THE JOURNAL FOR LIVE EVENT TECHNOLOGY PROFESSIONALS FEBRUARY 2017 | PROSOUNDWEB.COM

LIVE SOUND INTERNATIONAL

INSIDE

STAYING INSIDE THE LINES

A Real World Gear look at medium-format line arrays.

THE SEMI-SILENT STAGE: RISE OF THE GUITARS

TIPS, TRICKS & PUNTS FOR MONITOR ENGINEERS

ADDING DELAY LOUDSPEAKERS (OR NOT)

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From the Editor's Desk

THIS ISSUE TAKES US in a bit of different direction than usual, and it's all good. Mike Sokol continues his discussion of developing a semi-silent stage by focusing on electric guitars, their amplifiers, and related facets. While it isn't a topic we typically pursue, in this case it's more than justified because these instruments (and others) are often a significant factor in stage volume.

It also fits with our ongoing goal of seeking to improve communication between musicians and audio professionals, plus, the fact that many of you are also musicians makes it all the more relevant. Mike's focused on numerous topics of vital interest since joining us a couple of years ago, and we're really glad he's on board.

Veteran engineer Nicholas Radina returns this issue with a piece that's a nice complement to the topic of what's happening on stage. He looks at succeeding in the often challenging role of monitor engineer, providing a wealth of tips and tricks based upon decades of experience.

Also don't miss Merlijn van Veen's interesting dive into delay loudspeakers. As always, it's backed up with considerable data and research in making several important points. In addition, Kevin Young delivers a profile of noted loudspeaker designer Dave Gunness, focusing on his work, motivations, career path, and additional areas of significance.

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

Keith Clark

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ON THE COVER: A d&b audiotechnik Vi8 line array flying at Salle Wilfrid-Pelletier, Place Des Arts, Montreal, Canada. *(Credit: MILK Images)*

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DiGiCo SD12

A digital console utilizing the latest generation of Super FPGA technology and offering 72 input channels with processing, 36 aux/group buses with processing, a 12 x 8 matrix with processing, LR/LCR bus with processing, 12 stereo FX units, 16 graphic EQs, 119 dynamic EQs, 119 multiband compressors and 119 DiGi-TuBes, 12 control groups (VCA), and SD Series Stealth Core 2 software that makes it compatible with all other SD Series sessions. The SD12 is also outfitted with dual 15-inch touch screens that provide 24 channels in one view, dual operator mode, and the ability for the right-hand screen to be the master. On the back of the console, there are eight local mic/line inputs, eight local line outputs and eight AES/EBU in/out for local digital sources, as well as two MADI ports, plus a UB MADI connection for recording at 48 kHz; 48 tracks of recording are possible with the console clocking at 48 kHz and 24 tracks if it is clocking at 96 kHz. There's also connectivity via optional DMI cards, including Dante and Waves SoundGrid. The iPad SD remote app offers remote control, expansion and show control. ***digico.biz***

Shure GLX-D Advanced Digital Wireless

A digital wireless system line and suite of products that includes the GLX-D Advanced Frequency Manager, rack-mount receiver system, remote antennas, and accessories. Frequency Manager is designed to allow users to operate up to nine simultaneous systems in typical conditions (11 channels in optimal conditions). With a new rack-mountable configuration, up to six GLXD4R receivers can be linked to Frequency Manager via the RF ports, and it automatically assigns optimal frequencies to all six receivers utilizing patented data communication via the existing RF cables. Linking two Frequency Managers together enables the use of additional rack-mount systems. ***shure.com***

Renkus-Heinz T Series

The next generation of the company's TRX Series loudspeakers incorporate redesigned HF and LF drivers. They also include the latest generation of proprietary Complex Conic Horns that are engineered to provide consistent beamwidth over a wide frequency range, and with transparency. T Series models are available in powered and passive versions, built to order with a variety of horn patterns for optimal coverage control, and can be further customized with custom color matching and weather resistance options. ***renkus-heinz.com***

Radial Engineering Key-Largo

Combines a keyboard mixer, computer interface and foot pedal in a single unit that includes three stereo input channels. Each channel is equipped with a level control and an effects send. A fourth input for a USB connects to a laptop to play back audio files and mix them with live keys. A stereo effects loop allows the addition of a rack-mount delay or reverb to a vintage Mini-Moog or to share it with the others and remotely activate the effects loop using the left onboard footswitch. Stereo 1/4-inch outputs feed stage monitors while balanced isolated outputs feed the PA system. A momentary footswitch is assigned for sustain, making it easy to connect to a piano patch. Side access MIDI connectors are available to link MIDI controllers to a laptop. ***radialeng.com***

dBTechnologies B·H Series

A line of four compact loudspeakers driven by onboard class D amplification. The 2-way designs incorporate a 1-inch compression driver feeding an asymmetrical horn designed to provide wide, uniform coverage. Woofer choices include 8, 10, 12 and 15 inches. Onboard DSP provides a choice of two EQ presets, flat or boost. The composite enclosures range in weight from 14 to 38 pounds. Three handles (one per side and another on top) foster portability. Cabinets can also be used horizontally in stage monitoring applications (with two angles). A pole mount cup facilitates stand mounting, and the loudspeakers can also be stacked atop subwoofers. ***dbtechnologies.com, americanmusicandsound.com***

Sennheiser XS Wireless 1

Six wireless system sets utilizing Sennheiser evolution microphone capsules and antenna switching diversity to foster reliable reception. The individual sets provide up to 10 compatible, preset channels in eight frequency banks, and are available in a number of ranges across the UHF spectrum – A: 548-572 MHz, GB: 606-630 MHz, B: 614- 638 MHz, C: 766-790 MHz, D: 794-806 MHz, E: 821-832 MHz + 863-865 MHz, K: 925-937.5 MHz. The receiver includes balanced XLR and unbalanced jack outputs. All sets come with receiver, transmitter, mic capsule or instrument cable, power supply unit, and batteries. Options include e 825 or e 835 cardioid capsules, ME 3-II headworn mic, ME 2-2 lav mic, CI1 instrument cable, and e 908 T gooseneck mic that clips onto brass instruments. ***sennheiserusa.com***

RCF HDL6-A

The smallest member to date in the HDL line array family of active 2-way line arrays incorporates two 6.5-inch midrange drivers with a 1-inch compression driver on a 100- by x 10-degree waveguide powered by an onboard 1,400-watt, dual-channel amplifier. The cabinets are outfitted with fixed locking pins and two rear handles for added portability. Inter-cabinet splay angles are adjustable in 1-degree increments between 1 to 10 degrees. Rear-panel controls include input gain adjustment as well as two presets, one for linear nearfield when in a pole mount or ground stack application and the other adding an HF boost for longer throw applications. Accessories are available for flying, pole mount or ground stack applications. In addition, the HDL12-AS, a 12-inch bass reflex, 1,400-watt active subwoofer, is designed to be flown with the full-range system or used in a ground stack situation with M20 eyebolt thread pole mount. ***rcf-usa.com***

PreSonus StudioLive Series III

The third-generation digital console/recorder launches with two models: StudioLive 16 with 17 touch-sensitive, motorized faders, 17 recallable XMAX preamps, and 24 inputs; and StudioLive 32 with 33 touch-sensitive, motorized faders, 33 recallable XMAX preamps, and 40 inputs. Capture multitrack software with virtual sound check is installed directly, recording up to 34 tracks to the console's onboard SD recorder. The Fat Channel processing section has received an overhaul, including a plugin-style workflow that includes vintage EQ and compression options on every channel. In addition to channel processing, all mix outputs have 6-band, fully parametric EQ. Sixteen FlexMixes can be individually designated as aux, subgroup, or matrix mixes. This is in addition to the four fixed subgroups, for a total of 20 mix buses in addition to the main mix, effects mixes, and solo buses. An AVB Ethernet connection enables networking compatible computers and streaming up to 55 channels of audio to/from a Mac or Windows PC. ***presonus.com***

Meyer Sound Galileo GALAXY With AVB

The third generation of the company's Galileo loudspeaker processing platform has been certified by the AVnu Alliance as fully compliant with the open AVB/TSN networking protocols developed through IEEE. As a result, GALAXY can be integrated into larger networked audio and video systems involving input, playback and processing components from multiple manufacturers. GALAXY incorporates the latest FPGA-based processing with up to 64-bit resolution for increased dynamic range, a lower noise floor, and analog-in to analog-out latency of 0.6 ms. ***meyersound.com***

QSC TouchMix-30 Pro Version 1.1

A firmware update for the company's most recent TouchMix Series model enables the mixer to have direct download of upgrades, and it's also able to check for and download all future firmware updates. The new firmware also brings improvements to the parametric EQ (PEQ) operation. When the channel RTA is disabled, the EQ graph now expands vertically to occupy the space that was formerly used for the RTA. In addition, the Q parameter now responds to a pinching gesture on the touch screen. This function is supported on the mixer as well as the latest release of the TouchMix Control app. Also available is the channel safe function, which prevents an individual channel's settings from being changed by a scene recall, along with numerous additional enhancements. ***qsc.com***

P.Audio VIA

A line array series comprised of VIA-8 and VIA-12 full-range cabinets joined by the VIA-SUB cardioid subwoofer and the SB-218 sub. All models are self-powered with 1,500 watts of RMS power (3,000 watts peak) matched to the remote controllable DSP with AD/DA converters operating at 96 kHz. The DSP makes use of FIR technology to help ensure linear system phase. An onboard RS-485 connector in each component facilitates remote control with the included software. VIA-8 and VIA-12 enclosures are both specified with 15-degree vertical dispersion, differing only in terms of horizontal dispersion (80 and 120 degrees). They're loaded with dual 8-inch neodymium woofers and a 3-inch large-format compression driver. The VIA-SUB cardioid sub utilizes a rear-facing 12-inch woofer to cancel sound radiation behind the cabinet, and the forward-facing 18-inch woofer has double neodymium magnets for higher flux density to provide more impact. The SB-218 The SB-218 sub is loaded with dual 18-inch woofers utilizing neodymium magnets. The rigging is designed to support up to 16 VIA-8 or VIA-12 cabinets, or 8 VIA-SUB cabinets. ***paudiothailand.com, onesystems.com***

Soundcraft by Harman Ui24R

A rack-mounted digital mixer and multi-track recorder that can be controlled by up to 10 devices via Ethernet or built-in, dual-band Wi-Fi. Ui24R provides 24 input channels, including 10 combo/XLR, 10 XLR, two line level, and two digital mix channels for a total of 24 simultaneous inputs. Studer mic preamps enhance sonic quality while Lexicon reverbs, choruses, delays, and dbx compression offer tailoring for vocals, acoustic guitar, and more. dbx AFS2 automatic feedback suppression is available on all monitor outputs. ***soundcraft.com, harmanpro.com***

Lectrosonics IFBR1a Firmware v4.0

The latest firmware for the IFBR1a belt-pack receiver allows it to be operated in two user interface modes: scan mode, exhibiting the original behavior of the IFBR1a receiver where users can locate and store up to five different IFB carrier frequencies by scanning for them; and direct entry mode, which takes away the scan capability and replaces it with the ability to program channels into the memories directly via the push-button and hex switches. In direct entry mode, five additional channel memories are available for a total of 10. New IFBR1a receivers in the UHF blocks shipping from the factory already have the firmware upgrade installed, while users with older units can have it installed at the factory via the parts and repair department. ***lectrosonics.com***

Allen & Heath dLive C Class

A range of compact surfaces and MixRacks joining the dLive family. There are three MixRacks – CDM32, CDM48 and CDM64, plus three new control surfaces – 19-inch rack-mountable C1500, C2500, and twin-screen C3500. The MixRacks house the XCVI Core (96 kHz FPGA), providing capacity for 128 inputs with processing and 16 dedicated stereo FX returns, plus a configurable 64-mix bus architecture with processing on all mix channels. Each surface and rack has a 128 channel I/O port, supported by an array of networking cards, including Dante, Waves, MADI, fibreACE optical, and more. The C Class is also compatible with S Class hardware and the ME personal mixing system, and is supported by an ecosystem of apps, Director software, and accessories. Surfaces employ the dLive Harmony UI, offering gesture touch control via 12-inch capacitive screens allied to color-mapped rotary controls. It's also possible to mix "surface-less" using a MixRack with a laptop or tablet for control. ***allen-heath.com, americanmusicandsound.com***

Celestion FTX1025 & FTX1530

The latest additions to the company's FTX Series of common magnet motor coaxial loudspeakers, available in 10-inch (FTX1025) and 15-inch (FTX1530) diameter sizes. Both offer full-range frequency response in a single self-contained driver by concentrically aligning low and high frequency drivers, designed to provide improvements in signal alignment and off-axis response. The common magnet motor assembly – in which the same magnet is used for both LF and HF elements – enables the voice coils and acoustic centers of the drivers to be brought very close together in order to enhance signal coherence and time alignment. The single magnet assembly also lowers both size and weight. A polyimide film HF diaphragm is engineered for greater power handling. The proprietary, second-generation Sound Castle soft clamping assembly reduces stress on the diaphragm, providing less distortion and enhancing reliability. ***celestion.com***

PK Sound Gravity 218

Utilizing proprietary Integrated Powered Adaptive Loudspeaker (IPAL) technology, Gravity 218 is designed as an LF module with increased output per cubic volume. An unrestricted vent helps streamline airflow, eliminating port noise and increasing output. The onboard 8,000-watt class D amplifier provides power to two high-excursion 18-inch transducers. Gravity 218 is also cardioid arrayable. It's available in install and touring versions, both integrating with PK's Kontrol software that enables remote access to DSP and real-time monitoring. Proprietary Automatic-Array detection utilizes built-in IR sensors to map relative cabinet positions in Kontrol for set up of arrays. The touring model incorporates an integrated rigging assembly. ***pksound.ca***

Kaltman Creations RF-Compass

An automated antenna tracker system for wireless microphone and IEM systems. It consists of two parts: a servo-controlled automatic panning device and a 5.8 GHz multi-channel miniature transmitter called the RF-Beacon that is provided with a belt pack mounting clip and is also small enough to be carried in a performer's pocket. With the performer carrying the Beacon, the servo unit is able to bi-angulate the signal, panning around to follow the transmission source. The UHF antenna is mounted on top, allowing it to also rotate to follow the movement of the performer and its own wireless transmitter.

The servo unit mounts to any standard mic stand and is compatible with any paddle or flat-panel directional antenna. It's able to rotate up to 360 degrees, and is rated to track the location of the transmitter at an operating distance of more than 250 feet. Each RF-servo unit and directional antenna is dedicated to one specific performer, but up to 40 RF- units can be used simultaneously without interference. ***www.kaltmancreationsllc.com***

Eminence N151M-8

A 1-inch-exit ring radiator compression driver designed to deliver dynamic performance. It utilizes a phase plug design intended to improve the distribution of forces over the diaphragm surface to lower distortion, smooth frequency response, and increase sensitivity. It incorporates a 4.5-ounce neodymium magnet, a 1.5-inch voice coil, and is rated at 45 watts (AES). Weight is 1.7 pounds. ***eminence.com***

Yamaha TF-RACK

A rack-mount version of the company's TF Series digital consoles, carrying the same core engine as each of the other models, as well as Steinberg Nuendo Live recording software, apps for mixing, and expansion capabilities. It also ships with with firmware v 3.0 that provides full fader views on its touch screen, the ability to add an administrator password, and other enhancements. Facilities include 16+1 stereo inputs, 16 outputs, 1-knob comp and 1-knob EQ, GainFinder input setup, and QuickPro optimized microphone, music and output presets that include popular model choices from Audio-Technica, Sennheiser, Shure and Ultimate Ears. An optional NY64-D Dante I/O card and Tio1608-D I/O rack are also available. ***yamahaproaudio.com***

Countryman B6 Connectors

A selections of detachable connectors for the company's B6 omnidirectional lavalier microphone enables the mic to quickly switch between different brands of transmitters, recorders, and other audio equipment. The line spans more than 400 models of equipment from AKG, Shure, Sennheiser, Lectrosonics, Sony, Wisycom, and others. Connector types include 3.5 mm, Hirose, TA3F, TA4F/Tiny QG, TA5F, LEMO, and XLR. ***countryman.com***

AKG by Harman C636 Master Reference

A handheld condenser microphone outfitted with a custom-tuned, 24-carat gold-plated capsule and hand-selected components. It is designed to eliminate feedback by combining a uniform cardioid polar pattern throughout the entire frequency spectrum with a custom suspension and grille that avoids unwanted sound reflections on the back of the capsule. A double shock suspension reduces unwanted handling noise. The capsule sits on an absorbent rubber bearing that cuts structure-based noise, while an adjustable balancing network cancels vibrations over a wide frequency range. A multilayer protection system prevents unwanted pop noises, consisting of the grille, a foam layer behind it, and a magnetically attached computer-modeled mesh layer on top of the capsule. ***akg.com, harmanpro.com***

Cerwin Vega CV Series

Three lines (CVE, CVX and CVXL) of powered loudspeakers for portable and install applications. CVX models will be available first, with both 2-way systems – CVX-10 and CVX-15 – incorporating a compression driver feeding a 90- by 60-degree (h x v) horn. The cone and compression drivers are powered by a 1,500-watt class D amplifier. Stated frequency response for the CVX-10 is 62 Hz to 20 kHz, and for the CVX-15 it is 54 Hz to 22 kHz (both at +/- 3 dB). They're outfitted with a 3-channel mixer with two combo XLR/TRS inputs, a TS stereo input, and three XLR outputs. Main/monitor and live/playback EQ modes are also provided. The CVX-18 and CVX-21 subwoofers incorporate a woofer (18- and 21-inch, respectively) that is angle-fired in order to reduce port turbulence. Driven by a 2,000-watt class D amplifier, both subs include two XLR/TRS inputs and three XLR outputs. When multiple units are used close together and rear sound rejection is desired, selection of an included cardioid mode can increase the forward volume by 6 dB while reducing rear volume by 12 dB. The CVX-18 has a stated frequency response of 38 Hz to 146 Hz, while the response of the CVX-21 is 35 Hz to 154 Hz (both +/- 3 dB). ***cerwinvega.com***


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Vintage Trouble
Volbeat
Weezer
Will.I.Am
Whitesnake
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Zac Brown Band
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LEADERSHIP + EXPERIENCE = SUCCESS

Eight ways to create a positive work environment and generate better results.

by Karl Winkler

A few years ago, I wrote a piece titled "Setting The Pace: Thoughts On The Power Of Leadership" (*LSI May 2014*) that generated a positive response from many readers, and it's also been about 18 months since I moved into an executive role at my company, so I thought it might be time to revisit the topic.

After reviewing the earlier article, I've found that my perspective has changed a bit, but thankfully I still largely agree with myself. (There's a shocker!) Yet the passage of time in general – and gaining additional experience in particular – leads me to share the following points and priorities for any who are (or might in the future) end up in charge of a team, a department, a project or even a company.

1. Learn to listen well. Most of us have the unfortunate habit of listening in order to respond, and/or thinking of something we want to say while a subordinate is speaking. This is a big mistake because most of us have a deep need to be heard and understood, plus we may be missing out on crucial information. Anyone in the role of manager or boss needs to fulfill the need for others to be heard by listening attentively – and – not passing judgment until the expressed ideas, concerns, or requests are fully expressed.

2. Your team is likely to come up with some great ideas. And probably some boneheaded ones, too. It takes a big person to promote ideas that aren't our own, but the results can be spectacular. Sure, leaders are there to shape those ideas and provide input and advice, but there's no better way to motivate people than to give them a real shot. And as mentioned in my original article, don't forget to give them the credit!

3. A primary role of leaders is to make the tough decisions. There are times when those decisions won't be popular, but that's indeed why we "get paid the big bucks." Turning down a new proposal or a budgetary expenditure usually isn't fun, especially if there's at least some merit in the proposal and if the person proposing it has merit as well. However, being touch when it's important can also serve to engender respect from associates and subordinates. After all, few (if any) of us want to be known as a pushover.

4. Don't avoid "the talk" with certain employees when necessary. Many people aren't sure what they can get away with until they find out in the form of a closed door meeting where the boss makes boundaries very clear. Again, most folks won't like it, and sometimes they find it petty (hopefully not too often) but ultimately if the discussion is handled in a professional manner, the result is usually mutual respect. It's also very important to stick to the facts of the issue and the behavior of the person in an impartial and unbiased manner. Try to keep emotions out of it; there lies the road to further difficulties.

5. Give ownership to subordinates. In the previous piece, I made the point about not making people change their work based only on your opinion as long as the work is valid. By giving employees a reasonable degree of license in how they do their work and even what work they're doing, they feel the pride of ownership. This usually motivates them to come up with more and better ideas because they're truly contributing and not just spinning their wheels in being told what to do.

6. Allocate time and funds for team members to continue their professional development. For one thing, we should all want the best and brightest on our side. Another way to look at it is to think about succession. Who will take your place if you get promoted, retire, or (heaven forbid) get hit by a bus? Related, it's important to make sure our team has the tools to get the job done as efficiently and professionally as possible. Don't let the equipment languish until it falls apart. There's certainly pride in using gear – if not the latest and greatest – that's at least serviceable, reliable, performs well, and looks good.

7. Keep up the communication! In all of my years in business, the one single thing that's a common thread behind the majority of problems is poor communication. Managers must foster clear communication in terms of direction and comprehension on both sides. It's absolutely critical to success. Also, be careful to avoid "You" statements, as in "You need to coil those cables over/under." Sure, it's true, but the collaborative approach is much more effective: "It's really helpful to coil the cables over/under, and here's why."

By creating the right kind of environment for our associates and subordinates, the cream will definitely rise to the top.

8. The majority of people are capable of rising to the occasion. Not everyone, and that's OK ("too many chiefs"), but we're interested in those that do. Finding them is largely a matter of following the previous points listed here. By creating the right kind of environment for our associates and subordinates, the cream will definitely rise to the top.

That said, most teams include those that pull their weight (and then some) and those who slack off. Whatever the case, the best way to go is creating a positive, motivating environment. The few who don't want to get with the program either depart of their own accord or can be "encouraged" to do so.

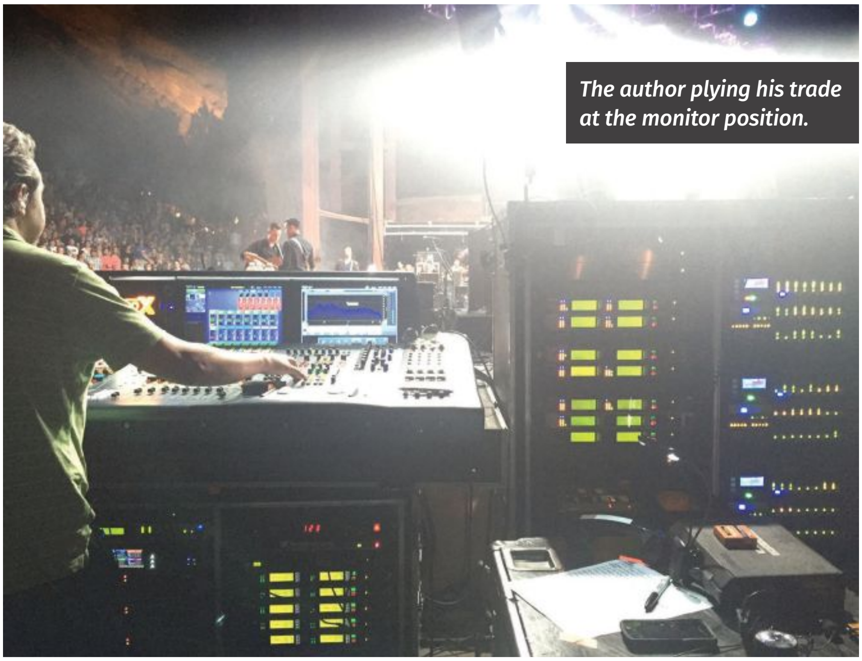
The ultimate goal is building solid team members as well as certain "star performers" will eventually be revealed and could, in the future, succeed us. There's no better feeling than handing the keys of a well-oiled machine over to the people that helped invent that machine and kept it oiled before you even thought to suggest it. LSI

***Karl Winkler** serves as vice president of sales/service at Lectrosonics and has worked in professional audio for more than 25 years.*

TIPS, TRICKS & PUNTS

A primer on the path to success as a monitor engineer.

by Nicholas Radina



The author plying his trade at the monitor position.

Working as a monitor engineer can be challenging, and I've had a fair share of hard lessons over the years with equipment (or lack thereof), facilities, artists, and more. Success is largely a matter of being able to manage a lot of moving parts.

I thought it might be helpful (and instructive) to share my experiences as well as some of the tips and tricks I've learned along the way — some at small gigs and others while touring with larger acts. Each has made me a better engineer as well as made my life easier. I'm going to focus primarily on general workflow ideas while mixing in-ear monitors (with sprinkles of other things that vibrate), as well as share a few of my favorite "punts."

THAT SOUNDED GREAT... I'VE HEARD BETTER

One big takeaway from early in my career, although seemingly obvious now, is that each of us hears differently. Aside from differing levels of hearing loss, we seem to have a different "curve" in relation to both frequency and taste.

On occasions where I'm working with a new act or using another engineer's file or console, I'm forced to learn very quickly what each artist wants and interpret how that artist hears. I literally "stand in their shoes" in wanting to hear what they hear (and feel), as well as checking the volume level setting on their belt pack for reference. A cue mix is no substitute for actually being in the same space — monitor world can be a bit of an island.

While mixing IEMs, one finger is always on the cue fader level, bumping up to listen to the mix at the artist's level and then quickly back down. I've found this technique gives me a fighting chance of "accurate" hearing throughout the gig. Listening at lower levels also seems to reveal issues much more quickly

Taking my first impressions of each mix into consideration, I begin to work on how I can make the artist more comfortable, confident and happier by using level, EQ, compression, panning and effects.

SPLIT FOR MORE OPTIONS

With both analog and digital consoles, I take advantage of splitting one input to multiple channels. I'll split the lead vocal input and other challenging lead instruments. This provides options when the singer and/or player hears their respective instrument in a way that's much different than the rest of the band.

I also do this when faced with one mic on the kick drum (as opposed to the common Shure 52/91A setup). I'll split the kick into two channels — one for the click and definition, the other is EQ'd for more low-end punch. When an artist needs more click or thump, I simply send more of whichever channel.

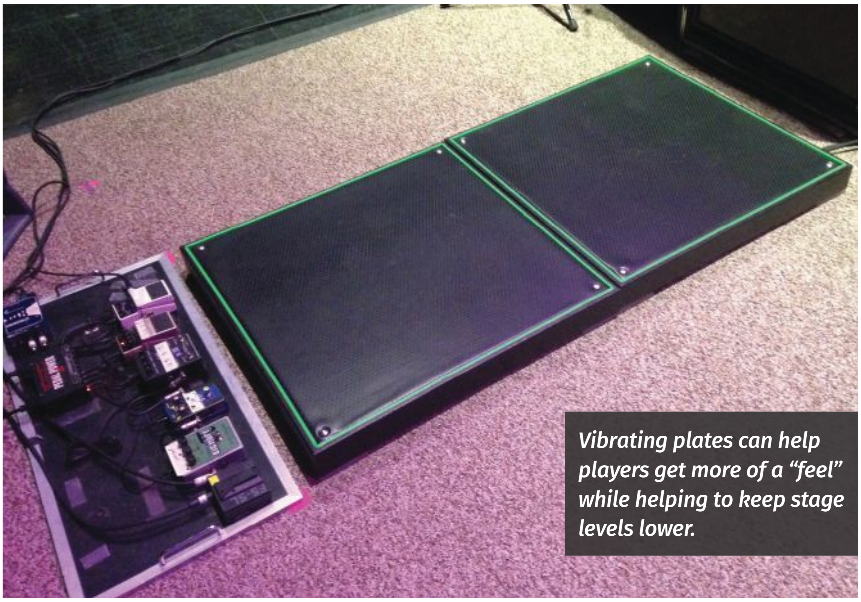
CHANNEL DELAY

Individual channel delay can really open up a mono source to feel more natural to the artist. With guitar amps, I view two mics as better (the whole "one is none" concept is in play) but when faced with only one mic, I'll split or soft patch one input to two channels. Delay the second channel by 10 to15 milliseconds, EQ a bit different, and pan to the opposite side of the first channel. Sneak in the delayed channel until the guitar sound widens. Also pay attention to where the guitar amp is in relation to the guitar player's ears.

A technique I learned from a talented engineer, Michael Larcey, to clean up and solidify drum sound is to align all of the kit's mics to the overheads. After learning this lesson, you can find me daily with a tape measure and a notepad, jotting down the distances. Solo-in-place is your friend, too. This concept can also be used with other inputs, but be careful of vocal mics and noticeable delay and phase abnormalities. By the way, while in position on stage, I often flip the polarity of an input or output, as well as delay, to see if it makes any improvement.

WHISH & WASH

The inevitable spill from cymbals and electric guitars that bleed into open vocal mics often gives me trouble. One technique I use is to take advantage of frequency dependent downward expansion. On a vocal channel, insert a BSS DPR901 EQ (hardware or software version) or a similar multi-band device and dedicate one of the bands — usually a wide Q around the 2 to 7 kHz region — to attenuate that frequency range and accompanying wash when the singer is not on the mic. I solo the channel or listen to the key on the expander, carefully paying attention to how the downward expansion behaves. It's a key to a natural sounding solution.



Vibrating plates can help players get more of a "feel" while helping to keep stage levels lower.

THINGS THAT GO BUMP

Stage volume is always a consideration and challenge, regardless of the music style being reinforced. Artists who learn to work within the space are a big help; with the proliferation of IEMs, the issue can get a bit out of hand due to inability to judge the true volume on the stage. Eliminating some sources of energy on the stage can surely help both monitor and FOH worlds. Common culprits are bass amp volume and drum fill bleed — both lower frequency generators. Things like drum set throne thumpers and vibrating plates can really help players feel those lower frequencies, much better than relying on loudspeaker fill.

STEREO PLEASE!

Over the years, I've worked with many artists who insist on using "one ear" and a wedge. I understand why — it feels more natural and less disconnected, at least first. The aversion to using monitors in both ears usually stems from having a poor experience early on with IEMs, either due to the lack of a monitor engineer, unsatisfactory mix experiences, or a lack of gear to provide a true stereo mix (sends, audience/room mics). I'm a big proponent of stereo and "both ears in" for three reasons:

1) The brain and auditory system function quite a bit more efficiently when listening in "stereo." (Experimenting with listing in mono, then again in stereo, makes this concept quite clear.)

2) Volume can be reduced.

3) Separation/masking. This is a big one for me. I always start panning, in broad strokes, the relative position of the instruments to the artist (lead singer), which helps him/her find physical "space" on the stage. Often one instrument is masking another, and panning slightly can help. (Tip: Pay attention to how one instrument or voice and its respective frequency ranges may mask another. Often, I gently reduce the 1 to 5 kHz region in other instruments to gain clarity in a vocal.)

AUDIENCE MICS

Wearing IEMs, even with a great mix and monitor engineer, can still produce a disjointed feeling in many artists. Introducing audience mics seems to help. Pay attention to how these are positioned and pointed (not directly at the close audience), and also provide a heavy dose of high threshold limiting. Putting them on a VCA and riding in between songs, as well as at the top and end of the show, can deliver additional needed energy to the artists. I also PFL (pre-fade listen) these inputs daily, listening for room abnormalities that need some EQ or polarity help.

COMPRESSION

I tend to add a bit of conservative soft knee compression, in varying amounts, to several inputs (such as vocals, acoustic guitars, horns, snare and keys). In addition, I'm looking periodically at the mix output levels from the desk and input levels on the wireless transmitters, watching for average level and clipping. Gentle, higher ratio, fast release compression on the outputs seems to help keep things in check. If running out of gas, I'll have a quick chat with the artist, asking him to turn up his pack, as I simultaneously turn the corresponding mix bus down, restoring a better gain structure.

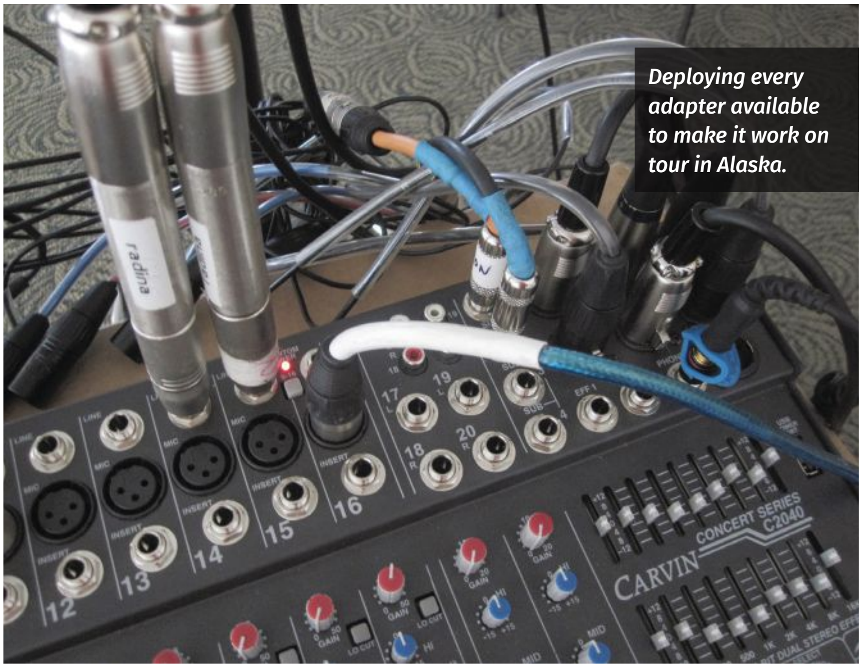
REVERB & EFFECTS

When available, I tend to dedicate individual reverbs to vocals and lead instruments, as well as drums. At the minimum, I give the lead vocal a dedicated effect unit/patch. Returning a small amount of a true stereo reverb, a touch of delay, as well as a tiny amount of a pitch shift patch, seems to help singers feel more comfortable. Adding a natural room or plate patch to drums seems to help solidify the sound. Often, I experiment with sending the drum reverb return to artists who don't like so much drums in their mix and sneak in very small amounts of the direct kick, snare and high hat channels for definition and time. I use a similar technique with background vocals — sending just the return to the lead vocal mix.

ALWAYS PREPARED

I have a soft spot for the (in)famous "punt" — a.k.a., a back-up plan, another option, what-if, and the like. This is where really creative solutions are born and, some days, what audio engineers are paid for. Aside from the obvious staples, such as a spare mic up and ready to go for the lead vocal and a mic cable always at the ready, console workflow can also provide safety nets.

Check out the photo at right: it's from a unique, albeit challenging, tour in Alaska. Advancing gear for several gigs in unorthodox, remote locations was quite an adventure. For most of them, the gear was delivered by an open pickup truck or four-wheeler (seriously). A daily supply of small analog mixers and less-familiar gear of many stripes set the stage for some unique punts. I used every adapter available to earn a fighting chance at success. The challenges usually revolved around finding ways to split and provide input strips for both monitors and front of house (I was handling both), which would give me with the ability to use the strip EQ (no graphs to be found most days) for wedges, IEM mixes and return reverb to one of the ear mixes.



Deploying every adapter available to make it work on tour in Alaska.

A "punt" workflow I use on most digital desks that have the mix bus capacity is to having a "spare" mix with transmitter and body pack assigned to a matrix. In the event of transmitter failure or pack failure, this matrix can then be fed whatever mix bus I need — either by spinning up the appropriate mix or simply feeding all mixes, in mute state, to that matrix and un-muting the one needed. I also create a guest mix with corresponding transmitter and pack for special guests who seem to always pop up last second, as well as other crew and tech packs/mixes.

My cue pack always operates in "engineer mode" or similar. This allows me, when issues arise, to "tune-in" and hear what the artist hears. In addition, I make this mode active on my spare pack as well as guest packs, providing the ability to feed them any mix I desire, including my own cue.

TALK BACK TO ME!

Talkback mics with corresponding "push to talk" foot switches like the Radial Hot-Shot line output selector, when strategically placed on the stage, allow the performers to talk to one another, as well as to me, other crew members and even the tour manager or lighting director if they have packs. When available, I assign a pack and mix for each crew member, also in engineer mode. This allows me to create a mix for each of them and also to talk directly to them if needed.

Since their packs are in engineer mode, they can switch back to their respective principals when necessary. Often the guitar tech or drum tech, while listening to their artists' mixes, will point out something that doesn't sound right and I'll revisit the mix and work out a fix.

ALWAYS AN ADVENTURE

What I enjoy most about living in monitor world is finding innovative solutions to challenges, learning how to keep my eyes on several people at once, and sharing the energy of the show – yet always being ready to duck if a '58 comes flying in my direction. Next time I'll address my workflow for mixing wedges, patching and some other goodies that happen post console. LSI

***Nicholas Radina** is a long-time audio engineer and musician based in Cincinnati who also tours as the monitor engineer with the band O.A.R. He invites your input via his website at NicholasRadina.com.*

THE BEAT GOES ON

Front of house engineer Stephane Plisson mixing with Waves on tour with Kids United.

Deploying the machinery of the craft to exceed expectations.

by Live Sound Staff

A TIGHT MIX PACKAGE FOR A GREAT CAUSE

Front of house engineer Stephane Plisson is utilizing a Waves Audio eMotion LV1 software mixer for the latest tour by Kids United - a French musical group comprised of six children - created to support UNICEF campaigns that's proven quite popular, with many concerts sold out in advance.

"I've been using Waves plugins for more than 10 years, and believe me, have been waiting for the eMotion LV1 mixer for a long time," Plisson says. "It was a sweet dream before just to use plugins in live mixing, but now, with the LV1, it's all there in one tight package."

His setup includes two Macs – one for the LV1, and the second for live recording using Waves Tracks Live software and for virtual sound check via SoundGrid, with two Waves Extreme servers. Additionally, he uses a DiGiGrid IOC audio interface for monitoring, an AES output for the system as well as for the Waves MaxxBCL insert, and a DiGiGrid MGO interface for the MADI preamp from the monitor desk's MADI stage box.

"A comfortable workflow is an extremely crucial issue for me. I was searching for a new and solid system, and then I discovered the LV1," he notes. "It allows me to use my beautiful Waves plugins, and in particular it lets me use them in ultra-low latency by using Waves servers. The combination of SoundGrid audio interfaces (DiGiGrid IOX + MGB) with Waves servers is absolutely remarkable."

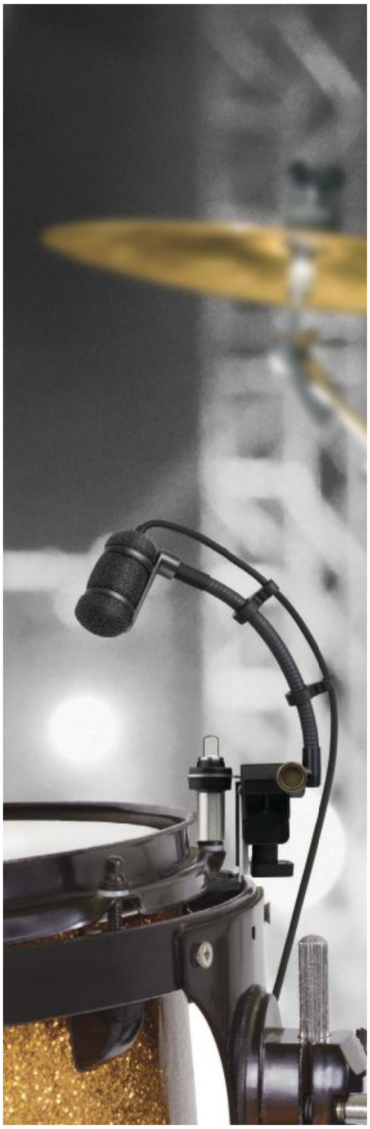
COVERING THE DISTANCE IN THE MIDDLE EAST

A daily crowd of more than 10,000 came together at the Bahrain International Circuit in Sakhir for the recent edition of the annual BIC National Day Festival, with Bahrain's Maestro Sound & Light providing sound reinforcement headed by the E-Series from Adamson Systems Engineering. The event at the "home of motorsport in the Middle East" featured the Kingdom of Bahrain's biggest fireworks display and performances from Arabic stars such as Khalid Fouad and Abdallah Al Rowaished.

"For the crucial mid-high requirements of Arabic artists these days, the choice of PA was obvious," states Kiran Tauro, technical director for Maestro Sound & Light. "We all leaned towards the Adamson E15, which in my opinion has the best possible mid-high performance of any other box on the market. Combined with Adamson's new E119 subs, which we used for the very first time, we had the perfect system for an event of this size and importance."

Using Adamson's Blueprint AV 3D modeling software, Tauro and his team designed a system that properly covered the rela-







woodwinds


piano


universal


drums


strings

Left to right, sound team members Kiran Tauro, Jerin Jose, Kamal Raj Burman and Kunal Jaiswal with some of the Adamson E-Series gear in Bahrain.

tively long and narrow audience area. "What was amazing was the fact that we only needed 24 E15 boxes to cover a significant distance with no delays," Tauro notes.

The PA was comprised of left-right arrays with 12 Adamson E15s and nine E119s per side, plus a complement of Metrix boxes for front fill. Speaking to his experience using the E119s for the first time, Tauro says: "Being used to the T21 subs in the past, I didn't know what to expect of the E119, having just the single 19-inch woofer. In the end, I was blown away. Those subs sounded orgasmic."

PLENTY OF FLEXIBILITY FOR A THEATRICAL PRODUCTION

Sound designer Gareth Owen has deployed an Avid VENUE | S6L live mixing system for a Broadway musical theater production of A Bronx Tale at the Longacre Theater in New York. Directed by two-time Academy Award winner Robert De Niro and four-time Tony Award winner Jerry Zaks, A Bronx Tale is based on the play that inspired the film. Owen turned to a 32-fader VENUE | S6L system with three Stage 64 I/O racks to support the 180 channels needed for the demanding production.

"Avid VENUE | S6L can become whatever you need it to be," says Owen. "It's quite revolutionary the way you can just bring up anything you want onto the encoders and custom fader banks, including plugins. Nobody else is doing that as impressively or as effectively. I love the fact that there's enough processing on board to be able to explore ideas that traditionally I haven't been able to do because of hardware or software limitations. I now have the resources to explore my creative ideas."

Sound designer Gareth Owen with the Avid VENUE | S6L mix system he selected for A Bronx Tale *in New York.*

He adds that the VENUE | S6L's new preamplifier design, 64-bit processing, and 96 kHz sample rate have also influenced his microphone choices. "It sort of sent me on a voyage of investigation to discover other things in my signal path that weren't as clean and clear as I thought they were, because the S6L is so transparent and so clean that it's exposing things that in the past I've never noticed."

MEETING BIG EXPECTATIONS ON THE EMERALD ISLE

Award-winning Irish singer-songwriter Gavin James recently played one of the biggest shows of his young life, at the 3 Arena in Dublin. It was also one of the biggest indoor shows of the year for Ireland-based rental company Sound Design, which deployed its new NEXO STM Series modular line arrays for the first time on an arena-sized live music event, including recently added M28 Omnipurpose cabinets.

Sound Design's NEXO STM arrays ready to go into the air at 3 Arena in Dublin for Gavin James.

Initial system designs for the production were generated on proprietary NEXO NS-1 modeling software by John Vickers and Eddie O'Brien of Sound Design as well as Val Gilbert from NEXO's engineering support team. The main system incorporated 18 STM M28 cabinets per side, with flown STM S118 subs to create a full-range left/right system. On the ground were another 20 subs forming a supplementary bass arc.

In the wide auditorium of the 3 Arena, out fill hangs consisted of STM M28 enclosures, while at the rear of the arena, under the balcony, Sound Design placed compact GEO M6 line arrays. "These cabinets were firing in towards the bars, which made a massive difference to the show and the atmosphere," states Vickers.

Monitor engineer Paul Moore also utilized of NEXO components for the stage system, including a dozen 45°N-12 array monitors on stage and two RS15s on the drum mix. Side fills used a stack per side, each with a STM M46, STM B112, and STM S118 sub, with audio power delivered by four NEXO NUAR racks.

ADVANCING ADVANTAGES AT FOH AND MONITORS

Pop artist Andy Grammer is a welcome addition on a range of tours, and last year saw him out on a co-headlined trek with Gavin DeGraw as well as opening for Train. The sound team keeps the equipment quotient light for Grammer and the four members of his band, carrying a DiGiCo SD10 console for front of house and an SD9 for monitors.

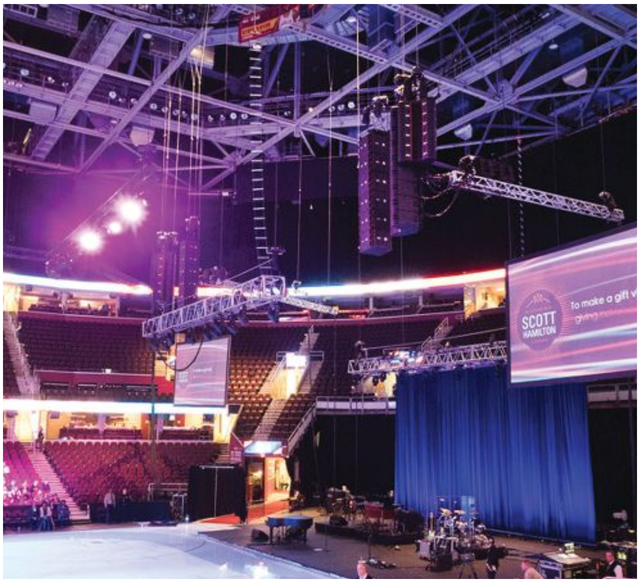
Both boards have been upgraded with Stealth Core 2 (SC2) for increased functionality, and the upgrade also added to the channel count of the SD9, enabling it to accommodate 96 channels for monitor engineer William Valentine.

"The SD9 with the Stealth Core 2 upgrade is a total space saver in an already great-sounding console," says Valentine. "We could not have done this tour on an SD9 before SC2, and had an SD8 on hold just in case. Thankfully, the software came out exactly when we needed it and we were able to be one of the first tours out with it. Opening up capacity from 48 flexi-channels to 96 total channels on this desk meant that we could fit everything we needed to and still have additional capacity to add things while on tour, if necessary. Having that many channels in such a small form factor is truly a game changer."

Meanwhile, front of house mixer Adam Robinson set up his SD10 with macros to do instant effects inserts on the fly. Both consoles, which were supplied by Clair Global, share a single SD-Rack. "Adam and I share eight channels over fiber with the SD-Rack, which is great because there's no copper to run," adds Valentine. "We have complete connectivity between the desks. And the gain tracking on the SD9 is second to none."

Andy Grammer monitor engineer William Valentine (left) manning the DiGiCo SD9, with playback tech David Reyes standing nearby.

CREDIT: ANDREW HAWKES



EAW Adaptive arrays flown by Anderson Audio at Quicken Loans Arena for Scott CARES in Cleveland.

HANDLING UNIQUE NEEDS AT AN ARENA FUNDRAISER

The recent iteration of skating champion Scott Hamilton's annual CARES fundraiser for the Cleveland Clinic, held at the Quicken Loans Arena in Cleveland, saw Harrisburg, PA-based Anderson Audio providing an Eastern Acoustic Works (EAW) Adaptive system. The event combines performances by live musical guests and champion figure skaters, with singer-songwriter Michael McDonald as this year's music headliner.

Anderson Audio, owned by Chris Anderson, has served Scott CARES for several years, but its the first time the company deployed its Adaptive rig for the event. The PA included left-right hangs, each made up of a column of eight Anya for main coverage and six Anna for out fill. Low end was provided by two columns of six Otto subwoofers flown next to each Anya/Anna array.

"We configured it to play to the seating area, which was roughly 180 degrees of the first two levels of the arena," Anderson explains. "Coverage-wise, the solution that Resolution software recommended – after we mapped the room – was quite good. Having provided systems for the venue before, I must admit that it was nice to have the room covered with such a simple solution. The quality of the bass from flown subwoofers was also quite impressive. We could keep the system high and out of sight, and still sound really great."

Scott Harvey, production manager for the event who recently became EAW Adaptive Certified, adds, "Our objective is to provide our musical guests with the quality of audio they expect and the audience with the best entertainment experience possible – all while keeping an eye on the bottom line. The sheer amount of power, coverage, and ease of rigging of the Adaptive system allowed us to run under very tight timelines and saved us considerable cost, which is particularly important for our fundraising event." LSI

MERGING WORLDS

Checking in with loudspeaker engineer/designer Dave Gunness.

by Kevin Young

NOTED FOR HIS WORK as an audio/electrical engineer, loudspeaker designer, and inventor over a career spanning more than three decades, if there's one constant in the professional audio career of Dave Gunness, it's innovation. And, perhaps more specifically, the fact that while the process of building something is valuable, the willingness and drive to continue to improve on your own creations is even more paramount.

The 56-year-old vice president of research and development as well as lead product designer at Fulcrum Acoustic – the rapidly growing loudspeaker manufacturer he co-founded several years ago – points to the creative aspects of design as always having been key to his work, and explains that his earliest creative impulses manifested themselves musically.

"I started out singing in church," he says. "I think the biggest audience I played to was at a Christmas service to something like 1,500 people." Stage fright was never an issue, he adds: "When you start out at six you get over it."

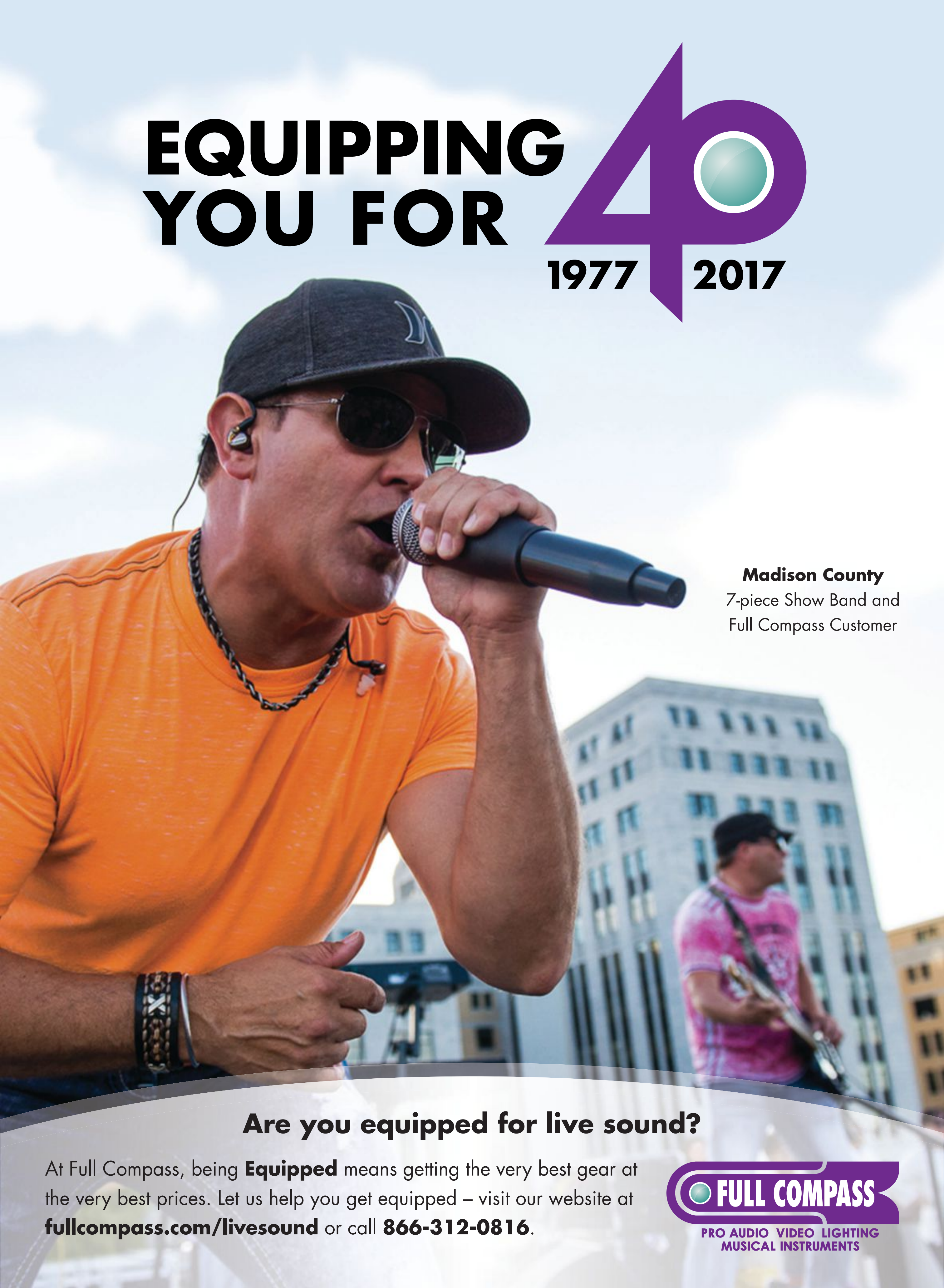
Growing up in Janesville, WI, he had an interest in audio and recalls buying multiple car speakers at a rummage sale, hanging them around his room and wiring them up in different combinations. His musical interests continued through college when he performed bar gigs as a solo acoustic guitar player/singer and in various bands to make extra cash. And, since he couldn't afford pre-built loudspeakers, he made his own.

"They weren't complex, but they worked, and since I built them they went through probably a dozen revisions by the time I was done," he notes. "I kept playing with the crossovers and changing out drivers and various things. It was a learning experience."

EARLY DIRECTIONS

Gunness' first career choice, however, was aeronautical engineering, which he studied at Purdue University in West Lafayette, IN. But it didn't last: "I went into the program thinking I'd design light airplanes. When I found out that nobody was actually doing that, and that aeronautical engineering meant you got to work on landing gear knuckles or something, I decided to go back to Wisconsin."

There, he enrolled at the University of Wisconsin-Madison in electrical engineering, which, by way of a "flat-out coincidence" led him to pro audio. "In my second year there I discovered they'd created a new emphasis in the department called electro-acoustics," he explains. "This was after Richard Greiner, who was famous for work he did on transistors in the early 60s, had gotten tenure. His emphasis was in semiconductors, but his passion was audio and he got this new program put in place, so I took on loudspeakers, acoustics and audio circuits.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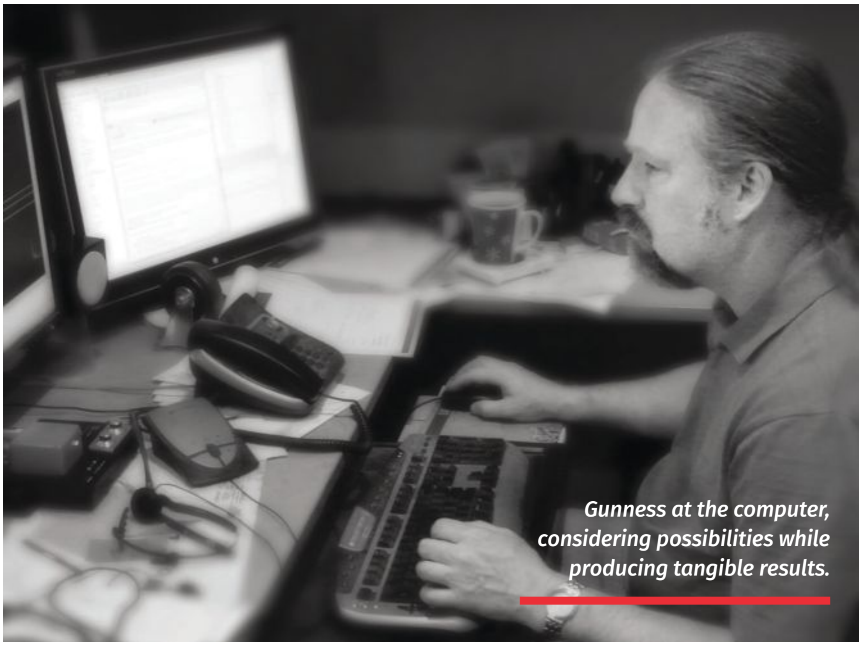


Gunness at the computer, considering possibilities while producing tangible results.

It was great."

His love for music, however, hadn't lessened. He still felt the lure of a career on stage. But what ended up tipping the scales in a different direction was a case of laryngitis. The story's a little more involved than that, however. "In 1982 I played in what I'll call a mercenary band; we had a booking agent, and whenever one of his bands canceled we'd substitute. One week we'd be a country band, the next a wedding or a rock band. It was a horrible experience," he says, laughing. "Everywhere we went people were disappointed that we'd showed up."

After a car accident enroute to one of the band's gigs left him with a serious concussion, he was unable to attend classes at the beginning of the next college semester, so he decided instead to sit it out and play full time. "I got a great gig: three nights a week, 75 bucks a night – good money in those days," he notes. "I played for a couple of months and came down with laryngitis. I couldn't talk, but I could still sing so I continued until I lost my voice entirely. So I'm sitting there, halfway through the semester – can't sing, can't go out and play – and I realized how fragile a music career is and that it might make a better hobby than a job."

Right after graduating in the mid-80s with a degree in Electrical and Computer Engineering from UW-M, he relocated to Buchanan, MI, accepting an engineering position with Electro-Voice. Even then he hadn't entirely shaken the music bug, however: "Early on I had the idea that I was going to work for EV for a couple of years, save a bunch of money, buy a good sound system and then play music," he says. Married by then, and with a family on the way, he realized it simply wasn't realistic and concentrated fully on serious loudspeaker design.

FURTHER PROGRESS

His first assignment at EV was to design the coaxial, all-weather Musicaster 100 voice/music loudspeaker, and then it was on to bigger things, literally. In 1984 he filed a patent for the core concept that ultimately became the MT-4 large-scale concert touring system, still fondly recalled by many old-time "road dogs" to this day. The "MT" stands for "Manifold Technology" and essentially it's a method for using a manifold to combine the outputs of multiple compression drivers for increased SPL and more coherent summation. Additional patents followed in conjunction with the development of horn improvements for both EV HP Series and Altec Lansing Vari-Intense loudspeakers.

In 1995, Gunness was named senior engineer at EAW, moving to company headquarters in Whitinsville, MA and joined by his wife Kathryn and their two children. Ultimately he rose to director of research and development, working with a talented engineering team headed up by Kenton Forsythe. After initial attention to creating custom loudspeaker designs for a variety of applications and clients, his research into directional control of concert loudspeaker clusters led to what became the KF900 Series, another large-scale system incorporating DSP for each individual row of drivers in a cluster, and with the ability to deliver targeted down fill coverage without altering the cluster structure. It resulted in two additional patents.

Concentrating on both individual and collaborative possibilities with DSP, coaxial loudspeakers and digitally steerable arrays led to the creation of FChart software and, later, Gunness Focusing, a significant impetus to his work. "The big one was using DSP to make things better," he notes. "That started in college. DSP for audio really wasn't a 'thing' yet. It was theoretical. There was a DSP chip that Texas Instruments came out with and my senior project was a paper about it. I didn't have the chip, just the manual, but I wrote code, counted cycle times and figured out how many times you could run a given length program in 22 microseconds at 44 kilohertz and evaluated the possibility of using a microprocessor to process audio.

"That was about 10 years before DSP really started gaining interest in audio. I felt people were using DSP to replace things they already could do in analog. It added convenience, but I felt there were things you should be able to do with it that went way beyond what you could do in analog."

VALUE IN THE CHASE

Gunness' need to innovate, he says, is a product of his personality, passion for music and audio, and the fact he was encouraged, particularly by his father, to be creative from an early age. "That's important," Gunness states. "If you grow up believing you can do anything, when you get an adult job you approach it from that point of view. My dad let me build a canoe when I was 10. I found a *Popular Mechanics* article that showed me how, he bought the materials, and I built it."

He only paddled it on open water once, explaining, "I had no way to get it to a pond, but it was the experience of building it that was the value." While that creative mindset is the engine that drives him, it took a bit of time to actually get it running. Technology needed to catch up, and he also needed organizational support to turn concepts into reality.

By 2002, still with EAW, he formulated a way to code and test his concepts, but he and his colleagues – Jamie Anderson (now with Rational Acoustics) and Rich Frembes (now with Fulcrum) – needed a specific DSP platform that just wasn't available at that time. Creating the NT Series of loudspeakers, he adds, finally provided the team with a suitable DSP platform to move forward.

Gunness had begun development of his proprietary FChart software while at EV, seeking a software tool that would, essentially, make him a better designer. That project, too, stalled for a time. "But I learned from it, and at EAW, I thought, 'Well, this time I'm not going to try to get funding. I'm just going to write it myself. It doesn't matter how long it takes."

For the record, it ended up taking 12 years and – unsurprisingly – he's continued to enhance and improve on those programming concepts since co-founding Fulcrum Acoustic in 2008. "People know about the specialized aspect of FChart – that I used it to do steering calculations for the KF900 system, and that it's the core of the (EAW) DSA Series – but it was also the integrated platform for core engineering. We'd measure loudspeakers with it, do polars, perform directional analysis, and obviously, DSP. I couldn't have done Gunness Focusing without that software. Fulcrum's engineering approach is structured the same way, using an equivalent software platform that we've dubbed 'Rayleigh' and a similar suite of DSP techniques called 'Temporal Equalization' or 'TQ'."

The essence of Temporal Equalization is incorporating DSP techniques into the design process from the outset. "Rather than choosing a compromise between two competing attributes, we physically optimize the attribute that can't be addressed with DSP and solve the other problem with DSP. Temporal Equalization is structured in such a way that it can be implemented on most modern system processors. In fact, Fulcrum provides verified TQ settings for more than 20 processing platforms."

SEEKING SINGULARITY

Engineers, Gunness believes, fall into one of two classifications: divergent or convergent thinkers. "Divergent thinkers generate new ideas/inventions. Convergent thinkers finish things." He's worked at residing in both camps, noting, "For me, divergent thinking was natural, but I had to learn to focus on finishing things, otherwise I'd be generating inventions that would never see the light of day."

Music's still a part of Gunness' world, even if it had to take a back seat role.

Convergent thinking played a role in his decision, in 2008, to go into business with partners Stephen Siegel and Chris Alfiero in creating Fulcrum Acoustic, which has production facilities in Whitinsville. His vision was to streamline and shorten the development process while also being able to fully concentrate on the technologies and products most appealing to his interests and the marketplace.

The TQ Install Series of coaxial loudspeakers for the installation market were the first result of the quest. "Coaxials are a common thing throughout my career," he says, citing the EV Musicaster 100 and at Fulcrum, his goal was to eliminate some of what he sees as the primary problems of traditional coaxial designs: intermodulation distortion, bulk, and weight. "It's not that we decided to make coax the core of the company," he adds, "It was just that once we figured out we could make multiple patterns work in the coaxial format, there was no good reason not to make the next thing coaxial."

Fulcrum's priorities in terms of R&D and product development varies from year to year. Most recently, attention has turned to overcoming excessive rear low-frequency radiation with what the company calls Passive Cardioid Technology, first introduced in the FL283 and FLS115 line arrays and now the basis of a line of cardioid subwoofers that includes CS118 (18-inch) and CS121 (21-inch) models. And yes, it's patent-pending.

As the company has grown, Gunness says, it's become increasingly difficult to spend as much time as he would like on advancing design concepts. While recent staff additions, such as the addition of director of engineering Nathan Butler, will ultimately open up his schedule, the real break won't come until Fulcrum completes its new 35,000-square-foot Whitinsville factory in 2017.

Even the briefest glance at the company's website will tip off anyone unfamiliar to a decidedly non-corporate culture and sense of humor that comes from the top down – Gunness, for example, in addition to being listed as vice president of R&D, is also touted as "lead coffee maker."

"It's important not to take yourself too seriously," he concludes. "We've all been through the corporate wringer, so we don't use terms like 'onboarding employees' and that kind of stuff. In fact, we cringe when we hear things like that – let's not forget, we're doing this for enjoyment and satisfaction, not just to make money." LSI

Based in Toronto, ***Kevin Young*** *is a freelance music and tech writer, as well as professional musician and composer.*

Rise Of The Guitars

Another important facet in the quest for a semi-silent stage.

by Mike Sokol

The author with the gear detailed in this article.

SMALL – BUT full-featured – guitar amplifiers can fulfill the mission of cutting the decibel level on stage while still providing the "bells and whistles" to be musically creative and retain feel, presence and tone. That's the focus in this third installment of the semi-silent stage series, where the mission is offering alternatives that allow musicians and engineers to work together to provide a better mix for the audience. *(Previous chapters were featured in the September and November 2016 issues and are also available on ProSoundWeb.)*

Note that this isn't an actual *review* of guitar amps but rather an *overview* of gear and techniques that work well in the quest for a quieter stage. In the interest of full disclosure, I'm not a guitarist, just a rock 'n' roll keyboard player of 40-plus years who still owns a Hammond B3 with Leslie, Mini-Moog synthesizer, and Rhodes 73 electric piano. But I began building tube-based stomp boxes and speaker-emulated DI interfaces for guitar players decades ago, and currently run production sound for shows and concerts of all sizes.

Most of the guitar amp suggestions that follow are for small- to medium-sized rooms found in music clubs and churches with capacities of 100 to

1,000 listeners. For larger outside stages, the proposals made here might need to be adjusted to fit expanded SPL monitoring requirements.

Some clarity before we proceed. When I'm talking about dB measurements/levels, all references are SPL, Slow-A. In addition, my terms of "semi-silent stage" or "low-dB stage," mean limiting stage wash being dumped into a room to less than 90 dB (and preferably lower), allowing the front of house engineer to deliver mixes at 90 to 95 dB that sound powerful and musical *throughout the entire room*. There can be areas on stage at more than 85 dB at the individual musicians, but the goal is to keep the collective stage volume as low as possible while giving the musicians enough "feel" to play musically.

Modern concerts can reach house volumes of 110 dB or more, and while that may be fun for a while, a steady diet of those levels will eventually leave everyone deaf. With that in mind, here are a few basic ideas on what it takes to build a low-decibel electric guitar rig.

TUBES RULE

While it's certainly possible to design and build a great modeling processor that will emulate a guitar tube amp, there are few modeling pedals you can plug-and-play right out of the box. They take time to try out, tweak for a specific playing style, and save the basic settings for quick access. However, a good tube amp is easy to understand for most guitar players, allowing them to quickly dial in their tone. So to get the party started I'm going with the tried-and-true signal path of an electric guitar feeding a few basic effects pedals, a 15-watt guitar amp with a tube output stage, and a 12-inch speaker cabinet miked with a Shure SM57, Sennheiser 609, or an XLR speaker-emulated DI output.

POWER TO THE PEOPLE

After 45-plus years of playing on stages, I've come to the conclusion that "tube watts" sound possibly 6 dB louder than an equal number of "transistor watts." That is, a 300-watt, all-tube Ampeg SVT (with its six 6550 output tubes) sounds just as loud as a 1,200-watt, transistor bass amp. That's probably because tube output stages can be driven into heavy clipping and still sound musical, while transistor output stages need to stay away from the distortion edge so as not to add their own harmonics.

So I'm giving transistor amps a 6 dB cushion just for headroom, without adding their own distortion. And, of course, 6 decibels is equal to 4 times the wattage power. On a semi-silent stage, my evaluations have shown that for electric guitar, either a 15-watt-class tube amp or a 60-watt-class transistor amp is more than adequate to get enough level for the musicians without blowing up the room.

to the guitar signal itself. It's exactly that musical individuality and tone that we want to keep while dropping about 12 dB of the level on stage.

To make a low-dB stage work for electric guitars, here's a suggestion that goes against the last 50 years of rock guitar traditional usage on stage. We don't want the guitar speaker cabinet sitting on the floor behind the musician, pointed straight at the audience. Rather, we want it tipped back like a wedge, pointing up at the player's ears to hear exactly what it sounds like, all the while using a lot less wattage. We then use a microphone or speaker-emulated DI output to feed the mixing console.

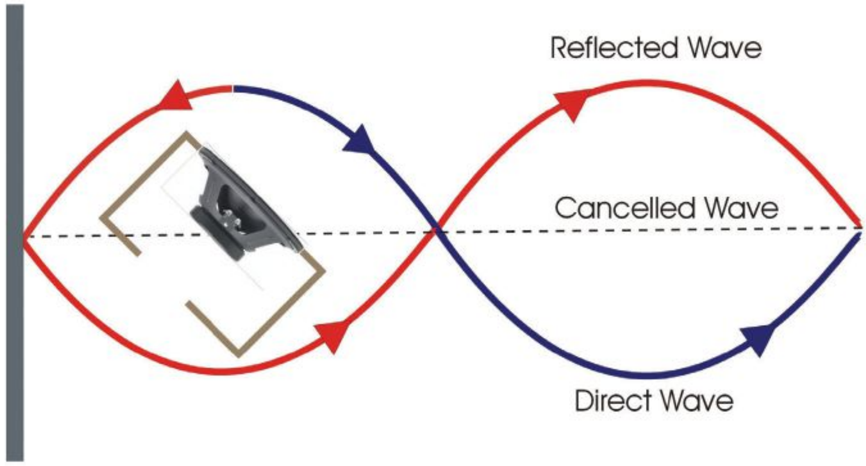


Figure 1: The boundary effect.

Remember, an 85-dB semi-silent stage doesn't require that everything be below 85 dB. A guitarist can certainly be hearing 90 to 95 dB of her/his own instrument at the playing position. We just want to reduce the stage spill into the room to less than 85 or 90 dB.

MONITORS: CAN YOU HEAR ME NOW?

OK, this requires a slightly different definition of a monitor wedge to work for backline instruments. Essentially, *all* loudspeakers on stage act as monitors for their respective musicians. It's just that these "backline" loudspeakers are under the direct control of the artist and are part of their musical creation signal chain. So a guitar amp is actually both a stage monitor as well as contributing harmonics and tone

Yes, I think that Leo Fender had it right 50 years ago with the tilt-back stand on his Super Reverb, but that idea wasn't accepted by those who needed to play large rooms and events without an equally large PA system. All of the sound of the guitar had to come from the stage, so it was big amps and speakers behind the players. (Think Woodstock with stacks of guitar and bass amps on stage pointed directly out toward the audience.)

DE-COUPLING & ICE PICKS…

Because cabinet position is important to the overall tone of a guitar amp, there are at least two arguments against putting the cabinet up on a stand, the first being that this reduces the speaker's coupling

Figure 2

to the floor, compromising the tone, particularly on the low end. However, I suggest that putting the speaker on a stand doesn't reduce the bass by somehow eliminating direct contact with the stage. While this effect of bass reduction is real, it's most likely caused by what's called a "boundary effect" (**Figure 1**). When positioning any sound source at some distance from a flat surface (such as a floor), there's a phase cancellation that occurs at frequencies around one-quarter wavelength of the distance between the speaker and the surface. So if we raise a cabinet to 2 feet above a floor, it creates a 4-foot return path, which works out to a quarter wavelength cancellation around 140 Hz. Raising the same speaker to 3 feet above the floor will cause a boundary effect cancellation around 90 Hz.

That's right in the "sweet spot" of bass frequencies for electric guitars and will definitely affect the tone. So depending how much bass "thump" is desired, it may be preferable to put the cabinet on the floor. Conversely, raising it up a few feet can help clean up the up the low end. Either way is OK on a semi-silent stage as long as the cabinet is tipped toward the player's ears and the speaker doesn't "beam" into the room.

There are two ways to accomplish this. In **Figure 2**, I'm using a portable kickstand called a Standback that allows most small-to-medium sized combo amps to be tipped back to a "monitor" position. Standback (standback.net) folds up and can easily fit in the back of a guitar amp.

Figure 3 shows a cabinet of my own design that I call the Iso-Wedge. It uses a 12-inch guitar speaker of choice (in this case, one from Celestion), and it includes a mic clip for either a Sennheiser 609 or Shure "bent" SM57. I've added a Flapjack beam blocker to this cabinet (more on this next).

The second argument is that listening to a guitar speaker on-axis is too shrill or an "icepick" in the ears. Any 12-inch speaker is going to get "beamy" on the center axis, which is why we like to move their mics off-axis and off-center, and why guitar players tend to listen to their cabinets the same way. Enter Flapjack (**Figure 4**), a simple rubberized disc that is positioned at the center of the guitar speaker, attenuating the shrill center tones while keeping the good stuff on the edges of the pattern. It includes a hanger on a little pin that allows it

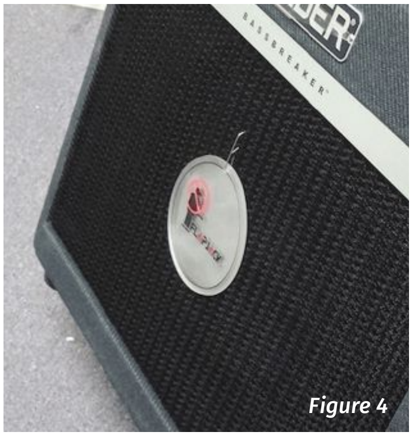


Figure 4

Figure 3

Figure 5

to be quickly added or removed from practically any cabinet with a metal or cloth grille.

ON TO THE AMPS

I selected four distinctly different types of guitar amps and asked the manufacturers to send them to me without any promises of glowing reviews, etc. Besides, guitar amp and speaker selection is a very personal thing to most guitarists, so the purpose is to provide basic guidelines for implementing a lower-dB rig. These are just the first four setups I've come up with, and all have been approved by Karl, a guitar player who's been on stage with me since the mid-70s. While your mileage may vary, these rigs are a great way to begin the discussion.

Fender BassBreaker 15. Nope, it's not for bass – it's a guitar amp. The name is derived from the original Bassman "bass" amps that have been coveted by blues guitar players for decades. However, the BB15 (**Figure 5**) has been upgraded with a digital reverb in its otherwise all-tube signal path, with 15 watts of valve-powered output.

Just like the original, this combo amp is super simple and gets the job done quickly. It's basic enough for blues players who wants to control their tone via playing style and guitar pickup selection, yet it still has enough extra controls to add tonal complexities for more modern sounds. It also includes a speaker-emulated XLR output and mute for connecting directly to a PA system without a mic (if preferred), or for practicing in an SPL-controlled apartment. And it's designed by Fender, one of the originators of the electric guitar amp, and is loaded with a 12-inch Celestion speaker. At home on just about any stage, I think that Leo Fender would be proud of this one.

Figure 6

Orange Tiny Terror. Those of us who grew up listening to Led Zeppelin in the late 60s and early 70s know that Orange amps were an integral part of the guitar tone of Jimmy Page. However, there's no need for 100 headache-inducing watts worth of tubes overpowering a small room. Enter the Orange Tiny Terror

Figure 7

(**Figure 6**). It's a really simple amp, with just input and output level controls plus a single tone control. Oh yes, there's also a 7/15-watt selector switch on the front panel. Set the input and output gain controls to your liking and add in a dash of treble. Shazamm!

Don't think that 15 watts is enough power? Well, plug the Tiny Terror into a speaker cabinet of choice, set it to 7 watts, and stand back. This thing is a serious pocket rocket that you'll likely need to turn down a bit, especially if its matching speaker cabinet is tipped back into wedge position. Want more gain? Try the Orange Dark Terror, a 7/15-watt, all-tube favorite of many metal/shred guitar players. It has crazy gain and distortion but with a very controllable volume on stage.

Hughes & Kettner Tubemeister Deluxe 20. While H & K doesn't have the heritage of Fender or Orange, the company's Red Box speaker emulator has been around for as long as I can remember, and was probably the first – and arguably the best – way to simulate a mic in front of a speaker. It emulates this important part of the guitar signal path by carefully equalizing the signal and conditioning it for the PA system.

The Tubemeister Deluxe 20 (**Figure 7**) has a full-featured Red Box with mic/line level output, which makes it perfect for connecting directly to a mixing console. Or do what I do and use both the emulated DI output in addition to a favorite mic on the speaker. The mixing choices become *huge*.

The Deluxe 20 itself is an all-tube amp with power-tap settings for 20 watts, 5 watts, 1 watt and 0 (zero) watts. So this compact (but still relatively heavy) amp head can be used anywhere from a larger stage to a studio at the flip of a switch. When we first fired it up, and since Karl already has a Tubemeister 18, he wisely selected the 1-watt setting. Talk about sounding fantastic with a lot of volume! (Yes, at 1 watt.) The 5-watt setting is right for medium-size stages, and I can imagine that the 20-watt setting would be perfect for powering a 4 x 12-inch cabinet for large shows or even a concert. It just sounds great at any volume.

This is a 2-channel unit with a dual foot-switch, which makes selecting between rhythm, lead and boost settings just a floor button push away. And it's also the primary amp selected by Karl for his wide range of playing styles. He can use it to play blues, metal, or 70s, 80s or 90s songs with just a few tweaks. Thus the Deluxe 20 will easily accommodate any of his 7-plus stage guitars and dozens of playing styles. I'm also going to have


Figure 8

Figure 9

him try it with his 1960 Rickenbacker lap steel guitar, and it will likely do great with that as well.

Tech 21 RK5 Fly Rig & Electro-Harmonix 44 Magnum. This is my own special home brew. Take a Tech 21 Fly-Rig RK5 designed by Richie Kotzen, run it into an Electro-Harmonix 44 Magnum pedal amp for power (44 watts), and send it to one of the Iso-Wedge cabinets of my own design that's outfitted with a 12-inch Celestion Creamback 70 speaker (**Figure 8**). The Tech 21 RK5 Fly Rig includes EQ, distortion, echo and reverb, plus it's small enough to fit into the strap compartment of a guitar case. Since it can fly on a plane with a guitar ("Fly Rig" – get it?), it can go to any gig and be plugged into the PA system.

This rig closely matches the original setup Karl and I built long ago when we were getting rid of the 100-watt guitar amps on stage. And this Frankenstein rig is his number 2 choice for practice and small room gigs. I only had to add a level attenuator between the output of the Tech 21 Fly Rig and the EHX 44 Magnum to reduce the gain a bit and make its volume more controllable. Thus it's actually a tiny guitar rig that's not really an amp at all (**Figure 9**).

Of course, the Tech 21 Fly Rig isn't limited to tiny amp status. The beauty of this design is that it can be scaled anywhere from a low-dB stage application all the way to full concert mode, depending only on how big of a PA system is at the gig. And players can always have the same guitar pedalboard at their feet, with all the settings they like already in place. Plus, it can be added to any of the above guitar amps to provide a huge amount of creative flexibility. (It's almost too much fun!)

EXPANDING HORIZONS

I regularly host semi-silent stage seminars and invite everyone to come check out this gear in person. (The schedule is on PSW and the Live Sound Co website listed below.) And if you're a player, be sure to bring along your guitar and pedal board to try things out. We're also going to experiment with all sorts of other low-dB setups, and I'll be sharing the highlights as we go.

In my next article, I'll be focusing on lowering bass guitar/amp levels without compromising tone or feel, as well as presenting several different solutions that include tactile monitoring when using in-ear monitors (IEM) on stage without a bass amp. Remember: you have nothing to lose but your excess stage volume. 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.*

THE ROAD LESS TRAVELED

Tips for touring in less developed countries.

by Becky Pell

Touring with full production in first world countries is, let's face it, easy street. Like a family of hi-tech snails, you carry your home and everything you need; and beyond power and rigging points, you don't need a whole lot from the local production/promoter. Even when you're not carrying all your gear, picking up what's needed locally in the western world is usually straightforward as long as it's been advanced properly, especially in countries where we all speak the same first language.

But what happens when traveling further afield, to countries where the language, culture, and wealth-status are very different? Touring in far-flung places is fabulous — you get to see parts of the world you might never have even considered visiting. You get to experience different cultures, meet people of all nationalities (and realize we're all the same), taste wonderful and unusual foods, explore cities where you can't read the road signs, see natural beauty and temples and monuments that you didn't even know existed, and step far, far outside your comfort zone and grow as a person more than you thought was possible.

You'll probably also get frustrated, feel out of your depth, and think you're being stared at a lot, especially if you're a woman working on a gig. Know what? It's all really good for you, both as a human and as an engineer.

My first world tour was 13 years ago, and I've wound up somewhere unusual every year since. Some tours have been big enough that we've carried everything except racks and stacks (PA, wedges and amps), and others we've carried little more than a multimeter and sense of humor — both of which are vital! I've picked up plenty of tricks and tips along the way to make life easier, which I hope you'll find useful the next time you find yourself on a long-haul flight to somewhere you never dreamed you'd find yourself!

Speak very plainly. When you start to write your spec, remember that English may not the first language of the people receiving your message, so work to keep it simple. Lose any slang, colloquialisms and unnecessary words in specs, emails and conversation. It avoids confusion and means you're more likely to get what you need with a minimum of fuss.

Make plain in the specs the things that would be taken for granted back home. For example, I offer a couple of alternatives for acceptable IEM systems, then add "all either x or y please, no mixtures." Likewise amps and crossovers — I once walked on stage to find the wedges I'd asked for, but all sounding completely different from each other. I asked my babysitter to show me the amps, which were buried under the stage, and sure enough, among a horrible snake pit of cable there were several different sorts of amps and crossovers. I certainly improved my chops that day! To things like mic stands, I add the words "clean and in good working order." This extends to production world too — things like toilet paper are not a given in some countries. Assume nothing and put it all on the rider!

When advancing, don't take a reply of "yes, everything is fine" as confirmation that everything is indeed fine. Many cultures are very concerned with "losing face" and want to be seen as stepping up to the mark in their dealings with you. Unfortunately this often means that they'll agree to everything on paper and wait until you're on site to tell you that this bit of kit is broken or that piece of gear is not available in their country. Ask them to list exactly what they have. You might be lucky or you might not, but better to know and have the conversation about substitutes and contingencies now, than when it's 10 hours until show time.

On a side note to this, always communicate respectfully, both because you're a nice human and because the "face" thing can't be overstated – if you make certain cultures feel disrespected, you'll make life very hard for yourself indeed.

Invest in a good quality multi-meter if you don't have one already, and take it everywhere. It's good practice to meter the power before plugging in wherever you are, but it can quite literally be life or death in less-developed countries. I've come across readings that could have killed someone had I blithely carried on without noticing. Local gear will often already be rigged and powered up — find an outlet and meter it anyway. If it's not what it should be, don't go any further until you get it sorted.

Make friends with the interpreter! If you're very lucky you might have a member of audio crew who speaks good English, or a technical translator, but the likelihood is you'll have a dedicated interpreter who doesn't have any technical knowledge. Nonetheless, they're going to be a big help, so learn their name and keep technical questions that need translating as simple as possible.

An hour later we returned to find a large dead chicken, several garlands of flowers, and a lot of incense at center stage.

Have a stash of wipes and a paintbrush for cleaning your gear and the desk. In a lot of places you'll be faced with gear that hasn't been well-maintained and sometimes is downright filthy – and there are few things grosser than a stinky vocal mic!

The food in some countries is fantastic, but that doesn't mean all crew catering

will be great. Some days it will rock your world, others it really won't. Having a supply of nuts or granola bars is a good idea for those days. Likewise, it's smart to carry a kit of basic medicines for common ailments, something I've done ever since trying to explain a UTI via sign language to a pharmacist in Russia!

Be sensitive to where you are. In very poor countries, the local crew might not be earning 1/100th of what you are each day — I'm not kidding. Understand that they don't have the same experience or opportunities as you; be kind and patient, and if they seem interested in what you're doing, take the opportunity to share some of what you know. There are often keen members of the local audio company who are eager to learn from you, and if you can teach them something that helps them, however simple, they'll never forget you.

There aren't many women doing what we do in a lot of the world, and staring is not considered rude in many cultures. This adds up to the fact that if you're a lady roadie, people may be more curious about you. You're going to get looked at, but 99 percent of the time it's completely innocent and they've just never seen a woman do what you're doing, so try to ignore it. (It goes without saying that if it gets creepy, don't stand for it — trust your instincts.) There's also less concept of personal space in some countries; for example, having people close behind me when I'm trying to mix a show is a personal pet hate. I deal with it by either creating a physical barrier, such as a cable trunk behind me, or if that doesn't get the message through, I smile and say, "I'm sorry, could you give me some more space please?" Again, the culprits are usually just trying to learn what you're doing — I've seen people take photos of channel EQs!

Pack your sense of humor. On one memorable stadium show, the stage was deemed unsafe by our production manager and we all had to walk away while it was put right. An hour later we returned to find a large dead chicken, several garlands of flowers, and a lot of incense at center stage – and for once the incense wasn't mine. When we asked the locals what was going on, they cheerfully explained that they had made an offering to the gods responsible, and the stage would now be fine. Needless to say, we went back to the dressing rooms for a little while longer!

Touring further afield is exciting, daunting, and a wonderful experience. It's a privilege that most people can only dream of — to travel the world with a bunch of buddies and get paid for it. Some days amazing things will happen, some days things will go horribly wrong. But you'll truly live life to the full — and if you ask me, that's what this whole rock 'n' roll business is all about! LSI

***Becky Pell** is a monitor engineer with more than 20 years of experience in live sound. She toured as a monitor and RF tech with Black Crowes, Travis and Kylie Minogue before moving behind the desk to mix monitors for artists such as Aha, Muse, Westlife, Anastacia and Take That. She also runs monitors annually on the main stage at the world's largest greenfield festival, Glastonbury. Read more from Becky at SoundGirls.org.*

Showcase

BUILT FOR A PURPOSE

Deployment and application of podium and lavalier mics.

by Craig Leerman

While most pro audio gear types are similar in terms of general design and application principles, each type also has its own traits and idiosyncrasies that need to be understood for them top operate optimally. In this regard, podium and lavalier microphones fit the bill. At first glance, they're relatively simple, but a closer look reveals that there's a lot more going on.

Let's start with podium mics, as they're commonly called, even though they're mounted to a lectern. Sure, many standard handheld mics on gooseneck stands can work well enough, but they don't look nearly as nice as their purpose-built counterparts, particularly on camera. Further, the typical 5/8"-27 gooseneck section designed to support handhelds are usually stiff when new, making them tough for presenters to adjust, and then lose their holding power after a relatively few uses and won't stay in position.

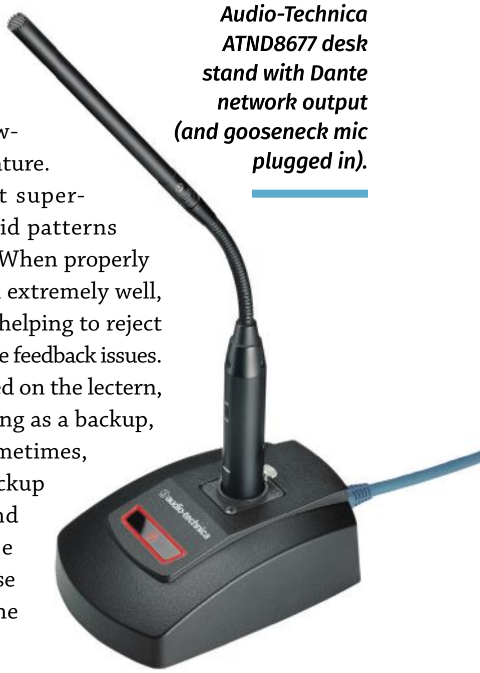
In addition, most gooseneck mounting arms are chrome plated so they tend to reflect bright stage lighting and/or shine on camera. I've also found that newer models may creak and make other undesirable noises when adjusted.

With all of that in mind, a far better option for live corporate events and meetings, as well as at places such as churches, is a compact gooseneck condenser mic that's purpose-designed. They're usually are dark in color and have a relatively small capsule/element positioned atop a long slender gooseneck that is easy to adjust. Because the element is small and light, the gooseneck stays where it's put and doesn't fail over numerous uses.

MYRIAD OPTIONS

Podium mics come in a variety of types and lengths as well as different mounting options. One way to categorize them is by their pickup patterns. Omnidirectional models are great for recording but don't work nearly as well for most PA applications, where more directional designs are the better choice.

Cardioid models are most popular because the wider pickup pattern is more forgiving when there are a variety of different presenters over the course of an event. Some presenters adjust the mic but many simply leave it in the same position as the previous person, which can result (for example) in a mic pointed at the chest of a presenter who's tall following someone smaller in stature.



Audio-Technica ATND8677 desk stand with Dante network output (and gooseneck mic plugged in).

This is not to say that supercardioid and hypercardioid patterns aren't worth considering. When properly positioned, they can work extremely well, with their tighter pattern helping to reject unwanted sounds and reduce feedback issues.

Often two mics are placed on the lectern, with the second one serving as a backup, usually not turned on. Sometimes, the mics have different pickup patterns (i.e., cardioid and supercardioid), with the engineer then able to choose which one is picking up the best. The second mic may also be dedicated to recording only, and with an omni pattern to pick up sufficient level even if presenters stray off axis.

Another way to categorize gooseneck models is length. Shorter goosenecks are great for use on desks where a presenter or presenters will be sitting. These typically run from 3 inches to 12 inches in length. Models for podiums/lecterns typically are a bit longer, running from 12 inches to more than 20 inches. A popular length is 18 inches, which accommodates a wide range of adjustment without being "too long."

Podium mics also come with a variety mounting options. Models with a built-in base can simply be placed where needed. Some may just an XLR connector at the base and no mounting. They can be placed in a standard mic clip and used with a desk stand, or more commonly, they're inserted into a base mount that accepts an XLR. Base units can be portable and offer extra features like shock mounting of the XLR as well as on/off switches and indicator lights to tell presenters when the mic is live. Some XLR base connectors can be built into a lectern, with most of these type permanently installed and offering some type of shock mounting.

The Microphone Base, a new stand for the DPA d:screet SC4098.

There's also flange mounting, where the end of the gooseneck is equipped with a built-in screw mount that can thread to a standard 5/8"-27 stand or to a flange mounted on a surface. They usually include an attached mic cable that exits the side of the gooseneck at the end, near the mounting screw.

Shure WL183 omni lav, also available in cardioid and supercardioid versions.

Yet another version is called "conference," and it may include a loudspeaker and/or a headphone jack in the base, along with a talk switch. These mics are usually part of a system, with the audio engineer – or even a panel chairperson/moderator – having the ability to turn off any open mics via a master station, limiting crosstalk and general noise. Many of these models operate on a mix-minus principle, allowing participants to hear others but not themselves. Another common feature is a lighted ring near the head/capsule mic that signifies if the mic is live.

LOOKING AROUND

Most mic manufacturers offer podium models, so I took a look around at what's available and here's what caught my eye. **Audio-Technica** offers a deep bench of gooseneck options, including the ATND8677 mic desk stand base, outfitted with a Dante output, that was introduced a few years ago. Any mic with an XLR connector can be inserted into the rugged base and join an audio network. The ATND8677 also has a mute/unmute switch, and it can be configured to trigger functions on compatible Dante-enabled devices such as a video camera's pan/tilt or a room's lighting preset.

Part of the Microflex series, the **Shure** MX412 and MX418 differ only in length and are a "go-to" choice. Several mounting options are offered, including versions with built-in desk bases. Interchangeable cartridges provide the optimum polar pattern for every application, while a locking flange-mount version is the right direction in permanent situations.

The **DPA** d:screet Series is comprised of miniature supercardioid models that can be mounted to booms ranging from 6 inches to 48 inches long, including common podium gooseneck lengths of 12 and 18 inches. There's also a wide variety of placement options. Accessories include shock-mount bases and the ability to terminate cable either to an XLR or Microdot that, when used with an adapter, allows the mics to work with all leading wireless system options. The company also recently debuted Microphone Base, a new stand for the d:screet SC4098 that's designed to be placed on a table or podium, or attached to a ceiling or wall. It's available in black and white, and comes with either a MicroDot connector, an XLR connector, or un-terminated leads.

Earthworks FlexMic Series with the optional Light Ring.

Countryman podium mics have long been a favorite with many of the audio folks I work with. ISOMAX 4RF's active vibration isolation technology detects and subtracts vibration from the signal, while a tightly-controlled pickup pattern reduces feedback. Building upon ISOMAX, the company recently introduced the A3 Series that's available in omni, cardioid and hypercardioid fixed patterns as well as selectable pattern versions, and they're offered in lengths of 12, 18 or 24 inches.

A look inside a Countryman A3 podium mic.

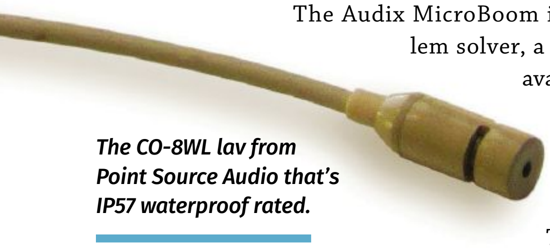
The **Earthworks** FlexMic Series has proven popular for good reason. All models are available with cardioid or hypercardioid patterns, and in 13-, 19-, 23- or 27-inch lengths. The standard series offers up a flat frequency response to 20 kHz and is specified to handle input of up to 139 dB SPL.

The Light Ring models add a red or green LED ring indicator near the business end, while the HD Series extends frequency response up to 40 kHz. One other nifty aspect is that they're available with a fully flexible gooseneck or a rigid center with flex at both ends.

The **Sennheiser** MEG 14-40 is outfitted with a KE 10 cardioid capsule with a flexible, streamlined gooseneck. Models MEG 14-40-L and MEG 14-40-L-II have a red or green integrated light ring which indicates that a presenter is allowed to take the floor. All versions come with XLR connector — an XLR 3M for the standard version and an XLR 5M for the other two that also delivers power for the light ring.

The PolarChoice Series from **Electro-Voice** were among the first podium models to allow the user to select the polar pattern. There's a choice of omni, cardioid, supercardioid and hypercardioid. Mounting options include XLR, flange mount and built-in base units.

In addition to a line of gooseneck mics, **CAD Audio** offers some innovative shock mounts for XLR. Also useful is the MB-1 Mini Boom, designed for use with the 1600VP variable pattern microphone system. It's a carbon fiber boom with a 56-inch maximum length that mounts to any standard 5/8" x 27 threaded mic post or stand.



The CO-8WL lav from Point Source Audio that's IP57 waterproof rated.

The Audix MicroBoom is a handy problem solver, a carbon fiber rod available in 24-, 50- and 84-inch lengths that attaches to any mic stand. The stand adapter provides control over angle, rotation and positioning. It's compatible with all of the company's Micro mics for a selection of pick-up patterns, output levels and frequency responses.

LET'S GET SMALL

Lavalier mics, usually feeding a wireless body pack, present a wealth of additional options. In recent years headworn and earworn designs have really come on, but lavs aren't on their way out any time soon. Theatrical productions like to conceal miniature lavs in costumes, hairlines and even beards, where they're virtually invisible. Lavs also make for quick and easy changes between presenters, and some presenters simply don't like wearing a mic on their head.

Many models also offer the ability to place two mics in a double clip so there is in place, at the ready. These dual setups are most commonly found in theater and broadcast. As to application guidelines for lavs, I encourage you to check out ProSoundWeb, which offers numerous in-depth articles on the subject, including several that I've authored.

The miniscule AKG LC82 MD omni comes in a choice of four colors.

Lavs are available in a variety of shapes and sizes, with the trend being as small as possible. Despite shrinking in stature, however, performance is bigger and better than ever. Here's a look at some interesting models.

Let's start with Shure WL183 omni, WL184 supercardioid and WL185 cardioid lavs, which folks who work in audio see often at shows (although the audiences barely see them). These workhorses are rugged, have adjustable clips and snap-fit windscreens, and pick up quite well.

Another predominant choice is the Sennheiser MKE2, which has been around almost as long as I have yet delivers thoroughly modern performance. It's compact and rugged with a wide frequency response (20 Hz to 20 kHz, +/-3 dB) and comes with a stated SPL rating of 142 dB.

The DPA d:screet line of omni types provides a palette of options, and a range of wireless system adapters ups the flexibility quotient even further. And the Slim 4061 really lives up to its name, so small (and very flat) as to be virtually invisible in many situations.

The LC81 MD from **AKG's** recently released MicroLite Series is a miniscule cardioid design with a diameter of 4.8 mm, while the LC82 MD omni lav is even smaller at a diameter of just 3 mm. Both are available in four colors.

Audio-Technica offers an impressive range of lav choices, including the AT898 (cardioid) and AT899 (omni), the latter measuring just 0.2-inch (5 mm) in diameter (and the former is not much larger). Both have a switchable low-frequency roll-off that reduces popping, with an array of accessories making it easy to meet virtually every application.

Countryman makes what's easily one of smallest directional lavs available in the form of the B2D. It can be ordered wired to an XLR or terminated for most popular wireless systems. There are two sensitivity ratings: standard, for most uses; and mid gain, optimized for head miking that's common in theatre.

A lav I've been referred to by colleagues several times is the CO-8WL from Point Source Audio, which is IP57 waterproof rated so it's built to withstand moisture (sweat, make-up, etc.) that can be quite harmful. An omni type, it's really low-profile (just 4 mm in diameter) and thus is easy to hide while being rated to handle SPL of up to 148 dB. LSI

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

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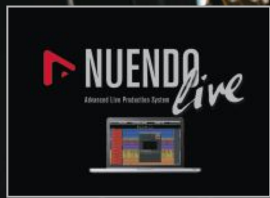

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TIME SENSITIVE

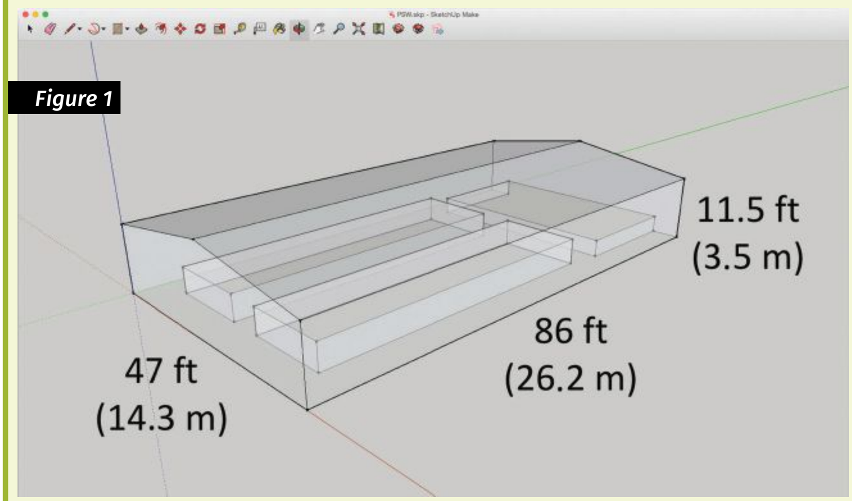
To delay or not, that is the question.

by Merlijn van Veen

I regularly read forum posts on ProSoundWeb questioning the necessity of adding delay loudspeakers to sound reinforcement systems. The default argument in favor of them appears to be level restoration, while the default objections are budget, sightlines, timing issues, distortion of the sonic perspective, logistics, and labor.

However, there is a serious but often overlooked advantage of deploying delay loudspeakers, bordering the effectiveness of absorption, that go beyond plain level restoration. The venue shown in **Figure 1** originates from an actual PSW post and is a small house of worship. The person who posted it was seeking input from the forum community regarding the best approach for designing a suitable sound system with limited means.

Figure 1



The walls and ceiling of the venue in Figure 1 are constructed of drywall (a.k.a., gypsum or Sheetrock) offering little to no mid- and high-frequency absorption. The stage and floor are covered with carpet. Typically, larger room volumes result in longer reverberation times unless the increase in size is accounted for by adding extra absorption. In this case. the volume is sufficiently small enough to get away with low absorption. And that doesn't even consider the typically beneficial effects of audience absorption.

That said, audience members sitting near low absorbent boundaries at moderate to far distances to a sound system are expected to suffer from strong reflections at near identical levels. If direct (as the crow flies) and reflected trajectories approach each other in path length, relative level offsets decrease and the frequency response ripple inherent to comb filtering becomes worse. Each time direct and indirect cancel each other out, all that's left is background noise. Signal-to-noise (SNR) ratios degrade and intelligibility suffers. There are three common ways to deal with this:

1. Steer clear of the offending boundaries with the main system by aiming the loudspeakers differently, and without missing the very audience members we're trying to serve who are located just in front or next to said boundaries. In practice, this presents a conflict of interest and is virtually impossible, especially at low angles of incidence.

2. If we can't avoid striking those boundaries, then absorb or scatter the sound on impact. Either approach will probably affect cosmetics in some way and is also likely to change the acoustics of the venue, which might be at odds with other applications like unamplified events that benefit from a certain amount of natural amplification and reverberation.

3. Deploy delay loudspeakers and exploit their directional properties by careful positioning and aiming in an attempt to effectively "bypass" boundaries. Before we look at the latter, let's start by considering the default argument in favor of delay loudspeakers, namely level restoration.

COMING UP SHORT

A section view of the venue (**Figure 2**) shows a 5.6:1 range ratio (15 dB of level variance from front to back) for a loudspeaker placed downstage at the "highest" possible position. A single loudspeaker, however, can only correct a range ratio of 2:1 at most (from on-axis at 100 percent relative distance to off-axis 50 percent closer). Even if we're willing to accept 6 dB level variance, this leaves us 3 dB short at the back of the audience.

At least four to five loudspeakers configured in an asymmetrical coupled point source or a "dash" array (a line array of six loudspeakers or less) would be required to deal with this kind of asymmetry in the vertical plane. Both solutions are beyond the scope of this article, as well as the available real estate and budget.

Another disadvantage of a single loudspeaker/main-only approach is tonal variance. The low-frequency transducer of a typical loudspeaker is incapable of introducing any directivity because it's producing wavelengths that exceed its

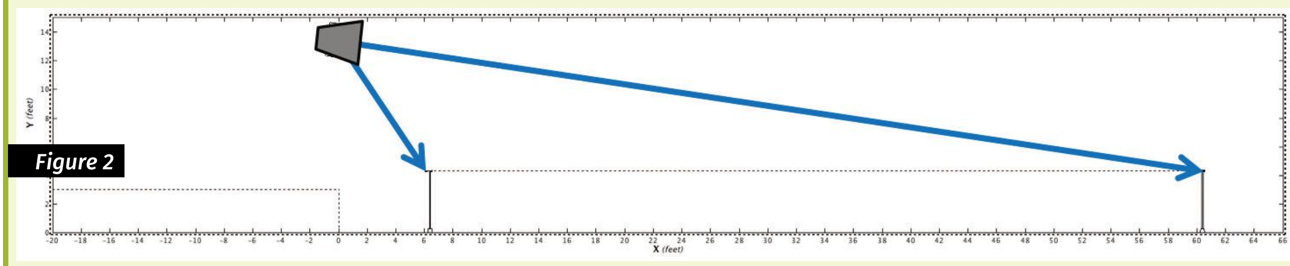


Figure 2

own diameter several times, rendering it immune to rotation. This is contrary to the mid and high frequencies, which can be controlled very well by a proper constant directivity horn, allowing us to direct the sound where we want it to go.

Figure 3 shows a single 50-degree loudspeaker aimed at the rear of the room. The front to back level drop in this part of the spectrum is 9 dB (15 dB of range minus 6 dB of angular attenuation), and it's overshooting the beginning of the audience. The latter issue affects only a minority of the audience and is best dealt with by a local solution, e.g., front fills.

Figure 4 depicts the low end of the same loudspeaker. Its lack of directivity and inherent immunity to rotation result in a 15 dB loss. There's only distance at play and no angular attenuation because there's no coverage angle to begin with in this part of the spectrum. Room gain favoring low frequencies, by accumulated reflections over distance (also known as LF buildup), is likely to decelerate the LF loss rate.

That being said, if left unaccounted for, different loss rates result in tonal variance. Should the result in the back of the venue be too dark (the rule and not the exception) because we simultaneously suffered from HF losses by air, the delay loudspeaker provides an additional bonus by restoring only those frequencies that are missing. This reduces the spectral tilt. Just be mindful that the hi-hat can't be traced back to the delay loudspeaker, focusing attention on its location.

ON THE PLUS SIDE

Regardless, level restoration of at least 3 dB in the rear of the room is required, the most common argument in favor of a delay loudspeaker, in order to place the entire audience within 6 dB of level variance or less.

Figure 5 offers a section view of this approach. The mains have been titled slightly down to limit their coverage to a

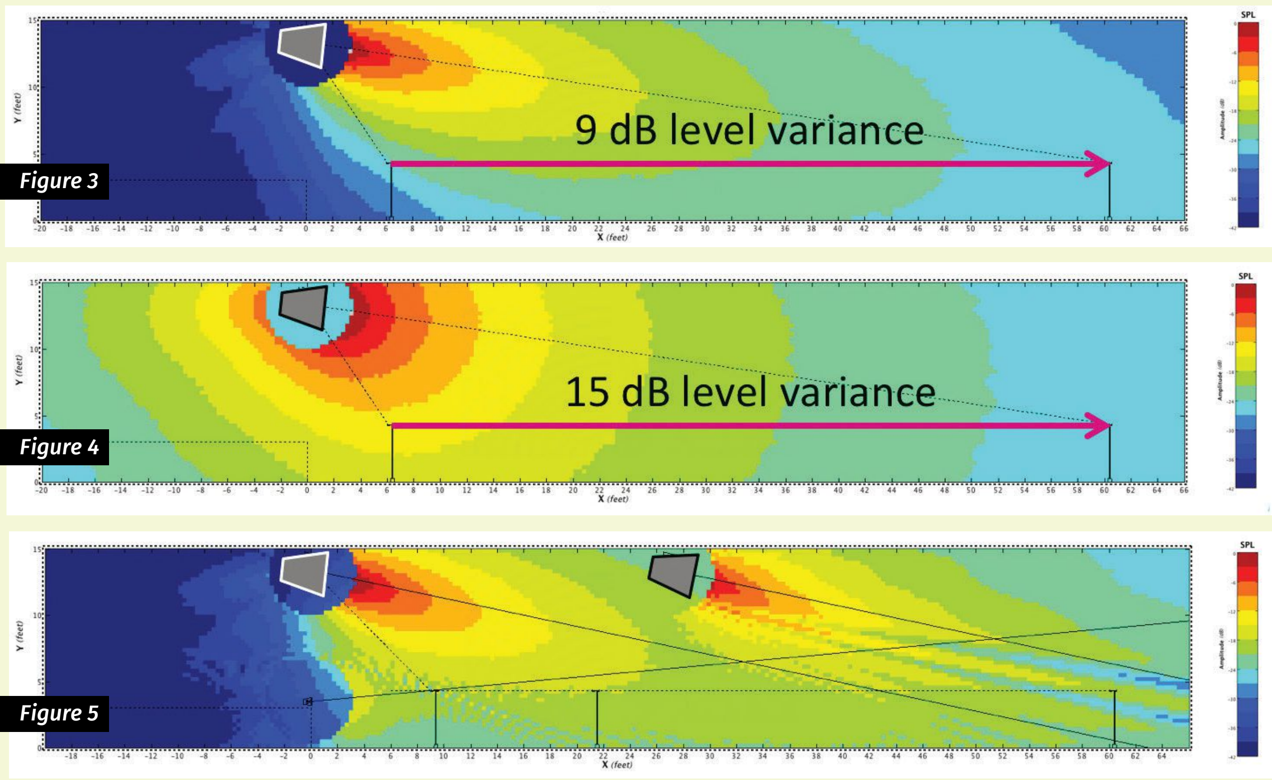


Figure 3

Figure 4

Figure 5

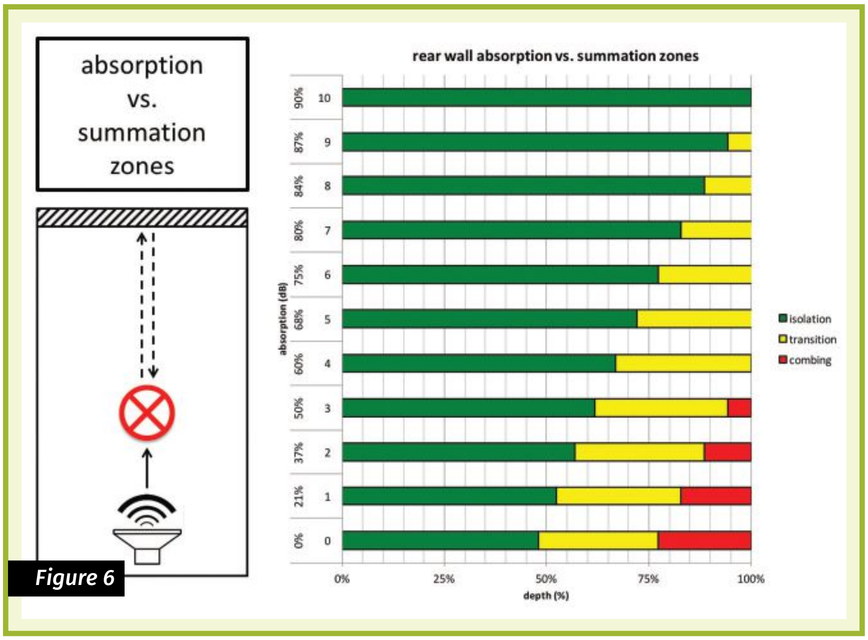


Figure 6

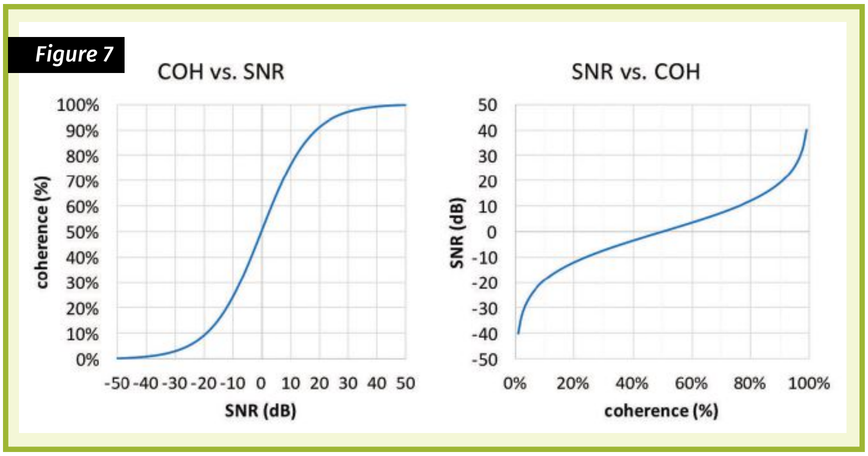


Figure 7

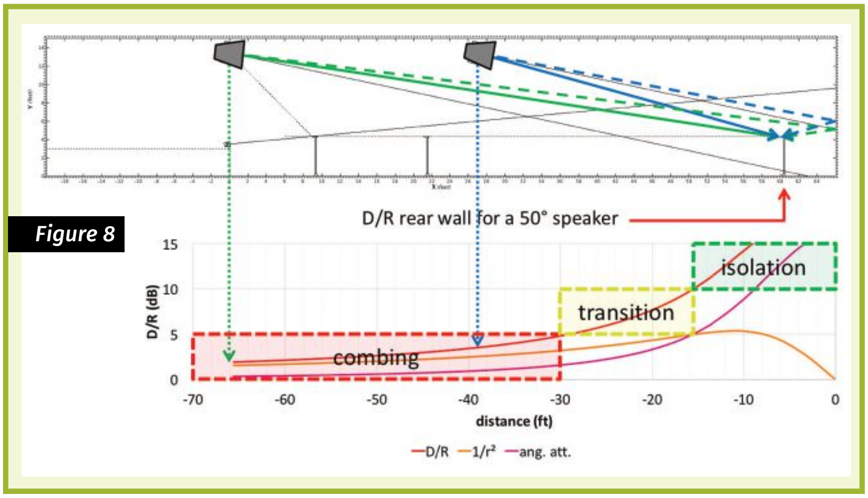


Figure 8

2:1 range ratio, reaching all the way down to the second row. A delay loudspeaker will take care of the remainder of the audience, piggyback-riding on the mains.

Figure 6 demonstrates the essence of this article and the often overlooked necessity of delay loudspeakers if re-aiming or absorption is not a viable solution. A single loudspeaker at a grazing angle of incidence, placing the audience in or near the propagation plane with respect to its reflected sound of a non-absorbent rear wall, can only maintain its level dominance in the first half of the audience.

Beyond that milestone, path lengths approach each other, resulting in strong reflections. We can't expect the main loudspeaker to reach the last row and magically avoid the rear wall. When we get closer to the rear wall its presence increases. On our dual-channel FFT analyzer, this will manifest itself as a decrease in coherence (**Figure 7**), a metric for SNR.

Contrary, the forward-positioned delay loudspeaker has a considerably different geometrical relation to that very same rear wall. Its increased down-tilt angle enters angular attenuation into the off-axis reflected path (**Figure 8**). The reflected trajectory traveled a longer distance compared to the direct sound and on top of that suffered an additional penalty at the start. Therefore, the delay loudspeaker will exhibit improved D/R (direct-to-reverberant ratio) that benefits the compromised main loudspeaker while simultaneously restoring level and reducing tonal variance.

The angle of incidence of the delay loudspeaker with respect to the main loudspeaker will determine the rate at which main and delay tear apart in terms of time (**Figure 9**). Evidently a properly delayed delay loudspeaker placed in-line with the main speaker will remain time aligned over distance, while delay and main opposing each other will create a stalemate situation.

When choosing the correct position for the delay loudspeaker(s), the coverage, level and tonal variance should be carefully balanced against the improved D/R of a more forward position in exchange for reduced "synchronicity." If room treatment is not an option (again, Figure 6), then delay loudspeakers are a viable alternative for improving D/R in the most vulnerable part of the audience.

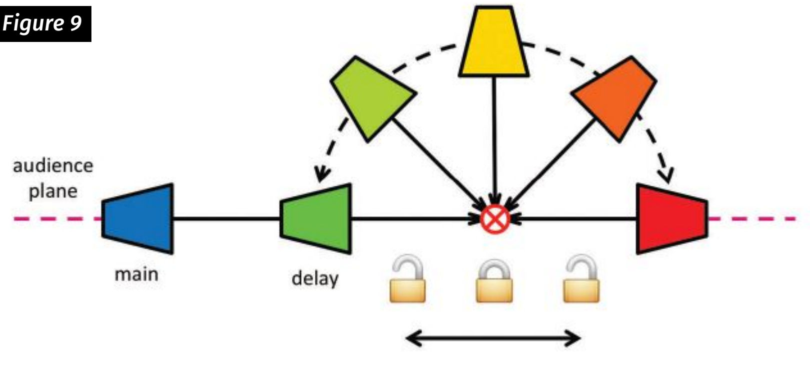
RULES OF THUMB

To insure that the output of the delay loudspeaker is as inconspicuous as possible, make sure:

1. It is equally loud as the main, at most.

2. It arrives on time. The Haas (or precedence) effect requires a differential system to detect an offset in arrival times. A system we do not possess in the vertical plane contrary to the horizontal plane. Over-delaying will result in an audible and measurable degradation, less efficiency, and artifacts ranging from strong tonal coloration to possibly discrete echoes, depending on the program material. DISCLAIMER: If big temperature swings are to be expected, affecting the sound speed and consequentially time alignment, delay times must be revaluated. If temperature swings can't be accounted for, the *relay* line offers better trade-offs.

Figure 9



3. It sounds the same. Typically, there is way less air between delay loudspeakers and the audience than there is between the main(s) and the audience. The main loudspeaker has suffered a bigger HF penalty than the delay loudspeaker. The latter, therefore, should be made equally dark. Alternatively, the output of the main could be made brighter as long as it doesn't make things worse in the front of the audience.

4. The low-end is shelved or ultimately even cut. Most mains will have suffered a substantial amount of low-frequency buildup in the back of room. Shelving out some low end in the delay loudspeaker will reduce tonal variance and simultaneously reduce LF backwash for the audience in front of the delay loudspeaker.

Figure 10

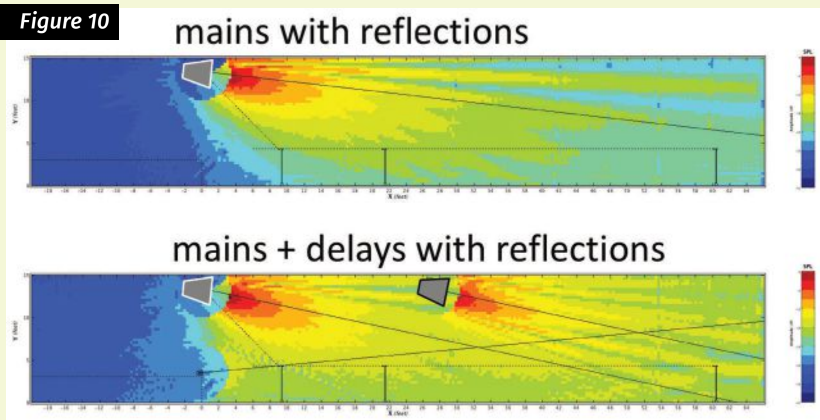
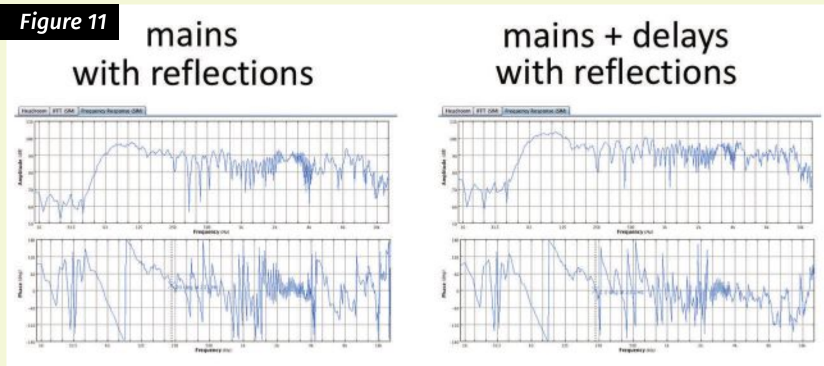


Figure 11



Figures 10 and **11** show the differences between both approaches with boundaries enabled. Notice the reduced comb filter in the back of the audience. In conclusion, properly deployed delay loudspeakers will:

- Restore level (which isn't always required)
- Decrease tonal variance
- Improve D/R
- Reduce comb filtering
- Improve SNR, coherence, and intelligibility LSI

Based in The Netherlands, ***Merlijn van Veen*** *(https://www.merlijnvanveen.nl) is a consultant specializing in sound system design and optimization, and he's also a noted audio educator.*

Project Memo

MAKING IT PERMANENT

Developing a new system for The Old Church.

by Live Sound Staff

THE OLD CHURCH is a non-profit entertainment and event venue in the West End cultural district of Portland, OR. Renowned as a space for acoustic and classical performances, it recently underwent a sound reinforcement upgrade in a project headed by Alcons Audio LR7 line arrays that has made a significant difference in attracting a wider range of performers and audiences.

Originally built in 1882, a dwindling congregation meant that in 1967 an alternative use had to be found for the building or it faced demolition. A group of Portland citizens formed a non-profit organization to preserve the building and use it as a live performance facility to enhance the local community.

FAVORED LOCATION

The venue presents a contemporary, comfortable and accessible space, with the board of directors always looking for ways to improve it. Recently they raised more than $270,000 with the aim of upgrading it's capabilities as a concert hall. As a result, Gary Stokes of Stokes Sound, who's also front of house engineer for k.d. lang as well as an audio engineer for artists such as Queen and Adam Lambert, was brought in to assist on the project, which included the design and installation of a permanent system.

"Throughout its life, the building has been a favorite location for classic and acoustic music in Portland," Stokes says. "It never had a permanent sound reinforcement system before; the only systems used were portable, which were brought in if needed."

In 1972 The Old Church was added to the National Register of Historic Places as an excellent example of Carpenter Gothic architecture. This, together with a recent internal restoration, made installing a permanent system more challenging.

A view of the coverage area from the front platform.

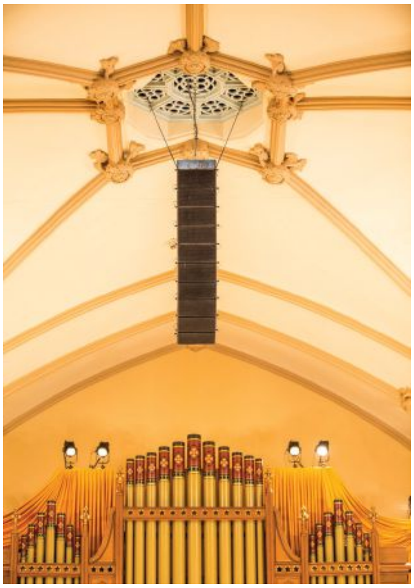
A perspective of The Old Church in Portland, now outfitted with a permanently installed house system.

"There are a lot of limitations regarding what you can do," Stokes explains. "It has no proscenium, the weight limits for flying loudspeakers are extremely low, the acoustics are very reflective and, because of its historic nature, you cannot do anything to dampen the room. In addition, the audience is seated in a 260-degree arc around the stage, so installing a system without getting speakers in sight lines or throwing a lot of energy at the walls was extremely difficult.

The venue first contracted a temporary system to see if adding a permanent solution could attract a wider range of performers. "It didn't look great, but it proved the point and we were given the go-ahead to investigate permanent systems," Stokes says.

SMALL & LIGHT

Having utilized Alcons Audio gear, including its pro-ribbon technology, on a U.S. tour by Jesse Cook, Stokes identified it as a potential solution for the unique nature of this project: "We looked at a range of



A closer look at the new main array, comprised of Alcons Audio LR7 modules.

options, but I had a feeling Alcons would be really well suited, thanks to the advantages offered by the pro-ribbon high-frequency driver technology."

Alcons Audio North America sales manager David Rahn arranged the loan of a demo system, which was set up on stands while Stokes took acoustic measurements. "It immediately proved its superior audio quality, which was very obvious to the board members," he notes. "It also proved that we could get a lot more speakers and high frequency control from much less weight than with any other manufacturer's products."

The final form of the installed system incorporates a mono central cluster of six LR7/120 (120-degree horizontal dispersion) and four LR7/90 (90-degree horizontal dispersion) compact line array modules, supplemented by fills of six compact VR8 monitors and two mid-sized VR12M monitors. Lower frequencies are enhanced with a pair of BF181i mkII compact subwoofers positioned left and right on the platform. Audio power is provided by Sentinel3 and Sentinel10 amplified controllers.

"The system delivers greater advantages than you would get from a stereo system," Stokes states. "Using the 90- and 120-dispersion LR7s, the center cluster covers 80 percent of the audience. Because it's such a live space, towards the back the audience normally gets less direct and more reverberant energy. The center cluster delivers more direct energy to the back of the room, meaning no need for delays. In addition, with the audience in a 260-degree arc, you would only give about 5 percent of them true stereo coverage anyway."

SPREAD THE WORD

An air return vent above the audience had enough of a wood beam structure above it for an additional beam to be installed, from which the center cluster was flown. Installed by Professional Sound and Lighting of Portland, the system was completed ahead of schedule, with Rahn on hand to provide direct support.

"David worked closely with the installers to get the system exactly right. He did a lot of work to ensure it was properly done," Stokes concludes. "The board of The Old Church is very happy; the system is fulfilling the mandate of raising the venue's profile very well." LSI

30 AND COUNTING

Yamaha marks three decades of digital mixing.

by Live Sound Staff

The DMP7, which made its debut in 1987.

In 1987, Yamaha unveiled the DMP7, a digital mixer equipped with channel parametric EQ, dual internal effects processors, a stereo compressor, scene memory that allowed instant recall of multiple mix setups, and unique motorized faders that moved with each recall.

The DMP7 also represented a significant company milestone — its first digital mixer, providing a new tool not only for professional keyboard players, but also for mixing in both live and studio situations. In the intervening 30 years, Yamaha has produced 22 unique series of digital mixers, helping significantly in paving the way for digital acceptance at the professional level while also takings its place as a leading manufacturer of the technology.

Yamaha got an early start in the field of pro mixing consoles with the introduction of the first PM Series analog mixer in 1972, which led to a string of successful launches spanning four decades. The company segued into the digital realm, proving the reliability of the concept, and along the way, pioneered many features now considered commonplace on today's digital mixers.

Engineer Michael Parker mixing monitors on a PM1D at the Grammy Awards in 2011

The past 30 years of digital mixer history at Yamaha is punctuated by several key products. Among them:

1995: The 02R, which soon became standard equipment in studios throughout the world upon its release, offering 44-channel mixing capacity with 4-band parametric EQ, dynamics processing, input delays, and more.

1999: The PM1D, the first of its kind in digital mixing systems, designed specifically for live sound reinforcement. It helped spark the digital revolution in the live realm after its debut at Carnegie Hall in New York City.

2004: The PM5D, which helped solidify that digital was well-suited for touring and festivals due to ease-of-use and intuitive operation. It was claimed by many rental companies to be the most "rider-friendly" digital mixer, and today remains as an acceptable substitute on many riders.

2005: The M7CL, designed for simple operation and creating a smooth transition for analog users and proving particularly popular in houses of worship.

In 2007, Yamaha was awarded a Technical Grammy by the Recording Academy, recognizing the company's long tradition of successful recording products, including REV Series digital reverbs, NS-10M studio reference monitors and HS monitors, as well the DMP7, DMC1000, ProMix01, 02R and DM2000 digital consoles. Today, the digital console legacy continues with:

- The CL Series, providing a comprehensive range of "coloring" options that foster a significant degree of creative freedom.
- The QL Series, offering all-in-one mixing, processing, and routing capability for small- to medium- scale live sound, corporate speech events and installations.
- The TF Series, equipped with the TouchFlow Operation interface to simplify digital mixing and make it more accessible, helping to open up engineering to amateurs and volunteers.
- The RIVAGE PM10, the latest flagship model that builds on the heritage of the PM1D and 5D models in significantly increasing the qualities and capabilities of both models.

A RIVAGE PM10 out with monitor engineer David Baker for Lynyrd Skynyrd.

"Yamaha has held a deeply-rooted commitment to the music products industry for 130 years, and for the past three decades, has built an extraordinary legacy for superior sound quality, innovation, craftsmanship and reliability in our digital mixers," states Alan Macpherson, professional audio general manager, Yamaha Corporation of America. "We are humbled by the accolades we have received from engineers around the world, which only inspires us to make quantum improvements and refinements for many years to come." LSI

THE FIELD IS SET

Manufacturers announced for LSI loudspeaker demo at USITT 2017.

by Live Sound Staff

The line-up of participating manufacturers at the LSI Loudspeaker Demo at the USITT 2017 Annual Conference & Stage Expo this March in St. Louis has been set, with the roster comprised of these market leaders:

- **Adamson Systems Engineering**
- **Alcons Audio**
- **D.A.S. Audio**
- **Electro-Voice**
- **Martin Audio**
- **Renkus-Heinz**

They'll be presenting both larger- and smaller-format systems in three full demo sessions each day, along with 30-minute individual demonstration sessions both days. Open and free of charge to all USITT attendees, audio professionals, church sound personnel and other interested parties, the demo is being held at the Ferrara Theatre at show site America's Center in downtown St. Louis on March 9-10 (2017).

The Ferrara Theater is a 1,400-seat, multi-level, fan-shaped, state-of-the-art live performance proscenium venue. One nifty aspect is that the demo is being presented in an actual working theatre and thus attendees will get the bonus of evaluating the loudspeakers in a truly relevant environment. The loudspeakers will reside on the stage, with the larger systems flown.

The event delivers a unique controlled environment demonstration providing side-by-side listening opportunities, in addition to offering further technical details and pricing information from qualified representatives of each company participating in the demo.

Specifically at USITT, each full demo session will offer two parts: first, a focus on smaller systems, followed by a focus larger systems. In each part, all participating systems are played in a round-robin, random format, supplied with identical audio tracks until the final round, when manufacturers have the opportunity to play their own select track through their systems.

The demo is under the technical direction of LSI/PSW senior contributing editor Craig Leerman, who will also serve as emcee utilizing a Lectrosonics Digital Hybrid wireless microphone system. Engineer Nicholas Radina will be plying his talents on a DiGiCo SD Series digital console to manage and deliver audio to each system.

To insure sound level conformity, all systems are fed pink noise, with the designated SPL set and confirmed. This process, managed by Leerman and Radina, is based on portable metering incorporating an iSEMic 725TR base kit-2 analyzer from iSEMCon. In addition, the 10EaZy measurement platform from Rational Acoustics, receiving input from a centrally located measurement mic, supplies constant SPL readings to attendees via a large video monitor during all demo sessions.

Go to livesound-demo.com to find out more about the demo and to register to attend. Again, specific demo dates are Thursday, March 9 and Friday, March 10, with three full sessions each day.

The United States Institute for Theatre Technology (USITT) connects performing arts design and technology communities to ensure a vibrant dialog among practitioners, educators, and students. This year's Expo in St. Louis promises to be one of the best to date, with pre-registration numbers and total exhibitors both at a very high rate. LSI

A panorama view of the Ferrara Theater, site of the upcoming loudspeaker demo at the USITT 2017 Expo.

L-ACOUSTICS KS28

Design overview and early deployments of a new large-format subwoofer. *by Mary Beth Henson*

In the early days of developing the yet-to-be-named KS28 subwoofer, the L-Acoustics research and development team's mantra wasn't just "more power," but rather "more resolution." The goal was to significantly exceed the performance of the SB28 in terms of additional dynamics, faster rig times, and lower weight, but the number one item on the list was improved musicality.

This effort ended up producing a new sub as well as a new amplified controller to drive it, the LA12X, which delivers more power and DSP resources. The KS28/LA12X combination, which launched at the Prolight + Sound show in Frankfurt last year, also benefitted significantly from a pilot phase that included key L-Acoustics partners in North America, Europe and Asia.

THE PARAMETERS

Simply, the KS28 is a reference subwoofer designed to extend the frequency response of large-format sound reinforcement systems. It incorporates two high-excursion, 18-inch direct-radiating cone drivers mounted in a bass-reflex tuned enclosure that's also equipped with laminar airflow L-Vents with a flared profile. A staple of L-Acoustics sub designs, L-Vents are designed to reduce turbulence and port noise at high levels while also increasing LF efficiency and performance, including maximized dynamics and power handling.

The KS28 operates from 25 Hz (-10 dB), with the excursion capability of the transducers, combined with L-Vents contributing to deliver rated SPL of 143 dB (peak level at 1 meter under half-space conditions). It can offer standard or cardioid directivity by combining physical deployment and suitable presets in the LA12X amplified controller.

In fact, the KS28 is the first L-Acoustics enclosure to fully utilize the LA12X, which offers advanced crossover functions, linearization and proprietary L-Drive protection of the transducers. An onboard library offers standard and cardioid presets, each available with two low-pass filters to accommodate various coupling conditions and LF contour requirements.

The cabinet is constructed of birch and beech plywood panels, and it measures 21.7 x 52.8 x 27.6 (h x w x d). Carefully optimized in terms of thickness, combined with stiffeners, the cabinet is engineered to produce maximized internal volume and mechanical integrity while keeping weight down to 174 pounds. Six ergonomic handles provide solid grip and efficient handling, with bottom and side runners to ensure safe stacking. A 2-point suspension system is flush-mounted into the cabinet.

EARLY DEPLOYMENTS

As noted, before the official launch of the KS28 (and LA12X), L-Acoustics worked with key partners around the world in a pilot phase. Special attention was paid to testing in all key regions, because the LA12X includes a universal Switch Mode Power Supply (SMPS), designed to allow it to travel the globe with immunity to unstable power mains.

Clearwing Productions (Milwaukee and Phoenix) was the North American pilot phase partner, receiving the subs in early 2016. Clearwing decided to put the KS28/LA12X combination to work right away, bringing it to a corporate event for the CBS network during Super Bowl 50 celebrations in San Francisco.

The team at Clearwing realized that the extended bandwidth and resolution of the box would make it valuable to all types of music, so in March, the decision was made to officially debut the KS28 on Pink Floyd alum David Gilmour's Rattle That Lock tour of North America. "When you get the call to support a superstar like David Gilmour, you know you need to be on your A-game," states Gregg Brunclik, president and CEO of Clearwing. "The pilot phase had given us the opportunity to confirm

The new LA12X amplified controller, which works closely with the KS28.

that KS28 is a marked improvement on the SB28, and I was not at all unhappy with the SB28 We knew that putting the new sub on a high-profile tour like this would be an added benefit both to the sound quality and to our logistics."

Brunclik adds that in his view, the LA12X played a significant role in the success of the KS28 on the Gilmour tour: "What L-Acoustics has done with the DSP is voodoo we are not privy to, but it works. It rounds out the bottom end of the system really nicely. It's 50 percent more amplifier than before."

Colin Norfield, front of house engineer for Gilmour, confirms that the KS28 was a value-add to the show. "They are half-again as loud and 30 pounds lighter (than the SB28)," he notes. "They have a tighter, deeper and richer sound, which helped to enhance David's show."

A row of KS28s in front of the stage, deployed by Clearwing for David Gilmour on tour.

CREDIT: PAUL WEBER

DIFFERENT GENRE

On the heels of that successful outing, Clearwing deployed the KS28/LA12X combo extensively throughout the summer festival season, including for the Electric Zoo Festival on Randall's Island in New York City that featured EDM artists such as Tiesto, Hardwell, Steve Aoki and The Chainsmokers.

While sound reinforcement for EDM is often characterized as an exercise in "more power," the music is actually quite intricate, with the resolution of the KS28 not lost on Clearwing's Joe Spitzer, who served as the Main Stage tech at Electric Zoo. "As DJs and producers create more and more complex electronic music, the sonic accuracy of sound reinforcement systems is extremely vital," he explains. "The KS28s bring definition that was missing. Even the most familiar songs and tracks now have a new life, as there are notes rarely heard reproduced so accurately.

"EDM festivals like Electric Zoo demand accurate, even, and powerful LF throughout the entire audience area," he continues. "The new LA12X-powered KS28s accomplished this task with ease, delivering extremely smooth frequency response and truly impressive SPL levels."

Robert Hegge, who served as the tech at Electric Zoo's Riverside Stage, also experienced the new subs. "Every time I take out the KS28s, I'm more impressed with them," he says. "I still find myself being caught off guard by the extended low-frequency response, which is a lot of fun. They have incredible transient response, and they get loud while still maintaining a tight, clean sound."

Bryan Baumgardner, head of audio operations and logistics for Clearwing, concludes: "Clearly, the added bandwidth and SPL allows us to bring fewer subwoofers to events. That saves on space both in the trucks and on-site, but also saves down the line with accessories like cables and amplifiers." LSI

Mary Beth Henson *serves as head of communications for L-Acoustics.*

MACKIE AXIS

Evaluating a new digital mixing system.

by Craig Leerman

The DC16 control surface DL32R rack-mount mixer that comprise Mackie AXIS.

AXIS from Mackie is a digital mixing system consisting of the DC16 control surface working with the DL32R rack-mount mixer. It provides a unique surface-to-wireless workflow, allowing users to switch between the DC16's hardware controls and wireless mixing.

AXIS provides 32 mic/line input channels, each with a recallable Onyx+ mic preamp along with 4-band parametric EQ, high-pass filter, gate, compression and RTA/spectrograph. The 28 output buses also offer 4-band parametric EQ, as well as 31-band graphic EQ, high- and low-pass filters, compressor/limiter, delay, and RTA/spectrograph.

The feature set also includes three onboard stereo processors, six VCAs, six mute groups, six stereo linkable matrices, 32 x 32 direct-to-disk recording, 32 x 32 networking for recording to Mac or PC, and 32 x 32 (Audinate) Dante network routing. I've previously reviewed the DL32R mixer (LSI February 2015), so the primary focus here will be on the DC16 control surface and how well the two units operate together.

The DC16, which operates in tandem with the workflow of Mackie's Master Fader app, comes with 17 Alps 100 mm motorized faders, a dedicated select channel section, full-color, backlit TFT (thin-film transistor) screens with variable brightness control, channel labeling, and 6-segment LED channel meters. Onboard connections include an XLR talkback input, 1/8-inch stereo input, and control room/headphones output. The chassis frame is steel with aluminum extrusions at front and rear, measuring 3.3 by 36.8 by 17.6 inches (h x w x d) and weighing 38 pounds.

SmartBridge accommodates up to three iPads of any size.

Up to three iPad devices can be added to the DC16, with charging built in. This is done via the integrated SmartBridge that provides simultaneous control over multiple channels and smart sensing that knows when an iPad is in place. SmartBridge also offers customization over each iPad view with both a fixed and history mode for workflow flexibility. One of the iPads can be utilized for wired recording and playback, while any of them can be used remotely as a controller.

The DC16 surface is very clean and clearly labeled, divided into logical sections. Every control and channel is instantly accessible. At the top are select channel controls, followed by each channel's backlit ID screen that are inline with their respective faders and meters. To the left of the faders is the groups select area with four TFT screens, modifiers (alt/shift/assign/lock buttons that allow other controls to do something else) and channel editing controls, while to the right is the mix select area with four more TFT screens, banking controls, snapshot control with TFT screen, and analog knobs for the talkback, monitor and headphone volumes.

GETTING ACQUAINTED

Out of the box I was impressed with the look and feel of the controller. It's a solid, rugged unit that would have no problems being taken out on a tour. As with all Road Test reviews, the first stop is my test bench.

The first order of business was installing the optional Dante card in the DL32R. I made sure the unit was unpowered then simply removed two screws, slid the included standard networking card from the slot, inserted the Dante card, and reinstalled the two screws. Then I connected a Wi-Fi router to the network port, grabbed one of my iPads that already had the Master Fader app loaded, and went to the App Store to upgraded to the latest version.

Next I linked the iPad (via Wi-Fi) to the DL32R, where the app then performed a firmware upgrade. Once the upgrade was complete, I connected the DL32R to some powered loudspeakers and quickly made sure the mixer was working. This was probably the easiest firmware update I've ever done with any piece of gear.

After the firmware update it was time to focus on the DC16. I connected a Cat-5 cable between the DC16 and DL32R and powered up the surface. The controller found the mixer and walked me through the setup process. When the DC16 was connected, my iPad could still control the mixer — perfect for having one person walk around and/or mix monitors while a second person mans the console at front of house.

Up to four channels at one time can be accessed and controlled by the DC16 or iPads. In addition, multiple DC16s can be configured to operate the DL32R rack mount mixer so one could serve as the FOH desk while another handles monitors. Also note that up to 20 iPads, iPhones or iPod touch devices can also

The backlit TFT screens on the surface are very clean and easy to read/ID.

be connected to the mixer for personal monitoring running the My Fader app.

I placed an iPad in SmartBridge of the DC16 and it automatically discovered the iPad. Up to three iPads of any size could be used, but I discovered that the thick cases on mine wouldn't quite fit. (Standard-size cases and of course un-cased iPads will fit easily though.)

With iPads in the SmartBridge, the operator can display the currently selected channel, channels that were formerly adjusted, or a fixed view of something, like the lead vocalists channel or an overview screen of levels and meters. iPads can be removed and used as a wireless control surface, and when placed back in the SmartBridge, they return to the last known follow state automatically. With the onboard TFT screens and up to three docked iPads, the DC16 displays a whole lot of information at a glance.

I also really like that the DC16 will charge iPads when they're in the SmartBridge via cables to USB connectors at the rear of the unit. It's always a pain charging tablets at gigs, having to use a "wall wart" plug and then find a power strip or outlet close by.

After spending a couple of hours working with the system I discovered two important things. First, the surface and screens can be set up in various ways depending on your workflow. Second, the DL32R really shines when you attach faders to the mixer. I do a lot of corporate shows where my finger stays on the podium microphone the entire presentation to ride levels, and I can't do that with a virtual fader.

PUTTING IT TO WORK

Confident that everything was working as intended and having a grasp on the basics, I took AXIS to work a couple of gigs. First up was a speech for a corporate event that needed a pair of podium mics as well as a lavalier for the presenter and playback from my computer for walk in/out music. A small pair of powered loudspeakers covered the room.

AXIS was a bit of overkill for this small application, but the DC16 didn't really take up much more space at the FOH table than the small rack-mount-sized mixer I usually deploy for these types of applications. I used a single iPad and set it to follow the currently selected channel, with the large screen making it quite easy for my old eyes to see what I was adjusting.

It was also great running a single Cat cable for the snake, and I loved the fact that I had a headphone jack at FOH. The headphone circuit gets quite loud and would have no problems getting above a loud band. Having an 1/8-inch jack made it easy to get music tracks via my PC into the system.

Next up was a more complex setup for a board meeting. Multiple tables were arranged in a U shape for 20 participants, each with a mic in front of them on a desk stand. In addition, there were two presenters and a moderator, each wearing a wireless lavalier. A wireless handheld for audience questions and my laptop for walk in/out music completed the inputs. Outputs included two powered loudspeakers and a recording feed to a client-supplied laptop as this was a government meeting and had to be recorded.

In addition to that "official" feed, I also recorded to a small hard drive plugged into the DL32R as a backup. I could also have recorded directly to an iPad docked with the DC16, and while I don't normally this, can clearly see what a great feature it is. I know many use iPads and tablets to record and even mix audio for events (or their own bands), and recording directly to an iPad at FOH would be a very easy way to make a "board tape."

The iPad can also be used for playback, making it simple to have some walk in/out music at an event without having to set up a laptop or other playback device. There is also the 32 x 32 routing to the Dante network, so recordings could be handled that way as well.

I'd also like to emphasize that the onboard 4-band parametric EQ and RTAs made it simple to get the most out of the system with absolutely no feedback.

Because I was located in a corner — a not-so-great mixing position — the monitor outputs on the DC16 allowed me to hook up a pair of small bookshelf-sized powered loudspeakers to clearly hear what was going on. Having a talkback input, stereo input, and monitor and headphone jacks at FOH is a necessity for me as well, and the DC16 delivers.

The system worked flawlessly for the meeting, and I'd also like to emphasize that the onboard 4-band parametric EQ and RTAs made it simple to get the most out of the system with absolutely no feedback. My client was happy and I had an easy gig, in no small way due to AXIS.

As a dedicated controller for the DL32R, the DC16 is fantastic. It provides all-important controls to engineers, and allows them to easily configure the surface to suit their workflows. Owners of a DL32R rack-mount mixer would do well in expanding with the DC16, while the overall AXIS package presents an excellent compact mix rig with the design and facilities to handle larger applications. LSI

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

STAYING WITHIN THE LINES

Medium-format line arrays and a roundup of current models. *by Live Sound Staff*

Large, small and anything in between, line arrays can provide many benefits, including more even audience coverage in terms of frequency response and SPL, control of vertical dispersion well into the lower midrange, improved sight lines, and ease of setup. Further, they're designed to be flown and taken down quickly, often in "blocks" of individual modules, and to be flexibly adjustable to different curvatures – and many can also be ground-stacked. This flexibility can be particularly useful in venues where arrays need to be adjusted regularly to accommodate different types of acts.

The definition of the specific characteristics that make a line array medium-format, as opposed to small- or large-format, is somewhat arbitrary. Is it how wide or high each element is, how much it weighs, how many components each houses, the diameter of the components, or how loud it gets? For this overview, as we've done for several years, we've based the selection on the size of the largest LF driver within the array, considering those with 8-inch to 10-inch cones to be medium-format.

Even within the medium-format category, there's a lot of variety. Among the represented brands and models, the horizontal coverage angle varies from 80 degrees to 150 degrees from a single array column, with most ranging between 100 to 120 degrees. Some manufacturers offer cabinets with the same "footprint" with differing horizontal coverage, allowing the user to better customize coverage for a particular venue. Enclosure width varies from a bit over 23 inches to over 30 inches, and weight for each cabinet ranges from a bit over 30 pounds to over 100 pounds. Many are self-powered, and others have dedicated external processing and amplification.

Lesser-scaled line arrays, in comparison to enclosures with 12-inch or 15-inch LF drivers, allow wider splay angles within the array elements, while still maintaining consistent coverage. This characteristic can be useful for covering smaller venues that have multiple levels.

The following Real World Gear tour of recent models covers a variety of design concepts based around LF components in the 8- to 10-inch range. Enjoy this look at medium-format line arrays. LSI

Alcons Audio LR18
alconsaudio.com

Configuration: 3-way
Dispersion (h x v): 90 x 10 degrees
LF: 2 x AMB8 8-inch cone drivers, vented
MF: 1 x 6.5-inch cone driver coaxially mounted behind HF driver
HF: 1 x RBN702rs 7-inch pro-ribbon transducer
Frequency Response: 70 Hz – 20 kHz
Maximum SPL: 140 dB
Power: Sentinel amplified controller (Sentinel10 recommended)
Rigging: Angle-setting on cabinets without lifting the array
Size (h x w x d): 8.9 x 31.1 x 17.2 inches
Weight: 61.7 pounds
Companion Sub: BC543 (triple 18-inch), BQ211 (single 21-inch)

RCF HDL 20-A
rcf-usa.com

Configuration: 2-way
Dispersion (h x v): 100 x 15 degrees
LF: 2 x 10-inch neodymium cone drivers
HF: 1 x 3-inch titanium compression driver on a custom waveguide
Frequency Response: 55 Hz – 20 kHz
Maximum SPL: 135 dB
Power: Onboard class D amplifier, DSP controlled input section with selectable presets
Rigging: Integral hardware with adjustable splay angles
Size (h x w x d): 11.5 x 27.7 x 17.5 inches
Weight: 64 pounds
Companion Sub: SUB Series (8006-AS, 8005-AS, 8004-AS)

L-Acoustics KARAi
l-acoustics.com

Configuration: 2-way
Dispersion (h x v): 110 degrees symmetric; vertical array dependent
LF: 8-inch cone drivers in bass-reflex tuned enclosure
HF: 1 x 3-inch compression driver coupled to DOSC waveguide
Frequency Response: 55 Hz – 20 kHz
Maximum SPL: 139 dB
Power: LA8 amplified controller, provides 3 operating modes
Rigging: 4-point system; angle increments of 0, 1, 2, 3, 4, 5, 7.5, and 10 degrees
Size (h x w x d): 9.8 x 28.1 x 15 inches
Weight: 51.7 pounds
Companion Sub: SB18 (single 18-inch)

Adamson S10
adamsonsystems.com

The new S10 is a 2-way, full-range line array enclosure ideal for a wide range of mid-size portable and install applications. It's loaded with two 10-inch ND10-LM Kevlar neodymium cone drivers joined by an NH4TA2 1.5-inch-exit HF compression driver.

A wave shaping sound chamber produces a slightly curved wavefront with a nominal dispersion pattern of 110 by 10 degrees (h x v). The chamber exhibits increased vertical response with minimal sacrifice of HF energy in the far field. Patent-pending Controlled Summation Technology further eliminates low-mid lobing. The S10 offers a maximum peak SPL of 141.3 dB, quite notable for such a compact enclosure.

The cabinet is made of marine grade birch plywood as well as aircraft grade steel and aluminum. The rigging system incorporates Adamson's proprietary Slidelock rigging technology for exceptionally easy setup and strike. An install-specific version (S10i) is also available.

OF NOTE: *The S10 (and companion S119 sub) are designed to be driven by the E-Rack unified rack solution, available in 8-channel and 12-channel configurations with Lab.gruppen PLM 12K44 amplifiers (with Lake processing) and supplied with a 20-port Ethernet switch to route dual-redundant Dante and control signal. Blueprint AV software is included.*

KEY SPECIFICATIONS:
Configuration: 2-way
Dispersion (h x v): 110 x 10 degrees
LF: 2 x 10-inch Kevlar neodymium cone drivers
HF: 1 x 1.5-inch-exit compression driver
Frequency Response: 60 Hz – 18 kHz
Maximum SPL (peak): 141.3 dB
Amplification: Designed for use with E-Rack
Rigging: Proprietary SlideLock rigging system allows angles to be set prior to lifting
Size (h x w x d): 10.4 x 29 x 20.7 inches
Weight: 60 pounds
Companion Sub: S119 (single 19-inch)

TECHNOLOGY FOCUS: *Controlled Summation Technology brings the LF drivers as close together as possible while symmetrically outwardly splaying them, which increases usable frequency range while decreasing summation at the crossover point, reducing interference. The LF drivers are also recessed behind the exit of the HF sound chamber so as to not limit their size and shape. Delay aligns the LF and HF.*

VUE Audiotechnik al-8
vueaudio.com

Configuration: 3-way
Dispersion (h x v): 90 x 10 degrees
LF: 2 x 8-inch neodymium cone drivers
MF: 4 x 4-inch Kevlar neodymium cone drivers
HF: 2 x 1-inch-exit Truextent beryllium-diaphragm compression drivers
Frequency Response: 75 Hz – 18kHz (+/-2.5 dB)
Maximum SPL: 136 dB
Power: External VUE V6 Systems Engine
Rigging: Integrated hardware, angles selectable in 1-degree increments
Size (h x w x d): 10.2 x 29.4 x 17.5 inches
Weight: 76.6 pounds
Companion Sub: al-8-sb (single 18-inch)

Martin Audio MLA Compact
martin-audio.com

Configuration: 3-way cellular drive
Dispersion (h x v): 110 x 10 degrees
LF: 2 x 10-inch neodymium cones, Hybrid slot-horn loaded
MF: 2 x 5-inch neodymium cones, horn loaded
HF: 4 x 0.7-inch neodymium drivers, horn loaded
Frequency Response: 65 Hz – 18 kHz
Maximum SPL: 135 dB
Power: Onboard 5-channel class D; DSP & networking
Rigging: Suspension of up to 24 enclosures
Size (h x w x d): 11 x 31 x 19.7 inches
Weight: 109 pounds
Companion Sub: DSX (dual 18-inch)

Meyer Sound LEOPARD
meyersound.com

Configuration: 2-way
LF: 2 x 9-inch cone drivers, Hybrid slot-horn loaded
HF: 1 x 3-inch compression driver coupled to constant directivity horn through patented REM manifold
Frequency Response: 55 Hz – 18 kHz
Maximum SPL: N/A
Power: Self-powered (class D), Galileo loudspeaker management
Rigging: Captive GuideALinks provide with splay angles from 0.5 to 15 degrees
Size (h x w x d): 11.1 x 26.9 x 21.6 inches
Weight: 75 pounds
Companion Sub: 900-LFC (1 x 18-inch)

RWG Spotlight Listing

d&b audiotechnik Y8/Y12
dbaudio.com

Whether for mobile deployment or fixed installation, compact Y8 and Y12 line array loudspeakers provide flexible and configurable solutions for a broad spectrum of performance needs. Intended applications include houses of worship, theaters and conference centers, clubs, trade shows and auditoriums for audiences of 50 to 2,000.

Y8/Y12 offer 80 and 120 degrees horizontal directivity, respectively, with remarkable dispersion control down to 500 Hz. The mechanical and acoustic loudspeaker design permits columns of up to 24 cabinets, which can comprise Y8 and/or Y12s as well as the dedicated cardioid Y-SUB.

Both models utilize the same patented 3-point rigging as the larger V-Series and J-Series for scalable, reliable and efficient solutions, supported by an extensive range of transport options and loudspeaker accessories. Y-Series Yi models are specifically designed for permanent integration and differ only in cabinet construction and mounting hardware.

TECHNOLOGY FOCUS: *Utilizing sophisticated horn geometry and an advanced bass-reflex port design, both Y-Series line and point source loudspeakers deliver full bandwidth capabilities with an extended LF output. In addition, a custom waveguide and new HF driver for Y8/Y12 models provide the renowned high directivity and smooth HF of the V-Series.*

COMPANION PRODUCT: *Y7P and Y10P point source loudspeakers also offer wide and narrow dispersion options (75 x 40 and 110 x 40 degrees, respectively) and share the same directivity and dispersion control toward low frequencies. They include 2 x 8-inch drivers in a dipole arrangement with a 1.4-inch driver on a rotatable CD horn.*

KEY SPECIFICATIONS:
Configuration: 2-way
Dispersion (h x v): 80 or 120 degrees; vertical is array dependant
LF: 2 x 8-inch cones, dipole arrangement
HF: 1 x 1.4-inch-exit driver with a wave transformer
Frequency Response: 54 Hz – 19 kHz
Power: d&b amplification (D6, D12, D20, D80)
Rigging: 0 to 14 degrees in 1 degree steps
Size (h x w x d): 10 x 24.8 x 14.8 inches
Weight: 45 pounds
Companion Sub: Y-SUB (cardioid)

QSC Audio WideLine-10
qsc.com

Configuration: 2-way
Dispersion (h x v): 140 degrees; vertical array dependent
LF/MF: 2 x 10-inch cone drivers
HF: 1 x 3-inch compression driver on proprietary slot waveguide
Frequency Response: 55 Hz – 18kHz
Maximum SPL (peak): 133 dB
Power: Biamp or triamp; external amplification
Rigging: Integrated hardware, adjustable from 0 to 10 degrees in 1-degree increments
Size (h x w x d): 10.8 x 27.4 x 20.7 inches
Weight: 83 pounds
Companion Sub: WL218-sw (dual 18-inch), WL118-sw (single 18-inch)

DAS Audio Event 210A
dasaudio.com

Configuration: 3-way
Dispersion (h x v): 90 degrees; vertical array dependent
LF: 1 x 10Mi4 10-inch cone driver
MF: 1 x 10Mi4 10-inch cone driver
HF: 1 x M-75 3-inch compression driver
Frequency Response: 70 Hz – 20 kHz
Maximum SPL: 134 dB
Power: Onboard 3-channel Class D amplifier (180 watts continuous/360 W watts peak per channel)
Rigging: Integrated rigging, angles adjustable
Size (h x w x d): 10.6 x 28.7 x 14.4 inches
Weight: 74.8 pounds
Companion Sub: Event 218A (dual 18-inch)

dBTechnologies VIO L210
dbtechnologies.com
americanmusicandsound.com

Configuration: 2-way
Dispersion (h x v): 100 degrees; vertical array dependent
LF: 2 x 10-inch cone drivers in V form housed in bass-reflex tuned enclosure
HF: 1 x 3-inch compression driver on new waveguide design
Frequency Response: 78 Hz – 18.1 kHz
Maximum SPL: 135 dB
Power: Onboard Digipro class D amplifier
Rigging: 3-point system; rear equipped with link to set splay angles
Size (h x w x d): 12.6 x 28.3 x 20.4 inches
Weight: 63 pounds
Companion Sub: S318 (triple 18-inch)

EAW Anna | *eaw.com*

Anna brings all of the benefits of Adaptive Performance to mid-sized applications requiring excellent sound quality, high output and precise coverage, as well as the ability to quickly adapt that coverage to any venue geometry.

Anna's smaller footprint and lighter weight make it ideal for permanent or temporary use in sheds, theaters, clubs, mobile staging and corporate AV. Anna arrays also interlock seamlessly with Anya arrays to horizontally expand Anya-based systems for out fill or delay. Both Anya and Anna also integrate perfectly with Otto (EAW's Adaptive subwoofer) to form a cohesive one-family solution for any application.

With the combination of Anna's ingenious enclosure design and the power of Adaptive Performance, users can be confident that the system will provide spectacular results at every seat in any venue, for touring or permanent installation.

TECHNOLOGY FOCUS: *Anna enclosures include 14 built-in amplifier and processing channels, independently powering and processing each loudspeaker component. Resolution 2 software controls the processing of each acoustic cell individually to generate the ideal coverage pattern for the venue while minimizing the impact of the venue's acoustics. The end result is extremely high fidelity, output and coverage control.*

OF NOTE: *Anna modules hang straight, without any vertical splay, and Resolution 2 software adapts performance to produce custom-tailored, coherent full-range coverage pattern to perfectly match the venue in seconds.*

KEY SPECIFICATIONS:
Configuration: 3-way
Dispersion (h x v): 100 degrees x Adaptive
LF: 2 x 10-inch cone drivers (proprietary Offset Aperture loading)
MF: 4 x 5-inch cone drivers (on proprietary Radial Phase Plugs and CSA apertures)
HF: 8 x 1-inch-exit compression drivers on proprietary horn
Frequency Response: 45 Hz – 18 kHz
Maximum SPL (Unadapted): 130/136/141 dB (LF/HF/MF)
Power: Self-powered
Size (h x w x d): 11.3 x 40 x 23.6 inches
Weight: 135 pounds
Companion Sub: Otto

Electro-Voice XLD291
electrovoice.com

Configuration: 3-way
Dispersion (h x v): 90 x 10 degrees
LF: 1 x DVN2080 8-inch cone driver
MF: 1 x DVN2080 8-inch cone driver
HF: 2 x ND2S 2-inch compression drivers
Frequency Response: 75 Hz – 18kHz (-3 dB)
Maximum SPL: 144 dB
Power: External, TG Tour Grade amplifiers recommended; LF/MF – 200 watts, HF – 80 watts (both RMS)
Rigging Angles: Integrated hardware, adjustable
Size (h x w x d): 9.9 x 28.6 x 14.5 inches
Weight: 48 pounds
Companion Sub: Subwoofer: XS212 (12-inch) & XLC215 (15-inch)

JBL Professional V20
jblpro.com

Configuration: 3-way
Dispersion (h x v): 105 x 0 – 12.5 degrees (inter-enclosure angles)
LF: 2 x 2261H 10-inch cone drivers
MF: 4 x 2164H 5-inch cone drivers
HF: 3 x D2415K compression drivers
Frequency Response: 60 Hz – 20 kHz
Maximum SPL: 133 dB SPL (MF)
Power: BSS OmniDriveHD V5 processing for use with Crown I Tech HD or VRack
Rigging: Angle Stop Mechanism (ASM) suspension allows tension or compression suspension
Size (h x w x d): 11 x 35.9 x 15.8 inches
Weight: 88 pounds
Companion Sub: S28 (18-inch) and S25 (15-inch)

WorxAudio TrueLine V8
worxaudio.com

Configuration: 2-way
Dispersion (h x v): 120 x 10 degrees
LF: 2 x 8-inch neodymium cones
HF: 1 x 3-inch voice coil titanium driver
Frequency Response: 65 Hz – 18 kHz
Power: PXD Series amplifier platform recommended
Rigging Angles: TrueAim Tour rigging adjustable in 1-degree increments
Size (h x w x d): 10.5 x 28 x 18 inches
Weight: 108 pounds
Companion Sub: TrueLine TL118SS & TL218SS (single and dual 18-inch, respectively)

RWG Spotlight Listing

NEXO STM M28 | *yamahaca.com*

The STM (Scale Through Modularity) M28 line array is an injection-molded, all-purpose 2-way loudspeaker providing 90/120 degrees of horizontal dispersion and 0- to 15-degree splaying angle between modules. It's the same width as the STM M46 main cabinet but two-thirds the height, and can be arrayed in the same column with M46 cabinets in a large-format STM system, arrayed in the same column as a B112 bass cabinet in a mid-format STM system, or used independently.

The M28 includes CompassRig and REDLock rigging that provides pre-setting of inter-cabinet angles, and it enables any size system to be flown safely by one person. The REDLock handle locks front rigging points from the rear of cabinet, and all adjustments can be made from one position at the rear of the array.

Recent applications of the M28 include the Lincoln Center Summer Series (Audio, Inc.); Battle of Bristol pre-game concert (pictured here) featuring Kenny Chesney, The Band Perry, and Old Dominion (Morris Light & Sound); Riot Fest Denver (Nomad Sound); Merchants & Music Festival (Event Enterprises); and Together 2016 (REACH Communications), which utilized 100-plus M28s as part of a 225-box NEXO rig.

TECHNOLOGY FOCUS: *The use of flat membrane MF drivers allows the HF exiting between the baffles to expand uninterrupted, ensuring smooth frequency response and consistent horizontal coverage.*

KEY SPECIFICATIONS:
Configuration: 3-way
Dispersion (h x v): 120 degrees x 0-15 degrees
LF: 2 x 8-inch cone drivers
MF: 4 x 4-inch flat membrane drivers
HF: 1 x 1.4-inch compression driver
Frequency Response: 60 Hz – 19 kHz
Maximum SPL: 139 dB
Power: NXAMP4x4 amplified controller
Rigging: 3-point system with intercabinet angle adjustments from 0.2 to 15 degrees in logarithmic steps
Size (h x w x d): 13.8 x 22.6 x 28.1 inches
Weight: 84 pounds
Companion Sub: STM B112 (single 12-inch), STM S118 (dual 18-inch)

Fulcrum Acoustic FL283
fulcrumacoustic.com

Configuration: 2-way
Dispersion (h x v): 90 degrees; vertical array dependent (20-degree max splay)
LF: 2 x 8-inch cone drivers, horn-loaded
HF: 3 x 1.4-inch compression drivers
Frequency Response: 54 Hz – 18.6 kHz
Maximum SPL: 139 dB
Power: Configured for full-range passive operation at 16 ohms; up to 8 units can be driven from a single amplifier channel
Rigging: Adjustable from 4 to 20 degrees in 2-degree increments; 0- and 2-degree splays with optional extended rear link bar
Size (h x w x d): 12.8 x 23.1 x 19.3 inches
Weight: 57 pounds
Companion Sub: FLS115 (single 15-inch)

Turbosound TFA-600HW
turbosound.com

Configuration: 3-way
Dispersion (h x v): 100 x 16 degrees
LF: 2 x 10-inch cone drivers, horn-loaded
MF: 1 x 6.5-inch cone driver on Polyhorn device
HF: 1 x 1-inch compression driver on Dendritic device
Frequency Response: 90 Hz – 18 kHz
Maximum SPL: 136 dB
Power: Triamp/biamp modes
Rigging: Vertical and horizontal flying systems integrated into cabinet
Size (h x w x d): 12 x 28 x 22 inches
Weight: 93.5 pounds
Companion Sub: TFA-600B/L (both single 18-inch)

Bose Professional ShowMatch DeltaQ
pro.bose.com

Configuration: 2-way
Dispersion (h x v): 100 or 70 degrees (changeable) x choice of 5, 10, or 20 degrees
LF: 2 x 8-inch cone drivers
HF: 4 x EMB2S compression drivers
Frequency Response: 69 Hz – 17 kHz
Maximum SPL: 127/131 dB (LF/HF)
Power: HF: 100 watts continuous, 400 watts peak; LF: 450 watts continuous, 1,800 watts peak
Rigging: 3-point "quick pin" system, support for up to 24 modules
Size (h x w x d): 11.1 x 31.2 x 18.3 inches
Weight: N/A
Companion Sub: ShowMatch SMS118

Renkus-Heinz VARIA VA101

renkus-heinz.com

VARIAi modular array systems are designed to be the most flexible and versatile loudspeakers available, employing powerful low-frequency woofers and lightweight neodymium compression drivers coupled to highly advanced cabinet and waveguide designs.

VARIAi 101 full-range models are available in three vertical cabinet angles – 7.5, 15 and 22.5 degrees – allowing vertical coverage to be easily tailored to meet shape and SPL requirements for every venue. For example, 15- and 22.5-degree cabinets can be used to cover greater vertical angles while 7.5-degree cabinets provide tighter control.

Taking the flexibility quotient even further, each cabinet can be ordered with one of five WaveGuides: 60, 90 or 120-degree standard WaveGuides or VARIAi's unique 60- to 90-degree or 90- to 120-degree Transitional WaveGuides.

TECHNOLOGY FOCUS: *VARIA101i DLL data in EASE and EASE Focus II simulation software allows users and system designers to quickly and accurately predict the response of the array.*

OF NOTE: *VARIAi can be configured in three basic system styles: As a Modular Point Source Array for applications with large vertical coverage angles; as a Modular Line Array for applications requiring line array throw and performance; and as a Horizontal Point Source Array for accurate horizontal coverage built in 22.5-degree increments.*

KEY SPECIFICATIONS:

Configuration: 2-way
Dispersion (h x v): 90 (also 60 & 120) x 7.5 (also 15 & 22.5) degrees
LF: 1 x 10-inch cone driver
HF: 2 x 1-inch neodymium drivers on proprietary Tuned Conic Diverter waveguide
Frequency Response: 60 Hz – 20 kHz
Maximum SPL: 126 dB
Power: Class D biamp (500 watts LF, 250 watts HF) with integrated RHAON networking; passive version also available
Rigging: Articulated hardware provides adjustment from 0 to 7.5 degrees
Size (h x w x d): 13 x 23.7 x 15 inches
Weight: 64 pounds
Companion Sub: VA15S (15-inch)

Clair Brothers i208

clairbrothers.com

Configuration: 3-way
Dispersion (h x v): 120 x 10 degrees (90 degrees h also available)
LF: 1 x 8-inch cone driver
LMF: 1 x 8-inch cone driver
HF: 1 x 1.4-inch-exit compression driver HF waveguide
Frequency Response: 60 Hz – 20 kHz
Maximum SPL: 132 dB
Power: Amplifier recommended at 1,000 to 1,300 watts, 8 ohms
Rigging: Integral bimodal rigging with angle adjustments between 0 and 10 degrees
Size (h x w x d): 9.2 x 28.9 x 23.9 inches
Weight: 62 pounds
Companion Sub: iS118 arrayable subwoofer

FBT MUSE 210LA

fbt.it

Configuration: 2-way
Dispersion (h x v): 90 degrees x 10 degrees
LF: 2 x 10-inch cones, bass-reflex
HF: 2 x 1-inch-throat B&C drivers on waveguide
Frequency Response: 55 Hz – 20 kHz
Maximum SPL: 135 dB
Power: Onboard class D, switch mode; onboard DSP
Rigging: 0 to 10 degrees in 2-degree steps
Size (h x w x d): 11.6 x 25.6 x 16.7 inches
Weight: 83.7 pounds
Companion Sub: MUSE 118FSA (single 18-inch)

K-array KH2

k-array.com

Configuration: 2-way
Dispersion (h x v): 110 x 10 degrees (preset dependent)
LF: 2 x 8-inch neodymium cone drivers
HF: 2 x 1.4-inch neodymium compression drivers
Frequency Response: 70 Hz – 19 kHz
Maximum SPL: 136 dB
Power: Onboard class D amplification; DSP controlled, digital steering
Rigging: Integrated hardware, adjustable
Size (h x w x d): 9.8 x 25.3 x 8.3 inches
Weight: 62 pounds
Companion Sub: KS5 (dual 21-inch)

Outline Butterfly C.D.H. 483
outlinearray.com

Configuration: 3-way
Dispersion (h x v): 90 degrees x 7.5 degrees
LF: 2 x 8-inch neodymium cones, band-pass loaded
MF: 2 x 8-inch cones, horn loaded
HF: 1 x 3-inch-exit neodymium drivers on Double Parabolic Reflective Waveguide
Frequency Response: 110 Hz – 18 kHz
Power: 800/120 watts (LF & MF/HF, continuous RMS)
Rigging: - 0 to 7.5 degrees, 0.125-degree increments
Size (h x w x d): 9.4 x 29.6 x 23.6 inches
Weight: 75 pounds
Companion Sub: C.D.L. 1815 (cardioid)

Ramsdell Pro Audio LA10-2
ramsdellproaudio.com

Configuration: 2-way
Dispersion (h x v): Option of 120, 90 or 60 x 10 degrees
LF: 1 x 10-inch neodymium cone driver, vented
HF: 1 x 1.75-inch voice coil neodymium compression driver
Frequency response: 65 Hz – 18 kHz
Maximum SPL: 121 dB
Rigging: Integrated hardware, 0 to 10 degrees in 2 degree increments
Power: Active and passive versions available; optional bi-amp
Size (h x w x d): 15.9 x 21.7 x 12 inches
Weight: 29 pounds
Companion Sub: LA12-S (single 12-inch)

ISP Technologies HDL 2208
isptechnologies.com

Configuration: 2-way
Dispersion (h x v): 100 x 10 degrees
LF/MF: 2 x 8-inch neodymium cone drivers
HF: 1 x 2.6-inch compression driver
Frequency Response: 68 Hz – 16kHz
Maximum SPL: 134 dB
Power: Onboard 3-channel DCAT amplifier (850 watts RMS total)
Rigging: Integrated hardware, adjustable from 1 to 10 degrees in 1-degree increments
Size (h x w x d): 9.1 x 24 x 19 inches
Weight: 62 pounds
Companion Sub: HDL118 (18-inch)

Verity Audio IWAC208
verityaudio.fr

Configuration: 2-way
Dispersion (h x v): 90 degrees x 8 degrees
LF: 2 x 8-inch cone drivers
HF: 2 x 1.7-inch compression drivers on proprietary waveguide/expanded horn
Frequency Response: 95 Hz – 19.5 kHz
Maximum SPL: 127/131 dB (LF/HF)
Power: Verity V4.25 amplification/DSP package provided in rack that can be ground-based or flown with arrays
Rigging: Adjustable 0 to 6 degrees in 1-degree increments
Size (h x w x d): 10.6 x 26.9 x 14.4 inches
Weight: 52.9 pounds
Companion Sub: SUB118T (single 18-inch)

Carvin Audio TRX3210
www.carvinaudio.com

Configuration: 3-way
Dispersion (h x v): 100 x 10 degrees
LF/MF: 2 x 10-inch neodymium cone drivers
HF: 2 x 1-inch-exit Mylar compression drivers on a PurePath lens system
Frequency Response: 80 Hz – 18.5 kHz
Maximum SPL: 131 dB
Power: External, 500 – 1,000 watts recommended
Rigging: Captive flypoints, optional SureFly rigging
Size (h x w x d): 11.5 x 23.5 x 14 inches
Weight: 50 pounds
Companion Sub: TRx118 (single 18-inch), TRx121 (single 21-inch)

AudioCenter V-HLA10+ MKII
audiocenter.net

Configuration: 2-way
Dispersion (h x v): 90 degrees x 15 degrees
LF: 1 x 10-inch custom cone driver
HF: 1 x 1.4-inch custom compression driver on proprietary T.A.C. waveguide
Frequency Response: 62 Hz – 20 kHz
Maximum SPL: 130/133 dB (LF/HF)
Power: CA450 touring power supply; active 2-way crossover
Rigging: Same frame can be used for flying or groundstacking; vertical angle of the frame is adjustable to 4, 2, 0, -2 and -4 degrees.
Size (h x w x d): 11.2 x 23 x 15.8 inches
Weight: 44 pounds
Companion Sub: V-HLA10+SUB MKII (single 15-inch)

Tarik Solangi has been named vice president of sales and marketing at **RCF USA**. Previously national sales manager, he started with the company as an independent consultant providing technical support and managing product demonstrations for the touring and performing arts communities.

"There are many reasons to move Tarik into this position," says **Roni Nevo**, CEO of RCF USA. "As our company has grown tremendously over the last few years, it's important to have the infrastructure in place for the continued management of the company for the future."

Paul Lamarre has joined **EAW**, managing sales and support for existing and future customers in the southern U.S. Previously he served as regional sales manager for EDA Pro AV, based in Duluth, GA, and was also general manager of Sam Ash Music in the greater Nashville area.

Rob Hofkamp is now the North American director of operations for **Void Acoustics**, which designs and manufactures loudspeaker systems, power amplifiers, control and accessories for the pro audio market. He brings decades of experience to the British company at a time of expansion that has also seen industry veteran **David Bissett-Powell** come on board.

(Hofkamp and Bissett-Powell are pictured above with Void Acoustics managing director **Alex Skan**.) LSI

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In Memoriam

Mark Herman, 1956-2016

Mark Herman, for decades a passionate, multifaceted member of the professional audio community, passed away December 17 due to a massive stroke just after a heart bypass procedure. He was 60 years of age.

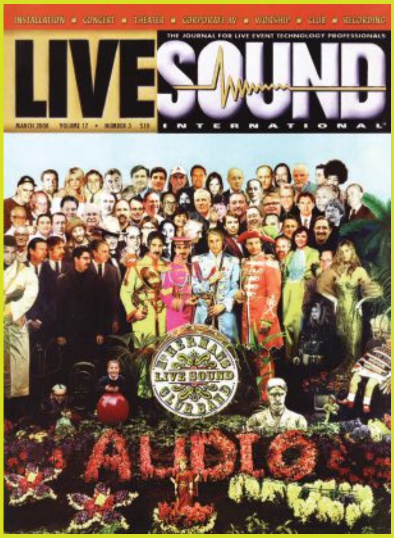
A resident of the San Francisco bay area, he attended UC Davis before starting the life-long audio career that would treat him very well. Starting in 1980, Herman worked with console design pioneer Jim Gamble as a technician and then founded Hi-Tech Audio Systems in 1983, a live sound console rental company that grew from a single console to the largest worldwide when Herman sold his stake in it in 1995. Following that, he launched Triton Network, which manufactured A-TECH loudspeakers in addition to handling the export of audio gear.

Herman's first dip in the audio publishing world was with *Mix* magazine and then *Recording Engineer/Producer*. He was also an original contributor to *Live Sound International (LSI)* when it launched more than 25 years ago, and remained heavily engaged with the publication before purchasing it in the late 1990s. Serving as president and group publisher of parent brand Huge Universe, in the early 2000s he also added *ProSoundWeb* to the stable.

Herman chose to remain on board after Huge Universe was acquired by current owner EH Media in 2007, serving as an advisor and consultant for several years. He also pursued other opportunities, such as being an original investor in DiGiCo, and founding AVL Factor, a video production and delivery company with more than 2,000 educational videos on audio, lighting and video production. Most recently he was focused on Big Buzz Factor, a website designed to offer audio product and industry news.

John Murray, former 11-year technical editor of *LSI*, relates, "I always looked forward to running around with Mark at all the trade shows, whether it was Musikmesse in Frankfurt, AES, InfoComm, or NAMM in Anaheim. I could always depend on a great group dinner or the last-call nightcap at the Anabella Hotel bar with Mark and a slew of his admiring followers, of which I was one. He was the business sage of the pro audio world, knew all the big-dog players, was the unchallenged bon vivant of our industry, and my personal friend. The trade shows will never be the same for me. God, I will miss him."

Herman is survived by his wife Lucy, daughter Holly, sister Julie, brother Matt, and mother Lynn. LSI



***Mark Herman and friends on the cover of the March 2008 issue of* LSI.**

IT'S A SMALL WORLD

And red flags are flapping in the breeze... *by Jonah Altrove*

My phone rings at midnight. I suspect that being violently awakened is somehow unhealthy, like red lining a car's engine in cold weather. In another five hours, I'll (gently) awaken and head to the site of a local music festival for load-in. Apparently being well-rested is too much to ask.

Now, pause a second. We're going to pull a Tarantino and jump back about two months, to when I'm handling arrangements for a different show. At 9 pm the night before the gig, I receive a call from the gentleman that the promoter hired to provide the generators.

Me: "Hello."
Generator Guy: "Hey, is this Jonah?"
Me: "Yep"
GG: "Hey! This is Larry. What kind of power do you need?"

This is kind of last-minute, but fine, I repeat the generator spec I provided to the promoter weeks ago.

GG: "Yeah, we don't have that."
Me: "OK, what do you have?"
GG: "Well, how many outlets do you need?"
Me: "We have our own distro. I just need about 25 kVA at 120/208, but I need it entertainment-related."
GG: "Right. So how many outlets, about?"

Deep breath.

Me: "I don't need outlets. We tie in with camlock."
GG: "At 120V?"
Me: "To ground, yes. But it's 208V phase to phase."

At this point, I realize I'm explaining 3-phase power generation to the guy selling me 3-phase power generators. Red flags are flapping in the breeze.

GG: "OK, no problem. Just let me know how many outlets you need."

Ah... Now I see what's going on here.

Me: "Larry, I'm going to have to call you back."

I call my boss, who's asleep.

Boss: "Hello?"
Me: "Get a pen."
Boss: "Why?"
Me: "Because if you don't deal with this, someone is going to be electrocuted at the show tomorrow."
(Pause.)
Boss: "Give me the number."

I provide the info and wait for his call back. It's not long.

Me: "So... How'd it go?"
Boss: "I fired him. Then called [friend at rental house]. He's bringing you a DCA45. You know how to work it?"
Me: "Yep."
Boss: "Good night."
Me: "One more thing –"
Boss: "What?"
Me: "How many outlets do you need?"

Onsite the next morning, the promoter wanders over as we're pounding in the ground rod.

Promoter: "What's that?"
Me: "Ground rod."
Promoter: "We didn't have that last year."
Me: "I didn't run your show last year. It's required by law. Fire marshal shows up, he'll shut you down."
Promoter: "Oh."

Now let's flash forward two months. Thanks to the "How many outlets?" guy, I now insist on speaking directly to the power provider for each event. The organizers of the show I'm about to load in have assured me they have a "great provider" they use often. No problem, I reply, ask the provider to call me, please.

And now you can probably guess how this story ends.

Me: "Hello."
Great Provider: "Hey, is this Jonah?"
Me: "Yep."
GP: "Hey! This is Larry. What kind of power do you need?" LSI

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