THE JOURNAL FOR LIVE EVENT TECHNOLOGY PROFESSIONALS
NOVEMBER 2016 | PROSOUNDWEB.COM

LIVE SOUND INTERNATIONAL

25th ANNIVERSARY

INSIDE

CONCERTED COOPERATION

Audio teamwork for Peter Gabriel and Sting on tour

THE CURRENT WIRELESS LANDSCAPE

INSIDE DIGITAL CONSOLE EFFECTS

PURSUING THE SEMI-SILENT STAGE


PRO SOUND
PURE EMOTION
HDL 50-A THE SYSTEM
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Anticipating the needs of the mobile production and system integration markets takes a terrific amount of user research and a delicate understanding of the multi-faceted problems that mid-size companies face in this economy. Simply put, they are doing more with less. Budgets are stretched and staffing can be slim.

So, when embarking on a joint project to help these companies succeed, EAW and Mackie focused on the number one pain point, knowing the right products could make a tremendous difference in day-to-day operations for countless hungry production and install companies. With the EAW RADIUS system, setup time and effort is dramatically reduced.

RADIUS™ with EAWmosaic
Smarter sound. Made simple.

- Compact array featuring OptiLogic™ array detection and optimization
- Powerful EAWmosaic™ control app for wireless system control with smart prediction and room design tools
- Hallmark EAW sonic quality featuring Focusing™ and DynO™ DSP
- Complete Dante™ integration

The RADIUS compact line array features OptiLogic™ array detection and optimization. A clever combination of integrated infrared transceivers and tilt sensors allows each array module to know its exact position and splay angle within the array. The modules are grouped accordingly and DSP optimizes the systems acoustical output to compensate for array size, audience geometry and throw. This can also be controlled using the EAWmosaic™ control app, featuring complete wireless system control plus highly approachable prediction and room design. All of this equates to quicker setup and the ability to dial in amazing sound with very little effort. Having the entire AXIS + RADIUS system on the Dante™ network also saves setup time and drastically simplifies routing.

At FOH and beyond lies the Mackie AXIS Digital Mixing System, a modular 32-channel design featuring the sleek DC16 control surface. With AXIS, Mackie concentrated on creating a product that delivers an incredibly fast workflow through visual feedback and customization to reduce setup time, allowing engineers at any level to dial in their mix quickly. The control surface layout is extremely intuitive and features tons of high-resolution, full-color channel ID screens. Having a multitude of small screens delivers information right where it's needed, so the engineer is never looking for anything, it is always there. Adding to the AXIS system flexibility is the SmartBridge™ design which allows up to three iPads® to dock in the control surface, allowing

MACKIE.
Mackie.com

"With the AXIS and RADIUS systems, Mackie and EAW have provided us with a complete set of tools to serve our clients with excellence."

APPLICATION PROFILE: BALLARD SEAFOODFEST

control over multiple channels at once. AXIS intelligently senses when an iPad is in place and its operation adjusts accordingly. A user can grab an iPad and walk the room for system tuning or dial in monitors. When they return to FOH and dock the iPad, it reverts to the previously chosen state. These are just a couple of the innovative features that help AXIS drive efficiency for any application.

AXIS

Better workflow through innovation.

- Flexible modular system with unmatched speed, visibility and customization
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- Flexible 32x32 recording/playback
- Complete Dante™ integration

The combination of AXIS + RADIUS is a powerful solution for any medium-sized mobile production or system integration company, delivering world-class sound and unmatched workflow speed to allow you to do more and do it all more easily and quickly. ■

For more than 42 years, the Ballard SeafoodFest has celebrated one of Seattle's unique neighborhoods with great food, beer and (of course) tons of live music.

"With 75,000 visitors over the course of a two-day weekend, this is a pretty complicated setup," remarks the event's Executive Director, Mike Stewart. "It's critical that the system can get setup quickly and efficiently in a dense urban environment."

To support the wide range of musical performances throughout the multi-day festival, EAW and Mackie debuted their new end-to-end audio solution made up of EAW RADIUS loudspeakers and Mackie's AXIS Digital Mixing System.

The EAW RADIUS PA consisted of left-right hangs of six RSX208L line array modules reinforced by 12 RSX18 subwoofers stacked six per side. Front fill was managed by two RSX89 loudspeakers. Each module provides up to 125 dB whole space SPL ensuring more than enough bandwidth to handle any performance type. The RSX18 delivered driving low end compliments of an 18-inch cone (3-inch voice coil) loaded in a vented enclosure.

And, at FOH, Mackie harnessed the power of their new modular AXIS system, combining the power of the 32-channel DL32R Rackmount Digital Mixer and innovative DC16 Control Surface to deliver a live sound solution with the efficiency fast-paced festival stages demand. Thirty-two remote-controllable Onyx+ mic preamps and 18 outputs with built-in DSP can handle the most rigorous schedules.

"This year everyone noticed the sound quality – which was pretty incredible," concludes Stewart. ■

~ Corner Audio & Video,
Nashville full-service design & build company

EAW
EAW.com

In This Issue

NOVEMBER 2016 | VOL. 25, NO. 11

FEATURES

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DEPARTMENTS

IN EVERY ISSUE

25th ANNIVERSARY
LIVE SOUND INTERNATIONAL

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From the Editor's Desk

AS MANY OF YOU KNOW, changes are ongoing in the U.S. with respect to the RF spectrum and wireless systems, and these changes continue to have impact. In this issue, Gary Parks provides a detailed account of what's happening now, what to expect, and how best to prepare for what's coming.

We've followed this situation from the outset and will continue to do so, both here and on ProSoundWeb.

Elsewhere in the issue, Jonah Altrove offers an interesting take on the world of digital console DSP and effects, touching on a wide range of areas. It's a dynamic and fluid state of affairs, changing and evolving as technology develops and bright minds (in the form of product designers as well as users inspiring new ideas and directions) continue to push the envelope.

Craig Leerman steps up with a wealth of solutions for dealing with tough acoustical environments, offering approaches that can be deployed with just a bit of forethought and planning. Meanwhile, Mike Sokol weighs in with the second installment on creating a semi-silent stage, furthering the discussion and also providing context on the musician side of the equation.

The Real World Gear focus this time out is large-format line arrays, a category that also continues to grow and evolve in a wide range of beneficial directions. As the marking of our 25th anniversary continues, we're pleased to present another "oldie but goodie" from Teri Hogan regarding the decision-making process when it comes to investing in new gear. Sage counsel then and now.

And as always, there's much more. Enjoy the issue.

Keith Clark

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ON THE COVER:
Peter Gabriel and Sting performing on the Rock Paper Scissors Tour.
(Photo by Peter Hutchins)

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Introducing the TouchMix™-30 Pro

With a host of all-new *Pro* features and even more "Results Made Easy" Presets and Wizards, the TouchMix-30 Pro will forever change your expectations of what a digital mixer should do. New *Pro* features include:

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Anti-Feedback Wizard

Room Tuning Wizard

Two Real-Time Analyzers (RTA)

USB interface for Mac® DAW Compatibility

32 Track Record/Playback Directly to USB Drive

MP3 Playback

BEYOND MIXING

www.qsc.com/beyondmixing

JoeCo BlueBox

A line of two workstation interface recorders providing individually switchable mic/line inputs with preamps on every mic channel. The BBWR24MP and BBWR08MP are both 24-bit/96 kHz models that have been designed and constructed to the same standards as the BlackBox range. The BBWR24MP has 24 switchable mic/line inputs, while the BBWR08MP supplies 8 switchable mic/line inputs and 16 channels of dedicated balanced line inputs. For larger applications, multiple BBWR24MP units can be connected to Mac (via Core Audio drivers) or Windows (via dedicated ASIO drivers). Both can also be controlled from the workstation itself with JoeCoControl or using the JoeCoRemote app for iPad. ***www.joeco.co.uk***

PreSonus StudioLive AR USB-series

Three mixer models that include the 18-channel StudioLive AR16 USB, 14-channel StudioLive AR12 USB, and 8-channel StudioLive AR8 USB. All are equipped with a Mac- and Windows-compatible, 24-bit, 96 kHz, USB 2.0 interface that can capture all input channels and the main mix. They also include a stereo SD recorder for recording the main mix without a computer; in addition, the recorder can also play up to 32 GB of MP3 and WAV files. The proprietary Super Channel facilitates playing audio from four stereo sources simultaneously. Bluetooth 4.1 is also included. The mixers are bundled with PreSonus Capture live recording software and Studio One 3 Artist DAW production software. ***www.presonus.com***

Radial LX3

A passive line splitter outfitted with a Jensen transformer that's designed to send one audio signal to as many as three destinations at once. It offers two isolated outputs to eliminate hum and buzz from ground loops, along with a third direct output. All three XLR outputs have ground lift switches for additional suppression of noise, and an input pad allows connection of very high line output levels without over-saturating the transformer. The input has an XLR/TRS combo jack that connects via balanced or unbalanced cables; the LX3 will automatically balance the signal at the isolated outputs, preventing interference and signal degradation over long cable runs. ***www.radialeng.com***

Waves Audio Greg Wells ToneCentric

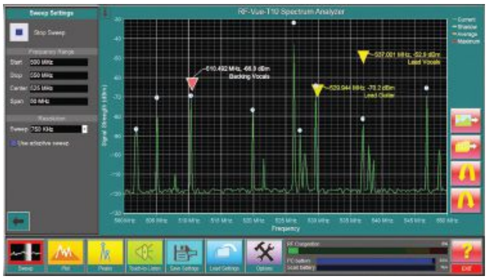
A musical harmonic enhancer plugin based on multiple Grammy-nominee Wells' favorite pieces of analog gear. Joining the Greg Wells Signature Series, it's designed to add analog tone to individual tracks or entire mixes (live and in the studio) with an easy-to-use single-knob interface. ***www.waves.com***

Eastern Acoustic Works EAWmosaic

An app providing control for the company's recently introduced RADIUS loudspeaker series via an iPad, along with system design parameters. In addition, a single cable can also be used to provide multi-channel audio from any Dante-enabled source (i.e., a mixer such as the Mackie DL32R). The app is available for free download at the App Store. ***www.eaw.com***

Danley Sound Labs J3-94 Jericho Horn

A loudspeaker stated to deliver 142 dB of continuous output using 18 drivers that combine via patented technologies to produce a single phase-coherent signal. Drivers include six 15-inch woofers, eight 6.5-inch mids, and four 1.4-inch highs with built-in overload protection. Dispersion is stated as 90 degrees by 40 degrees (h x v). The tri-amped cabinet uses four amp channels: two for the lows, one for the mids, and one for the highs. It can be flown with several different standard lifts/towers and comes standard with 3/8-inch fly points on top/bottom and L track on the sides. ***www.danleysoundlabs.com***



Kaltman Creations RF-VUE

A tablet-based RF spectrum analyzer that allows users to listen directly to RF signals and identify interference. The built-in RF Congestion Scale gauges local severity, and up to 100 color-coded custom markers are available to monitor the performance of transmitters and to track potential interference sources. RF-VUE is available in frequency ranges up to 2.5 GHz and offers adaptive sweep functionality to help foster quick and accurate scans. Two versions are available: pre-configured Windows tablets with permanently-mounted RF analyzer hardware or a stand-alone hardware kit for users' own compatible tablets or touch screen-capable computers (permanent mounting of the hardware to the tablet is optional). ***kaltmancreationsllc.com***

d&b audiotechnik 24S, 24S-D & 21S-SUB

Two point source loudspeakers and a subwoofer designed for permanent installations such as live performance venues and churches. Differing in horizontal dispersion (75 by 45 degrees and 110 by 45 degrees), the 24S and 24S-D house two 12-inch LF drivers in a dipolar arrangement and a horn-loaded 1.4-inch-exit compression driver. The HF horn is rotatable. Frequency response is stated as 55 Hz to 18 kHz. The 21S-SUB incorporates a single 21-inch driver in a bass-reflex design. Stand-alone frequency response is 35 to 105 Hz, and as a supplement to other d&b subwoofers in INFRA mode, response is 33 to 85 Hz. In combination with the d&b 30D amplifier, the 24S, 24S-D and 21S-SUB are rated to produce 138 dB SPL, 137 dB SPL and 134 dB SPL, respectively. ***www.dbaudio.com***

Yamaha MonitorMix For Android

An app for the company's CL, QL and TF Series consoles that facilitates individual wireless monitor mixing for up to 10 iOS and Android devices simultaneously. It enables performers to have control over the monitor buses assigned to them, and also allows them create personal group settings; for example, all instrument levels on just one fader. The app is available for free download at Google Play. ***www.yamahaca.com***

Shure Wireless Workbench 6.12

The latest version of the company's wireless system control software includes Timeline, a logging utility designed to capture channel status information over time, including RF level, antenna status, audio level, interference, ShowLink Remote Control status, and battery level. Timeline can record data from all networked channels for an indefinite amount of time, given sufficient system resources. Version 6.12 also offers scan peak management to process and classify peaks more accurately, the ability to manually ignore the intermod (IMD) spacing parameters with a given set of frequencies, and customizable coordination to adjust the order in which frequencies are coordinated. The software is available for download from the company website. ***www.shure.com***

BEHRINGER X32 v3.0

A firmware and application software update for the X32 console consolidating the new versions of the X32-Edit (PC/Mac/Linux/RPi) and X32-Mix (iPad) remote control apps. There's also a new Automixing option, support for the new X32-EDIT 3.0 remote control application, networked and MIDI remote control directly from X-TOUCH control surfaces, and output phase inversion on all outputs, aux outs and Ultranet channels. Also included are crossover filter options for the main L/R and mono/center bus EQs. ***www.behringer.com***

Renkus-Heinz ICONYX Gen5 & IC Live RD Versions

Dante-enabled models of ICONYX Gen5 and IC Live loudspeakers; specifically, RD versions provide dual redundant Dante connectivity with audio transport and configurable sample rates up to 96 kHz. RHAON 2.1, an update to the company's RHAON II software, now supports Dante connectivity and incorporates more than 100 software updates and enhancements, including support for multiple zones, as well as a new device "ccon view" mode for organizing larger systems. ***www.renkus-heinz.com***

MIPRO ACT 2400 Series

Several models of wireless systems encompassing both single- and dual-channel half-rack receivers with a variety of transmitter options, all operating in the 2.4 GHz band. They utilize four frequencies for each channel in an adaptive tracking algorithm that avoids interference from 2.4 GHz products on channels 1, 6, and 11. They also employ dual-tuner true digital diversity reception and supply 12 compatible simultaneous channel operations at receiving distances stated as being up to 100 meters (330 feet) with no dropouts. Both receiver types include the MP-80 smart charging cradle. The ACT-24HC handheld transmitter (with condenser capsule) and ACT-24TC miniature bodypack transmitter use ICR18500 rechargeable lithium batteries rated to provide 1-hour operation on a 25-minute quick charge or 10 hours of service on a 4-hour charge. The MT-24 wireless digital guitar transmitter is also available. MIPRO is distributed in North America by Avlex Corporation. ***www.avlex.com***

Sennheiser SpeechLine

An expansion of the company's digital wireless system includes two new base units (SL Tablestand 153-S DW and133-S DW) with wireless charging by induction. Both can either be charged wirelessly with the new CHG QI 2 wireless charging base or via USB, and can also be monitored and controlled with Sennheiser Control Cockpit. The CHG QI 2 provides wireless charging for two mobile devices. The line also now includes the CHG 4N, a network-enabled charger with four individual charging bays that can either be used for the SL Bodypack DW or the SL Handheld DW. Each bay has four dedicated LEDs to indicate the corresponding battery status. ***www.sennheiserusa.com***

Eminence Alpha 3

A 3-inch full-range loudspeaker driver suited for tight-fitting line array and column array applications. Rated at 8 ohms, the program power rating is listed as 60 watts, with sensitivity of 88.6 dB. An 8.3 ounce ferrite magnet in conjunction with an internal neodymium bucking magnet increases sensitivity and extends response to a stated 20 kHz. The tightly pressed cone sandwiches paper with a layer of water-resistant polypropylene for additional stiffness and lower distortion. ***www.eminence.com***

Turbosound TFM122M-AN & TFM152M-AN

Two models joining the existing Flashline stage monitor range that can also serve as full-range loudspeakers. Both have an integrated class D amplifier rated to provide 1,100 watts of power (peak) joined by Klark Teknik DSP, with ULTRANET networking for connectivity to mixing consoles and other compatible digital devices. They deliver 60- by 40-degree (h x v) dispersion, incorporating a titanium dome 1-inch compression driver and either a 12- or 15-inch ferrite woofer. ***www.turbosound.com***

ELEPHANTS & YARDSTICKS

The importance of revisiting our ever-shifting knowledge base.

by Jonah Altrove

When I was growing up, my mother would stand me up against the frame of my closet door and mark my height on the wood. This chart of my growth during the first dozen years of my life is still there.

Nowadays, I'm not sure my mom could reach high enough to mark my adult height, but what's strange is that I don't really remember getting any taller. In my mind's eye, I've always been the same height, and things around me just got smaller.

Since we always experience things from our own perspective, we're maybe not the best judges or our own changes, especially when that change is gradual. ("When did I go gray? When did I get fat? Is that really what my voice sounds like?") My closet doorframe was a "yardstick" of reality, an objective record of what really happened, not tainted by my own perception.

FILLING THE GAPS

I jumped into my first touring gig as an FOH engineer with the typical young-gun hot-shot attitude that comes as standard issue for fresh college graduates. During the tour, I mixed with a couple dozen different consoles, from Profiles to powered mixers, in venues ranging in size from 35 to 3,500.

I also mixed on line arrays for the first time, and had to solder one venue's decrepit rig back together in order to limp through a show. In other words, I got my butt kicked in the best possible way. As the saying goes, "Experience is something you get just after you need it."

Upon returning home, I thought, "Wow, OK. So there's a lot more to this that I didn't know about." I logged on to Amazon and depleted my (meager) checking account on as many audio (and related) books as I could, and when they arrived, I devoured them.

I learned acoustics from F. Alton Everest, psychoacoustics from Floyd E. Toole, the principles of rigging from Harry Donovan, and the essentials of power from Richard Cedena. I read Bob McCarthy's text on system optimization so many times that it completely fell apart. (My second copy is currently held together with packing tape.) It seemed as if the authors were speaking directly to me through their books.

MAKING THE MOST

Of course, this is only one side of the coin. Reading about how a GEQ works doesn't help in deciding which filter to cut. Understanding the steps to make a basket hitch around a beam is not the same as fumbling with a shackle while balancing on that beam, 60 feet above the ground.

Books prepare us for practical experience but they don't replace it. However, they're still quite valuable. They can help us make the most of the practical experiences that follow, particularly in areas where "guess, test and revise" isn't the healthiest approach, such as power distribution.

Books can also teach us why we do things the way we do, and they can give us the vocabulary necessary to discuss these ideas with others. But most of all – and now we get to my primary point – books are yardsticks, just like my closet doorframe.

One of the books I purchased dealt entirely with signal metering. It was a small, thin, non-threatening thing, and it burst my bubble. I found the book confusing, frustrating, impenetrable. I remember thinking, "Who could possibly want to read this?" as I put it back on the shelf.

REVISITING THE PREMISE

The book was never at fault, of course. It was an advanced topic, and I simply

The contents of the book hadn't changed; the contents of my brain had.

lacked the prerequisite knowledge. Rather than recognize this for what it was, I blamed the book. A couple of years later while rearranging my shelves, I came across it and flipped to a random page, realizing, "Hmm, this doesn't look too bad." I ended up reading it from cover to cover, grasping the discussion perfectly well.

In the intervening years, I'd deepened my understanding of the relevant topics to the point that I could understand what the author was saying. All the while, I had gone right on thinking that it was beyond me because I never checked. The contents of the book hadn't changed; the contents of my brain had. This change was so gradual, as with my height when growing up, that I either hadn't noticed or taken time to reevaluate.

A friend once told me an Indian folk tail that makes the point better than I can. A trainer buys a baby elephant and chains her leg to a metal stake in the ground to keep her from wandering away. At first the elephant tugs at the stake but eventually learns that she's not strong enough to pull it out of the ground, so she stops trying. Years go by without the elephant ever tugging at the chain.

Later, when the elephant is fully grown, a passerby stops to talk to the trainer. "You fool!" he says. "Your elephant will escape. Why, she can pull that stake right out of the ground!"

"Ah," the trainer replies. "You know that, and I know that." He then gestures to the elephant. "But she doesn't."

PUTTING IT IN PRACTICE

Another friend, this one a physicist, once bought me a book about the physics of musical instruments. The bits I can understand are fascinating, but much of it is too advanced for me, so mostly this book sits on the shelf.

But because I don't want to end up like the elephant, once in a while I retrieve it and flip it open – to test the stake. If you've got your own archnemesis – a book, a skill, a fitness routine, whatever – that's defeated you in the past, I encourage you to give it another shot. What you find may come as a pleasant surprise. LSI

__Jonah Altrove__ is a veteran live audio professional on a constant quest to discover more about the craft.

WHERE THINGS STAND

The current and future landscape of the RF/ wireless world.

by Gary Parks

The much-discussed auction of the 600 MHz frequency band is happening in the U.S., and it may well affect present wireless systems as well as related issues such as frequency planning/coordination. It's important for everyone who works with creating the content that will stream on the mobile devices when the spectrum is cleared to understand the present situation and to be planning for the transition to different frequency bands.

Several leading pro audio wireless system manufacturers have taken the lead in working with the U.S. Federal Communications Commission (FCC) as well as other wireless stakeholders to insure that sufficient spectrum remains available for pro audio. They (and others) have also been highly involved in providing education on the coming changes and in developing products to meet future requirements.

Let's take a look at the likely end results of the auction, the possible timeline of when the 600 MHz band will no longer be usable for wireless systems, alternative frequency bands, the benefits of licensed operation, new technologies that wireless manufacturers are developing, and what actions users can take to smooth the transition.

But first, note that the category of "wireless microphones," per the FCC, covers a variety of wireless audio devices typically used in concert and event production, churches, public facilities, performing arts venues and even local night clubs. This equipment includes wireless handheld mics, headset or lavalier mics with bodypack transmitters, guitar transmitters, intercoms, in-ear monitors (IEM), and IFB systems used to cue and provide program feeds to on-air talent. Each uses a certain portion of RF spectrum to function.

THE PROCESS IS UNDERWAY

Officially called the Broadcast Incentive Auction, the 600 MHz spectrum is being repurposed from television broadcast to mobile broadband and similar telecom applications. Earlier this year, broadcasters were invited to provide a price for which they would be willing to give up the spectrum they occupy, and agree to either move to a less desirable band, share spectrum with other broadcasters, or cease broadcasting. This provided the pool of available spectrum for the telecom companies to bid for.

Initially, more than 4,000 "blocks" of spectrum throughout the U.S., consisting of paired 5 MHz bands for upstream and downstream broadband transmission, were offered. This spectrum spans 126 MHz and has an asking price of approximately $88 billion. Bidders include companies such as AT&T, Verizon, Comcast, and Dish Network. Purchasing the right to use this spectrum more or less involves a process of multiple bidding "rounds," with each stage of bidding offering less spectrum for sale – and eventually concluding when the total amount bid is equivalent to the price that broadcasters had agreed to.

As of mid October of this year, the first stage had been completed, while the second stage offering 114 MHz just closed without meeting the target. Joe Ciaudelli, director of U.S. Spectrum Affairs at Sennheiser, states, "The worst-case scenarios didn't come to fruition," adding that he anticipates that the amount of spectrum up for auction will be less going forward, and that a third bidding stage and possibly more will be necessary.

Mark Brunner, senior director of Global Brand Management at Shure, notes that some industry observers expect the end result might be approximately 86 MHz of spectrum becoming cleared and unavailable for use by wireless microphones. "This is a better outcome for pro audio as far as more UHF spectrum remaining available," he says, adding, "However, it may also change the size of the guard bands" which are potentially open for shared use. Karl Winkler, vice president of Sales & Service at Lectrosonics (and LSI author), concurs and suggests that, "The floor for new services will be the top of TV 37."

TIMING ISSUES

Once the auction is complete (it may extend into 2017), wireless microphones and related devices operating in the cleared bands will have 39 months to make the transition, after which

The Word Is Out

VENUE | S6L - the next stage in live sound is here

"Everything is right in front of me—it resonates with my style of mixing. And the sound is absolutely stunning. I've never heard Ozzy's vocal quality like this!"

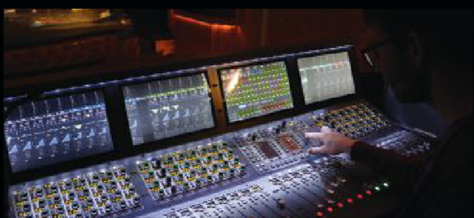
–Greg Price, FOH engineer, Black Sabbath, Ozzy Osbourne, Van Halen

"As soon as we went into band rehearsals on the first day, John Taylor, the bass player, turned around and said, 'It sounds incredible!' They noticed straight away."

–Charlie Bradley, monitor engineer, Duran Duran, Annie Lennox, Snow Patrol

"After just two weeks on tour, I can say that S6L has the fastest and easiest workflow of any console, with the best sounding preamps and a fantastic control surface."

–Gerard Albo, FOH engineer, a-ha, Patti Smith, Tom Jones



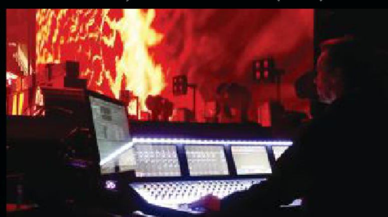
National Theatre of Iceland, Reykjavík

Rich Steeb, Blue Rodeo (FOH)

Rob Allan, Massive Attack (FOH)

Myles Hale, Black Sabbath (monitors)

Transmix Oy OB Truck, Helsinki, Finland



Robert Nevalainen, Bryan Adams (monitors)

Discover VENUE | S6L, now with I/O sharing, 128 Pro Tools recording tracks, and Dante and MADI connectivity: **avid.com/S6L**

their use in that spectrum is banned. During this time period, "Telecom services will be moving into their newly acquired space on a market-by-market basis," according to Brunner. "This is what is challenging for wireless mic operators, in that the 600 MHz band spectrum that is auctioned will not all be declared off-limits at a consistent time; it will also vary market by market. The only requirement is that they don't interfere with newly licensed services that come on the air."

So during the transition, it will be necessary for users to use RF scanning hardware and software to conduct local environmental scans before setting channels in cleared spectrum. They will also need to consult the nationwide geo-location databases, which will reflect the new services commencing operations in those markets. Complying with cases of interference will continue to be done as it is currently, which is on a complaint-driven basis.

"I don't want to sugar-coat things – the transition is not going to be easy," Ciaudelli says. However, he points out that the broadening of licensing categories for larger users and opening new bands of spectrum are offsetting aspects of the 600 MHz clearing.

LICENSED WIRELESS

In recognition of the widespread and essential use of wireless mics in event production, and pushed by the lobbying efforts of wireless manufacturers and other industry leaders, the FCC in 2013 broadened the types of users who are able to obtain licenses to encompass sound companies, live theatre, churches, and many other entities – going beyond the original categories of broadcasters and film production.

Larger users who typically occupy 50 or more channels will be able to obtain a Part 74 license, which will give them priority to use the RF spectrum at specific times and locations. This means that unlicensed users of white space devices or other wireless mics in those particular frequency bands will have to turn them off if they're causing interference.

Ciaudelli points out that the FCC has made a clear distinction between unlicensed and licensed users, defined as those who routinely operate 50 or more channels. "There are many entities who are eligible to become licensed operators, but they haven't so far because functionally it wasn't a necessity. Going forward, licensing will be vital," he says, adding that, "It's very important now to determine whether you're eligible, and begin the licensing process. It provides rights and privileges over unlicensed operation of any device – not only other wireless mic users. You want to make sure that you're higher on the totem pole."

Additionally, licensed operators will be allowed the use of alternative bands outside of the traditional UHF, VHF, and ISM spectrum.

SPECTRUM OPTIONS

While pro audio is losing some UHF channels, other prime portions of spectrum are now open to licensed wireless microphone operators. For example, the mid-900 MHz band that was previously available only to broadcasters has been expanded from its previous 8 MHz bandwidth to almost 19 MHz. This 941 to 960 MHz spectrum provides another option for those who have been operating in the 600 MHz band, especially if they've already used up all the space available to them in the 500 MHz band.

The technology to use this newly opened spectrum is already in development and is very likely to come to market soon. And note that this additional bandwidth is in addition to the current unlicensed 902 to 928 MHz swath, with some professional systems, such as the Shure ULX-D, already operating in this range.

"Power users" at mega-events such as political conventions and sporting events will also be able to use 1.435 to 1.525 GHz. However, they will need to coordinate with a government agency called AFTrack because that frequency range is used for flight testing, typically Monday through Friday during daylight hours. The product development process to operate in this range incorporates some completely new technology, so it's likely to be a few years before we see commercially available systems.

A new and critical component required to operate in this band is that equipment will need an electronic key to function. Users will be required to apply with AFTrack for the specific events where their wireless mics are to be used and will be given a code to enter into the equipment that allows it to operate at the specified time, date, and location.

NEW TECHNOLOGIES

As noted, manufacturers are adding new products and technologies to accommodate the transition. The Lectrosonics IFB4-VHF system offers 239 selectable frequencies between 174 and 216 MHz. In addition, the tuning ranges of the Venue 2 and SRc modular wireless receivers have been tripled to 75 MHz, and additional filtering technology has been added to reduce interference when multi-channel systems are used in a crowded RF environment. The company is also beginning development on equipment for the new frequency bands opened by the FCC for licensed users.

Audio-Technica is exploring other frequency bands as well as adding new products in the lower part of the UHF band, along with incorporating a variety of innovations into its wireless for non-licensed frequencies. Jim Lappin, product manager for wireless microphones at A-T, points to the System 10 PRO, which operates at 2.4 GHz while incorporating frequency and time diversity technologies, the ability to place modular receivers remotely via Cat cable, and linking technology that allows up to 10 receivers to be synched for simultaneous use. In addition, A-T recently announced the formation of a new subsidiary company, Alteros, with a dedicated focus on providing a range of solutions to directly address the issues surrounding the ever-shrinking frequency spectrum (see sidebar).

Shure is approaching the transition on many fronts. Brunner states that these efforts include improving spectral efficiency, advocating for frequency bands that pro users can migrate to and then developing the necessary technology and products, and creating reliable wireless to meet the needs of unlicensed, semi-professional users. The company has also

CAPITALIZING ON Audio-Technica's many years of ultra-wideband (UWB) and RF technology research, Alteros is focused on the ongoing issues of frequency spectrum allocation, auctions, coordination and more. The new company is under the direction of president/CTO Jackie Green, U.S. VP of R&D/Engineering at A-T, who led the first commercial sound implementation of UWB technology with the SpectraPulse wireless microphone system that bypasses the increasingly congested RF environment. The Alteros management team also includes Brian Fair, executive VP of Digital Engineering, and Bob Green, executive VP of Product Engineering.

"Alteros represents Audio-Technica's absolute dedication and commitment to developing the highest-level technical tools in support of the high-end audio market. Our first product addresses the issues facing broadcasters and audio professionals operating in the wireless realm," states Philip Cajka, A-T U.S. president and CEO. "Jackie and her team have been working closely with leading broadcast engineers and their technical staff, who are all providing input which ultimately will result in a line of market-driven products under the Alteros brand. This is an exceptionally exciting time for Audio-Technica, and we expect great things to be coming from Alteros in the near future." LSI been exploring the VHF band while expanding the ULX-D line into the 900 MHz and lower UHF bands down to 470 MHz. Digital transmission technologies such as the ULX-D operating in "high density mode" can allow more channels to share a given band of frequency, with Brunner adding that "antenna technology has improved, enabling VHF as a viable option for professional use."

From Sennheiser's perspective, two critical aspects of future wireless designs are tuning flexibility and spectral efficiency. The ability of receivers and transmitters to operate over a wide band of frequencies allows users to avoid areas where other licensed or unlicensed equipment may create interference, while still providing sufficient spectrum. Sennheiser also recently added the A1 band, which starts at 470 MHz.

GETTING THE WORD OUT

With a transition of this magnitude, reliable information along with advice and technical assistance from manufacturers is essential. Winkler has been heavily involved in a variety of educational and training efforts around the coming changes. Along with Tim Vear of Shure and Eric Reese of Sennheiser, he's served as a co-instructor at SynAudCon's "Making Wireless Work" educational seminars. He also notes that the company's sales managers consult with customers to evaluate the frequency bands of their current wireless equipment, help them understand the auction process and how it is proceeding, and offer input on how they can adapt for the future.

Along with other educational outreach, Sennheiser has published white papers that detail the likely impact of the spectrum auction for wireless microphones, along with providing informed counsel to dealers and customers. The company has also instituted a "trade-in" incentive program on certain models for affected wireless.

Shure has been an active advocate as well, sending out press announcements that help keep the pro audio public informed, hosting phone conferences, providing insights and updates to dealers and customers, and participating in forums such as the "Current & Future State Of Wireless/RF In The U.S." podcast currently available on ProSoundWeb. The company also began discontinuing products in the upper portion of the 600 MHz band in 2015, recommending that customers opt for lower UHF or unlicensed bands.

MAKING THE TRANSITION

As Brunner notes, "Until the auction is complete, we won't know what the actual band plan is." However, in distilling all of this the input, there are several actions wireless users can take to minimize the effects of the transition, including:

- Inventory current wireless equipment to see which ones are operating in soon-to-be-cleared bands, and predict future requirements such as how many channels, what mix of mics, IEM, and intercoms will be needed, and so on.
- Know the RF environment in the locations where they operate, and become familiar with likely future shifts and interference sources. Reference the FCC's TV Query Broadcast Database and other information sources.
- Determine whether they qualify to become a licensed user of wireless microphones, and begin the licensing process. It's likely that the manufacturers of the users' wireless equipment can provide guidance.
- If a significant number of current channels will only operate in the 600 MHz band, team with those manufacturers to smooth the transition.
- When purchasing additional channels, get them in the lower UHF bands and supplement with VHF and the unlicensed 900 MHz and 2.4 GHz bands. For example, the Radio Active Designs UV-1G wireless intercom operates in the VHF band, Clear-Com's Freespeak II operates at 1.9 and 2.4 GHz, and the RTS BTR-240 at 2.4 GHz. Wireless mics and IEMs suitable for semi-pro and pro applications are also widely available in these bands.
- Use available scanning tools and planning software to visualize the RF environment and efficiently place wireless into service. Many systems are complemented with management software that helps with RF planning, advancing shows, and monitoring systems during events.
- Finally, be prepared to share wireless and frequency coordination information with other operators in nearby venues, especially in high-density areas like Broadway's Theatre District.

"The future is solid for licensed wireless users, yet may be tougher for smaller non-profit performing arts centers and others that aren't eligible for licensing," Ciaudelli concludes. Knowing what's coming will help in making the best of the situation while keeping productions running without interference. LSI

***Gary Parks** is a writer who has worked in pro audio for more than 25 years, holding marketing and management positions with several leading manufacturers.*

QSC TOUCHMIX-30 PRO

Detailing the expansion of a compact digital mixer platform.

by Mark Frink

The new QSC TouchMix-30 Pro digital mixer.

Following the success of the category-bending TouchMix touchscreen "laptop" portable mixers, QSC moves up-market with the new compact TouchMix-30 Pro, adding over a dozen new features. Those familiar with QSC's TouchMix-16 and -8 collaboration with Greg Mackie and Peter Watts immediately note its new form factor. The wedge-shaped chassis weighs 17 pounds, is 18 inches deep and its 17-inch width accepts an optional rack-mount conversion kit.

The TM30's 7-inch height creates room for all its connectors on the rear. Two rows of a dozen XLR connectors for its 24 preamps line the top half, with the last four inputs being TRS/XLR combo-jacks. Below are two rows of XLR male connectors for its 14 auxiliary send outputs, stereo left-right main and stereo monitor outputs. Next to a TRS headphone jack, auxiliary mixes 11 & 12 and 13 & 14 are duplicated on stereo headphone outputs for easy, direct connection for two hardwired IEM mixes.

Objectors to the number 30 in its name must take note of three pairs of TRS connectors – labeled 25/26, 27/28 and 29/30 – for three stereo line inputs providing a total of 30 analog inputs, with the last pair duplicated as a convenient TRS-mini jack beside a second 1/4-inch TRS headphone jack on the front. A fourth stereo input (for a total of 32 inputs) accommodates playback of stereo MP3 files from a flash drive.

The internal 100 to 240-volt, 85-watt international power supply employs a blue Volex mains connector on its lower left that allows either a locking or a standard IEC cable to be used, regardless of international regional outlet plug type. When released at the LA AES show, it shipped worldwide and has language choices for English, French, German, Spanish, Chinese and Russian. Though still small enough to bring in an airplane cabin, you will find TouchMix-30 in every corner of the globe.

Above the AC mains is an Ethernet connection, a USB Type B connection for Mac-hosted DAWs and two USB 2.0 Type A connectors. A momentary Standby button on the top left is hidden from view but easily reached by an operator's hand. Finally there's a K & Lock security slot compatible with Kensington's MicroSaver security cable.

ON THE SURFACE

The control area above the armrest is dominated by a horizontally oriented 10-inch diagonal (8.75 x 4.75) 1024 x 600 (WSVGA) color touch-screen towards the left, almost twice as much screen as the original TM models. The larger screen now not only has enough real estate to show a row of fader layers across the top, but also a column of mix tabs down the left side. Touching each tab quickly navigates between mixes the same way the top row navigates fader layers. These color-coded tabs provide instant access to any mix's color-coded faders, with the main L/R mix at the top and a paired stereo aux mix becoming its two mono tabs combined.

The screen also allows a new, comprehensive channel overview that provides all its controls at once: gate and compressor, parametric EQ, FX sends and auxiliary sends, with all parameters directly adjustable.

Right-hand controls are similar to earlier TM mixers, with the addition of dedicated RTA, anti-feedback and rec/play buttons beside the screen for more new features. Besides the usual four user-defined buttons around the encoder, four more UDBs across the top have factory defaults of "play/stop" and "rec/stop" for quickly recording and playing back multi-tracks for virtual soundcheck and "copy" and "paste" for quick duplication of an entire channel's parameters or just a section.

GOING DEEPER

Each output's anti-feedback section now has a dozen filters with individually adjustable frequency and depth. Filter width is globally adjusted from a Q of 10 to 30 (1/7th to 1/20th of an octave) and overall depth of up to 20 dB can be adjusted as a percentage from 0 to 100 percent, allowing filters to be globally backed off after they're set.

A 28-band real-time analyzer (RTA) can be assigned to any mix, to follow the cue or fed from the talkback mic, which has phantom power, allowing it to be used with a measurement mic. Also, each TM30 output provides an RTA display above its parametric and graphic EQ filter controls.

The rec/play button provides quick access to one of three modes: stereo playback of MP3 files or multi-track recording and playback of 30 inputs plus the stereo mains either to a DAW or directly to a USB drive. In short, TM30 is also a 32-channel Mac interface that can be used to both record and mix a show, a rehearsal or a recording session.

Across the top of the console are two rows of a dozen analog trim pots for 24 class-A microphone pre-amps, providing gain adjustment from 10 to 60 dB. Other than the single Master Encoder, TouchMix consoles have no faders or knobs, making dedicated gain controls necessary. "Beta" vocal mics sit nicely at "12 o'clock" with about 25 dB of gain, while SM vocals need a bit more gain, positioned at "2 o'clock."

FLEXIBILITY & EFFECTS

TM30 increases auxiliary bus count to 14 with stereo-linkable pairs providing up to seven stereo mixes, all on XLR connectors with the last two pairs duplicated on TRS connections for directly driving hardwired IEMs. Each mix can be globally selected as post- or pre-fader as well as pre-dynamics and even pre-everything.

The optional tablet holder supporting an iPad.

Expanded I/O capability on the TM-30 rear panel.

Also provided are six parameter adjustable effects – including mono and stereo delays, both a "dense" and a "lush" reverb, a chorus and a stereo pitch shift – each with its own dedicated aux send and dedicated stereo returns.

In addition to eight DCA groups, TM30 adds eight subgroups, each with its own parameter-adjustable limiter and 6-band PEQ with high-pass and low-pass filtering, plus auxiliary and FX sends. Being able to assign a sub group to an aux send allows that aux to be used as a super matrix. A sub group preference automatically takes input channels assigned to a sub group out of the main L/R mix. Any input or output can now be delayed up to 100 milliseconds (113 feet), as can the stereo monitor (control room) output, which can be assigned either the cue or the main mix.

Like its predecessors, the TM30 employs 32-bit floating point processing, with 44.1 or 48 kHz sampling and specs include a S/N ratio of 95 dB, dynamic range of 105 dB and latency of 1.6 milliseconds, comparing favorably to mid-range professional touring desks.

REMOTE CONTROL

Also as with the TM-8 and -16's TouchMix app, the TouchMix-30 iOS or Android app lets you control the console from one or more tablets or a single mix from a smartphone. The free tablet apps not only duplicate all the controls on the console's screen, but also the console's right-hand buttons, allowing it to be operated entirely remotely, as well as providing an offline demo.

An optional tablet holder can be added on top of the console, providing a resting place where it serves as a second control surface. Other optional accessories include a compact Wi-Fi dongle instead of a router and a soft-sided tote bag.

Remote control over Wi-Fi is the intended primary method of using TM consoles. Whether remotely from a tablet or locally from its touchscreen, mixing occurs without mechanical faders, by touching and moving on-screen fader icons. These are adjusted with either a sliding motion, or a rolling motion of the finger to produce small, precise gain changes beneficial for typical 'bumps and rides.' The mobility of mixing from the app frequently outweighs its lack of physical faders.

While any selected parameter can be adjusted with the console's rotary encoder, the app replaces it with up and down "nudge" buttons that produce 1 dB fader moves, a handy and exact tweak for fader changes. A "fine" control changes this to 0.1 dB adjustments. Beside it, a "zero" button instantly bumps a selected input fader all the way off, or restores an output master or DCA to unity.

The new TouchMix-30 Pro expands and refines a console category that QSC defined two years ago, combining professional console quality and features with mobility and modern mixing convenience. LSI

__Mark Frink__ is a regular contributor to LSI and spent the summer beta testing the TM30 at the South Shore Music Circus in Cohasset, MA, where it served as the utility mixer for announcements, support acts and family shows.

VDO Face 5

COMPLETE THE PICTURE

The VDO Face 5 offers high performance through a 5.2 mm HD screen. Driven by the same P3 system controller as all other Martin LED Video Products the VDO Face 5 is the perfect fit for integration in a total video solution. The consistent pixel-to-pixel calibration does not only give the best image quality, but also enables panels from different batches and customers to be combined quickly and hassle-free.

Concerted Collaboration

Audio teamwork for Peter Gabriel and Sting on the Rock Paper Scissors Tour.

by Gregory A. DeTogne

Throughout the tour, Firehouse Productions flew a 270-system primarily comprised of L-Acoustics K1, K2 and Kara arrays.

PRIOR TO THIS YEAR, the last time Peter Gabriel and Sting toured together was in 1988 as part of a string of Amnesty International concerts. Also featuring Bruce Springsteen, Tracy Chapman, Youssou N'Dour, and a bevy of other guest artists, the 20 benefit shows brought a stellar array of talent together as distinct co-headliners.

In contrast to those dates, this year's Rock Paper Scissors Tour stands completely apart. Rather than co-headlining this time out, it would be more accurately stated that the pair amalgamated their best efforts and coalesced into a new whole. Sharing the same stage and collaborating throughout a two-hour-and-40-minute set, Gabriel and Sting— as well as their individual bands — swapped lead parts, harmonized on choruses, blended, weaved, and intermingled as the concert galloped through a jointly-authored songbook of hits that played to sold-out audiences for several months.

Just as Sting did an admirable job of taking over lead vocals on Gabriel's "Shock the Monkey" and the latter added darkly rich vocals to many verses within the former's "Englishman in New York," the audio crew for both acts melded in mutual cooperation to develop a sonic blueprint that had the front of house engineers for both bands mixing simultaneously from their respective consoles.

"The way this show worked is that whatever we did as individuals we did for the betterment of the entire show," Sting FOH engineer Howard Page related a few months after the tour launched on June 21 in Columbus, OH. "It wasn't a competition to see whose mix was better or louder. It was the whole that counted. You had to set aside all ego. Everything had to be understood from this perspective, and we had ground rules by which everyone abided."

BLENDING IT TOGETHER

Page, who mixed Sting from behind a Studer Vista 5 console using control provided by Clair Global, was joined at FOH by Richard Sharratt at the Gabriel helm, utilizing a Solid State Logic L500 Plus desk for his work. Sharratt relied upon a control schematic supplied by Firehouse Productions (Redhook, NY), the same entity that took on the task of building the PA.

With the capability to distribute audio evenly across a 270-degree path, the house system was headed by L-Acoustics K1 and K2 enclosures supplemented by K1-SB and SB28 subwoofers, joined by a contingent of Kara and ARCS loudspeakers. Sixty LA8 amplified controllers delivered amplification, DSP, and network functions to the design using L-Acoustics proprietary L-DRIVE system protection and IIR/FIR filter algorithms.

Firehouse system engineer Jamie Pollock recently noted on ProSoundWeb not long ago that one of his biggest challenges on the tour was configuring the system to accommodate both Page's and Sharratt's individual styles of mixing. "Each FOH console had eight sends that went into a pile of Lake DLP processors," he says, explaining how he managed the twin-console quandary, "which is where all the summing happened. From there it went into a (Riedel) RockNet (networking) system that sent everything to the LA8 amplified controllers."

From his standpoint, Sharratt concedes that while he and Page worked extremely well together, it was Pollock who had the ultimately responsibility of keeping the pair happy. "The two different styles of music had very different requirements," he notes. "So Jamie would find us a good middle ground each day, and then Howard and I would EQ our respective PA feeds to meet our individual needs.

"An internal eight-way GEQ inserted over the PA outputs proved to be the most effective and quickest method for me to rise to the task. Generally speaking, for each gig I would end up pulling out 1.6 to 3.15K to lose some of the edginess, and Howard would put a slight low/mid shelf on his output to achieve the clarity he desired. For outdoor shows, having an EQ like that to grab as the atmospherics changed was also very useful."

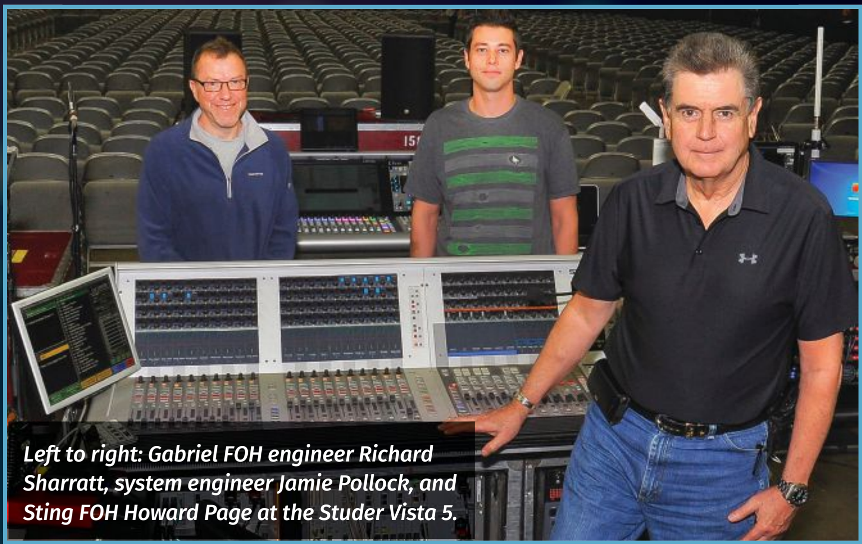
Within the Lake-based summing matrix everything was active all the time. This was made possible by configuring the board groups on each of the FOH consoles to deliver separate feeds to the PA so that at any given moment any combination of the performers could be heard.

"There were no crossovers, no system EQ, tuning, or anything else involved within the summing matrix," Page relates, describing the process further. "On my console I only had Sting inputs. But once I fed the PA, everything I sent summed with the content of Richard's desk. I did my thing and Richard did his, and provided we had an understanding between us of how the overall dynamics and volume would go, the interplay heard among the performers onstage was seamless.

"It was complicated to set it all up the

Co-headliners Sting and Peter Gabriel were joined by up to 14 musicians.

PHOTO CREDIT: PETER HUTCHINS



Left to right: Gabriel FOH engineer Richard Sharratt, system engineer Jamie Pollock, and Sting FOH Howard Page at the Studer Vista 5.

PHOTO CREDIT: STEVE JENNINGS

PA tech Erik Sandberg, Gabriel band monitor engineer Dan Ungaretti, Sting monitor engineer David Staub and Sting monitor tech Ken McDowell (left to right) prior to a show.

PHOTO CREDIT: STEVE JENNINGS

An SSL L500 Plus and Avid S6L awaiting instructions at stage left.

Gabriel monitor engineer Dickie Chappell at his SSL L500 Plus console.

PHOTO CREDIT: STEVE JENNINGS

first time — you have to be very careful to match levels and whatnot — but once everything is dialed-in it's set it and forget it for the rest of the tour," he continues. "You never have to touch it again. There were a lot of songs where both bands played together. At that point Richard and I were mixing not only to the system, but to each other. In situations like this it blends very well together, as long as everyone including the performers knows what they're doing."

IN THE SPIRIT OF THINGS

On stage, vocal microphones included Audix OM6s, a choice made based upon their rejection of background noise, clarity, and accuracy. Drum mics were what one might expect plus a little more: Sennheiser E901 and Audix D6 on kick, SM57s and Beta 57s on snares, Audix D4 and Sennheiser MD 421 on toms, Schoeps CMC6/MK4 on hat and rides, AKG C414s on overheads. Longtime Gabriel guitarist David Rhodes miked his cabinets with Sennheiser MD409s, while Tony Levin's bass and bass keys were on DI. Everything else was run direct.

Continuing in form and function with the tour's spirit of collaboration, the monitor set on Rock Paper Scissors spread itself across a sprawling sonic landscape where David Staub (Sting), Dan Ungaretti (Gabriel's band), and Dickie Chappell (Gabriel) held court.

Sting and his band (including Vinnie Colaiuta on drums and vocalist Jo Lowry) received their stage mixes from Staub, who rode herd over a Clair Global rig with the help of an Avid S6L mix system, Clair 12AM and CM22 wedges, R4 side fills, and 12 channels of Sennheiser 2000 Series IEM units with double beltpacks for each channel. Noel White, Colaiuta's drum tech, mixed the drummer's IEMs on a separate Yamaha CL5 console, receiving a full Sting input split plus stems of Gabriel's band generated by Staub.

"I'd been mixing Sting on an Avid Profile," Staub notes, "so it seemed like a logical move to use the S6L, as I could load my Sting band show file onto it, and then just simply add all the extra inputs and outputs I needed for this larger show. As we configured it, anyone in either Sting's or Peter Gabriel's band could hear anyone else from either band. The Peter Gabriel monitor engineers had a full split of both bands, as their split was done in the digital domain. I chose to receive as many as 22 analog stems from Dan Ungaretti — we were both on stage left right next to each other — and that allowed me to offer my band individual control over each Peter Gabriel band member's vocals and instruments."

EVERYTHING IN ITS PLACE

Over on the Gabriel side of monitors, both Ungaretti and Chappell used SSL L500 Plus consoles to direct mixes across 16 channels of Sennheiser 2000 IEMs and 14 channels of Shure UHF-R wireless mics and instrument beltpacks. With a Dante network running between the two expansion card-equipped monitor desks and the FOH console, submixes, effects, and shout mics could easily be shared using 16 channels of send/return for each console pair.

"We may have had full IEMs on the Peter Gabriel side of the stage," Chappell says. "But with the wedges and side fills on the Sting side, plus a large bass speaker he uses, we had a louder stage than normal for us. We got it all to work regardless, and kept to our core commitment of having all of the stage inputs coming to our SSL consoles from both bands. We didn't want to run any groups or stems, as Peter's detailed approach to things requires that kind of flexibility. He wants to be able to hear any individual input or mic, and regularly asks to."

Back out front, Sharratt's SSL L500 Plus was called upon for internal effects such as band reverb, tap delays, backing vocal reverb, SansAmp distortion, and more. He also kept a pair of Bricasti M7s at hand, one for Gabriel's vocals and another for drums, both of which were fired by MIDI program change.

"I think it's fair to say that we all entered this project with some trepidation," Sharratt states, underscoring its complexities and sheer scope. "At first none of us knew how it was going to work out on any level. Right from day one in rehearsals, however, Howard and I agreed to give each other the space we needed. If it was a Peter Gabriel song I would call the shots, and if it was a Sting song Howard had final say. The bands were simultaneously working things out among themselves in a similar fashion, and often what worked best for them was a case of less is more.

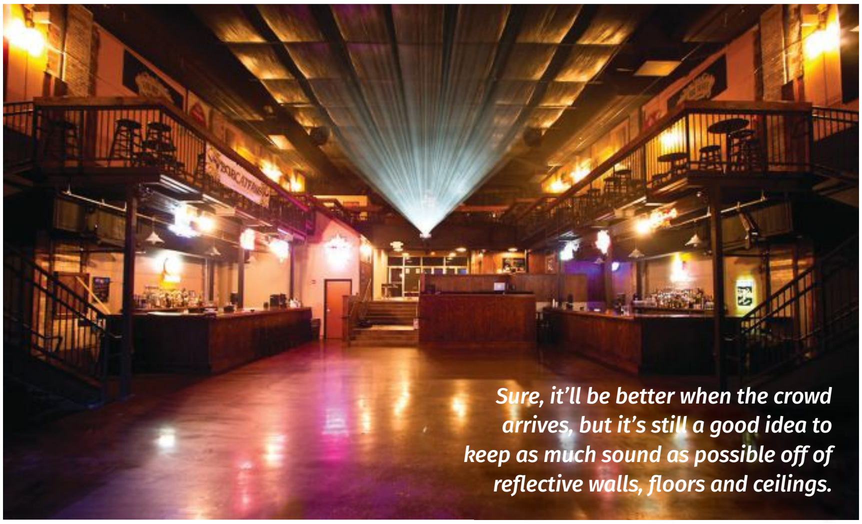
"Rule number one was to leave room for the two lead vocals, then to use caution when adding things into the other person's mix. In the end, I kept my vocal upfront, maintained clarity without being abrasive, and kept things warm, with body and depth. I had great inputs to work with, and a little EQ was generally all that was needed to find a place for each element in the mix, along with light compression on the backing vocals and staying on top of Peter Gabriel's vocals." LSI

Gregory A. DeTogne *is a writer and editor who has served the pro audio industry for the past 32 years.*

TAMING THE BEAST

Getting acoustically difficult spaces under control.

by Craig Leerman



Sure, it'll be better when the crowd arrives, but it's still a good idea to keep as much sound as possible off of reflective walls, floors and ceilings.

We've all been there. The room is highly reflective or the PA can't be placed in a preferred location for whatever reason. Or both.

Ideally, you've noted the challenges during a walk-through and can put some thought into an overall approach and a system design that will produce success despite the obstacles. My most recent "nightmare room" serves as an example of what can be done.

The space was square with two glass walls of full-length windows, two more walls covered with mirrors, a marble floor and a flat plaster ceiling. (I told you it was a nightmare!) Of course, the corporate client for the awards banquet my company was supporting expected great sound from both the presenters and the band.

The first thing I noticed during the walk-through was that the two walls of windows had drapes, so I asked the client if we could close the drapes. She allowed one wall to have the drapes drawn, but the other, opposite the stage (of course), was to be left unobstructed so that attendees could enjoy the view. Still, the drawn drapes on the on wall helped (quite a bit, actually).

Next, I asked if we could run some pipe and velour drape (provided by my company) behind the portable stage, which resided in front of one of the mirrored walls. She liked the idea, so we got a double bonus: improved acoustics and a fee for the pipe and drape.

I was then informed that a large video screen would be placed along the other mirrored wall. Checking in with the video company later, I was informed that a dress kit (velour drapes that frame a screen) would be provided, and while the kit's main function is to make a bare screen look better, it also would provide some damping.

Still, the PA and stage were facing the uncovered wall of windows, so even with the modest improvements elsewhere, there was still the potential for major refection and slap-back issues. Our solution was to place the main loudspeakers high up on my custom stands and point them down toward the audience, enhancing direct coverage while also helping to keep energy off the glass. In addition, the subwoofers were placed at stage left only – at stage right they were too near a reflective wall.

Well prior to the event, I contacted the band for some advance information. Significantly, I learned that the drummer has a big kit with large tom sizes and the guitar player uses a half-stack with a 4 x 12 cabinet.

We mitigated the drum sound with a plexiglass shield around the drummer, and with the guitarist, we spoke with him about turning down his amp and also pointing it sideways. But at sound check he insisted on having his amp face the audience, so with the mix, I brought the rest of the band up around him, and pretty much left him out of the PA. And, the bass player was kept in check by having him turn his amp down to a reasonable volume level, and to compensate, we put more bass in his monitor.

The space was still highly reflective but would have been unusable for an amplified system if we'd not been proactive. The client was happy and a couple members of the venue staff told us that it sounded better than any other event held in that room.

With that anecdote in mind, here are some tips and tricks that I've picked over the years that can come in handy facing acoustic challenges.

KEEPING IT CONTAINED

Start by working with the band. Control the stage volume so you don't have to fight against it. Packing blankets can be placed in front of loud instruments or amplifiers, as well as behind loud items onstage to absorb some sound and limit it from reflecting off a back wall. A mic boom stand with a horizontal boom (T shape) provides a quick and easy place to drape a blanket.

Plexiglass shields are another way to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contain sounds onstage. They work well for isolating instruments from bleeding into different stage areas and also from bleeding into the audience, while still allowing the performer to be seen. Most often seen in front of drum sets, shields can also work well in front of any instrument onstage. We often employ them to contain stray (and/or too loud) percussion sound, and they also help keep other sounds on stage from getting into the percussion mics, which are usually left on as the musician may switch between several instruments and "toys" during a song.

Digitally steered approaches, such as this Renkus-Heinz IC Live system, can be quite helpful in providing extra control in reverberant spaces.

Stage amplifiers can be loud. Turning a closed-back cabinet around to face away from the audience can reduce the bleed from the stage, especially if it plays into a blanket. Amplifiers placed at the side of the stage and aimed at the musicians still allows them to hear while limiting the "blare" into the audience. Angling the cabinet upward to point at a musician's head can mean turning it down, thereby cutting stage volume.

Simply rearranging the performers onstage can also help. At one show I convinced the two guitar players to stand next to each other instead of at opposite sides of the stage. The result was they lowered their stage volume because now they could clearly hear each other.

Of course, using in-ear monitors instead of wedges can reduce stage volume immensely. Not all performers are comfortable with in-ear monitors, so I've developed some approaches to help change them over.

Most people use ear buds with their phones/personal music players, so they're already familiar with the sound and feel of IEM. While custom-molded ear pieces provide the best fit, sound and isolation, standard phone ear buds can work. If a performer doesn't have his/her own ear buds or doesn't want to wear hard plastic earpieces, I offer a set of mine that use replaceable foam tips. In addition, we set up a few wedges and/or side fills with a basic mix of kick, piano, guitar and vocal, so if they decide to pull out their "ears," they'll still have monitors.

Adding the input of an open mic pointed at the crowd to a mix can lessen the sense of isolation felt by performers wearing IEM. It's also a good idea to make sure everybody is happy at sound check, even if it takes an extra song or two. The additional time spent can make all the difference.

Another way to eliminate monitor bleed is to place the main loudspeakers behind the band so they also serve as monitors. This can work well with performers who do not have a loud stage volume or with artists used to working with just a few monitors. Locate the loudspeakers so they won't generate feedback yet close enough for monitoring.

Unless the audience is located a decent distance from the stage, reinforcement is the name of the game when dealing with a loud stage volume. Only put instruments into the PA that are not heard well in the listening area. If the guitar player is loud enough for the room, simply bring the band up around the guitar. Same with some drummers, as many tend to be loud enough in small rooms.

Column loudspeakers deployed behind the band to also serve monitoring needs.

ALTERNATIVES TO THE NORM

For highly reflective rooms, the first priority is keeping the sound focused on the audience as much as possible while avoiding the walls and ceiling. This can be accomplished in a few ways. For large events, one of our primary solutions is a digitally steered PA that can be configured to direct its output, and without having to tilt or aim the cabinets. Asymmetrical coverage patterns can be used to further tailor coverage.

Of course, not every situation has the budget or need for a digitally steered system. For smaller shows in highly reflective rooms, loudspeaker tilting can be employed. Just as in my example earlier, use tall stands to get the loudspeakers higher in the air and then tilt the cabinets down to point at the audience. This helps eliminate reflections off the rear wall in comparison to loudspeakers deployed in a traditional upright position on a shorter stand.

Yoke brackets or stand tilters can be used, and some loudspeakers even have dual pole sockets, one for straight ahead mounting and one with a downward tilt. Make sure the stand is sturdy enough to hold a tilted loudspeaker as the center of gravity of the loudspeaker may not be directly over the pole. Sandbags can be placed on the stand base to provide additional ballast.

For loudspeakers stacked on top of subs, "Acoustic Aiming Devices" (AADs, a.k.a., small blocks of wood painted black) can be used to tilt the cabinets downward toward the audience. Make sure the stack is secured with straps. AADs can also help

tilt cabinets upward toward a balcony or to position floor wedges at a better angle.

Sometimes a room is so live and/or oddly shaped that the best thing to do is add loudspeakers on delay to supplement the main PA. This way, coverage is extended and clarity is often enhanced while keeping overall levels down. It's common at corporate events to utilize delay loudspeakers even in medium-sized breakout rooms so it won't be too loud in the front rows of the audience. Delays also reduce chances of feedback because the mains – located near the stage – are at a lower volume.

Another technique often used at corporate events with mostly speech programming is to roll off the lows below 150 Hz and the highs at 9 kHz to reduce the chance for feedback outside the human speech range (loosely 200 Hz – 8 kHz). If music is to be played through the same system, the speech channels can be routed through a submix to be band-passed.

QSC K Series loudspeakers offer a dual pole socket to attain a downward tile when desired.

FURTHER IDEAS

Column loudspeakers have made a big comeback in recent years. Their narrow vertical dispersion, coupled with a skinny form factor that can blend in aesthetically, has made them a hit with both install companies and live event professionals. Unlike the columns of yesteryear, today's models offer a wide frequency response and can get quite loud.

The décor and "look" of a show can take precedence over audio quality, and many event planners don't want "big ugly speakers" to be located where they need to be for coverage. Modern columns can provide a winning (and elegant) solution. Some models utilize small full-range drivers rather than woofers and compression drivers on horns, and they can sound very smooth with vocals and speeches because there's no crossover in the vocal region.

Stereo may be great when the audience is located between left and right loudspeakers, but a distributed mono approach may work better for a challenging space, particularly if a room is oddly shaped or the audience is in two (or more) separate areas. One zone may be next to the stage, or even just one side of the stage, with another zone on the other side of the stage and/or with another zone near the bar in the back of the room. The lobby could be another zone, and so on.

The first few seating rows at center stage might be the best place to see a performance but not necessarily to hear it. In this region, loudspeakers can be deployed for zoned fill (a.k.a., front fill). Any size of full-range cabinets can be used, but slim designs with rotatable horns, laid on their side, are optimum for this application. The object is to use low-profile loudspeakers no taller than the stage wedges to leave sightlines clean.

Underbalcony seats present another problem area to cover. Many theaters have built-in underbalcony fill loudspeakers that can be tied into, but if that's not available, small loudspeakers on stands can be placed in the outside aisles, pointed at the seating. Also, sometimes compact loudspeakers can be suspended from the balcony above to cover the center area or the entire under balcony.

Some events may not require more than a single loudspeaker location. If it can be covered with one loudspeaker, it should be. Adding loudspeakers generates interference between them, as well as being extra equipment that has to be hauled, loaded, set up and taken down. Sometimes just a second loudspeaker in a small, highly reflective space might generate enough reflections to compromise intelligibility.

The same goes with subwoofers – positioned on each side of a stage, they can create a "power alley" in the middle where their output couples. Placing a pair of subs on one side or next to each other in front of the stage will still produce plenty of low end but without the power alley effect. Further, a cardioid or steered subwoofer array can help keep low end from spilling on stage and reflecting off walls. Placing a single cardioid array on one side of may be all that's needed.

As noted at the outset, the best way to deal with a tough acoustical environment is to perform a walk-through of the site in advance. This leads to a far better understanding of the unique challenges, which in turn will result in a system and overall sonic presentation that meets client and audience expectations. LSI

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MAGICAL MYSTERY TOUR

A look at the functions, features and future of console DSP.

by Jonah Altrove

Ladies and gentlemen, step right up! I'll be your guide as we survey signal processing capabilities of modern digital consoles. We'll focus on several recent trends, check out the approaches of various manufacturers, and even speculate a bit on things to come.

DIVING IN

Let's get our feet wet with a look at the concept of "pooled" DSP. "Pooled" is my term for one of the two main approaches that govern various manufacturers' allocation of DSP in their desks, the other being "designated" DSP. You're fee to call these approaches whatever you like (that's what I've done), but the basic idea is this: The vast majority of consoles on the market have a fixed configuration – a fixed number of input channels, mix buses, matrixes, and effects slots. You know exactly what you're getting.

However, a handful of models take a different approach. For example, the DiGiCo SD7, SSL L500 and Roland Pro AV M5000 provide a certain number of "audio paths" or "mix paths," freely configurable as mono or stereo inputs, groups, auxes or matrixes as required by the show. Maybe we have an input-heavy performance (like an orchestra) one night and an output-heavy concert (30 pairs of in-ear monitors!) the next. No problem – changing the board's input-output structure is as easy as loading a show file.

Channel View on a DiGiCo S21 console's touch panel.

There are many variations on this concept. Roland's Open High Resolution Configurable Architecture (OHRCA), which is the backbone of the M5000, pulls everything from the same pool – input channels, buses, matrices, master buses, solo buses, talkback, oscillator, etc. The Midas PRO2 defaults to six effects rack slots and eight graphic EQs, but each effect slot can be "traded in" for an additional four GEQs if more are needed. Plus, the six-band parametric EQ on each output bus can be swapped for a GEQ if desired.

The Yamaha LS9 and Behringer X32 offer eight effects rack slots that can host effects, GEQs, or any combination of the two. Yamaha CL Series models and the M5000 have a designated graphic EQ rack but allow users to swap each graphic for an eight-band parametric. All of these are examples of how flexible configuration allows us to allocate the DSP where it's needed most.

If you're still fuzzy on the pooled versus designated idea, let's look at DiGiCo's "Flexi Channels," which are a great example of designated DSP allocation. The S21 sports 40 Flexi Channels, any or all of which can be made stereo with no loss of channel count, which is certainly convenient. (The same goes for the 16 buses.)

This means that the desk can process the equivalent of 80 inputs (40 stereo pairs), so if you're mixing 40 mono inputs, half of that DSP is going unused. A pooled approach would let you allocate the leftovers by "cashing in" the unused processing power into additional inputs.

Likewise, some engineers (myself included) prefer not to use graphic EQs (GEQs), so a rack dedicated to graphics represents "wasted DSP" for that workflow. I applaud manufacturers allowing the ability to swap these for parametric EQs (PEQs) and encourage others to consider following suit.

As you can see from these examples, the flexibility and utility of a pooled DSP approach is obvious – just make sure you plan ahead and allow for the unexpected (which should be expected in our line of work) to avoid coming up short. You don't want to realize too late that you need more inputs after you've allocated that DSP elsewhere.

SIZING IT UP

Just as you shouldn't judge a book by its cover, it's worth investigating the relationship between the physical size of the system and the capabilities of its DSP. We would expect the two to scale correspondingly, as higher channel counts are better managed with a larger control surface.

This is probably mental conditioning from the "golden era of analog," where gigantic desk equals tons of channels (and higher risk of getting a hernia flipping it over). And, if fact this is the case for some consoles, such as the Yamaha CL Series – the CL1, CL3 and CL5 get progressively larger and more powerful. Equipped with 18, 26 and 34 faders, respectively, the DSP scales accordingly – 48 mono inputs on the CL1, 64 on the CL3 and 72 on the CL5.

However, we also see systems that physically scale as DSP remains unchanged. The Roland M5000C is physically more compact than the M5000, but both sport identical DSP (128 audio paths). Same goes for the Midas PRO2 and PRO2C. A prime example is the Behringer X32 and Midas M32 families, which run the gamut from full-size consoles to 1U rack-mount DSP cores, all with identical DSP capabilities. (Note that there are significant differences in terms of onboard inputs and outputs, but our discussion here pertains to the system's ability to process the signals, now how we get them into or out of the mixer.)

Conversely, we also see configurations in which the DSP scales up or down independent of the control surface. Generally these systems physically separate the DSP into a rack-mount "mix engine" and use the console surface for control only – all of the actual processing happens inside the mix engine, while the control surface just tells the engine what to do. Examples include the Allen & Heath iLive, Midas 3/6/9/X, Yamaha RIVAGE PM10, and the Avid VENUE | S6L and S3L systems.

Interestingly, although the stand-alone DSP approach has historically been reserved for large-format systems, it has recently exploded into the compact market, thanks to approaches that eschew control surfaces entirely in favor of now-pervasive smartphones and tablets. Self-contained systems complete with I/O are available from a large number of manufacturers, including Yamaha, Soundcraft, Mackie, PreSonus, RCF, Midas and Behringer.

If your input requirements are minimal, you'd do well to consider one of these systems, as they completely eliminate the requirement of running a snake to FOH. Remember, the processing is done onboard, not by the tablet/phone, so a wireless dropout won't take down the whole show.

STANDARD ISSUE

These days, you can pretty much bank on the fact that a digital desk will offer a sweepable high-pass filter (HPF), four or more bands of PEQ, and a compressor and gate on every input channel. Consider re-reading that last sentence to fully comprehend how amazing this is. If you're a relative newcomer to the trade and cut your teeth on digital, you may not even give this a passing thought, but those of us who carted around racks full of analog output gear still feel somewhat spoiled by this modern development. For some perspective, imagine the stack of analog gear you'd need to duplicate the functionality of even the most basic 24-channel digital desk. Yikes.

Another welcome trend we're seeing in this regard is the bolstering of "stock" EQ and dynamics by offering some alternatives. Yamaha CL firmware v4 offers four distinct algorithms for channel EQ: precise, aggressive, smooth and classic. (Differences between EQ filters generally rest in technical facets such as filter slopes, relationship between gain and Q, ripple and overshoot in the passband, and the behavior of the interaction between bands.) You can usually swap the default comp and gate for expanders, de-essers, and duckers, and many desks also offer a compression sidechain and a key filter for the gate.

A couple of manufacturers offer some alternatives to the "vanilla" utility compressor for times when you want a little "character." For example, the Midas PRO Series offers a choice between "vintage," "adaptive," "creative" and "corrective" compressors for input channels, plus a fifth ("shimmer") on output buses.

Similar features are found on several other desks, which I find very helpful. Note that the differences between compressors is a complex topic in its own rite, but include whether the compressor responds to peak or RMS levels, whether the attenuation is applied in a linear or logarithmic fashion, knee curves, and distinct time-domain and distortion characteristics of certain circuit types.

Virtually every digital desk carries some form of onboard effects processing, and standard fare includes a healthy selection of reverbs and delays along with some tried and true modulation effects (chorus, tremulo, flanger). The past couple of years have seen an increase in the inclusion of "useful stuff: like amp sims, rotary speaker emulators, phase alignment tools, RTA/spectrographs, and some truly solid emulations of well-known studio units – BBE Sonic Maximizer, EQs from Neve and Pulse Technologies (Pultec), and compressors like the famous Fairchild, Universal Audio (UA) 1176, Teletronix LA2A, and more. Yamaha CL and QL consoles sport of "premium rack" of DSP dedicated to hosting these types



Yamaha CL Series consoles get progressively larger and more powerful.

of effects.

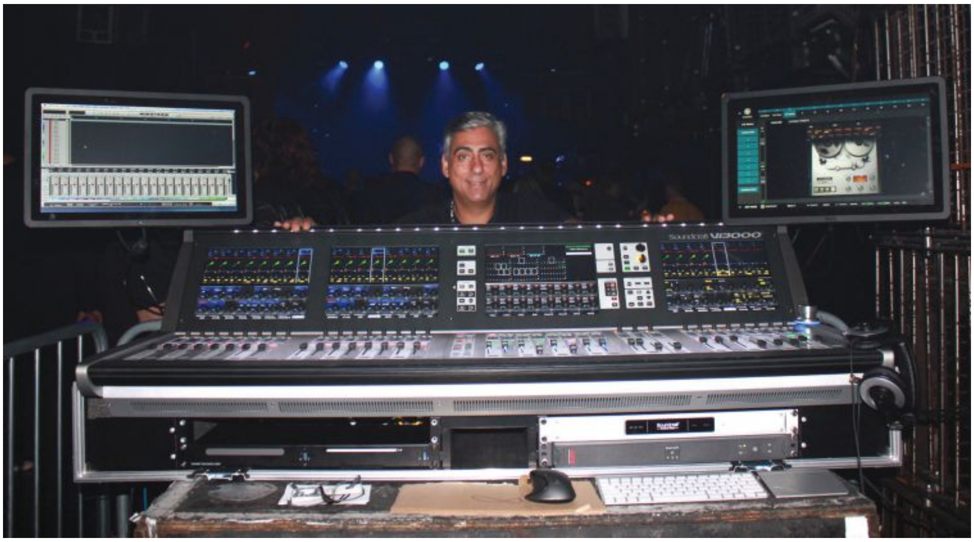
DiGiCo's recent Stealth Core 2 update brings dynamic EQ, multiband compression, and DiGiTuBe tube emulation to every input and output bus on every SD Series desk, along with massive increases in channel counts. DiGiCo also offers optional software upgrades that add features specifically designed for broadcast and theatrical applications.

Speaking of improvements, one thing I'm glad to see dwindling away is weird restrictions on which effects can be placed in certain rack slots. I once spent more time that I'd care to admit trying to add a Ditch Shifter to a desk's virtual rack, only to learn from the manual that the effect could only be placed in rack slots 5 or 7.

GET IN LINE

Just as important as the processing itself is the order of the DSP operations, or "modules." For example, you may want the channel's compressor located before or after the EQ, depending on what you're trying to accomplish, so most consoles allow you to swap two modules on a per-channel basis. Be sure to double-check the order when working on an unfamiliar desk, something I've overlooked more than once.

Something else that I find important to know is the location of the input channel's delay module in the signal chain. Allen & Heath GLD consoles place the delay after the pre-fader bus sends, which allows the user to delay drum inputs through the FOH mix, even when mixing monitors from FOH.



FOH engineer Jimmy Sarikas on tour with Culture Club a couple of years ago, where he deployed a Soundcraft Realtime Rack with his Vi3000i console.

The most common placement for the input delay is right at the beginning of the chain, which drags the stage monitors back in time – no good! It's common for consoles to offer multiple optional tap locations for inserts and direct outs, something to be aware of.

The Avid VENUE | S6L physically separates the DSP into a rack-mount "mix engine" and console surface.

NEED MORE PLUGINS?

Don't worry, it happens to the best of us. Sometimes the onboard effects just don't cut it, and we need a little something extra. There are several ways to skin this metaphorical cat. Software platforms like Waves MultiRack and Audioström LiveProfessor allow your laptop to act as a plugin host, whereas DiGiGrid SoundGrid and Soundcraft's Realtime Rack (which has the excellent Universal Audio UAD powered plugins) are two examples of dedicated hardware solutions.

Thanks to the wide acceptance of USB, Dante and MADI interfaces, it's usually pretty simple to connect an external plugin platform to your console of choice. Some manufacturers have designed their desks to run Waves plugs onboard. Such an integrated solution can save a lot of time and headaches if the plugin requirements are fairly complex. Your best bet is to consult with specific manufacturers to see what options they recommend.

I think it bears mentioning that some users have achieved a limited amount of success using digital audio workstation (DAW) software (Pro Tools, Logic, GarageBand, etc.) as a live plugin host. Do so at your own peril. These programs are designed for recording and editing audio files, so using them as real-time plugin hosts means having the software do something for which its not intended – never a good idea in a critical situation.

There's the risk of latency and stability issues, and DAW-based setups always seem to fail at the most important moment. If you seek DAW-style functionality, a popular solution is Ableton

Live, a full-featured DAW designed and optimized for from the ground up (and appropriately named) for live audio processing.

RUNNING LATE

Keep in mind that any DSP operation, no matter how basic, requires time for the calculation to be carried out. For example, a level change of -6.02 dB requires the console's engine to divide every sample value by 2. Of course, modern processors are blazingly fast, we're talking microseconds here, but it adds up.

This processing time, called latency, is a special concern when using third-party plugins, as many of them were developed for studio use, where latency is not a pressing concern. But even using only the onboard processing can still lead to trouble.

Imagine that you have two snare drum microphones into two console channels. Now let's say you add an insert effect on one of them. The input with the inserted effect is going through more steps of DSP, which means that its signal ends up slightly delayed in comparison to the non-processed input. Most of us are familiar with the delay caused by acoustic propagation – a snare hit arrives at the snare mic before it reaches an overhead, and a bass DI will always lead the miked cabinet – and that's why consoles offer input delay, to offset these effects.

Perhaps less recognized are the relative timing differences between channels that arise due to mismatched processing as they move though the desk. Whenever a signal is recombined electronically at the main mix bus with a delayed version of itself, a comb filter results – a series of alternating summations and cancellations that in many cases can be quite audible.

An important distinction here: We're not talking about absolute latency (the total time it takes a digital console to pass signal). This issue arises from mismatched relative latency; when some signals take longer to process than others. Differences on the order of 50 microseconds will create comb filtering in the audible spectrum.

How do manufacturers manage this? (Glad you asked!) Solutions to latency management range from just ignoring it (which was a common criticism of many early DAWs) to sophisticated compensation systems, such as the one employ by Midas PRO Series desks. If you tried our dual-snare-mic scenario a PRO console with input insert compensation enabled, for example, the DSP will automatically add the appropriate delay to our non-processed snare channel – and all other inputs – so that when our insert offset finishes with the processed snare send, everything else is waiting, and the whole mix remains phase-matched and coherent.

Again, remember that these types of design facets only deal with latency mismatches created by the DSP. They don't address sources that come into the desk already mismatched – for that, use a plugin such as Waves InPhase.

IT'S GONNA BE THE FUTURE SOON

The beautiful thing about DSP is that it's largely a function of software code, so we don't have to physically rebuild the desk to add new features. Bug fixes are the most critical part of firmware updates, but the exciting part is the new features that manufacturers dream up. I often find myself wishing a console would support a certain function. Here are some excerpts from my wish list:

- A good vocal doubler or "Abbey Road Doubletrack" effect.
- A grid-style matrix with delay and polarity inversion at each crosspoint. This would greatly simplify zoned musical theatre systems and foster driving delay clusters and sub arrays straight from the desk at smaller events.
- A convolution reverb effect.
- Re-allocatable channel DSP. If I'm not using the EQ or dynamics modules on a certain channel, let me double up somewhere else. One of my "go-to" studio techniques is to use several compressors in series, separated by EQs. I would love to be able to do this on a live desk.
- More flexible routing and busing. Let's say I want to send a vocal group to an IEM mix. Most consoles don't like this, as both buses are at the same level of hierarchy in the DSP. The DiGiCo S21 allows sending any bus to any other bus, which can be a real problem solver. Similarly, the Midas PRO Series allows input channels to be sent straight to matrices. This saves a lot of resources and also effectively provides extra monitor buses.
- Selectable high-pass filter slopes. Every FOH engineer's best friend, the HPF does a lot of heavy lifting. I'd love to see more desks allow me to swap between 2nd- and 4th-order filters.

None of these ideas are flaky or revolutionary, but they'd all be truly useful in my workflow. If you have a wish list of your own, why not drop an email to your favorite console manufacturer? Hey, you might just get lucky! LSI

Platforms such as Waves MultiRack can provide a welcome expansion of the console DSP palette.

Jonah Altrove *is a veteran live audio professional on a constant quest to discover more about the craft.*


woodwinds
piano
universal
drums
strings

PROVIDING ALTERNATIVES

Continuing the discussion of working toward a semi-silent stage.

by Mike Sokol

The first installment of this series (*September 2016 LSI*) drew some pretty interesting feedback. One reader called me a "Nazi" who was trying to interfere with his tone, adding that he would *never* turn down his guitar amp. On the other hand, a pretty well-known producer/engineer told me I was exactly on-point, and that some notable touring acts are beginning to implement the sort of monitoring techniques that I first began working with nearly 40 years ago.

In a nutshell, this project is not about forcing anyone to do anything. The mission is to provide monitoring alternatives that will allow the musicians and engineers to work together to provide a better mix for the audience. While some musicians feel that they're only playing for themselves, those who want to enjoy steady work understand that they're really playing for the audience.

Let's explore a bit of the psychology of volume levels on stage and why it's such a touchy subject. As a preface, please remember that I'm a child of the 1960s and 70s and played in some of the loudest bands imaginable. So if I can do this, then you can too.

CONDITIONED BEHAVIOR

Why do musicians want to play loud? And why do they get so upset when they're asked to turn it down? Growing up, there's a time when the "music bug" bites many of us. Mine happened at about age 14, which made it 1968. So it was "Mony, Mony" and "Lady Madonna" 45s being played in my room. Not terribly loud at that point, but then came 1969 and Led Zeppelin.

At that point, something ignited and I felt that this music needed to be played as loud as the phonograph would go. Deep Purple and Santana soon joined my record collection, as did Black Sabbath. So I crammed a full drum kit and a Kustom 100 guitar amp into my bedroom along with a Hammond M3 organ and started learning a lot of rock tunes.

However, while it was fun jamming along with the records, I soon found that my parents did not share my enthusiasm for loud music. Just as I would get playing a really cool (and loud) lead, they'd start pounding on the bedroom door and telling me to "turn that garbage down." Many of you probably recall similar episodes.

So now when someone tells us to turn it down, we recall those parental battles. Don't believe it? Try the opposite. Tell a musician to "turn it up" and it's almost 100-percent likely that they'll be happy with the acknowledgement that you like their playing. So telling them to turn it down must mean the opposite, correct?

It's one of the biggest sticking points with sound engineers asking a musician to turn down particular instruments. We have to get around the "parental reflex" we've been conditioned to accept as normal.

ASSEMBLING THE ELEMENTS

How can we convince musicians to try smaller amps, monitors, and so on? I recall exactly how I did it 35 years ago, but your mileage may vary. In the '60s and '70s there were few sound reinforcement systems powerful enough handle all of the stage instruments, so instead we used Marshall stacks, SVT cabinets, multiple Leslie cabinets, and really loud drummers.

But by the mid-'70s, after a few years of playing in a really loud rock band, I figured out that a steady diet of double SVT cabinets and full Marshall stacks was going to eventually kill my hearing. So I came up with a plan to reduce stage volume by putting everything directly into the PA.

In those days a lot of us used something like a Kustom 100 PA, driven by all of 50 watts and really only suitable for vocals. So I took my four (count 'em, four) Ampeg SVT 8 by 10-inch cabinets and turned them into subwoofers for the PA. Next I installed Gauss cone drivers in my EV Voice of the Theater cabinets, put Atlas PD60 drivers on the horns, and then added banks of piezo tweeters for everything above 10 kHz. This package was completed by wiring electronic crossovers with adjustable frequencies using 531 op-amps, joined by a bunch of hand-built Heathkit and Dynaco power amplifiers.

After a year of experimenting, it evolved into a pretty solid 5 kW system that would easily handle all of the instruments without resorting to big stage amps. I then built power soaks to tame our 100-watt Marshall amps down to 20 watts or so, and designed guitar wedge cabinets that would direct the sound of the Jensen or Celestion speakers right back at the musicians. I even designed and built DI boxes with speaker emulation so to eliminate mics and cross bleed on stage.

Finally, I put together effects pedals for the guitar players that had 12AX7 and 12AU7 tubes creating all of the desired compression and distortion, plus I added overdrive tubes with a master volume control into the Hammond B3 so it could scream at a lower volume. This was before any of these products existed commercially, so I was in uncharted territory.

A modern version of the author's ISO-Wedge guitar cabinet design. It's powered with an Orange Tiny Terror 7/15 watt all-tube amp, and is currently loaded with a Celestion 12-inch Cream Back speaker rated for 70 watts, so there's no power soak required. Note that this guitar wedge is positioned like a vocal wedge but it's definitely a guitar speaker pointed up at the musician's face. This improves sustain and feedback, plus it reduces the power alley that often occurs with a guitar cabinet on the back wall facing the audience. It includes a mount for a standard mic clip, so any type of mic can be used to attain the desired tonal coloration.

MIXED REVIEWS

Were the musicians happy? Not by a long shot. I soon had a revolt on my hands. For example, one of the guitar players admitted that while adding a power soak and small speaker/mic wedge sounded great, he couldn't get over the fact that nobody would see a stack of speakers behind him.

The argument went along the lines that unless he had his Marshall cabinets behind him (now unplugged, I might add), the crowd coming into the bar wouldn't recognize him. Now, he was a great player and didn't need any props, but he insisted that he couldn't play as well without a big speaker stack, even one that was disconnected. And after a while, he left the band.

To convince potential replacement guitar players that I wasn't crazy, I set up the 5 kW sound system in a field outside and let them bring their amps. That's when I found Karl. He had a guitar cabinet cut in half so he could put a 12-inch speaker on each side of the stage. I showed him that he could either play through his stage amp all by itself, or play through an off-stage cabinet running on a power soak with an SM57 feeding into the big PA.

Karl stepped out in front of the system to play a few leads, and almost immediately asked, "Why would I want to play though just the stage amp when I could be playing though all of the loudspeakers of this big PA?" I had my first convert.

THE MIX COMPONENT

How do we best choose what actually goes into which monitor mix? My view is that single-send monitor mixes are evil. I see them at churches and small clubs all the time, where a single monitor send drives

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several monitor wedges for multiple singers and musicians.

It usually doesn't work very well since everyone wants to hear themselves louder than the other musicians, and can lead to an "arms race" where everyone on stage keeps playing louder and asking for more monitor level, with the sound bouncing off the back wall louder than the PA. The rule we use in our own productions is one monitor mix per musician whenever possible. So if there are four players on stage, we put up four monitor mixes, and so on.

What each musician really wants to hear is "me" (themselves), so we start there, and then fill in with whatever other instrument or vocal they need to cue from. Simply putting everything in every monitor generally doesn't work well because now we're back to each "me" not being loud enough in the mix problem.

Separate monitor mixes also allows each monitor to be smaller and lower wattage, especially since they can now be mounted much closer to their respective musician. This isn't as complicated or expensive as it may seem. A modern digital console typically will have 8 to 12 spare aux channels that can drive active monitor cabinets, and the mixes can be easily setup with an iPad right on stage.

I especially like the recently released Galaxy PA6S for distributed personal monitoring since it has a real crossover and tweeter, and it's powerful enough for vocals on many small stages. It's an updated version of the original Galaxy Hot Spot with on-board amplification and

A Galaxy PA6S can come in handy for distributed personal monitoring.

equalization. Yes, there's not significant bass response from the small woofer, but there is usually already a ton of bass bleed on stage from the PA.

PHYSICAL REALITIES

Why does turning down the monitors make musicians turn up their stage amps? If they're used to straining to hear themselves due to excessive stage volume, then they'll fight for every decibel of monitoring. This contributes to even higher levels on stage.

Also, it's sad but true that many musicians have been deafened by decades of playing on loud stages. This isn't only rock musicians with loud amps – I've met a number of church organists that are hard of hearing after regularly sitting close to really loud organ pipes. To entice them to turn down the volume they're sending from an electronic organ to the sound system, I generally provide a personal monitor positioned as close to their face as practical. This can be done with something along the lines of a Galaxy PA6S on a mic stand right at keyboard level.

I've found that you can actually "gaslight" them if you want. Watch them react as you turn the keyboard level up in the system and down in their personal monitor cabinet. As the monitor level goes up, they tend to back off on the volume pedal. Provide less monitor and they step down harder on the pedal. No, this is not a passive-aggressive thing (but it could be). The goal is finding the "sweet spot" of monitor SPL that they're comfortable with.

One time, I dealt with a keyboard/organ player who kept moving her stage amp farther and farther away; eventually it ended up on the other side of the stage. Of course, all the other musicians were complaining that all they could here was her keyboard. So when she took a break I moved her monitor directly beside her keyboard and tipped it back to point at her face, turning turned it down to a moderate monitoring level and then listening to it myself.

When she returned, she complained that it would be too loud and that's why she kept moving it farther away. I explained that there was a DI send from her amp and we could turn it up in the house mix to whatever it needed to be. Nobody had ever told her that before, and she assumed that she needed to play loud enough to "reach the back of the room." Again, a lot of this is about educating musicians.

The same sort of thing happened at a church that had a Hammond B3 organ and twin Leslie speaker cabinets. The organist had turned down the volume on the cabinet right beside him and turned it up on the cabinet across the stage. He was pushing the swell peddle to floor so he could hear himself play, but it was really loud in the room and drowned out the choir. I convinced him to reverse the volume levels between the two cabinets. Problem solved and everyone was a lot happier.

Next time we'll begin to explore ways to control guitar amp levels on stage without compromising tone or playability. Several solutions will be presented, and you'll be able to pick and choose the one (or ones) that work best for your situation. LSI

***Mike Sokol** is the lead instructor for Live Sound Co, an AV integration and installation company in western Maryland. Visit www.livesoundadvice.com for Mike's educational articles and videos, and email him at mike@livesoundco.com with comments and suggestions.*

WORTHY DEPLOYMENT

A range of recent sound reinforcement system and technology applications. ***by Live Sound Staff***

A dynamic scene with JBL VTX arrays at Lollapalooza 2016 against the backdrop of Chicago's nighttime skyline.

BRINGING THE THUNDER TO LOLLAPALOOZA IN CHICAGO

Thunder Audio (Livonia, MI and Nashville) recently deployed a large-scale JBL Professional VTX Series-based main system at Lollapalooza 2016 in Chicago's Grant Park for a diverse lineup at the festival's Bud Light stage, including Ellie Goulding, Major Lazer, Lana Del Ray, The 1975, Future and many others.

"We always set up the system as flat as possible out of the box without a lot of processing or equalization – the system already sounds great with the JBL voicings," says Greg Snyder, vice president of sales at Thunder Audio. "We're handing the front of house engineers a very flat and neutral system that works well for any type of music, so they don't have to do a lot of work to get the sound they want for their artist. This makes it much easier to meet the artists' expectations and everyone had a great time using it."

The system incorporated two main arrays of 18 VTX V25-II modules, two arrays of eight VTX V25-II modules for out fill, another S28 subwoofers flown per side, and 30 S28 on the ground deployed in 10 clusters of three. The loudspeakers were driven by 72 Crown I-Tech 12000HD amplifiers, with Thunder Audio utilizing JBL HiQnet Performance Manager 2.0 software to configure, optimize and monitor the system.

"When we built the system, we chose to use Crown I-Tech 12000HD amplifiers for the power," Snyder adds. "There's so much power on reserve, and we get clarity out of the full spectrum of the system, whether it be low, mids, highs or subwoofers. If a dynamic drum roll or a downbeat occurs, the power is always there and it's always clean."

CRAFTING THE PJ HARVEY LIVE VOCAL SIGNATURE

Following the release of her UK number one album The Hope Six Demolition Project, English singer-songwriter and multi-instrumentalist PJ Harvey has been on the road headlining some of Europe's most prestigious music festivals. Her vocal microphone of choice on the shows is a supercardioid MD 431 from Sennheiser.

Harvey's front of house engineer, Howard "Head" Bullivant, who has been working with her since 1990, notes, "I originally chose the Sennheiser MD 441 for her vocal simply because it's a fantastically versatile microphone, great for vocals and just about everything you choose to point it at," he says. "Polly [PJ Harvey] used it on some of the vocal recording on her latest

PJ Harvey singing on her latest tour with a Sennheiser MD 431 microphone.

Hope Six Demolition Project album and in rehearsal, so it was always an option for the live shows, but in the end, when the whole PJ Harvey production was set up with nine musicians on stage and 23 monitors, we thought that the MD 431 had slightly better rejection, and Polly choose it above the MD 441 to use for the festival shows.

"This is a show where the audibility of the vocal is crucial," he continues. "I constantly recommend the MD 431 to other engineers. That's the best I can say." Monitor engineer Magali Couturier adds, "We use the MD 441 as vocal mic because of the sound and the look of the mic. The filter and boost have very nice attributes. We also use the MD 431 because, as Head says, it's easier to use in loud environments and with her band. And I'm using Sennheiser's 2000 IEM system because they are the best, with precise sound reproduction and no drop outs.

SEWING A SONIC SOLUTION AT THE KNITTING FACTORY

An audio upgrade at the Knitting Factory Concert House in Boise, ID is headed by a new d&b audiotechnik-based house system, designed and installed by the 3G Productions (LA, Las Vegas) install division. The 1,000-capacity room hosts a wide range of live performances – the first five artists after the upgrade included country, EDM, hip-hop, classic rock and heavy metal.

The new system comprises arrays of five d&b Yi8 and Yi12 cabinets per side with four d&b B22-SUB subwoofers under the stage. Two d&b E8 loudspeakers deliver front fill and two 12S-D are in the mezzanine for delays. d&b 30D four-channel amplifiers provide the audio system power.

"Bottom line, we couldn't be happier with the d&b system," states Knitting Factory COO Greg Marchant. "It's like night and day. The clarity is remarkable. Audio technology has come a long way since our original system and this is the perfect rig for the space. We're getting full frequency sound and so much more volume from a system that's half as heavy as the original cabinets. Same thing goes for the number of subwoofers that produce far greater low impact.

One of the new d&b audiotechnik Y-Series arrays in silhouette at the Knitting Factory in Boise.

"3G has a great eye for detail," he adds. "They measured the performance space and not only evaluated what brand would work best, but the specific deployment to best cover all of the areas in the room."

NO TEARS, NO FEARS WITH THE MIXING APPROACH

A recent tour by British duo Tears For Fears supported by Clearwing Productions (Milwaukee, Phoenix) utilized a Yamaha RIVAGE PM10 console for front of house duties, provided by Hi-Tech Audio

The Yamaha RIVAGE PM10 on the road with front of house engineer Doug Kimball and Tears For Fears.

(Hayward, CA).

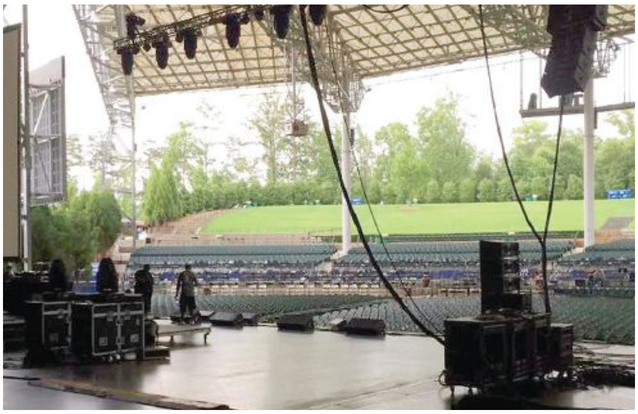
The PM10 was requested by front of house engineer Doug Kimball, who made the selection after a brief demo in Las Vegas and then received more training on the console from Yamaha prior to tour rehearsals. "I knew the Yamaha name, console reliability, and stable mixing platform," he states. "The PM10 is laid out in a very smart 'engineer-mixer' style."

Kimball also cites the encoder section of the console as being very helpful. "I can grab knobs quickly without having to page through to find what I need," he explains. "The custom fader banks help me put everything that I want, where I want it. And, the Rupert Neve Designs SILK processing is amazing; just inserting them adds more depth to the audio. People are stunned by the clarity and tonality of my mixes. Even our band mentioned it to me as their friends told them the same thing: 'sounds like a record.'"

REVIVING THE 90s IN GEORGIA

For Live Nation's recent I Love The 90s concert at the Verizon Amphitheater in Alpharetta, GA with more than 11,000 in attendance, AEE Productions (Atlanta) deployed EAW KF line arrays for main reinforcement. The concert offered live performances by a number of popular 1990s artists, including Salt-N-Pepa with Spinderella, Coolio, Tone Loc, Young MC and several others.

Specifically, AEE production manager Tracey Towne and his team deployed left-right arrays of eight KF740s to cover the farther seating areas joined by four KF730s for the seats closer to the stage. Side fill coverage was handled by left-right arrays of three KF730s stacked on two SB730 (subs), while the venue's installed system covered the lawn.



A portion of the EAW KF Series system providing coverage in Alpharetta for I Love The 90s.

"The updated voicings really make using the KF730 and KF740 together seamless," Towne notes. "When you were walking around the venue you didn't hear any difference from front to back. They also work well with the SB730 and SB1000 subwoofers that drove the low end."

The 16 SB1000z subs were stacked four by two under each array, while the four SB730s were ground stacked in front of the stage. "The SB730 is a longer throw sub that allowed us to get a longer, punchier throw to the back seats," Towne adds. "That way we didn't have to drive the SB1000z subs too hard while still distributing serious low end throughout the venue."

PHOTO CREDIT: ZACHARY BELCHER

ATTAINING NEW REACH (AND MORE) ON THE BEACH

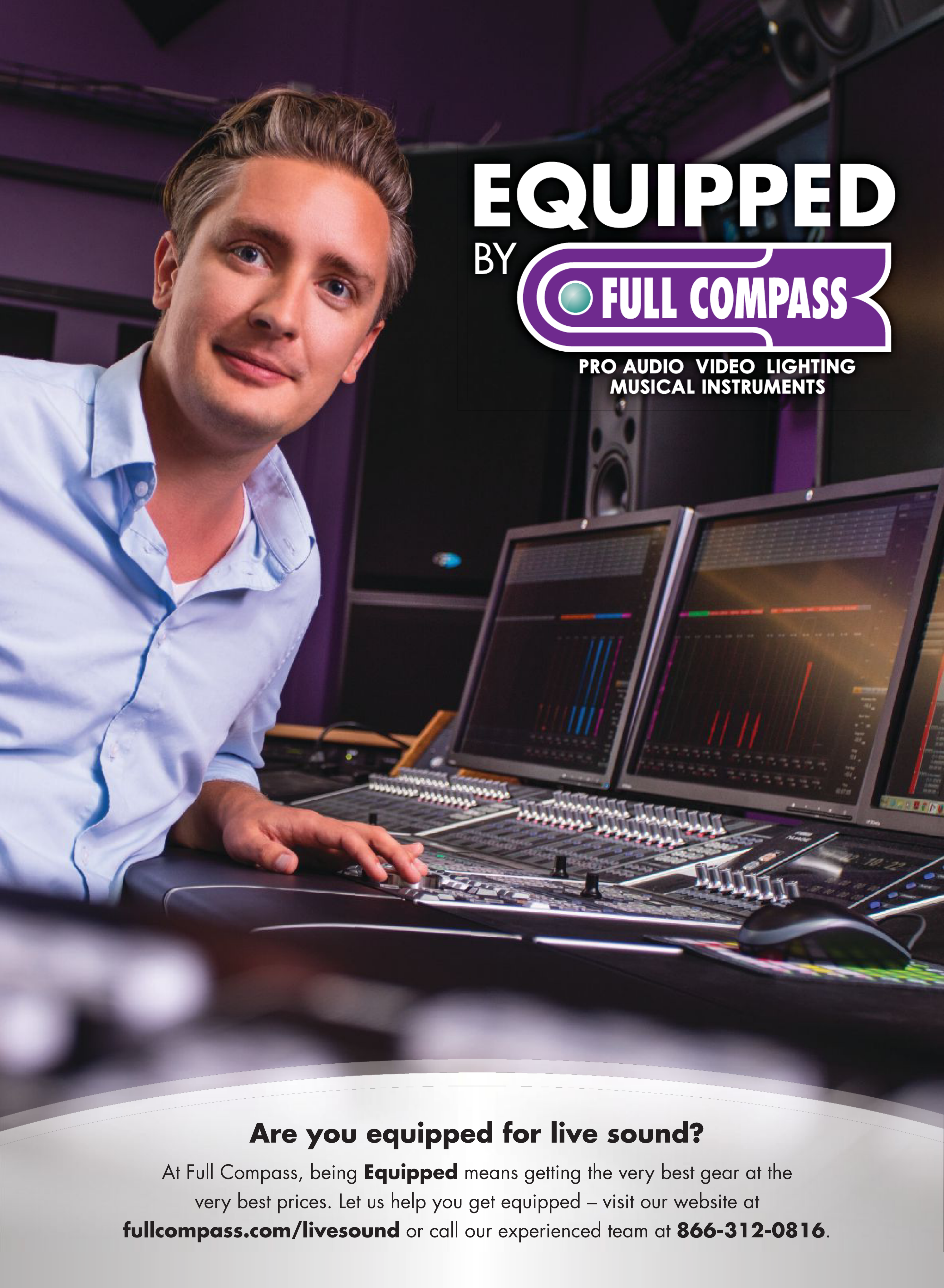
Meyer Sound LEO and LYON arrays deployed at a stop on Dierks Bentley's Somewhere On A Beach tour.

The Somewhere On A Beach tour by four-time CMA nominee Dierks Bentley has James "Pugsley" McDermott, longtime front of house engineer for Bentley and system engineer for VER Tour Sound (worldwide), utilizing Meyer Sound LEO and LYON arrays.

The system includes main hangs of 12 LEO with four LYON-Ms underhung and six flown subs in cardioid. There are two stacks of two on the ground per side, 1100-LFCs in cardioid, and then a dozen LYON-Ms for side fill. Eight UPAs and eight UPJs round out the package. The system has provided the versatility to cover arenas ranging from 5,000 seats to over 17,000.

"I'm able to accomplish dynamically much more effectively than I have with previous rigs," McDermott states. "With its consistency and range, the rig is a perfect marriage with Bentley's show – a reliable counterpart to the singer's vivacious country spirit."

For a show at the Shoreline Amphitheatre in Mountain View, CA, McDermott was particularly satisfied with the rig's reach. "I could very easily leave the long delays off because it has that reach that I can throw it 500 feet plus without really losing a whole lot of gain," he explains.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FUN WITH PURPOSE ON THE ROAD WITH KENNY CHESNEY

Backstage jams among the core band and many in the audio crew are a regular part of life on tour with Kenny Chesney, where a Mackie DL32R digital mixer plays a key role. "We have a lot of fun out on the road," reports drummer Sean Paddock, a 20-year veteran with Chesney. "Whenever we get a big enough dressing room, we set up our little studio space. The band and the crew love to come in and jam, so it's all-in, like a pickup basketball game."

The backstage setup is heavy on Mackie gear, including several portable loudspeakers and subwoofers, along with the rack-mount mixer. "We use the DL32R for our mixdown in the band room," explains Phillip "Sidephill" Robinson, Chesney's monitor engineer for more than 16 years. "It's so great to not have to drag out a snake; you can set the DL32R up, plug in your mics, fire up the iPads and away you go."

Phillip "Sidephill" Robinson doing some tablet mixing via a Mackie DL32R digital mixer backstage prior to a Kenny Chesney show.

Robinson points to the DL32R's ease of use as another asset: "We have a lot of different levels of technical expertise among the crew, and it's great to have a mixer that's intuitive enough for anyone to get their head around quickly." Paddock also cites to the DL32R's recording capability. "Since it has the ability to record," he says. "We can record the jams, and then come back a week later and listen to it and remember, oh yeah, that's what we did."

FACILITATING FESTIVAL MAIN STAGE CONTROL & ZONING

Returning to its new home in Douglas Park for the second consecutive time, the Chicago site for this year's Riot Fest had four stages served by Martin Audio components provided by Technotrix (Calumet City, IL), including MLA rigs for the Riot and Roots main stages. The eclectic lineup for the three-day punk/ska/metal festival included The Flaming Lips, Morrison, Julian Marley, Sleater-Kinney, Death Cab for Cutie, Rob Zombie, The Original Misfits and many more.

Julian Marley performing with an assist from Martin Audio monitoring at Riot Fest in Douglas Park.

For the main stages, the Technotrix crew needed to provide front to back coverage for the audience area while cutting off coverage at different points during daylight and evening hours. "Festival management wanted us to have the PA throw different distances during the day," explains Technotrix audio manager Brent Bernhardt. "Zone 1 (day zone) was about 50 feet past front of house with coverage dying hard beyond that so they could draw fans into the stage area and not have them spread out all over the lawn. Early evening or zone 2 was for direct support bands, and we'd open up MLA another 100 feet Then zone 3 would be for headliners in the evening and that was the system full on.

"We needed to maintain levels at front of house of 104 dB A-weighted and have these hard die-offs happen segmentally without affecting the rock show up front. To do that, we used a combination of different adjustable deltas and 'Hard Avoid' in the MLA control scheme to create the noise cancellation in specific zones during those times of the day. The good thing about MLA was that we could fearlessly deploy these presets without changing tonality, something you can't do with other PAs."

Both the Riot and Roots stages were outfitted with 11 MLA and 1 MLD modules per side with 18 MLX subwoofers ground-stacked in front of the stage in a cardioid deployment of six stacks of three each evenly spaced four feet apart to allow for camera platforms, cryogenics and other special effects. Each stage also had Martin Audio LE2100 monitor wedges, dual WS218X with W8LCs on top per side for side fill subs and two more WS218X subs with W8LCs on top per side for drum fill. LSI

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THE ALL-IN-ONE TECH

Taking on more production roles for fun and profit.

by Samantha Potter

I meet a lot of sound engineers and techs around my age (mid-20s) that strictly stick to audio and don't dare to touch anything else, whether it be to stay focused on audio or to not get roped into something they don't particularly want to do.

There isn't anything wrong with this. Laser-focusing on one discipline can yield some great results. I know that my passion lies in audio, live sound especially. I study it, read about it, train myself, and I find it the most thrilling of all of our disciplines. But I've noticed something: very few clients want to pay audio engineers properly. It's an endless undercutting massacre of our income.

As a result, something that I've done to make sure I get my preferred rate of pay is to learn other aspects of event production. It's hard to justify less pay for someone who does more than "just" sound. Again though, there's nothing wrong with sticking with and focusing on audio exclusively. Each of us must follow our own particular passion.

SETTING PRIORITIES

I'd like to present some of the other things I do to supplement my audio work. I'm a full-fledged member of a band called Funk Syndicate here in my home base of Kansas City. I serve as the sound engineer, although I typically tell people I'm the production manager. Getting the sound dialed in is my number 1 priority, but I do so much more. In addition to FOH and monitor engineer duties, I also handle lighting and run the band's social media efforts.

All eight band members use in-ear monitors, so getting their monitor sound correct is an ongoing and delicate task. As much as it hurts to admit it, the monitors are arguably more important than FOH (please don't throw anything at me). I have plenty of time to get FOH dialed in, but need to get the monitors optimized ASAP in addition to checking in regularly during the gig (they often request changes throughout every set).

There are many, many times that I'm trying to fix something with FOH when one of the musicians needs an immediate monitor change. I drop everything and fix that first. When the monitors are precise, the performance is better, which makes my FOH role easier. Musicians finding the 'pocket' can take a performance from OK to incredible.

In designing the band's lighting. I don't do anything crazy because I don't consider myself a lighting engineer. We've got LED bars and some other lights in the works, as well as DMX lighting software. I've bothered to learn all I can about the lights and the software so that this aspect of a show is as polished as possible.

I want them to look great, and further, enjoy seeing an entire show come together. I also know every song forward and backward, who is singing what parts, who's got solos, the length of each song, and so on. Knowing this so well is important because it helps me with not only mixing FOH, but also which FX and lighting presets to deploy, and when. "Uptown Funk," in particular, is a real hoot (people go crazy for this song, it's ridiculous). Doing three different things at once – FOH, monitors and lighting – is something I've found to be very thrilling.

CONSTANTLY BUSY

Now, it gets more daunting, because I'm also handling photos, videos and social media. Here it's important to add the caveat to make sure one thing is as perfect as it can be in a certain moment before moving on to the next thing. I don't give four mediocre performances. That's unacceptable. I strive for four fantastic performances.

The key is staying calm and knowing the technology as thoroughly as possible. Don't take on more tasks if you can't handle them, and don't sacrifice quality for ego. I see this way too often.

I frequently do more than just audio outside of working with the band. I also serve as an IT media supervisor at a large church and a graduate seminary, where I take care of all media-related needs like chapel production, videos, and large events.

What I've discovered as that I'm most excited when handling more than one thing because there isn't a single moment where there isn't something to be done. Then it becomes a challenge to wrangle every single thing in and see if it can all be made calm for a few precious moments. It's a game for me.

For some of the simpler events, I can set up audio and keep referencing back to it, but then also run graphics, and cameras, and streaming switches. "Why do one thing when I can do all of them?" is a joke I frequently tell myself.

I also have the luxury of working with the same musicians every week and mostly the same people outside of the band. I know these people, and thanks to digital consoles, the work gets saved for each event so that I'm not starting from scratch every time. Doing four things at once can be a little more complicated when there's potential

The author pursuing her primary passion, mixing live sound.

for total chaos in working with three different bands you just met two hours ago.

CONTINUAL GROWTH

Still, I'm an audio person first, and if I needed to pick one thing to do for the rest of my life, that would be it. But I've made my value higher and am much more indispensable in developing skills to do other things that nobody else can do.

So at the very least, if you love FOH, learn monitors. And vice-versa if you love monitors. This doubles the number of potential gigs you can work. If you're OK (or better) with lighting, seek to become the same with video. And so on...

Not only is it fun to learn about new things, but it makes you far more valuable. Why, when all other things are equal, would a company/client hire someone who can only do one thing when they could hire someone that can do two? Or three? Or four?

The bottom line is to never quit aspiring for more information and more experience. Never become stagnant. Always be moving, and always be striving. LSI

Samantha Potter *is an IT media supervisor and audio engineer for the largest Methodist Church in the U.S. and a production manager for Funk Syndicate in Kansas City. In addition, she's head of the Kansas City chapter for SoundGirls.org, where you can read more from her.*

DECISIONS, DECISIONS

Bringing common sense to gear purchasing.

by Teri Hogan

EDITOR'S NOTE: This fine article was featured in the September 2005 issue. We reprint it here in celebration of our 25th anniversary. The article was also accompanied by a fun cartoon from Frank Frombach, so that too is presented.

So you need a new "thingamajig" for your system?

The problem: there are so many brands and types of this thingamajig available, how do you best go about the process of choosing the one is exactly right for your situation? Sometimes buying a new toy – rather than being fun and exciting – is instead fraught with frustration and indecision. And the need rarely comes at a "financially convenient" time.

Purchasing professional audio gear can't be an impulse decision, because nothing in this business is cheap. If you guess wrong, it's very difficult (and likely financially not possible) to justify buying yet another unit. Therefore, the first rule is to determine right up front that there will be no hurry, and that emotions won't rule the day. "What do I *really* need?" is the key question.

Start by evaluating your market and considering your chosen place in that market. The needs of a small company primarily providing systems for local bands in clubs and smaller events are going to be quite different from a regional provider working concerts and larger festivals. Meanwhile, a regional provider shopping higher end pro gear still might not actually need the level of gear required by a large company specializing in touring packages.

A personal example: My company is a regional provider whose primary clients are promoters of concerts attracting crowds of 2,000 to 4,000, as well as festivals that host "rising and falling" national acts. Our current primary consoles are excellent desks that have all the necessary features, but once or twice a year, we encounter an artist that insists on more features, so we've lined up a reliable source (several hundred miles away) that cross-rents us the required desk. It's a much more economically prudent approach for our particular business.

Note that this approach keeps emotion and ego out of the equation, leaving a carefully considered solution meeting the real need.

O.K. – let's go shopping!

PESKY NUMBERS

Budget is the very first aspect to be determined:

- How much can you afford to spend on this thingamajig?
- Is there cash on hand or will a loan be necessary?
- How much should *realistically* be invested?
- And, will buying this item impact the ability to purchase other needed items now or in the near future? (This is what I call the "other needs" question, and it's a big one.)

Right off the bat, narrow down the money issue. With, say, $500 to spend, all of the "Cadillac" brands and units are likely off the table. But with $5,000 to spend, the MI stuff offered at retailers and the like probably isn't a consideration. Where to start looking depends upon what you have to spend.

Now, enter the "other needs" question, which helps further clarify the situation. In our business, for example, we've often had to settle for something a little more cost-effective to keep our inventory intact. Checks and balances.

The focus then becomes compiling a list of possible gear choices. There are several resources to access in compiling this list, topped by talking with fellow sound people, and even better, visiting them to see what they're using and how they're using it.

Also be sure to ask how they like a component and whether they'd replace it with something else if they had the chance. But keep this input in perspective – we'd all like to have the latest and greatest, but think budget first, not emotionally, and apply that filter to any input provided by others.

Two other important aspects to be sure to ask about: reliability and service. Has the component been reliable, and if not, what were/are the problems? And is the company responsive to answering questions and providing all assistance possible if a problem occurs? Remember that with some components of a lesser price, you get what you pay for, and this goes well beyond feature sets.

Browse pro audio websites and magazines, noting the advertisements and product reviews, while, of course, keeping these two aspects in their proper perspective. Make note of candidates that pique your interest while realistically fitting both system and budget needs.

Visit the web sites of manufacturers who offer possible choices, and by all means, explore the web sites of all pro audio manufacturers you can think of in order to see if they offer any units you weren't aware of. None of us can ever possibly know everything that's available.

WHAT IS REQUESTED

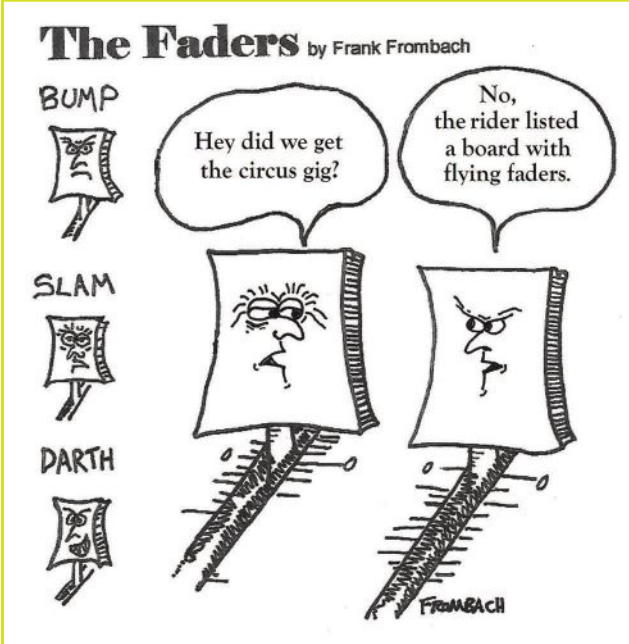
Your list should now have a number of viable "possibles" on it – time to start narrowing the focus. Here's where keeping all those riders of past bands/gigs can come in handy. (You do keep old riders, don't you?)

Search these riders and see what the band engineers have requested in the recent past. (Don't go too far in the past, because technology keeps advancing, especially these days.) Are there trends? Does anything stand out as requested repeatedly? And: Is there anything that some have specifically listed as taboo?

Once all of the information sources have been exhausted and there's been an opportunity to fully digest the research, it's time for a decision.

At this point, your list should be reduced to a much more manageable size. Time to compare apples to apples, and I do this by carefully reading the published specifications for all units on the list.

For more information, look up manufacturer sales representative firms (reps) serving your region. If this isn't provided on the web site, call the manufacturer to ask. You're likely going to place an order with one of these rep firms anyway, so it's good to make contact before your final decision. Besides, any good rep will bend over backwards to do whatever it takes to get your business.



Your list should now be quite short, and after consideration of the various bells and whistles on each unit, you might be down to the final candidate. But what if there are two candidates and you just can't pull the trigger on a final decision?

MORE SOURCES

Time to access additional recommendations from colleagues. This can be done via conversations (phone or in person), but also keep in mind there are several on-line pro audio communities where folks get together to air likes and dislikes.

Most notable among these are community forums such as the LAB (Live Audio Board) on ProSoundWeb. Before diving in to ask your questions, be sure to do a search and review previous conversations that could well address what you're looking for. If not, post your inquiry, and you're bound to get a least a few replies with first-hand, specific information.

Further, if there are any hidden foibles in any of these devices, someone is bound to bring it up. Issues of reliability and service will also likely be addressed, particularly if someone has had a bad experience or if a company has gone "above and beyond." Just don't forget that with open forums, a lot of what you get is opinion.

Once all of the information sources have been exhausted and there's been an opportunity to fully digest the research, it's time for a decision. Now, and only now, is the time to dive in!

Whether you float or sink depends on how well you've done the homework. And in the end, only you (and you alone) can decide which item is right. Happy shopping! LSI

Teri Hogan *is a long-time audio professional and was also the co-owner of Sound Services Inc., a sound company based in Texas.*

CADLIVE 3000 SERIES

Putting a wireless microphone system through its paces.

by Craig Leerman

CADLive 3000 Series wireless system with handheld transmitter.

The new CADLive 3000 Series from CAD Audio is a wireless microphone system that currently operates in two bands of the UHF spectrum (R band – 580 to 600 MHz; S band – 655 to 679 MHz) and will soon be available in the 470 to 489 MHz and 514 to 542 MHz bands. Available with both handheld and bodypack transmitters, the system incorporates features such as true diversity that's designed to minimize multipath interference and proprietary CAD-Lock Automatic Tone Encoded Squelch intended to eliminate unauthorized transmissions in the signal path.

The frequency agile design, when partnered with proprietary ScanLink technology, is designed to scan, select, and link to a clear channel. According to the company, up to 15 systems can be used simultaneously per frequency band. Stated frequency response of the system is 40 Hz to 15 kHz and dynamic range is quoted as up to 110 dB.

The WX3000 handheld transmitter has metal construction, incorporating the CADLive D90 supercardioid dynamic mic capsule. The WX3010 bodypack transmitter is supplied with both Equitek E19 earset and E29 lavalier mics as well as a WXGTR guitar cable.

Transmitters include SoftTouch multi-function on-off/mute switches, and the bodypack is also equipped with CAD-Tone circuitry to foster accurate reproduction of Hi-Z guitar and Lo-Z mic inputs. Both transmitters are stated as providing up to 15 hours of operation with AA batteries. Transmitter power is adjustable (10, 30, and 50 mW).

Transmitters and receivers are equipped with high-definition LCD displays as well as RF, AF, battery life, mic sensitivity and RF power metering. Receivers are housed in a half-rack, all-metal chassis and supplied with single/dual rack ears and a BNC relocation kit so antennas may be mounted in the front. Outputs are both XLR and 1/4-inch.

FIRST IMPRESSIONS

The system I was provided with for this evaluation included the receiver and the handheld transmitter, operating in the 580 to 600 MHz band. Out of the box, I was impressed that the system ships in a heavy-duty, foam-lined plastic case, great in general and especially for musicians who carry their own wireless.

I've previously done a Road Test on the D90 microphone (December 2015 LSI) and was so impressed that I added some units to my company's inventory. It's a great sounding mic, outfitted with a thin True-Flex diaphragm for articulate response and a PowerGap neodymium magnet for a "hotter" signal.

The handheld transmitter also has the rugged grill of the D90 and the display is nicely sized and easy to read at a glance. A single push button controls the mute function with a short push and the on-off/mute function with a longer push and hold. This button is slightly recessed so it won't be accidentally pushed. The mic feels very nice in hand and is comfortable to hold.

The receiver's front panel display is also plenty large and easy to read. Of particular note, it displays both group/channel and frequency in MHz at the same time. The display is flanked by set and scan buttons, up and down select buttons, and a power button. There's also a link button under a small IR ScanLink window.

The rear of the unit provides two BNC jacks for the antennas, a power jack, and the XLR and 1/4-inch output jacks. The receiver is solid and well built.

I set up the system on the bench, plugging it into my test PA, easily finding a free frequency and linking the transmitter with the IR ScanLink function. The transmitter's gain and transmit strength is set at the receiver, along with frequency. After the parameters have been set, the transmitter's IR window (located by unscrewing the battery compartment handle section) is held near the IR port on the receiver, and a single press on the receiver's link button sends all parameters over to the transmitter. It could not get any easier.

The mic sounds as good as the hard-wired D90, and there were no drop-outs at a distance of more than 100 feet, in a space with lots of loaded metal pallet shelving between the transmitter and receiver. Satisfied that the unit was working correctly, we took it out to a few gigs.

IN THE REAL WORLD

Our first application of the system came at the Live Sound Loudspeaker Demo at the WFX convention in Louisville. The receiver was set up next to the front of house console, and I was testing the system when I was informed that it was time to start a demo session. With transmitter already in hand, I used it for the entire session (more than two hours of constant use), with no dropouts anywhere throughout the 150- by 150-foot listening area. It also sounded quite good through all 26 of the ground-based and flying loudspeaker systems that were played multiple times.

Next up was a corporate gig at a Las Vegas casino ballroom located right on the strip. It's an area noted (some might say notorious) for RF congestion, and this property was no exception, particularly with an in-house show using more than 100 wireless frequencies right down the hall.

Our event had a wireless coordinator and I received a frequency from him, and again I noted how easy it was to change the frequency and link the mic. (The video guy adjacent to me was also impressed with this aspect.) We used the system for the moderator who led a panel discussion, and it worked without a glitch. The supercardioid pattern was a great help in picking up her soft voice.

The next day we replaced an audience Q+A (question and answer) mic with the WX3000 handheld. No problems, and it sounded great with a wide variety of male and female voices.

ONE MORE TIME

A few days later, the system was deployed for a Vegas-based band with both female and male singers that performs at casinos and corporate meetings. First up was the male singer, with a voice that was a little soft at sound check. I boosted the mic's gain a touch and it was dialed in.

In the next set we switched the system to a female singer. The mic worked well with her voice too, with the supercardioid pattern also keeping out some of the loud stage volume. After the set, she told me that she didn't want to give up the mic, noting that it "sounds so much better" in her monitor than her usual system. She was also really surprised when I informed her of the attractive price range.

The bodypack ships with E19 earset and E29 lavalier mics as well as a guitar cable.

The CADLive 3000 Series system is a winner in my book. It's rugged and feature laden, topped off by the proven D90 mic capsule. While I didn't get the opportunity to test it, the beltpack should also perform well – we've previously used the E19 earset mic and it's a quality unit. This is a professional package, with an added bonus of being relatively easy on the wallet.

U.S. MAP: $599 (with WX3000 handheld); $649 (with WX3010 bodypack, E19/E29 mics and guitar cable) LSI

Senior contributing editor **Craig Leerman** *is the owner of Tech Works, a production company based in Las Vegas.*

GOING BIG

The latest large-format line array systems.

by Live Sound Staff

It all started with the large-format genre, the place where the "big guns" kicked off the modern line array revolution about two decades ago. Since then we're witnessed the development of scores of models from manufacturers around the globe.

It's still surprising to some that line arrays have been around for more that half of a century in the guise of column loudspeakers, and most of them in earlier days were voice-range only. L-Acoustics V-DOSC was the first (in the mid-1990s) to show the concert sound world that more level and smoother frequency response can come from fewer drivers in a line array. Now, large-format line arrays (which we loosely identify as models with 12-inch and larger woofers) exist in a wide range of types and options. They're ideal for large-scale concert and event applications, really delivering in terms of output, and in many cases, are now much more than simply loudspeakers, better described as fully integrated systems.

The vast majority of the models we're presenting here incorporate dedicated power and DSP packages (onboard or rack mounted), as well as being outfitted with increasingly sophisticated control, networking, and optimization packages. Integral rigging is user-friendly and marked by precision and ease of use.

Driver configuration and enclosure design remains a defining factor. Some employ dual woofers with a center high-frequency section to provide horizontal symmetry. The simplest systems may just have a single cone and high-frequency driver. Quasi 3-way solutions use dual woofers, but low-pass one woofer at a lower frequency than the other, thereby eliminating cancellations at higher frequencies where their acoustic centers are farther apart than the wavelengths being reproduced.

True 3-way designs operate separate low-, mid- and high-frequency drivers, each in their own band. There are several approaches to horn-loading, which can provide higher sensitivity for additional power. And now, we're seeing some 4-way systems, with the mid-frequencies subdivided into two sections.

In the listings that follow, note that the horizontal dispersion spec is provided for all models. Vertical dispersion varies, and in most cases is dependant on array structure and configuration. Also, several companies offer more than one choice in this category, and for the most part, we've selected the largest model and/or the most recently introduced model, with the other options referenced.

Enjoy this look at the latest large-format line arrays.

d&b audiotechnik J-Series
www.dbaudio.com

Configuration: 3-way
LF: 2 x 12-in woofers
MF/HF (coaxial): 1 x 10-in cone (horn-loaded), 2 x 1.4-in-exit drivers on a dedicated waveshaping device
Frequency Response: 48 Hz – 17 kHz
Horizontal Dispersion: J8/J12 – 80/120 degrees
Size/Weight: J8/J12 – 14.2 x 43.3 x 22.4 in; 132 lbs
Electronics: d&b D Series amplifiers/DSP/loudspeaker management, accessible via d&b Remote network

Adamson E15
www.adamsonsystems.com

Configuration: 3-way
LF: 2 x 15-in neo Kevlar woofers
MF: 2 x 7-in neo Kevlar cones
HF: 2 x 4-in-exit voice coil neo Kevlar drivers
Note: MF/HF coaxially mounted on Co-Linear Drive Modules
Frequency Response: 60 Hz – 18 kHz
Horizontal Dispersion: 90 degrees
Size/Weight: 15.4 x 51.4 x 21.4 in; 176 lbs
Electronics: E-Rack with Lab.gruppen PLM amplifiers, audio & AC panels, Ethernet switch
> *Adamson also offers the E12 12-in line array*

RCF HDL50-A
rcf-usa.com

Configuration: 3-way
LF: 2 x 12-in neo woofers
MF: 4 x 6.5-in neo cones, symmetrical
HF: 2 x 3-in voice coil neo drivers
Frequency Response: 40 Hz – 20 kHz
Horizontal Dispersion: 90 degrees
Weight: 106 lbs
Electronics: Self-powered (4400 total watts) with 32-bit DSP and low noise converters, RDNet for remote monitoring and control

NEXO STM | *www.yamahaca.com*

The STM Series (Scale Through Modularity) large-format line array is ideal for sound company rentals in the touring and festival markets, as well as large corporate A/V rentals. STM combines the best of Alpha functionality with the technical innovation of GEO waveguide designs to deliver a powerful, flexible, and easy to use system. The system can scale up or down depending on audience size, from 1,000 to 100,000.

The M46 Main can comprise arrays of main cabinets only, or main plus bass, or bass plus main plus bass. The ability to add extra bass cabinets in order to increase power and headroom without introducing unwanted phase anomalies makes STM unique, as it's the first vertical array to offer scalable LF. Deploying the dedicated sub-bass cabinets, either in arrays or as ground-stacks, further increases options. Omni fill loudspeakers complete the coverage pattern, whatever the scale of the system.

An STM system (pictured above) was recently used for the Battle of the Bristol Tailgate Concert at the Bristol Motor Speedway in Tennessee, supporting performances by Kenny Chesney, The Band Perry, and Old Dominion for more than 50,000 fans.

TECHNOLOGY FOCUS: *Both M46 Main and B112 Bass modules share the same dimensions, weight and gravity center, forming seamless arrays. The STM S118 extends LF response down to 25 Hz.*

OF NOTE: *Deployed for Kenny Chesney's most recent tours, the STM system configuration consists of main hangs: 24 x M46 + 24 x B112 + 2x M28 x 2; sub bass: 24 x S118 flown x 2 (24 ground-stacked across the center); aux hangs: 15 x M46 + 15 x B112 + 3 x M28 x 2; delays: 12 x M28 x 2.*

KEY SPECIFICATIONS:

Components: 3-way
B112 Bass LF: 1 x 12-in neo woofer, hybrid horn design
M46 Main LF/MF: 4 x 6.5-in flat-membrane cones
M45 Main HF: 4 x 2.5-in neo drivers on Hyperbolic Reflector (4 x HRW waveguides)
Frequency Response: 85 Hz – 19 kHz (M46); 55 Hz – 200 Hz (B112)
Horizontal Dispersion: 90 degrees (M46)
Size/Weight: 13.8 x 22.6 x 28.1 in; 130 lbs (M46 & B112)
Electronics: NXAMP4x4 amplifiers/processors are outfitted with precisely matched presets for the STM Series

Martin Audio MLA
www.martin-audio.com

Configuration: 3-way
LF: 2 x 12-in woofers on Hybrid bass horn
MF: 2 x 6.5-in cones, horn-loaded
HF: 3 x 1-in drivers, horn-loaded
Frequency Response: 52 Hz –18 kHz
Horizontal Dispersion: 90 degrees (usable to 120 degrees)
Size/Weight: 14.6 x 44.7 x 26.5 in; 193 lbs
Electronics: Self-powered (6 channels), power & DSP control for up to 144 individual cells

Meyer Sound LEO-M
www.meyersound.com

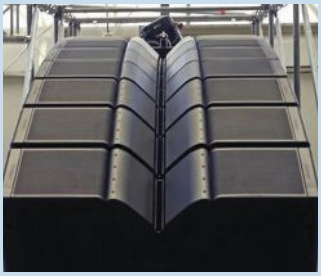
Configuration: 2-way
LF: 2 x 15-in woofers, vented
HF: 2 x 4-in drivers coupled to CD horn through REM manifold
Frequency Range: 55 Hz – 16 kHz
Horizontal Dispersion: N/A
Size/Weight: 17.8 x 44.4 x 23 in; 265 lbs
Electronics: Self-powered (3 channels), Galileo Callisto processing, RMS remote monitoring

> *Meyer Sound also offers LYON 12-in line array*

EAW Anya
www.eaw.com

Configuration: 3-way
LF: 2 x 15-in woofers, vented, Phase Aligned and Offset Aperture loading
MF: 6 x 5-in cones, horn-loaded with Radial Phase Plug & CSA apertures
HF: 14 x 1-in-exit drivers, horn-loaded
Frequency Response: 35 Hz – 18 kHz
Horizontal Dispersion: 70 degrees
Size/Weight: 16.8 x 45 x 30 in; N/A
Electronics: Self-powered (22 channels, each w/DSP), EAW Focusing & Adaptive Performance processing, Resolution 2 control software

Turbosound Flashline TFS-900H
www.turbosound.com



The TFS-900H is a true 4-way flown line array incorporating a total of 11 custom-designed drive units precisely configured over four frequency bands. The rigging hardware is fully integrated into the end-cheeks on each side of the TFS-900H cabinet, which also provides grab handle positions. Drop links at the front and rear of the box engage in the flygear of the box above in the array to give a range of inter-cabinet angles from 0 to 5 degrees.

TFS-900H cabinets are normally transported four-up on the TFS-DOLLY with a heavy-duty, shower-resistant cover, and can be flown directly off the dolly in blocks of four, with the rigging hardware already pre-configured for use. A simple TFS-GRID fly frame and TFS-TIP system is used to fly a typical array; no additional external parts are required to fly the system.

The cabinet is constructed from 15 mm birch plywood and finished in black shower-resistant TourTough finish with IP54 rated stainless steel perforated grilles. A recessed rear panel carries two parallel-linked Neutrik speakON NL8 connectors for input and link-through connections.

TECHNOLOGY FOCUS: *Amplification is provided by an advanced, powerful Lab.gruppen 20000DP 4-channel amplifier with Lake processing and Dante networking built in. The power capability of the TFS-900H's four frequency bands exactly matches the output of the 20000DP's four output channels.*

OF NOTE: *Driver maintenance can be performed without removing cabinets from the dolly because all the drivers can be accessed either from the front (the entire mid/high section is removable) or from the rear via the driver access doors.*

KEY SPECIFICATIONS:

Configuration: 4-way
LF: 2 x 12-in neo woofers, horn-loaded
LMF: 4 x 6.5-in neo cones, horn-loaded
HMF: 2 x 6.5-in neo cones on Dendritic waveguide
HF: 3 x 1-in neo drivers on Dendritic waveguide
Frequency Response: 70 Hz – 18 kHz
Horizontal Dispersion: 90 degrees
Size/Weight: 14.2 x 48.8 x 22.4 in; 224 lbs
Electronics: Self-powered (4 channels), Lake processing, Dante networking

Renkus-Heinz VerSys VL3
www.renkus-heinz.com

Configuration: 3-way
LF: 2 x 12-in neo woofers, vented
MF & HF: 2 x CDT1.5TN CoEntrant drivers with 2 x 6.5-in cones and 2 x 2.5-in drivers. Devices work with proprietary Isophasic Plane Wave Generator
Frequency Response: 45 Hz – 18 kHz
Horizontal Dispersion: 60, 90 and 120 degrees (user specified)
Size/Weight: 17 x 38.5 x 26.5 in; 175 lbs
Electronics: Passive or upgradable to self-powered RHAON-enabled version.

dB Technologies DVA-T12
www.dbtechnologies.com
www.americanmusicandsound.com

Configuration: 3-way
LF: 1 x 12-in neo woofer
MF: 2 x 6.5-in neo cones, sealed basket phase plug, horn-loaded
HF: 3 x 1.4-in-exit neo drivers, integrated CD horn
Frequency Response: 60 Hz – 19 kHz
Horizontal Dispersion: 100 degrees
Size/Weight: 15.4 x 23.2 x 17.2 in; 64 lbs
Electronics: Self-powered (3 channels), DSP w/presets, networking for RDnet remote control

L-Acoustics K2
www.l-acoustics.com

Configuration: 3-way
LF: 2 x 12-in woofers, bass-reflex
MF: 4 x 6.5-in cones, bass-reflex
HF: 2 x 3-in drivers on DOSC waveguides
Frequency Response: 35 Hz – 20 kHz
Horizontal Dispersion: 110/70 degrees symmetric; 90 degrees asymmetric (35/55 or 55/35 degrees)
Size/Weight: 14 x 52.7 x 15.8 in; 123.2 lbs
Electronics: LA-RAK with LA8/LA4 amplified controllers, includes signal, network & power distro panels

L-Acoustics also offers the K1 15-in line array

VUE Audiotechnik al-12
www.vueaudio.com

Configuration: 3-way
LF: 2 x 12-in neo woofers
MF: 6 x 4-in neo Kevlar cones
HF: 2 x 1.4-in-exit beryllium drivers on a precision waveguide
Frequency Response: 62 Hz – 19 kHz
Horizontal Dispersion: 90 degrees
Size/Weight: 14 x 42 x 16.8 in; 129 lbs
Electronics: Integrated VUEDrive System Engines with SystemVUE control, Dante networking

K-array KH5
www.k-array.com

Configuration: 2-way
LF: 2 x 15-in neo woofers
HF: 4 x 1.4-in-exit neo drivers
Frequency Response: 50 Hz – 19 kHz
Horizontal Dispersion: 110 degrees
Size/Weight: 20.5 x 47.9 x 8.3 in; 134.5 lbs
Electronics: Self-powered, integrated DSP and digital steering, multiple analog and digital inputs, onboard touch screen with matrix control

JBL Professional VTX V25-II
www.jblpro.com

Configuration: 3-way
LF: 2 x 15-in Differential Drive woofers
MF: 4 x 8-in Differential Drive cones on RBI (Radiation Boundary Integrator)
HF: 3 x dual-diaphragm/dual-voice-coil drivers on 4th generation waveguide
Frequency Response: 35 Hz – 18 kHz
Horizontal Dispersion: 90 degrees
Size/Weight: 16.3 x 48.1 x 24.1 in; 182 lbs
Electronics: VRack with Crown I-Tech HD amplifiers, BSS processing, HiQnet Performance Manager configuration/control software

Clair Brothers i218-M
www.clairbrothers.com

Configuration: 3-way
LF: 2 x 18-in woofers
MF: 6 x 6-in cones
HF: 3 x 1.4-in-exit drivers, horn-loaded
Frequency Response: 38 Hz – 20 kHz
Horizontal Dispersion: 90 degrees
Size/Weight: 19.9 x 54.5 x 30 in; 275 lbs
Electronics: Proprietary DSP and amplification packages offered; recommended power: LF – 2,000-4,000 watts, MF/HF – 1,000-2,000 watts (both 8 ohms)
Clair Brothers also offers the i218-LT long-throw model

Electro-Voice X2-212/90
www.electrovoice.com

Configuration: 2-way
LF: 1 x 12-in neo woofer
HF: 2 x 3-in titanium drivers coupled to pair of PDH Plane Wave Generators on a waveguide
Frequency Response: 52 Hz – 19 kHz
Horizontal Dispersion: 90 degrees
Size/Weight: 13.5 in x 28.8 in x 12 in; 93 lbs
Electronics: TG7 amplifier, RCM-28 OMNEO network/DSP module for amps, NetMax controller (& DSP-2 extension), Dante network module
Electro-Voice also offers X1-212/90 12-in line array

Alcons Audio LR28
www.alconsaudio.com

Configuration: 3-way
LF: 2 x 14-in neo woofers, reflex-loaded
MF: 4 x 6.5-in neo cones, slot-loaded configuration
HF: 1 x 14-in ribbon driver
Frequency Response: 50 Hz – 20 kHz
Horizontal Dispersion: 80 degrees
Size/Weight: 15.8 x 47.2 x 21.1 in; 138.9 lbs
Electronics: Sentinel10 amplified loudspeaker controller (4 channels) with specific drive processing with array-compensation, filter presets, system EQ, phase-matching, etc.

Outline GTO C-12
www.outlinearray.com

Configuration: 3-way
LF: 2 x 12-in woofers, hybrid band-pass loaded
MF: 4 x 6.5-in cones drivers, partially horn-loaded
HF: 2 x 3-in drivers on Double Parabolic Reflective Waveguides coupled to V-Power baffle
Frequency Response: 65 Hz – 17 kHz
Horizontal Dispersion: 90 degrees
Size/Weight: 14.3 x 44.1 x 25.8 in; 160 lbs
Electronics: T-Series amplifiers, iMODE processing platform, connection via standard Ethernet fosters system control

Verity Audio IWAC212
www.vaudio.fr

Configuration: 2-way
LF: 2 x 12-in neo woofers
HF: 2 x 1.4-in-exit neo drivers on a waveguide
Frequency response: 50 Hz – 18 kHz
Horizontal dispersion: 90 degrees
Size/Weight: 14.8 x 37.8 x 19.2 in; 105 lbs
Electronics: Requires outboard power and processing. Stated power handling (program): LF – 1,600 watts, MF/HF – 400 watts (both 8 ohms)

Coda Audio AIRLINE LA12
www.codaaudio.com

Configuration: 3-way
LF: 2 x 12-in neo woofers
MF: 4 x 3.5-in neo mid drivers loaded to advanced hyperbolic lenses
HF: 3 x 4-in neo planar wave drivers
Note: Tri-axial V-arrangement of LF, MF & HF transducers
Frequency Response: 50 Hz – 20 kHz
Horizontal Dispersion: 100 degrees
Size/Weight: 13.3 x 43.5 x 19.5 in; 147 lbs
Electronics: LINUS RACK40 with four amplifiers, DSP platform, interconnect; Coda EASE Focus simulation software
› *Coda Audio also offers AIRAY 12-in line array*

D.A.S. Audio Aero 40A
www.dasaudio.com

Configuration: 3-way
LF: 1 x 12-in neo woofer, rear-loaded
MF: 1 x 8-in cone/horn assembly w/ phase plug to enhance coupling
HF: 2 x drivers on waveguide
Frequency Response: 60 Hz – 20 kHz
Horizontal Dispersion: 90 degrees
Size/Weight: 12.4 x 37.1 x 15.7 in; 158 lbs
Electronics: Self-powered (3 channels), DSP, remote control via DASnet management platform

Ramsdell Pro Audio LA-15-3
www.ramsdellproaudio.com

Configuration: 3-way
LF: 1 x 15-in neo woofer, vented
MF: 2 x 12-in neo cones
HF: 2 x 2-in-exit neo drivers
Frequency Response: 50 Hz – 18 kHz
Horizontal Dispersion: 130 degrees
Size/Weight: 15.7 x 42 x 19; 106 lbs
Electronics: External amplification/processing; program power: 1200 watts LF, 600 watts MF, 300 watts HF
› *Ramsdell also offers LA-1-15 15-in line array and two 12-in models*

PK Sound TRINITY
www.pksound.ca

Configuration: 3-way
LF: 2 x 12-in neo woofers, horn-loaded/rear-vented
MF: 4 x 6-in CMI-loaded drivers
HF: 2 x 4-in coaxial drivers
Frequency Response: 40 Hz – 22 kHz
Horizontal Directivity: 120 – 60 degrees (intercabinet adjustment: 0 – 5.5 degrees)
Size/Weight: 13.8 x 54.8 x 23 in; 259.1 lbs
Electronics: Self-powered (6000 watts, RMS), DSP, rear-panel LCD or Kontrol software for control of DSP and vertical/horizontal settings LSI

NewsBytes THE LATEST NEWS FROM PROSOUNDWEB.COM

EAW has announced the addition of **Bryan DiFabio** to its Application Engineering team, where he is providing technical support and expertise to the company's global customer base.

"We're thrilled to welcome Bryan to the team," states **Adam Shulman**, director of marketing and application engineering for EAW. "His experience and background will be extremely valuable to our customer base. He is a great example of the new directions we will be taking the applications team in the coming year."

Big House Sound (Austin) has taken delivery of an **Adamson** S-Series rig, including a full complement of S10 line array enclosures and S119 subwoofers. Adamson application engineer **Brian Fraser** also provided two days of training that included an extensive overview of Blueprint AV simulation software as well as hands-on rigging and alignment of the new system.

"Our business services events from festivals that draw in 50,000 all the way down to clubs with a capacity of 100," explains **Roy Kircher**, owner of Big House Sound, picture here (at right) with Rod Nielsen (left) and some of the new inventory. "So when Adamson came out with the S-Series it caught our eye. It fits nicely with the demand we get for sound systems for mid-sized to smaller venues. Between the sound quality and how easy it is for one person to handle? It's a perfect fit for our business model."

Audio Visual Live (AVL), based in Lancaster, CA, has added **VUE Audiotechnik** al-12 large-format line arrays in upping its VUE inventory to more than 60 elements. AVL president **Dennis Layton** (pictured here with VUE president Ken Berger), whose company is noted for its work with HBO Boxing, took delivery of the new al-12s in time for the second annual Johnson Smooth Jazz Festival in Lancaster.

"VUE Audiotechnik manufactures some of the best speakers on the market, and any new VUE product release has my attention," Layton notes. "Adding the al-12 to my existing al-Class inventory was a no-brainer. The full-sized beryllium-infused line array system scales well to the needs of my clients whether for a 1,500-person casino show or a 10,000-person live and broadcast sports event."

Long-time **Avid** VENUE provider **Dimension Audio**, a UK-based rental company that supports a variety of areas of the live sound market, recently added two Avid VENUE | S6L mix systems to its product offerings.

"Avid live sound solutions make any kind of event easy, and after a demo of VENUE | S6L, I was convinced it would be our new desk of choice," says **Mark Boden**, director at Dimension Audio. "It's vital for us to offer the right solutions to our clients, and to me, Avid solutions mean longevity. I'm confident VENUE | S6L will perform to meet any criteria in a live situation, resulting in a solid return on investment along with customer satisfaction all round."

Martin Audio has appointed **Brad Stephens** to the position of southeast regional sales manager. In the newly created post, Stephens is engaging closely with the company's sales reps and customers in the region, improving relationships and helping to further penetrate vertical markets. He has previously worked in a variety of sales and product management positions for leading brands such as Avid, Bose and Turbosound.

"We're delighted to welcome Brad to the team," says Martin Audio vice president **Lee Stein**. "Having worked with him previously, I know his depth of knowledge and relationships within the industry are impressive, and believe that his skillset and credibility will enhance Martin Audio's presence in the U.S. and drive value for our partners in the territory."

Wolfgang Leute has joined **Meyer Sound** as sales director, Germany. His background includes more than 26 years of pro audio experience, most recently as the head of international sales and distribution at beyerdynamic GmbH. "Germany has always been an important market for Meyer Sound," says Helen Meyer, company co-founder and executive vice president. "We are pleased to welcome Wolfgang to the team to lead sales in this key market. His experience and network will add value for our existing customers and will grow our business in this region."

In addition, Meyer Sound's **Chris Mead** has been promoted to sales manager, Middle East, Africa and India, while **Sandeep Braganza** has been appointed sales manager, India, reporting directly to Mead. And, **Brian Chow**, based in Hong Kong, has been promoted to sales manager, Japan and South Korea. LSI

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THE RIDER GAME

Adventures in system requirements.

by The Gang in the LAB

W*e recently came across this entertaining yet instructive thread on the Live Audio Board (LAB) forum on ProSoundWeb, and present it here with a light copy edit and trim for space.*

Question posted by Ivan: I've been out of the direct rental business for 16 years now, so I don't often see riders anymore – but somebody passed this on to me and I was wondering how often you see this, and how it's "checked." I totally understand the whole "rider game," but has anybody actually insisted on something like the following: "The frequency response of the system shall be equal to or greater than 20 Hz to 20 kHz and must provide an SPL of 120 dB (no weighting specified) over 95 percent of the audience.

I'm not aware of systems that can actually get all the way to 20 kHz and down to 20 Hz, especially at the rear areas of the audience, at 120 dB "continuous." This was for a "bass heavy" live act. I guess I'm just wanting to "update" my knowledge on what people are requesting these days.

Other interesting things on a couple of riders were "NO DIGITAL CONSOLES" or "NO ANALOG CONSOLES" (both of those were all caps and bold – different riders, of course). With the "no analog consoles," there was lots of outboard gear – mic pres, compressors, effects, etc.

And one band had "No XYZ Speakers." XYZ in this case would be the most recognized brand and most rider-friendly speakers out there. I just found it "interesting."

Any thoughts would be appreciated (specifically on the frequency response). I know it's a "wish list" but how often do people actually meet it?

Reply by Brian: Unless you're dealing with A-list bands, the rider is a general guideline at best. At a festival situation, there is only so much that you can do. It's usually best to have a discussion before they arrive as to what on the rider is not realistic. Many times, they won't have a clue as to what the technical things in the rider mean anyway.

I highly doubt that there has been *any* system at a regional festival that could produce 20-20K at 120 dB over 95 percent of the audience. That would take one heck of a system, unless the audience was a crowd of 12.

Reply by Jamin: That's when you say, "Sure, we can do that. Here's what it will cost for what the rider requires." $$$$$$$$$$$ "Here's what it will cost for something adequate." $$$$ "Let me know which one you choose."

Reply by Steve: You missed one! "Here's what it will cost for something inadequate." $

Reply by Tim: I recently got one for an old country act that required one stack a side of any PA (not specified) that could do 150 dB at front of house!

Reply by Ivan: Maybe it was really tall.

Reply by Jamin: An old country act that needs 150 dB? Crazy. Just locate FOH a foot in front of the mains…and turn it up really loud.

Reply by Stephan: 20 kHz at 120 dB keeps all the dogs for miles around out of the venue. The old 20-20 was a hi-fi axiom from back in the day when no transducer could achieve it. Amazing that people in production would still dredge it up.

Reply by Bob: I can hear a dog whistle at 100 yards. Ever hear a dog whistle? They usually only know one song.

Reply by Tim: They're deaf. Or will be soon.

Reply by Steve: 20 Hz to 20 kHz… +/- how many dB? And how many of us can hear much over 16 kHz?

Reply by John: I can't hear a test tone over 11K. Checked it with a 30-year-old dancer and she topped out at 14K.

Reply by Steve: Mine drops off at 14 kHz and is almost non-existent at 16 kHz.

Reply by John: Are you a 30-year-old dancer? LSI

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