels decreased between 1977 and 2003 and then leveled out

between 2003 and 2013. These findings are different from the well-cited 3 dB/decade increase in Pacific Ocean sound levels, but they did align with recent studies that also show declines in sound levels in other regions. Therefore, another comparison between acoustic recordings 40 years apart near Bermuda and from an ADEON site (October 1980 and 2018–2020, respectively) was performed to facilitate a better interpretation of how the soundscape is changing over decadal timescales in the Atlantic Ocean.

**10:00–10:15 Break**

**10:15**

**5aUW5. Basin-scale propagation modeling of MLS signals between Kauai and Monterey.** Nicholas C. Durofchalk (Phys., Naval Postgrad. School, 2788 Defoors Ferry Rd., Apt 325, Atlanta 30318, Georgia, ndurofchalk3@gatech.edu), Kay L. Gemba, Kevin B. Smith, and Paul Leary (Phys., Naval Postgrad. School, Monterey, CA)

Long distance underwater acoustic propagation is of interest for a variety of applications including underwater navigation, yet modeling such propagation is challenging due to the large degree of environmental uncertainty. In this presentation, the propagation of 75 Hz center frequency maximum length sequence (MLS) signals emitted from a submerged source near Kauai and received at the Monterey Accelerated Research System (MARS) observatory are modeled with the Bellhop ray tracing and the Monterey-Newport Parabolic Equation (MNPE) models. The range-dependent sound speed environment is based on historical profiles from the World Ocean Atlas database and reanalysis data from the hybrid coordinate ocean model (HYCOM). Bathymetry profiles along the axis of propagation are interpolated from the General Bathymetric Chart of the Oceans (GEBCO) 2021 database and the Naval Oceanographic Office Digital Bathymetric Data Base Variable Resolution (DBDB-V) data product. Simulated channel impulse responses are compared to observations for transmission loss and signal coherence.

**10:30**

**5aUW6. Implementation and validation of a three-dimensional semi-coherent energy flux model for underwater acoustic propagation.** Mark A. Langhirt (Penn State, 201 Appl. Sci. Bldg., Graduate Program in Acoust., University Park, PA 16802, mal83@psu.edu), Charles W. Holland (Elec. and Comput. Eng., Portland State Univ., Portland, OR), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Sheri Martinelli (Appl. Res. Lab., The Penn State Univ., State College, PA), and Daniel C. Brown (Penn State Univ., State College, PA)

The energy flux method uses a continuum of adiabatic Wentzel–Kramers–Brillouin modes to integrate intensity directly as a function of propagation angle. Work published a decade ago extended the energy flux method to include semi-coherent interference between neighboring modes to capture convergence zones. This study seeks to generalize the energy flux method to three-dimensional inhomogeneous underwater environments by converting a cross-product of double mode-sums to a double integration over solid angle. Theoretical derivations for a proof of concept model were presented previously, and current work has been focused on the numerical implementation of the model. Model validation will be discussed with comparisons to analytical solutions and other three-dimensional underwater acoustic propagation models.

**10:45**

**5aUW7. Underwater soundscapes at critical habitats of the endangered Hawaiian monk seal.** Kirby Parnell (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Univ. of Hawaii Manoa, 46-007 Lilipuna Rd., Kaneohe, HI 96744, keparnel@hawaii.edu), Karlina Merkens (Saltwater, Inc. in support of NOAA Fisheries Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), Aude Pacini, and Lars Bejder (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Univ. of Hawaii Manoa, Kaneohe, HI)

Describing underwater soundscapes of critical habitats of marine mammals can provide valuable information on the acoustic environment utilized by sound-reliant animals. For the endangered Hawaiian monk seal *(Neomonachus schauinslandi*), whose underwater hearing abilities and vocal communication were recently described, the soundscapes of their aquatic habitats are poorly understood. We measured ambient noise levels and identified acoustic signals that contribute to the underwater soundscape at four critical habitats of the Hawaiian monk seal. We deployed SoundTrap 500HF acoustic recorders, at sites with varying habitat types, and recorded continuously for 179 days. We measured broadband (20–24 000 Hz) and octave-band (31.5, 250, 500, and 16 000 Hz center frequencies) sound pressure levels (SPLs) in hourly intervals at each site. Average hourly broadband SPLs ranged from 107.8–123.4 dB re 1 $\mu$Pa. Octave-band SPLs confirmed diel patterns associated with biological and anthropogenic sources. We recorded two large-scale geophysical events: Hurricane Douglas (Category 4) and a 6.2 magnitude earthquake which increased the 31.5 Hz octave-band SPL. This study provides the first description of underwater soundscapes at critical habitats of the Hawaiian monk seal across its expansive range. These measurements serve as a baseline for future studies to understand the impact of human activity on underwater soundscapes.

# Session 5pPA

## Physical Acoustics and Biomedical Acoustics: Ultrasonic Assessment of Properties in Complex Materials II

Andrea P. Arguelles, Cochair
*Engineering Science and Mechanics, Penn State University, 212 Earth-Engr Sciences Bldg, University Park, PA 16802*

Marie Muller, Cochair
*MAE, North Carolina State University, 911 Oval Drive, Engineering Building III, Raleigh, NC 27606*

### *Contributed Papers*

**1:00**

**5pPA1. Simulation of a tissue backscatter coefficient measurement experiment using the finite element method.** George West (Radiotherapy and Imaging, Inst. of Cancer Res., 15 Cotswold Rd., London SM2 5NG, United Kingdom, george.west@icr.ac.uk), Emma Harris, Jeff Bamber (Radiotherapy and Imaging, Inst. of Cancer Res., London, United Kingdom), Michael J. Lowe, and Peter Huthwaite (Mech. Eng., Imperial College London, London, United Kingdom)

Ultrasound has been shown to be an effective imaging modality for investigating tissue health. For example, the backscatter coefficient (BSC) has been shown to be a biomarker of tumour response to therapy but is limited in its application clinically due to the difficulty of accurately acquiring attenuation and diffraction corrections. To assess the variability in BSC assessment, we present a finite element tool simulating a singleelement planar reflector substitution method estimate of the BSC of a set of simulated phantoms. BSC estimates were computed with errors <5% of the theoretical value for a range of source apertures and over a set of phantoms with varying scatterer number densities to within 10%. In addition, the firstorder amplitude envelope statistics of backscattered waves were shown to be commensurate with Rayleigh scattering models. These results indicate that the tool is accurate in its replication of soft tissue-like scattering, suggesting that it could be an opportune testing ground for investigating sources of variability in BSC measurements and the testing of algorithms and hypotheses for interrogating the scattering behaviour of soft tissues.

**1:15**

**5pPA2. Using random matrix theory to quantify pulmonary fibrosis: Investigating the effect of time window duration.** Azadeh D. Cole (Mech. and Aerosp. Eng., North Carolina State Univ., Eng. Bldg. III (EB3) 3141, Raleigh, NC 27695, adashti@ncsu.edu), John Blackwell (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Stephanie A. Montgomery (Dept. of Pathol. and Lab Medicine, Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Thomas M. Egan (Surgery and Biomedical Eng., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Marie Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

Random matrix theory (RMT) exploits the distribution of singular values of the inter-element response matrix (IRM). If multiple scattering dominates the propagation, the singular value distribution follows the quarter circle law. However, dominance of single scattering results in Henkel function behavior of the singular value distribution. In our previous work, we have shown that this can be exploited to estimate severity of bleomycin-induced fibrosis (measured by histology) in rodent lungs. We showed that $E(x)$, the expected value of the singular value distribution, as well as the singular value with the highest probability, correlated significantly with histology scores. Here, we investigate the sensitivity of these metrics to the time window duration used for the Singular Value Decomposition of the IRM, which is performed in the frequency domain, using overlapping time windows. A linear transducer with a central frequency of 7.8 MHz and a Verasonics scanner were used to obtain IRMs in 24 rat lungs. Different degrees of pulmonary fibrosis were induced using bleomycin in 18 rats while 6 rats were left as controls. The IRMs were time-windowed with different duration of 2T, 4T, and 6T where T is the transmitted pulse period. $E(x)$ and were evaluated for each time window duration for all rat lungs. For all time window durations significant correlations were observed between and $E(x)$, and histology scores. Wilcoxon ranksum tests show that the distributions obtained for and $E(x)$ are not affected by the time window duration.

**1:30**

**5pPA3. Mechanical and acoustical characterization of elastic properties for additively manufactured polymers.** Celeste A. Brown (U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, celeste.brown@nrl.navy.mil), Amelia Vignola (U.S. Naval Res. Lab., Washington, DC), Luz D. Sotelo (U.S. Naval Res. Lab., Lincoln, NE), Grant Warner (Mech. Eng., Howard Univ., Washington, DC), and Matthew D. Guild (U.S. Naval Res. Lab., Washington, DC)

Recent advances in the additive manufacturing of photopolymers has opened new opportunities for the design and realization of novel acoustic materials, such as metamaterials. Despite the significant advantages additive manufacturing offers for photopolymers, a significant lack of characterization data inhibits their use in design for acoustics applications. This work provides characterization data and establishes a relationship between static and dynamic properties: specifically, the mechanically and ultrasonically derived damping factor and loss tangent for additively manufactured photopolymers. To achieve this, a range of stiff and compliant photopolymers are fabricated using a commercially available Polyjet 3-D printing platform. The elastic loss and storage moduli for each material are characterized under static uniaxial loading conditions. Furthermore, shear and compressional phase velocity and attenuation are measured for each material using ultrasound. The damping factor and loss tangent for each material is then calculated for both the static and ultrasonic case using the complex elastic moduli, and a frequency-based relationship is drawn between the two. The presented results are expected to contribute to improved design and broadband characterization of additively manufactured acoustic materials using photopolymers. [Work funded by the Office of Naval Research.]

1:45

**5pPA4. Ultrasonic characterization of compressional and shear wave properties for additively manufactured photopolymers.** Amelia Vignola (Code 7165, U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, amelia.vignola@nrl.navy.mil), Celeste A. Brown (Acoust. Div., U.S. Naval Res. Lab., Washington, DC), Luz D. Sotelo (Code 7165, U.S. Naval Res. Lab., Lincoln, NE), and Matthew D. Guild (Naval Res. Lab, Washington, DC)

Advanced applications of polymer additive manufacturing (AM) require knowledge of the material acoustic and complex elastic properties. Recently, ultrasonic nondestructive evaluation (NDE) has been applied to the characterization of AM polymers; however, the available information is limited and there are little data on shear-wave and complex elastic properties. In this study, a two-sample method using ultrasonic measurements of compressional and shear phase velocity and attenuation was used to experimentally determine the complex elastic properties of 14 AM photopolymers manufactured using PolyJet systems. In this method, two samples of identical material properties but different thicknesses are measured using ultrasonic contact transducers, with the change in the time of arrival and amplitude providing the phase speed and attenuation. In order to determine the optimal thickness difference between the two samples, preliminary sets of samples were printed in a range of thicknesses, without a priori knowledge of the material properties of the AM polymers. Using a subset of these results, further analysis and optimization was conducted to better select the two thicknesses to use for each material sample. The results provided here are expected to aid in experimental design for characterization of polymers. [Work supported by the Office of Naval Research.]

2:00

**5pPA5. The Doppler ultrasound twinkling artifact on crevices etched in silicon wafers.** Eusila C. Kitur (Penn State Univ., 170 East 6th St., Claremont, CA 91711, eusila.kitur@gmail.com), Eric Rokni (Penn State Univ., State College, PA), and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The twinkling artifact, or rapid color shifts, are observed when imaging kidney stones and other mineralizations. The current theory is that scattering from bubbles in micron-sized crevices causes twinkling. Our objective is to investigate the effect of ultrasound frequency and crevice size and number on twinkling of silicon wafers. Ten or 100 randomly positioned cylindrical crevices with diameters of 1, 10, or 100 $\mu$m and depths of 10 $\mu$m were dry-etched on wafers. A research ultrasound system was used to image wafers at 5, 7.8, and 18.5 MHz; IQ data were post-processed to calculate twinkling or Doppler power. Twinkling for all crevice sizes and numbers followed the trend 5 MHz>7.8 MHz>18.5 MHz. For crevice size, at 5 and 7.8 MHz, twinkling was higher for 1- or 10-$\mu$m crevices compared to 100 $\mu$m; at 18.5 MHz, twinkling was only observed on 100-$\mu$m crevices. Increasing the number of crevices increased twinkling for 1 $\mu$m crevices at 5 and 7.8 MHz and 10 $\mu$m crevices at 7.8 MHz; no differences were noted at 18.5 MHz. These results support previous work showing twinkling is frequency-dependent and suggest that even 10, 1-$\mu$m crevices are sufficient to cause twinkling. [Work supported by SURIEA, NSF CAREER-1943937.]

**2:15–2:30 Break**

## *Invited Paper*

2:30

**5pPA6. Realistic modelling of microstructural features in numerical simulations of wave propagation in metals.** Michael J. Lowe (Mech. Eng., Imperial College London, Dept. of Mech. Eng., Imperial College, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk)

Research in numerical modelling in recent years has enabled realistic simulations of wave propagation in three dimensions in large volumes of material, while including features at small scale. A critical enabler has been the growth of computer power, especially Finite Element computations on GPUs. The presentation will start with the development of modelling capability for polycrystalline materials, for which extensive recent work has created realistic simulations for wave speed and grain-scattering attenuation, representing the microstructure at grain scale. This topic has received much attention in theoretical work over several decades, so the simulations have been helpful to evaluate and understand the theoretical models and the physics of the behaviour. Subsequently, the modelling has addressed several other microstructural interests, including simulations for creep damage and fatigue damage, each of which cause reductions of wave speed; models of equivalent macro material properties for these are developed based on studies of the microstructural information. Finally, recent work will be presented on wave propagation and scattering in Titanium alloys containing Macro Textured Regions (MTRs/Macrozones); these comprise microstructures of mixed scale, for which there is interest to use ultrasound to characterise the MTRs (large scale) against a background of a smaller scale regular microstructure.

**2:55**

**5pPA7. Investigation of the planar reflector substitution method using the finite element method.** George West (Radiotherapy and Imaging, Inst. of Cancer Res., 15 Cotswold Rd., London SM2 5NG, United Kingdom, george.west@icr.ac.uk), Emma Harris, Jeff Bamber, Peter Huthwaite (Radiotherapy and Imaging, Inst. of Cancer Res., London, United Kingdom), and Michael J. Lowe (Mech. Eng., Imperial College London, London, United Kingdom)

The backscatter coefficient (BSC) has been shown to be an indicator of tissue state in assessing tumour response to therapy, but it is limited in its clinical applicability by the difficulty in acquiring the appropriate reference spectrum. The accuracy of a BSC estimate made by the substitution method is influenced by the quality of the reference spectrum, which normalises the measurement to the source's diffraction field and spectral characteristics. Among the reference objects employed for normalisation, a planar reflector has been popular. With the sample region at the focus of a focused source or the last axial maximum of an unfocused source, the reflector has been placed either at this depth or half this depth by different authors. This work explores the effect of planar reflector positioning on BSC estimation. FE models were constructed to simulate BSC measurements of virtual phantoms with various sources, and the estimates compared to scattering theory. The results indicate that unfocused sources provide more accurate BSC estimates when the reflector is placed at half the distance to the last axial maximum (d), while focused sources provide better estimates with the planar reflector at the focal plane, and that focal-plane reflector positioning is preferable for source radii-of-curvature $\leq$ d.

**3:10**

**5pPA8. Finite element studies to understand reflection of nonlinear waves in a continuous functionally graded material.** Pravinkumar R. Ghodake (Dept. of Mech. Eng., Indian Inst. of Technol. Bombay, Mumbai, Maharashtra 400076, India, mech7pkumar@gmail.com)

The nonlinear pulse-echo ultrasonics technique is effective when only one side of pressure vessels mainly used in thermal and nuclear power plants is assessable. As another side is exposed to continuous thermal and nuclear damages, the early-stage damages accumulate as a function of position. Concluding experimental results from such a complex interaction of reflected nonlinear waves during pulse-echo testing is difficult. Obtaining analytical solutions to the reflected waves from a functionally graded nonlinear region with an extended linear part is challenging. This complexity in understanding is resolved through various computational studies presented in this article. The linear and exponential variation of nonlinear parameters, loss in elastic stiffness, and density of the damaged region are modeled by updating linear and nonlinear constitutive equations. The results demonstrate higher energy of reflected longitudinal harmonic waves (2f, 3f, 4f,….) from a fixed boundary than a free boundary. Amplitudes of higher harmonic responses as a function of the position are verified. Harmonically reflected higher harmonics are more sensitive to the spatial distribution of nonlinear parameters than the loss in linear elastic stiffness, whereas a significant change in the overall shapes of the pulses in time domains is observed. Self-wave mixing increases the number of higher harmonics.

**3:25**

**5pPA9. Time-varying elastic wave mode conversion in vibrating elastic beams with subwavelength nonlinearity.** Samuel D. Parker (Walker Dept. of Mech. Eng., The University of Texas at Austin, Austin, TX 78712, sdparker@utexas.edu), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Daniel R. Roettgen (Structural Dynam., Sandia National Labs., Albuquerque, NM)

Introduction of time-varying properties in a background medium can enable unconventional wave behavior. Furthermore, it is well-known that subjecting nonlinear materials to time-varying stress can be used for material characterization [Appl. Phys. Lett. **94**, 011905 (2009)]. We propose a technique called Dynamic Asymmetric Transmission Measurement (DATM) that combines structural dynamic (SD) and ultrasonic (US) testing modalities in an elastic waveguide to detect and characterize changes in the global stress state of a structure that result in local time-varying stress conditions due to local nonlinearity. Asymmetric geometric features in beams and plates, paired with time-varying stress conditions, result in asymmetric mode conversion of guided US waves that depend on large-scale structural dynamics. The DATM technique is explored through finite element modeling, semi-analytical methods, and experiments and discussed in a structural health monitoring context. We also discuss the use of dynamic structures with engineered defects to introduce time-varying stress conditions for the purpose of manipulating elastic waves. [Sandia National Laboratories is a multimission laboratory managed and operated by the National Technology & Engineering Solutions of Sandia, LLC, a wholly owned subsidiary of Honeywell International Inc., for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-NA0003525.]

**3:40**

**5pPA10. A technique for measurement of ultrasonic waves propagating in time-varying media.** Samuel D. Parker (Walker Dept. of Mech. Eng., The University of Texas at Austin, Austin, TX 78712, sdparker@utexas.edu), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Daniel R. Roettgen (Structural Dynam., Sandia National Labs., Albuquerque, NM)

Laser Doppler Vibrometry (LDV) is often used in nondestructive testing to make high-fidelity, non-contact structural dynamic measurements. However, synchronized measurement on slow, structural dynamics (SD) and fast, ultrasonic (US) time scales can yield additional useful information about the overall condition of elastic structures. This work presents a technique, Dynamic Asymmetric Transmission Measurement (DATM), that enables time-aligned measurements of large-scale structural dynamics via LDV and small-scale ultrasonic signals to provide information at a particular global dynamic state of an elastic structure. This approach is applicable to structural health monitoring and generalizable to the measurement of guided ultrasonic waves propagating in structural components subjected to time-varying stress states. We demonstrate the utility of DATM by detecting and characterizing time-varying stress conditions caused by local nonlinearity due to surface-breaking cracks resulting in time-varying asymmetric transmission of US signals in beam and plate structures. The test methodology to synchronize LDV and US data is presented together with example measurements. [Sandia National Laboratories is a multimission laboratory managed and operated by National Technology & Engineering Solutions of Sandia, LLC, a wholly owned subsidiary of Honeywell International Inc., for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-NA0003525.]

## Session 5pUW

## Underwater Acoustics: General Topics in Underwater Acoustics II

Christina Frederick, Chair
*Department of Mathematical Sciences, NJIT, 323 M. L. King Boulevard, Newark, NJ 07102*

### *Contributed Papers*

1:00

**5pUW1. Seabed classification and source localization with Gaussian processes and machine learning.** Christina Frederick (Dept. of Mathematical Sci., NJIT, 323 M. L. King Boulevard, Newark, NJ 07102, christin@njit.edu) and Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ)

Workshop ’97 data are employed for seabed classification and source range estimation. The data are acoustic fields computed at vertically separated receivers for various ranges and three different environments. Gaussian Processes are applied for denoising the data and predicting the field at virtual receivers, sampling the water column densely within the array aperture. The enhanced fields are then used in combination with machine learning in order to map the signals to one of 15 sediment-range classes (corresponding to three environments and five ranges). The classification results after using Gaussian Processes for denoising are superior to those when noisy workshop data are employed.

1:15

**5pUW2. Machine learning for point-to-point transmission loss estimates in ocean acoustic waveguides.** Mark Kelly (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr NW, Atlanta, GA 30318, mkelly75@gatech.edu), Brian O’Donnell (Georgia Tech Res. Inst., Atlanta, GA), Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Marsal Bruna (Georgia Tech Res. Inst., Atlanta, GA)

Underwater acoustic system performance depends on several complex and dynamic environmental parameters, and simulating such performance is vital to the success of development and implementation of these systems. Because of the complexity of the environment and governing physical equations, realistic simulations can become computationally prohibitive. This is especially true of for large environments with many active systems being assessed. By utilizing convolutional neural networks (CNNs) trained on data generated by well-established physics based models (such as BELLHOP’s ray tracing algorithm), network predictions can be used lieu of physics-based models to significantly reduce the computational burden in the loop for system performance simulations. In this paper, the usefulness and limitations of using CNNs to estimate transmission loss (TL), which is a key element in determining system performance, is explored. Using BELLHOP’s ray tracing algorithm as a baseline, CNN’s were able to produce TL results with significantly lower errors than those estimates made using other estimation methods such as spherical spreading and K-nearest neighbors. This indicates that the computational costs of large underwater acoustic simulations may be shifted from inside the simulation to network training, thus allowing for more efficient traditional and Monte Carlo style simulations.

1:30

**5pUW3. Investigating cepstral methods for blind deconvolution.** Alexander S. Douglass (Oceanogr., Univ. of Washington, 1501 NE Boat St., MSB 206, Seattle, WA 98195, asd21@uw.edu) and Shima Abadi (Oceanogr., Univ. of Washington, Seattle, WA)

Cepstral methods are homomorphic signal processing techniques in which signals are transformed into the cepstral domain, typically to be averaged or filtered. In the cepstral domain, the convolution of the source signal and impulse response changes to a sum of the two parts by taking a logarithm in the frequency domain, followed by an inverse Fourier transform. Cepstral methods have been utilized in speech processing to separate vocal tract information from speech excitation, in marine mammal bioacoustics to classify vocalizations, and in seismic surveys to estimate source wavelets from airgun arrays, but do not appear to have been considered for blind deconvolution in other underwater acoustic applications. By assuming either the source signal or impulse response is stationary in time and/or space, averaging cepstral domain windows from a receiver array can suppress the non-stationary term. Here, we investigate the capabilities of cepstral averaging techniques for the purposes of blind deconvolution. We will exploit the spatial dependence of the impulse response across a receiver array to suppress this non-stationary term, while constructively combining the source signal term. We will consider the effects of different signal types, environmental characteristics, signal-to-noise ratios, and array design on its success. [Work supported by the ONR.]

1:45

**5pUW4. Automatic detection and classification of baleen and toothed whale calls via machine learning approaches over instantaneous wide areas in the Gulf of Maine received on a coherent hydrophone array.** Hamed Mohebbi-Kalkhoran (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, mohebbikalkhoran.h@northeastern.edu) and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Large acoustic data sets are typically generated from ocean observations with a 160-element coherent hydrophone array and correspondingly larger volumes of acoustic detection events stem from coherent array processing. Beamforming enhances detection signal to-noise ratio, significantly improving detection ranges, as well as providing signal bearing. Here, we develop and train algorithms for the automatic detection and classification of baleen and toothed whale calls present in multiple beamformed spectrograms spanning 360 degree azimuths generated via the passive ocean acoustic waveguide remote sensing technique in the following six categories for the Gulf of Maine: Fin, Sei, Minke, Humpback, unidentified baleen whale downsweep chirps, and general toothed whale encompassing echolocation clicks and whistles below 4 kHz. The classifiers include random forest, support vector machine (SVM), and decision tree applied to hand-engineered features, as well as Convolutional Neural Network (CNN)-based model on the per-channel energy normalization transform (PCEN) applied directly to beamformed spectrogram imagery. Total accuracy of 95% and average F1-score of 85% are achieved using random forest classifier. The processing flow, including beamforming, PCEN extraction and call classification, run in real-time making the methods suitable for real-world applications, such as marine mammal monitoring and mitigation in ocean hydrocarbon prospecting and wind farm installations.

**2:00**

**5pUW5. Underwater acoustic classification using masked modeling-based swin transformer.** Kang you (Tonguing Univ., Shanghai, China), Kele Xu (National Key Lab. of Parallel and Distributed Processing (PDL), Changsha, China, 107, Yanwachi, Changsha 410073, China, kelele.xu@Gmail.com), Ming Feng (Tonguing Univ., Shanghai, China), and Boqing Zhu (National Key Lab. of Parallel and Distributed Processing (PDL), Changsha, China, Changsha, China)

Underwater acoustic classification is a challenging task due to complex background noise and complicated sound propagation patterns. How to represent the signals is important for the classification task. In this paper, we propose a novel representation learning method for the underwater acoustic signals, leveraging the mask modeling-based self-supervised learning paradigm. Specifically, we first explore modifying the Swin Transformer architecture to learn general representation for the audio signals, accompanied with random masking on the log-mel spectrogram. The main goal of the pretext task is to predict the masked parts of Log-mel spectrogram and the gamma-stone spectrogram, so that the model can not only learn the local and global features but also learn complementary information. For downstream task, we utilize the labelled datasets to fine-tune the pre-trained model. On DeepShip datasets which consist of 47 hand 4 minof ship sounds in four categories, our model achieves state-of-the-art performance compared with competitive approaches. Our method obtains a classification accuracy of 78.03%, which is better than the separable convolution autoencoder (SCAE) and using the constant-Q transform spectrogram. This work demonstrates the potential of the masked modeling based self-supervised learning for understanding and interpretation of underwater acoustic signals.

**2:15–2:30 Break**

**2:30**

**5pUW6. An end-capped lead zirconate titanate broadband hydrophone theoretical calculation and electroacoustic measurement for towed array applications.** Matthew E. Schinault (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave. 126 Egan / 409 Dana M.S., Boston, MA 02115, schinault.m@husky.neu.edu) and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

Hydrophone transducers for towed array applications require low self-noise, small form factor and a linear frequency response below 50 kHz for general purpose long-range ocean sensing. The design specification for this hydrophone application has several aspects including acoustic sensitivity, frequency response, dissipation factor, and physical size. A comprehensive noise model informs filter design for pre-amplifiers after the piezoelectric element. Theoretical calculation of pressure to voltage conversion is made here by estimating the Free-Field Voltage Sensitivity (FFVS) using methods from R.A. Langevin and G.W. McMahon. Using these methods to achieve the desired design specification, we use two cylindrical ceramics with a split electrode poled in the radial direction with series stacked ceramics using end caps with air backing to increase sensitivity and reduce overall size. An experiment is carried out to measure the resonant modes using a simple impedance measuring circuit which then can be used to estimate FFVS linearity. These results are compared with theoretical calculation for hydrophone response of resonant modes and results from measurement circuit.

**2:45**

**5pUW7. Development of a large-aperture 160-element coherent hydrophone array system for instantaneous wide area ocean acoustic sensing.** Max Radermacher (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave. Boston, MA 02115, radermacher.m@northeastern.edu), Sai Geetha Seri, Matthew E. Schinault, Hamed Mohebbi-Kalkhoran (Elec. and Comput. Eng., Northeastern Univ., Boston, MA), Chenyang Zhu, Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

A large-aperture coherent hydrophone array and processing system has been developed in-house at Northeastern University (NU). The overall acoustic aperture length is 194 m with 160-elements nested into four subapertures each with 64 hydrophones and spacing corresponding to half-wavelength at design frequencies in Hertz of 250, 500, 1000, and 2000. Hydrophones with integrated broadband preamplifiers designed with a linear frequency response from 10 to 50 kHz sending differential pair amplified and filtered analog signals to 24-bit, 32-channel Analog to Digital Converters (ADC) with sampling rate programmable up to 100 kHz per channel. Array internals are designed using field replaceable pressure tolerant components including hydrophones, preamplifiers, power modules, telemetry units, and ADC verified by testing in an in-house assembled pressure chamber. Forward and aft modules are equipped with non-acoustic sensor elements to provide depth, heading, pitch, roll, and temperature measurements. The NU array was made without specialized facilities by utilizing modular interchangeable array interconnects allowing for conventional array populating and filling methods. The system implements real-time array data acquisition and processing including beamforming, beamformed spectrogram generation and signal detections. The NU array has been sea tested in 2021 in both shallow and deep water environments of the US Northeast coast.

**3:00**

**5pUW8. Adapting ocean surface scattering models to accept *in-situ* environmental data.** Anthony Eller (Appl. Ocean Sci., 5242 Port Royal Rd., Springfield, VA 22151, anthony.eller@appliedoceansciences.com), Paul Hursky (Appl. Ocean Sci., Springfield, VA), and Kevin D. Heaney (Appl. Ocean Sci., Fairfax Station, VA)

A renewed examination of the theoretical basis of rough sea surface scattering has unearthed uncertainty over what theoretical approach is best suited for realtime applications, as evaluated in terms of speed, accuracy, and ability to allow greater use of *in situ* environmental data. The investigation focused primarily on applied models, as related to signal scattering losses and to reverberation modeling. Modeling approaches compared include perturbation theory, the approach by Eckart, and the approaches examined at the University of Washington and the Naval Research Laboratory. Validation by comparisons to data will be cited as available. A design goal of many models is to require only a small number of easily observed environmental parameters, such as wind speed and significant wave height. This presentation explores the insertion of additional inputs that would support predictions more relevant to the specific time and place of the application. These inputs would include measured or simulated surface wave spectrum data and estimates of surface height coherence length.

**3:15**

**5pUW9. Refining the metavariables of convolutional autoencoders for better performance on littoral acoustic backscattering data.** Timothy Linhardt (College of Elec. and Comput. Eng., Univ. of Iowa, 103 South Capitol St., Iowa City, IA 52240, tlinhardt@uiowa.edu) and Ananya Sen Gupta (Dept. of Elec. and Comput. Eng., Univ. of Iowa, Iowa City, IA)

Convolutional autoencoders have been used in an effort to learn latent encodings of littoral accoustic backscatter, but previous work did not explore a wide range of metavariable variation. Only the dimension of the encoding space, and the learning rate of the training optimizer were adjusted to achieve a functioning initial result. The other metavariables were assigned with some justification but optimality was not verified. We explore the effects of changing the method by which random weights are initialized as well as doubling the amount of random kernels per convolutional layer of the autoencoder. Additionally, we vary the layer count of the network, and the shapes of the kernels in the layers. Finally, various nonlinear activations are compared against the activation used initially. The results based on experimental field data from the TREX13 experiment will be presented. [This work is sponsored by ONR grant N00174-20-1-0016.]

**3:30**

**5pUW10. Hologram formation by using vertical antenna in a shallow water waveguide.** Sergey A. Pereselkov (Mathematical Phys. and Information Technologies Dept., Voronezh State Univ., Russia, Voronezh, Universitetskay pl, 1, Voronezh 394018, Russian Federation, pereselkov@yandex.ru), Venedikt Kuz'kin (Hydrophysics Lab., Prokhorov General Phys. Inst. of the Russian Acad. of Sci., Moscow, Russian Federation), Ilya Kaznacheev, Sergey Tkachenko, and Pavel Rybyanets (Mathematical Phys. and Information Technologies Dept., Voronezh State Univ., Voronezh/Russia, Russian Federation)

Holographic signal processing method by using a vertical linear antenna in shallow water is presented in the paper. Offered method of moving source sound field hologram formation by using vertical antenna is based on holographic signal processing of each element of vertical antenna. Sound field of source moving in shallow water waveguide creates a stable interference pattern (interferogram) in the frequency-time domain on each element of vertical antenna. The 2D-FT (2D Fourier transform) of the interferogram is antenna element hologram (AEH). AEH allows to coherently accumulate the sound intensity of the interferogram in a relatively small area as focal spots for each element of antenna. It is shown that the focal spots coordinates on AEH depends upon the source range, velocity, and motion direction. The focal spots coordinates on AEH are same for each element of antenna. The output antenna hologram is formed by superposition AEH. Relationship between focal spot coordinates on the antenna hologram and the source range, velocity, and motion direction are derived in the paper. The results of the numerical experiment of antenna hologram formation for low-frequency (100200 Hz) sound field of source moving in shallow water are presented. The results presented in the paper significantly expand the efficiency of interferometric signal processing in shallow water. [This study was supported by RFBR 19-29-06075, MK-6144.2021.4, and MK-4846.2022.4.]

# ETHICAL PRINCIPLES OF THE ACOUSTICAL SOCIETY OF AMERICA FOR RESEARCH INVOLVING HUMAN AND NON-HUMAN ANIMALS IN RESEARCH AND PUBLISHING AND PRESENTATIONS

**The Acoustical Society of America (ASA) has endorsed the following ethical principles associated with the use of human and non-human vertebrate animals in research, and for publishing and presentations. The principles endorsed by the Society primarily follow the form of those adopted by the American Psychological Association (APA), along with excerpts borrowed or modified from the Council for International Organizations of Medical Sciences (CIOMS) and International Council for Laboratory Animal Science (ICLAS), and the American Institute of Physics Publishing (AIPP). The ASA acknowledges the difficulty in making ethical judgments, but the ASA wishes to set minimum socially accepted ethical standards for publishing in its journals and presenting at its meetings. These Ethical Principles are based on the principle that the individual author or presenter bears the responsibility for the ethical conduct of their research and its publication or presentation.**

**Authors of manuscripts submitted for publication in a journal of the ASA or presenting a paper at a meeting of the Society are obligated to follow the ethical principles of the Society. Failure to accept the ethical principles of the ASA shall result in the immediate rejection of manuscripts and/or proposals for publication or presentation. False indications of having followed the Ethical Principles of the ASA may be brought to the Ethics and Grievance Committee of the ASA.**

## I. USE OF HUMAN SUBJECTS IN RESEARCH-Applicable when human subjects are used in the research

The ASA endorses the view that all research involving human subjects requires approval by an existing appropriate governing authority (e.g., institutional review board [IRB], Health Insurance Portability and Accountability Act [HIPPA], or by other governing authorities used in many countries) whose policies are consistent with the Ethical Principles of the ASA and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, the research should have met the following criteria:

### Informed Consent

When obtaining informed consent from prospective participants in a research protocol, authors must have clearly and simply specified to the participants beforehand:

1. The purpose of the research, the expected duration of the study, and all procedures that were to be used.
2. The right of participants to decline to participate and to withdraw from the research in question after participation began.
3. The foreseeable consequences of declining or withdrawing from a study.
4. Anticipated factors that may have influenced a prospective participant's willingness to participate in a research project, such as potential risks, discomfort, or adverse effects.
5. All prospective research benefits
6. The limits of confidentialit .
7. Incentives for participation.
8. Whom to contact for questions about the research and the rights of research participants. That office/person must have willingly provided an atmosphere in which prospective participants were able to ask questions and receive answers.

Authors conducting intervention research involving the use of experimental treatments must have clarified, for each prospective participant, the following issues at the outset of the research:

1. The experimental nature of the treatment;
2. The services that were or were not to be available to the control group(s), if appropriate;
3. The means by which assignment to treatment and control groups were made;
4. Available treatment alternatives if an individual did not wish to participate in the research or wished to withdraw once a study had begun; and
5. Compensation for expenses incurred as a result of participating in a study including, if appropriate, whether reimbursement from the participant or a third-party payer was sought.

### Informed Consent for Recording Voices and Images in Research

Authors must have obtained informed consent from research participants prior to recording their voices or images for data collection unless:

1. The research consisted solely of naturalistic observations in public places, and it was not anticipated that the recording would be used in a manner that could have caused personal identification or harm, o
2. The research design included deception. If deceptive tactics were a necessary component of the research design, consent for the use of recordings was obtained during the debriefing session

### Client/Patient, Student, and Subordinate Research Participants

When authors conduct research with clients/patients, students, or subordinates as participants, they must have taken steps to protect the prospective participants from adverse consequences of declining or withdrawing from participation.

### Dispensing with Informed Consent for Research

Authors may have dispensed with the requirement to obtain informed consent when:

1. It was reasonable to assume that the research protocol in question did not create distress or harm to the participant and involves:
   a. The study of normal educational practices, curricula, or classroom management methods that were conducted in educational settings
   b. Anonymous questionnaires, naturalistic observations, or archival research for which disclosure of responses would not place participants at risk of criminal or civil liability or damage their financial standing, employability, or reputation, and confidentialit
   c. The study of factors related to job or organization effectiveness conducted in organizational settings for which there was no risk to participants' employability, and confidentialit .
2. Dispensation is permitted by law.
3. Research involving the collection or study of existing data, documents, records, pathological specimens, or diagnostic specimens, if these sources are publicly available or if the information is recorded by the investigator in such a manner that subjects cannot be identified, directly or through identifiers lin ed to the subjects.

### Offering Inducements for Research Participation

(a) Authors must not have made excessive or inappropriate financial or other inducements for research participation when such inducements are likely to coerce participation.

(b) When offering professional services as an inducement for research participation, authors must have clarified the nature of the services, as well as the risks, obligations, and limitations.

### Deception in Research

(a) Authors must not have conducted a study involving deception unless they had determined that the use of deceptive techniques was justified by the study's significant prospective scientific, educational, or applied value and that effective non-deceptive alternative procedures were not feasible.

(b) Authors must not have deceived prospective participants about research that is reasonably expected to cause physical pain or severe emotional distress.

(c) Authors must have explained any deception that was an integral feature of the design and conduct of an experiment to participants as early as was feasible, preferably at the conclusion of their participation, but no later than at the conclusion of the data collection period, and participants were freely permitted to withdraw their data.

### Debriefing

(a) Authors must have provided a prompt opportunity for participants to obtain appropriate information about the nature, results, and conclusions of the research project for which they were a part, and they must have taken reasonable steps to correct any misconceptions that participants may have had of which the experimenters were aware.

(b) If scientific or humane values justified delaying or withholding relevant information, authors must have taken reasonable measures to reduce the risk of harm.

(c) If authors were aware that research procedures had harmed a participant, they must have taken reasonable steps to have minimized the harm.

## II. HUMANE CARE AND USE OF NON-HUMAN VERTEBRATE ANIMALS IN RESEARCH-Applicable when non-human vertebrate animals are used in the research

The advancement of science and the development of improved means to protect the health and well-being of both human and non-human vertebrate animals often require the use of animals in research, education, and testing. The ASA remains committed to ensuring the health and welfare of vertebrate animals used for these purposes. Vertebrate animal experiments should have been undertaken only after due consideration of the relevance for health, conservation, and the advancement of scientific knowledge. (Modified from the Council for International Organizations of Medical Sciences (CIOMS) and International Council for Laboratory Animal Science (ICLAS) document: "International Guiding Principles for Biomedical Research Involving Animals-2012").

The ASA endorses the view that all research involving non-human vertebrate animals, hereinafter referred as "animals," requires approval by an existing appropriate governing authority (e.g., an institutional animal care and use committee [IACUC]) whose policies are consistent with the Ethical Principles of the ASA and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, the research should meet the following criteria:

1. Animals have been used only when necessary and when no alternative methods, such as non-animal approaches, mathematical models, or computer simulation, are available to achieve the scientific goals
2. Investigators have handled all animals in compliance with all current federal, state, and local laws and regulations, and with professional standards.
3. Investigators have made all reasonable efforts to minimize the number of animals used in research to achieve the scientific goals.
4. Investigators are experienced in the care of laboratory animals, supervise all procedures involving animals, ensure all subordinates who use animals have received proper training in methodology and animal care, and assume responsibility for the comfort, health, and humane treatment of experimental animals under all circumstances.
5. The health and welfare of animals are the primary considerations in making decisions of animal care including acquisition, housing, veterinary care, and final disposition of animals.
6. All surgical procedures have been conducted under appropriate anesthesia and followed techniques that avoided infection and minimized pain during and after surgery.
7. Investigators have made all reasonable efforts to monitor and mitigate any possible adverse effects to animals as a result of the experimental protocol. Strategies to manage, mitigate, and minimize any pain and/or distress in animals should be developed in consultation with a qualified veterinarian or scientist. Animals that suffer chronic pain, distress or discomfort that cannot be relieved should be removed from the study and/or euthanized using a procedure appropriate for the species and condition of the animal.
8. Investigators proceed to rapidly and humanely terminate an animal's life when it is necessary and appropriate, always minimizing pain and always in accordance with accepted procedures as determined by a veterinarian and/or appropriate review board.

## III. PUBLICATION and PRESENTATION ETHICS-For publications in ASA journals and presentations at ASA sponsored meetings

### Statement of Ethics and Responsibilities

The mission of the ASA is to generate, disseminate, and promote the knowledge and practical applications of acoustics. To that end, it is essential that all authors of papers in ASA journals and presentors at ASA-sponsored meetings conduct themselves in accord with the highest level of professional ethics and standards.

By submitting a manuscript to an ASA journal, each author explicitly confirms that the manuscript meets the highest ethical standards. The same is required for material presented at meetings. Authors submitting to ASA journals should also adhere to the policies included in the particular journals' Instructions for Contributors.

This section is mainly based on the policies of the American Institute of Physics Publishing.

### Plagiarism

Plagiarism is the unauthorized and unacknowledged use of someone else's words, ideas, processes, data, or results in a manner that can mislead others into thinking the material is your own. Plagiarism can also be in the form of text recycling, also called self-plagiarism, where an author reuses portions of text from their own work that isn't properly credited. Plagiarism or self-plagiarism constitutes unethical scientific behavior and is never acceptable.

### Publication Credit

Authorship should be limited to those who have made a significant contribution to the concept, design, execution or interpretation of the research study. All those who have made significant contributions should be offered the opportunity to be listed as authors. The author who submits a paper for publication or an abstract for presentation and publication should ensure that all coauthors have seen the final version of the paper or abstract and have agreed to its submission. Other individuals who have contributed to the study should be acknowledged, but not identified as authors.

Proper acknowledgment of the work of others used in a research project must always be given. Information obtained privately, as in conversation, correspondence, or discussion with third parties, should not be used or reported without explicit permission from the investigator with whom the information originated. Information obtained in the course of confidential services, such as refereeing manuscripts or grant applications, cannot be used without permission of the author of the work being used.

Authors must obtain permission when reproducing or adapting any previously published materials from the original copyright holder. Proper credit lines for all previously published material must be included in the manuscript.

### Reporting Research Results

The results of research should be recorded and maintained in a form that allows analysis and review, both by collaborators before publication and by other scientists for a reasonable period after publication. Exceptions may be appropriate in certain circumstances in order to preserve privacy, to assure patent protection, or for similar reasons.

### Reporting Errors in Publication

All coauthors have an obligation to provide prompt retractions or correction of errors in published works.

### Fabrication of Data and Selective Reporting of Data

Fabrication of data is an egregious departure from the expected norms of scientific conduct, as is the selective reporting of data with the intent to mislead or deceive, as well as the theft of data or research results from others.

### Disclosure of Conflicts of Interest

A conflict of interest is anything that interferes with, or could reasonably be perceived as interfering with, the full and objective presentation of articles in the ASA journals and presentations at the ASA meetings. Author(s) have the obligation to disclose any personal interest or relationship that has the potential to be affected by publication of the submitted manuscript or presentation at ASA meeting:

1. The complete affiliation(s) of each author and sources of funding for the published or presented research should be clearly described in the paper or publication abstract.
2. If the publication or presentation of the research would directly lead to the financial gain of the author(s), then a statement to this effect must appear in the acknowledgment section of the paper or presentation abstract or

in a footnote of a paper. Authors must report any financial interest in corporate or commercial entities dealing with the subject matter of the manuscript or presentation.

3. If the research that is to be published or presented is in a controversial area and the publication or presentation presents only one view in regard to the controversy, then the existence of the controversy and this view must be provided in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper. It is the responsibility of the author to determine if the paper or presentation is in a controversial area and if the person is expressing a singular view regarding the controversy.

Authors must submit corrections if conflicts of interests are revealed after publication.

Approved by the Executive Council on 9 December 2019.

# Sustaining Members of the Acoustical Society of America

The Acoustical Society is grateful for the financial assistance being given by the Sustaining Members listed below and invites applications for sustaining membership from other individuals or corporations who are interested in the welfare of the Society.

Application for membership may be made to the Executive Director of the Society and is subject to the approval of the Executive Council. Dues of $1000.00 for small businesses (annual gross below $100 million) and $2000.00 for large businesses (annual gross above $100 million or staff of commensurate size) include a subscription to the *Journal* as well as a yearly membership certificate suitable for framing. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year.

Additional information and application forms may be obtained from Elaine Moran, Office Manager, Acoustical Society of America, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: elaine@acousticalsociety.org

### Acentech Incorporated
*www.acentech.com*
Cambridge, Massachusetts
Consultants in Acoustics, Audiovisual and Vibration

### ACO Pacific Inc.
*www.acopacific.co*
Belmont, California
Measurement Microphones, the ACOustic Interface™ System

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*www.acousticsfi st.com*
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Career resources, undergraduate education, science policy, and history

### BBN Technologies
*www.bbn.com*
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R&D company providing custom advanced research based solutions

### GRAS Sound and Vibration
*www.gras.us*
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Measurement microphones, intensity probes, calibrators

### Kinetics Noise Control, Inc.
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Kinetics manufactures products to address vibration and noise control, room acoustics, and seismic restraint concerns for almost any building application

### Massa Products Corporation
*www.massa.com*
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Indianapolis, Indiana
An Association of Independent Firms Consulting in Acoustics

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Products for personal and environmental safety, featuring E·A·R and Peltor brand hearing protection and fit testing, Quest measurement instrumentation, audiological devices, materials for control of noise, vibration, and mechanical energy, and the E·A·RCALSM laboratory for research, development, and education, NVLAP-accredited since 1992.
**Hearing conservation resource center**
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*www.wengercorp.com*
Owatonna, Minnesota
Design and Manufacturing of Architectural
Acoustical Products including Absorbers, Diffusers, Modular Sound
Isolating Practice Rooms, Acoustical Shells and Clouds for Music
Rehearsal and Performance Spaces

### Wyle Laboratories
*www.wyle.com*
Arlington, Virginia
The Wyle Acoustics Group provides a wide range of professional services focused on acoustics, vibration, and their allied technologies, including services to the aviation industry

ACOUSTICAL · SOCIETY · OF · AMERICA

# APPLICATION FOR SUSTAINING MEMBERSHIP

The Bylaws provide that any person, corporation, or organization contributing annual dues as fixed by the Executive Council shall be eligible for election to Sustaining Membership in the Society.

Dues have been fixed by the Executive Council as follows: $1000 for small businesses (annual gross below $100 million); $2000 for large businesses (annual gross above $100 million or staff of commensurate size). Dues include one year subscription to *The Journal of the Acoustical Society of America* and programs of Meetings of the Society. Please do not send dues with application. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year. If elected, you will be billed.

Name of Company ______________________________

Address ______________________________

Telephone: ______________________________ Fax: ______________________________

E-mail: ______________________________ WWW: ______________________________

Size of Business: ☐ Small business ☐ Small business—No Journal ☐ Large business

Type of Business ______________________________

**Please enclose a copy of your organization's brochure.**

In listing of Sustaining Members in the Journal and on the ASA homepage we should like to indicate our products or services as follows:

______________________________

(please do not exceed fifty characters)

Name of company representative to whom journal should be sent:

______________________________

It is understood that a Sustaining Member will not use the membership for promotional purposes.

Signature of company representatives making application:

______________________________

Please send completed applications to: Executive Director, Acoustical Society of America, 1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300, (516) 576-2360, asa@acousticalsociety.org

## MEMBERSHIP INFORMATION AND APPLICATION INSTRUCTIONS

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of the application; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out arts I and III.

| **BENEFITS OF MEMBERSHIP** | full Member | Associate | ce-Associate | Student |
|---|---|---|---|---|
| *JASA* Online–Vol. 1 (1929) to present | * | * | * | * |
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| Society Membership Directory | Online | Online | Online | Online |
| Electronic Announcements | * | * | * | * |
| *Physics Today* | * | * | * | * |
| Eligibility to vote and hold office in AS | * | | | |
| Eligibility to be elected Fellow | * | * | | |
| Participation in ASA Committees | * | * | * | * |

### QUALIFICATIONS FOR EACH GRADE OF MEMBERSHIP AND ANNUAL DUES

**Student:** Any student interested in acoustics who is enrolled in an accredited college or university for half time or more (at least eight semester hours). Dues: $50 per year.
**Associate:** Any individual interested in acoustics. Dues: $115 per year. After fi e years, the dues of an Associate increase to that of a full Member.
**Corresponding Electronic Associate:** Any individual residing in a developing country who wishes to have access to ASA's online publications only including *The Journal of the Acoustical Society of America* and Meeting Programs [see http://acousticalsociety.org/membership/membership_and_benefit ]. Dues $50 per year.
**Member:** Any person active in acoustics, who has an academic degree in acoustics or in a closely related field or who has had the equivalent of an academic degree in scientific or professional experience in acoustics, shall be eligible for election to Membership in the Society. A nonmember applying for full Membership will automatically be made an interim Associate Member, and must submit $115 with the application for the first year's dues. Election to full Membership may require six months or more for processing; dues as a full Member will be billed for subsequent years.

### JOURNAL OPTIONS AND COSTS FOR FULL MEMBERS AND ASSOCIATE MEMBERS ONLY

- ONLINE JOURNAL. All members will receive access to the *The Journal of the Acoustical Society of America (JASA)* at no charge in addition to dues.
- PRINT JOURNAL. Twelve monthly issues of *The Journal of the Acoustical Society of America*. **Cost: $35 in addition to dues.**
- **EFFECTIVE DATE OF MEMBERSHIP**. If your application for membership and dues payment are received by 15 September, your membership and Journal subscription will begin during the current year and you will receive all back issues for the year. If you select the print journal option. If your application is received after 15 September, however, your dues payment will be applied to the following year and your Journal subscription will begin the following year.

### OVERSEAS AIR DELIVERY OF JOURNALS

Members outside North, South, and Central America can choose to have print journals sent by air freight at a cost of $185 in addition to dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

# ACOUSTICAL SOCIETY OF AMERICA

**1305 Walt Whitman Road, Suite 110, Melville, NY 11747-4300, asa@acousticalsociety.org**

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## APPLICATION FOR MEMBERSHIP

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of this form; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

### PART I. TO BE COMPLETED BY ALL APPLICANTS (Please print or type all entries)

| | | | |
|---|---|---|---|
| CHECK ONE BOX IN EACH COLUMN ON THE RIGHT | ☐ NON-MEMBER APPLYING FOR:<br>☐ MEMBER REQUESTING TRANSFER TO: | ☐ STUDENT MEMBERSHIP<br>☐ ASSOCIATE MEMBERSHIP<br>☐ CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP<br>☐ FULL MEMBERSHIP | Note that your choice of journal option *may* increase or decrease the amount you must remit. |

### SELECT JOURNAL OPTION:

**Student members** will automatically receive access to The Journal of the Acoustical Society of America online at no charge in addition to dues. Remit $45.

**Corresponding Electronic Associate Members** will automatically receive access to The Journal of the Acoustical Society of America and Meeting Programs online at no charge in addition to dues. Remit $50.

Applicants for **Associate or full Membership** must select one Journal option from those listed below. Note that your selection of journal option determines the amount you must remit.

[ ] Online access only—$150
[ ] Online access plus print Journal $185

Applications received after 15 September: Membership and Journal subscriptions begin the following year.

**OPTIONAL AIR DELIVERY:** Applicants from outside North, South, and Central America may choose air freight delivery of print journals for an additional charge of $185. If you wish to receive journals by air, remit the additional amount owed with your dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

| | | | |
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| HOME ADDRESS (STREET & NUMBER) | | | |
| CITY | STATE OR PROVINCE | ZIP OR POSTAL CODE | COUNTRY |
| NAME OF ORGANIZATION OR BUSINESS | | | |
| DEPARTMENT | | | |
| ORGANIZATION ADDRESS (STREET & NUMBER) | | | |
| CITY | STATE OR PROVINCE | ZIP OR POSTAL CODE | COUNTRY |
| BUSINESS TELEPHONE: AREA CODE/NUMBER | FAX: AREA CODE/NUMBER | HOME TELEPHONE: AREA CODE/NUMBER | |
| E-MAIL ADDRESS: (PRINT CLEARLY) | MOBILE PHONE: AREA CODE/NUMBER | | |
| DATE AND PLACE OF BIRTH (Req'd for Awards and Emeritus Status) | SEX: ☐ Female ☐ Male ☐ Non-Binary ☐ Transgender ☐ Prefer not to answer ☐ | | |
| HIGHEST ACADEMIC DEGREE | DATE OF DEGREE | FIELD | INSTITUTION GRANTING DEGREE |
| OTHER DEGREE | MONTH/YEAR | FIELD | INSTITUTION GRANTING DEGREE |

CHECK PERFERRED ADDRESS FOR MAIL: ☐ HOME ☐ ORGANIZATION

**Part I Continued ➔**

**PART I CONTINUED:** ACOUSTICAL AREAS OF INTEREST TO APPLICANT. Indicate your three main areas of interest below, using 1 for your main interest, 2 for your second, and 3 for your third interest. (DO NOT USE CHECK MARKS.)

- ☐ ACOUSTICAL OCEANOGRAPHY **M**
- ☐ ANIMAL BIOACOUSTICS **L**
- ☐ ARCHITECTURAL ACOUSTICS **A**
- ☐ BIOMEDICAL ACOUSTICS **K**
- ☐ COMPUTATIONAL ACOUSTICS **O**
- ☐ ENGINEERING ACOUSTICS **B**
- ☐ MUSICAL ACOUSTICS **C**
- ☐ NOISE & NOISE CONTROL **D**
- ☐ PHYSICAL ACOUSTICS **E**
- ☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
- ☐ SIGNAL PROCESSING IN ACOUSTICS **N**
- ☐ SPEECH COMMUNICATION **H**
- ☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
- ☐ UNDERWATER ACOUSTICS **J**

## PART II: APPLICATION FOR STUDENT MEMBERSHIP

| | | |
|---|---|---|
| NAME AND ADDRESS OF COLLEGE OR UNIVERSITY WHERE PRESENTLY ENROLLED | | |
| DEGREE EXPECTED | MONTH & YEAR DEGREE EXPECTED | NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER |
| PRINT NAMES & E-MAIL ADDRESSES OF TWO FACULTY MEMBERS CERTIFYING THAT YOU ARE REGISTERED FOR AT LEAST ONE-HALF OF FULL TIME | | |
| SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME | | |
| SIGNATURE OF APPLICANT | | DATE |

## PART III: APPLICATION FOR ASSOCIATE MEMBERSHIP, CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP OR FULL MEMBERSHIP (and interim Associate Membership)

SUMMARIZE YOUR MAJOR PROFESSIONAL EXPERIENCE on the lines below: list employers, duties and position titles, and dates, beginning with your present position. Attach additional sheets if more space is required.

| |
|---|
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| |
| |
| |
| CONTRIBUTIONS TO ACOUSTICS: LIST MAIN PUBLICATIONS, PATENTS, ETC. Attach separate sheets if required. |
| |
| |
| |

SPONSORS AND REFERENCES: An application for full Membership requires the names, and email addresses of two references who must be **full Members or Fellows** of the Acoustical Society. Names and signatures are NOT required for Associate Membership, Corresponding Electronic Associate Membership or Student Membership applications.

| | |
|---|---|
| PRINT NAME OF REFERENCE (required for Full Member applications only) | PRINT NAME OF SECOND REFERENCE (required for Full Member applications only) |
| EMAIL ADDRESS OF REFERENCE | EMAIL ADDRESS OF SECOND REFERENCE |
| SIGNATURE OF APPLICANT | DATE |

**MAIL THIS COMPLETED APPLICATION, WITH APPROPRIATE PAYMENT TO:** ACOUSTICAL SOCIETY OF AMERICA, 1305 WALT WHITMAN ROAD, SUITE 110, MELVILLE, NY 11747-4300; FAX: 631-923-2875

## METHOD OF PAYMENT

☐ Check or money order enclosed for $ ____________ (U.S. funds/drawn on U.S. bank)

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# Regional Chapters and Student Chapters

Anyone interested in becoming a member of a regional chapter or in learning if a meeting of the chapter will be held while he/she is in the local area of the chapter, either permanently or on travel, is welcome to contact the appropriate chapter representative. Contact information is listed below for each chapter representative.

Anyone interested in organizing a regional chapter in an area not covered by any of the chapters below is invited to contact the Cochairs of the Committee on Regional Chapters for information and assistance: Evelyn Hoglund, Ohio State University, hoglund1@osu.edu and Sandra Guzman, Shure, Inc., guzman_sandra@shure.com

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to papers presented at

## 183rd Meeting of the Acoustical Society of America

# INDEX TO ADVERTISERS



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