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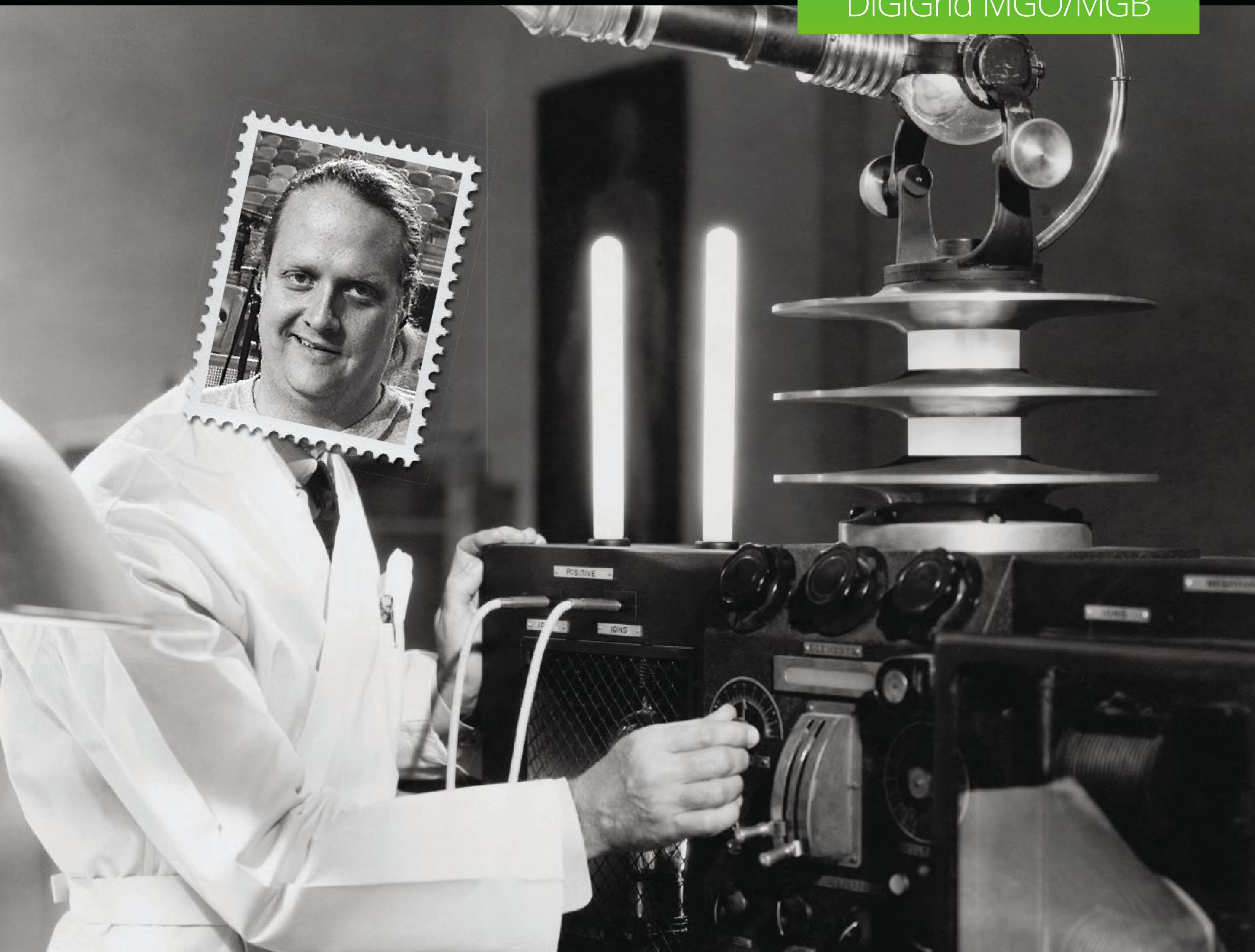



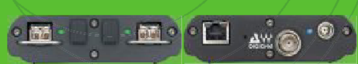



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# From the Editor's Desk…

Last month's Winter NAMM show in Anaheim drew a record crowd (well over 99,000 registrants) along with a dramatic increase in exhibitors.

At about the same time, we presented a poll on the home page of ProSoundWeb asking readers how their business prospects are looking for this year. A whopping 82 percent responded that they expected it to be the same or better than last year, with 64 percent choosing the latter response.

Clearly there's a lot of optimism going around in the pro audio industry, a nice and welcome change from the uncertainty of the past several years. At NAMM we saw an even greater flow of new products and technologies than usual, and I'm hearing about sound companies growing and adding inventory, manufacturers adding staff, new distribution channels opening up, and numerous projects (large and small) underway or about to kick off.

You'll see some evidence of this in the Loading Dock section of this issue, which is much longer than usual but is still just the tip of the iceberg of all of the great stuff hitting the market. For even more, I encourage you to check out ProSoundWeb.

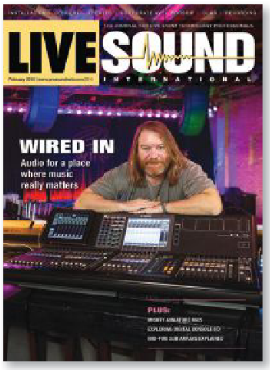
Also in this issue, Craig Leerman details rapidly expanding options in terms of digital console I/O (and other nifty developments), as well as applications of the growing number of miniature microphones on the market.

Meanwhile, Bob McCarthy provides a useful graphic-based presentation to explain end fire cardioid subwoofer arrays, while Ike Zimbel offers useful instruction on avoiding common wireless mistakes that can lead to interference.

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

Keith Clark

Keith Clark
Editor In Chief, Live Sound International/ProSoundWeb
kclark@livesoundint.com




**ON THE COVER**: Partner Paul Bolger at the Yamaha CL5 console, with Bose Pro RoomMatch arrays flying behind him, at Wire in Berwyn, IL. Coverage begins on page 22. *(Photo by David Kindler)*

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Publisher | **Kevin McPherson** | kmcpherson@ehpub.com
Editor-In-Chief | **Keith Clark** | kclark@livesoundint.com
Senior Contributing Editor | **Craig Leerman** | cleerman@livesoundint.com
Senior Technical Editor | **Ken DeLoria** | kdeloria@livesoundint.com
Church Sound Editor | **Mike Sessler** | msessler@livesoundint.com
Europe Editor | **Paul Watson** | pwatson@livesoundint.com
Technical Consultant | **Pat Brown** | pbrown@synaudcon.com

Art Director | **Katie Stockham** | kstockham@ehpub.com
Associate Art Director | **Dorian Gittlitz** | dgittlitz@ehpub.com

**ProSoundWeb.com**
Editor-In-Chief | **Keith Clark** | kclark@prosoundweb.com
Product Specialist | **Craig Leerman** | cleerman@prosoundweb.com
Webmaster | **Guy Caiola** | gcaiola@ehpub.com

**Chris Huff** | **Bob McCarthy** | **Ike Zimbel**
**Mark Frink** | **John Schauer**

**Live Sound International**
111 Speen Street, Suite 200
Framingham, MA 01701
Phone: 800.375.8015
www.livesoundint.com

**Jeff Turner** | Account Executive
415.455.8301 Fax: 801.640.1731
jturner@livesoundint.com

**Mark Shemet** | Associate Publisher Online, ProSoundWeb.com
603.532.4608 | Fax: 603.532.5855
mshemet@prosoundweb.com

**Manuela Rosengard** | Ad Production Director
508.663.1500 x226 | mrosengard@ehpub.com

**Jason Litchfield** | Ad Production Manager
508.663.1500 x252 | jlitchfield@ehpub.com

**Rachel Felson** | Jr. Production Designer
rfelson@ehpub.com

Circulation and Customer Service inquiries should be made to:
**Live Sound Customer Service**
EH Publishing
Phone: 800-375-8015, ext 294
(Outside the U.S.: 508.663.1500 x294)
Fax: 508.663.1599
customerservice@livesoundint.com
111 Speen Street, Suite 200
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Circulation & Subscriptions | circulation@livesoundint.com
Loading Dock Submissions | kclark@livesoundint.com
World Wide Web Inquiries | webmaster@livesoundint.com
Advertising Rate Information | adinfo@livesoundint.com

REPRINTS: **Erica Halloran**
508.663.1500 x265 | ehalloran@ehpub.com

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# LOADINGDOCK


## FaitalPRO HF108 ↑

A 1-inch-throat, high-output compression driver with a 1.7-inch (44 mm) voice coil and Ketone Polymer diaphragm, designed for both far- and near-field applications. A neodymium ring makes the unit very compact and light in weight (1.8 pounds), and it measures just 3.4 inches in diameter and 1.6 inches in depth. AES power handling is stated as 60 watts, with maximum power handling of 120 watts and sensitivity of 109 dB. Nominal impedance is 8 ohms, and minimum impedance is 6.8 ohms.

**www.faitalpro.com**

## Audio-Technica System 10 PRO Rack-Mount →

A digital wireless system operating in the 2.4 GHz range with a rack-mount chassis housing one or two receiver units that can be operated locally within the chassis or be removed and mounted remotely (up to 300 feet away) via Ethernet cable. This approach increases the versatility of the system while also enhancing wave propagation without needing an antenna distributor and corresponding cables. Up to five chassis (10 receivers) can be linked together using the included RJ12 cable, creating a multichannel system with the simultaneous use of up to 10 channels. An LCD dual display shows RF signal level, system ID, transmitter battery level, and link status for both channels. The system offers 24-bit/48 kHz wireless operation and three levels of diversity: frequency, time, and space. Included are the receiver chassis, one or two receiver units, one or two receiver mounting brackets, RJ12 cable, two rack mounting brackets, joining plate, AC adaptor, and one or two UniPak body pack and/or handheld transmitters.

**www.audio-technica.com**


## Allen & Heath Qu-Pac ↑

A free-standing or rack-mount mixer offering 16 mono inputs, 3 stereo inputs and 12 mix outputs on the rear panel. This can be expanded up to 38 in/8 out by connecting to Allen & Heath's family of remote AudioRacks over Cat-5. The mixer offers total recall of settings and preamps, multitrack recording to USB via Qu-Drive, a choice of personal monitoring, channel ducking, multichannel USB streaming and the iLive FX Library. The Qu-Pad iPad app provides wireless control of the mixer's key parameters and settings. A 5-inch high-resolution color touchscreen offers access to all functions, providing a simple interface for day-to-day operation and a backup in the event of Wi-Fi connection problems. As well as stereo recording or playback from a USB key, the built-in, 18-channel Qu-Drive can record and play back multitrack and stereo WAV files to a USB key or drive.

**www.allen-heath.com, www.americanmusicandsound.com**


## Radial Engineering Decoder ↑

A self-contained MS (mid-side) interface for creating stereo imaging when combining a stereo microphone with a second middle mic. It has three XLR inputs, with input-1 for a center mic and input-2(a) for a single output figure-8 mic or 2(b) for two cardioid mics. Individual on-off switches and level controls for all three inputs are located on the front panel. A recessed 48-volt phantom power switch can be activated to supply DC for condenser mics, and a high-pass filter can also be engaged to eliminate excessive low frequency rumble and improve stereo focus. When using two separate mics to create a figure-8 pattern, a 180-degree switch is engaged to reverse the polarity of mic 2(b). The enclosure is made of rugged 14-gauge steel, and a unique book-end design creates a protective zone around the front panel switches and potentiometers. **www.radialeng.com**

## PreSonus StudioLive AI Mix Systems →

Available in 48- and 64-channel frame sizes, Mix Systems consist of two cascaded StudioLive AI consoles, a joining bracket that locks the two together, a PreSonus PRM1 reference microphone, and a custom dust cover. Every channel is routed through the bus outputs of the master mixer and every global setting is controlled from the master to integrate the two cascaded consoles. Connecting each mixer to a wireless router with a standard Ethernet cable makes them ready for remote control from an iPad or iPhone, or wirelessly using the included USB WiFi LAN adapters. The StudioLive 64 AI has an 80 x 66 continuously bidirectional FireWire 800 recording interface, 26 mix buses (including 8 effects buses), and 558 discrete EQ and dynamics processors. The StudioLive 48AI has an onboard 64 x 50 FireWire 800 recording interface, 22 total mix buses (including 8 effects buses), and DSP processing on every channel and bus for a total of 438 EQ and dynamics processors. Both include the StudioLive Software Library. **www.presonus.com**

## Kaltman Creations CPArray ↑

An antenna for wireless microphone systems with a circular polarized pick-up pattern in the 470 MHz to 960 MHz range. In application, two CPArray antennas are mounted on a mic stand Tee bar, joined by a low-loss antenna combiner that offers the ability to selectively cover in opposing, off-set, and multi-elevation directions. This configuration provides the ability to focus reception in selected areas for maximum efficiency, and to avoid extraneous and interfering RF. The antennas are sold in a twin-pair configuration with the Tee bar for traditional "diversity" connections or with the combiner for multi-directional focusing. The antennas are painted theater black with a 50-ohm, low-loss BNC connection, measuring 6.7 x 6.2 x 1.6 inches and weighing 1.2 pounds. **www.KaltmanCreationsLLC.com**


## Waves Audio dbx 160

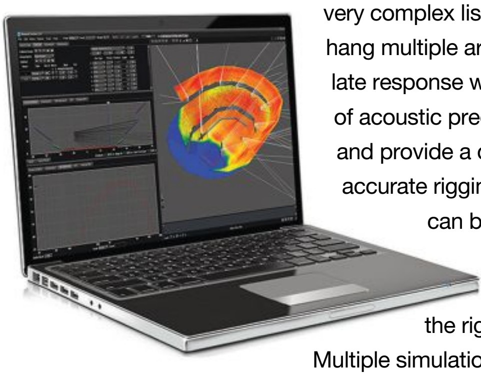
A plug-in created in collaboration with dbx and designed with advanced modeling to produce an authentic emulation of the 160 compressor that's been heavily utilized for decades, particularly in drum applications. It provides very clean sound with minimum THD, and also includes an MS matrix, mix and noise controls, and a stereo component (the original hardware was mono only). The plug-in is available in both Native and SoundGrid versions. **www.waves.com**

## JBL Professional SRX800 Series

Portable powered loudspeakers incorporating JBL transducers and built-in Crown DriveCore amplifiers as well as user-configurable DSP tuning capability. Models include the 12-inch 2-way SRX812P, 15-inch 2-way SRX812P, dual 15-inch 3-way SRX835P, 18-inch SRX818SP subwoofer and dual 18-inch SRX828SP sub. High frequencies are handled by waveguides that provide accurate horizontal and vertical pattern control with smooth frequency response over a wide area. The high-excursion woofers deliver bass with minimal dynamic compression. SRX800 loudspeakers and subwoofers are rated at between 135 dB and 141 dB, depending on model. Onboard DSP includes 20 parametric equalizers, 96 kHz FIR (Finite Impulse Response) filters for improved crossover tuning, delay adjustment, and a signal generator to aid in system calibration. All models also incorporate JBL Application Engineered preset tunings that facilitate fast system setup, and they're compatible with Harman HiQnet network communications protocol, enabling control via Audio Architect or JBL SRX Connect, a new iOS and Android app. The loudspeakers have M10 suspension points and standard 35 mm pole cups, and the rear panels offer an LCD screen and glow-in-the dark ink to facilitate setup on dark stages. **www.jblpro.com**

## Adamson Systems Blueprint AV

The company's multi-use predictive software suite that operates both in the 2D and 3D realm has been released via a licensing system that will allow end users to distribute a pre-determined number of copies to their teams of technicians. Designed to provide a user-defined level of detail, the software provides precision in any operating mode. Users can build a simple or very complex listening area, hang multiple arrays, simulate response with a variety of acoustic predictive tools, and provide a detailed and accurate rigging plot that can be exported to DXF files for transfer to the rigging team.



Multiple simulation options exist, including SPL measurements as well as standard 1/3-octave frequencies or a full-range average. Delay, directivity and virtual microphone responses can also be calculated. The software comes pre-loaded with acoustic data from Adamson's entire line array product range, with plans to add all point source loudspeakers in the near future. **www.adamsonsystems.com**

## Crown Audio XLi Series

The XLi3500, XLi2500, XLi1500 and XLi800 are rated to deliver 1,000, 500, 330 and 200 watts per channel respectively (into 8 ohms; 2,700, 1,500, 900 and 600 watts in bridged mode). All models have user-selectable input sensitivity to accommodate a variety of program sources that have low output signals, and they also include balanced XLR and unbalanced RCA inputs. XLi models can be operated in stereo, parallel or bridged-mono mode, providing binding post and speakON output connectors to facilitate use with virtually any type of passive loudspeakers. All have front panel level controls, signal presence, and clip and fault LED indicators. They're also designed to provide comprehensive protection against shorts, no-load conditions, and power on/off thumps, and in addition, they're shielded against radio frequency interference (RFI). **www.crownaudio.com**

## Shure PG ALTA

A microphone line with 14 models, including vocal and instrument mics, drum kits, and a studio kit. The PGA181 is a side-address condenser mic suited for acoustic and amplified instruments, vocals, and more. Two gooseneck condensers include the PGA98D for drums and the PGA98H for horns. Three drum and studio kit options all come with carrying cases and cabling, and two also include drum mounts. PG ALTA mics are available with various cable and connector options. All models can be purchased without cables, and all (except PGA27) are available with XLR cables. The PGA48 and PGA58 are available with XLR and QTR cables. **www.shure.com**



## VUE audiotechnik al-8SB

A flyable subwoofer designed for integration with the company's al-8 line array. It's rated to provide LF reproduction down below 35 Hz. The 18-inch long-throw transducer employs a 4-inch voice coil and dual spider for minimal power compression, housed in an enclosure that is the same height as two al-8 line array elements. This provides visual symmetry between flown arrays of al-8and al-8SB, as well optimal enclosure volume for the woofer. The enclosure is constructed of birch plywood coated in a 12-step Dura-Coat LX finish, and outfitted with recessed handles. Extensive interior bracing eliminates resonance. The 4-point symmetrical rigging allows the al-8SB to be flown in reverse fashion with other al-8SB subs to create a cardioid bass arrangement. VUE's new fly beam can suspend the al-8SB enclosures or full-range al-8 arrays. A flyable end-fire grid fosters assembly of end-fire arrays for improved directionality. Optimal performance is achieved when the al-8SB is used with one of the company's V Series systems engines. **www.vueaudio.com**

## D.A.S. Audio Aero 20A

A compact self-powered line array incorporating a new 12-inch woofer with light aluminum voice coil bonded to a fiberglass reinforced cone, an optimized magnet circuit, and a new suspension design. In addition, the woofer's venting design dissipates voice coil heat to foster a high thermal rating and low power compression. The neodymium magnet compression driver (3-inch voice coil) works with a wave-guide assembly optimized in conjunction with the driver and developed specifically for the Aero 20A. The Class D amplifier combines the power supply, output stage and connectors in a single chassis. The 2-channel (800 watts plus 400 watts) amp makes use of switch-mode technology. FIR filters have been used to provide optimized alignment for uniform coverage all the way down to the crossover point. The Aero 20A is DASnet capable, allowing for remote monitoring and control. Enclosures are constructed of birch plywood and finished with an ISO-flex coating. A captive rigging mechanism allows angle selection to be made while stacked on the transport dolly. **www.dasaudio.com**

## Countryman Associates I2

A low-profile microphone designed for a variety of instrument applications, available in four polar pattern configurations: omnidirectional, cardioid, hypercardioid, and bidirectional/figure-8. Frequency response is stated as 20 Hz to 20 kHz. A combination of small size and sonic precision makes the mic suitable for X-Y, M-S, and other techniques that rely on accurate coincident positioning and consistent patterns. Close miking techniques are aided by the company's new line of vibration-isolating magnetic instrument mounts. The I2 ships with 10 feet of black aramid-reinforced high-strength cable, terminated for wireless transmitters or XLR connections. It's also compatible with several hundred transmitters from dozens of wireless manufacturers. **www.countryman.com**

## Point Source Audio CM-i3 ➔

An intercom headset weighing two ounces that's available with varying male/female XLR connectors as well as a 3.5 mm option to support intercom apps for iPhones and iPads. All CM-i3 packages include the company's EM-3 in-earphones, microphone assembly, three sets of eartips, windscreen, and a custom case. Binaural functionality reduces noise and increases intelligibility. Earphones offer switching between three listening modes (left, right, or left and right) without having to remove or adjust the entire headset. The supplied three sizes of eartips lets users design a personal fit. The earphones are modular, allowing for on-site service replacement or swapping out with other limited varieties of in-earphones. Several CM-i3 models are available in varying terminations compatible for Clear-Com, Telex, RTS, and other intercom systems. **www.point-sourceaudio.com**

## BASSBOSS DV8

A self-powered loudspeaker incorporating dual 8-inch neodymium woofers and a 1.7-inch-diaphragm compression driver on an isophasic waveguide that fosters wide horizontal coverage of 120 degrees. Vertical coverage is a tight 15 degrees. Components are driven by a 1,500-watt amplifier, with DSP also onboard. The woofers have octagonal baskets so they can be mounted directly adjacent to each other, minimizing cabinet frontal area. The waveguide is mounted forward of the loudspeaker baffle and above the grille to minimize edge diffraction. The Baltic birch cabinet construction incorporates a 4-degree down-angle when deployed on a pole stand or resting on a flat surface, helping to ensure even coverage over the additional distance afforded by the tight vertical pattern and waveguide. **www.bassboss.com**

## Grund Audio GP Series

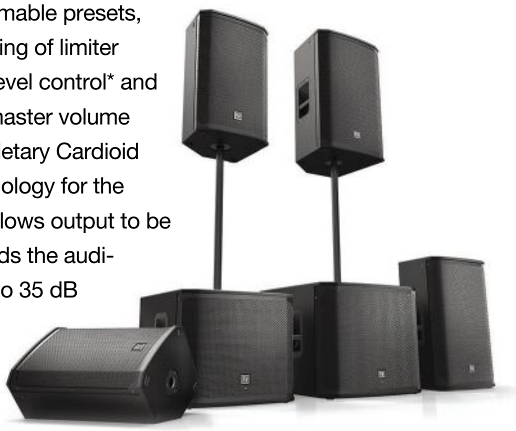
Four self-powered 2-way loudspeakers (GP-08A, GP-10A, GP-12A, and GP-15A) with a 1-inch compression driver mated with an 8-, 10-, 12-, or 15-inch weather-treated LF transducer in a reinforced, ribbed, 2-piece molded, low-flex enclosure. The onboard power amplifiers range from 200 watts to 700 watts of output. Included are XLR mic input, CD player, and line level XLR and 1/4-inch inputs, as well as individual volume controls for each input and master tone controls for output. In addition, they're outfitted with isolated outputs for connecting multiple enclosures. The injection molded enclosures incorporate M8 rigging points, a pole mount cup for use with floor stands (available separately), and floor monitor feet. **www.grundaudio.com**

## Celestion FTX

A range of cast aluminum, ferrite magnet coaxial loudspeaker drivers available in 12-inch, 8-inch, and 6.5-inch diameters. They offer combined LF and HF components powered by a "common magnet motor" assembly (where the same magnet is used for both LF and HF elements), enabling the voice coils and thus the acoustic centers of the two drivers to be brought closer together to improve signal coherence and time alignment. The use of a single magnet assembly also means lighter weight and a more compact profile. Polyimide film diaphragms provide greater HF power handling. A Sound Castle soft clamping assembly decreases diaphragm stress for reduced distortion and enhanced reliability. Both HF and LF voice coils are edge wound using lightweight copper or copper clad aluminum to increase barrel stiffness and enable a closer coil wire packing density, improving cooling and increasing motor strength. Demodulation rings minimize the effects of power compression and reduce harmonic and intermodulation distortion. **www.celestion.com**

## Electro-Voice EKX Series

A portable loudspeaker line comprised of eight models (four powered and four passive), including 12-inch and 15-inch two-way models as well as 15-inch and 18-inch subwoofers. All models incorporate EV-engineered loudspeaker, and powered models incorporate Class D amplifiers as well as signal processing and application presets. Full-range models have a Signal Synchronized Transducers (SST) waveguide design for consistent coverage. QuickSmartDSP includes processing, EV's single-knob user interface, and menu navigation via LCD. Easy setup is supplied via four presets (music, live, club, and speech*), sub/top system-match crossovers, 3-band EQ*, five user-programmable presets, visual monitoring of limiter status, input level control* and meters, and master volume control. Proprietary Cardioid Control Technology for the subwoofers allows output to be steered towards the audience with up to 35 dB reduction on stage when multiple subs are deployed (powered models only). Enclosures are made of 15 mm wood with EVCoat finish, and include hardware. (*Full-range models only.) **www.electrovoice.com**



## Meyer Sound CAL AVB

The company's column array loudspeakers have received certification by the AVnu Alliance, the industry consortium that certifies Audio Video Bridging (AVB) devices. The AVnu certification is given to devices that have implemented the IEEE AVB standards and passed AVnu Alliance's rigorous testing for interoperability and compliance. With the certification, CAL provides interoperability with AVnu-certified AVB devices from other vendors to simplify network implementation for the user, while also offering advantages of the open IEEE AVB standards, such as streamlining network infrastructure by combining audio signal transmission with system control and monitoring using cabling such as Cat-5e and Cat-6. CAL is offered in three models. **www.meyersound.com**

## AKG DMS800

A digital wireless microphone system that builds on the DMS700 V2. It offers two digital audio outputs for Dante and AES EBU, along with improved design and mechanics for the handheld transmitter. Additional components are a stationary receiver and a wireless body pack transmitter. The system offers two balanced XLR and two unbalanced jack connectors, as well as digital wireless transmission with low-cut filter, 3-band EQ, and dbx compressor and limiter. Mic heads are interchangeable on the handheld, with users offered a choice of the D5 WL1, D7 WL1 and C5 WL1. Up to 40 channels are available. Built-in Harman HiQnet network remote control and monitoring can be done from a PC via HiQnet Audio Architect software, Apple iPhone/iPad/iPod or Soundcraft Vi Series consoles. The DMS800 is specified as having a frequency range of up to 150 MHz, with 512-bit encryption ensuring protection of sensitive audio information. **www.akg.com/pro**


## dbx Professional DriveRack VENU360

A loudspeaker processor that's the successor to the DriveRack 260, offering additional features such as the latest advancements in dbx's AFS (Advanced Feedback Suppression) and AutoEQ algorithms (first introduced in the DriveRack PA2). Other input processing includes 31-band graphic EQ, 12-band parametric EQ with narrow-notch capabilities, dbx compression, subharmonic synthesis for enhanced LF impact, backline time delay, and noise gating. Multi-crossover configurations are possible with support for full-range, 2-way or 3-way operation (custom mono 4-way, 5-way or 6-way configurations are also available). Output processing is provided with 12-band AutoEQ, compression, automatic gain control, subharmonic synthesis, noise gating, tower delays (up to 1000 ms per output), 8-band parametric EQs (used for loudspeaker tunings), limiting and driver alignment delays. Connecting a dbx RTA-M measurement microphone (sold separately) to the front-panel RTA mic input allows proprietary Level Assist and AutoEQ algorithms to be utilized. **www.dbxpro.com**

TECH TOPIC

# Whoa, You Can't Do That!

## Common wireless mistakes that can lead to interference.

*by Ike Zimbel*

ONE OF THE MAIN CAUSES of RF (radio frequency) interference is intermodulation products created by our own wireless equipment. In this piece, I will outline some of the common setup and handling errors that contribute to this problem.

First up is increased noise floor and intermodulation (intermod or IM) products due to transmitters being in very close proximity. If you work with a single band and just put mics up on stands every day, you might not encounter this. But at festivals, broadcast events, churches and theatrical productions, there are situations where a number of active transmitters (i.e., microphones, belt packs) must be marshaled somewhere and handed out to talent at various intervals.

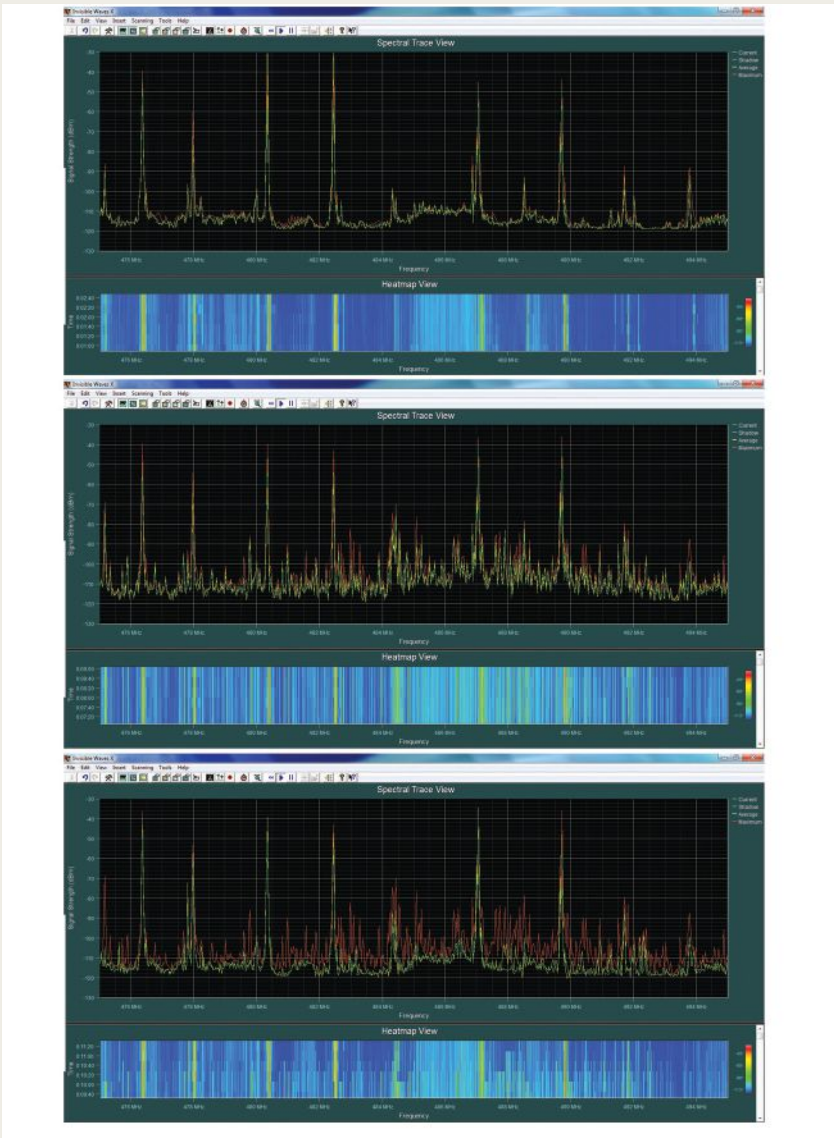



Figure 1

**Figure 1** shows the effects of handling six handheld wireless microphones in different ways. In the top image they're on stands, about 4 feet apart. In the second image, they're lying side-by-side on the fader bay of a console. In the third image, they're back on stands, with the red trace showing IM and increased noise floor (the "after" result).

We can clearly see that it goes from a very clean RF spectrum to a very noisy RF spectrum, and then back again. There are several ways to deal with this common problem:

- If laying the transmitters out on a table (or a road case), arrange them so that the antennas are pointing in opposite directions for each mic (one up, one down, etc.).
- If this is something you encounter regularly, invest in some aluminum loaf pans and place the transmitters in them, with the antenna end in each pan.
- Monitor engineers who keep their lead vocal mic and the spare on the console should place them either windscreen-to-windscreen, or both facing the same way, so the antenna end of one is facing the windscreen of the next.

The same problem occurs with in-ear monitoring systems, because out of necessity there are usually a number of transmitters in close proximity to each other. The trick here is to use antenna combiners, which have varying degrees of intermod suppression built into them.

**Figure 2** shows a rack of six IEM systems with whip antennas fitted (top image) and then routed through an antenna combiner (bottom). Note that

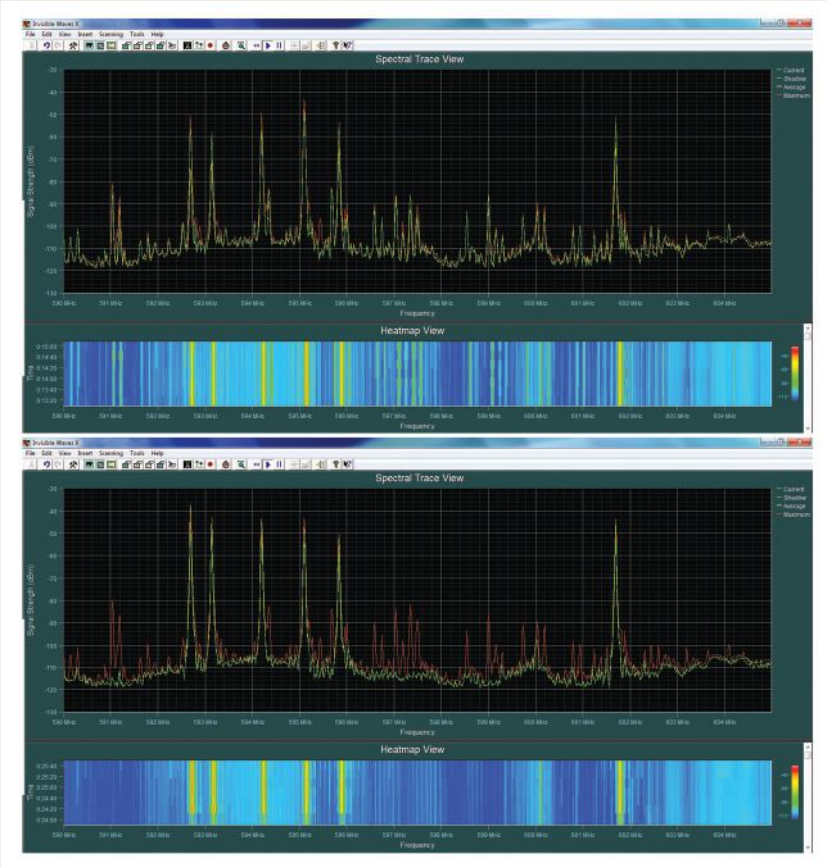



Figure 2

the bottom image is also the "after" shot and clearly shows all of the intermod products in red.

Note that five of the IEM frequencies shown are on top of a weak DTV (digital television) channel, channel 34 in this case. This accounts for the 6 MHz wide "hump" that the five channels are sitting on. I didn't program or coordinate any of the frequencies shown in these plots, but merely grabbed a couple of racks that had come back from a show and analyzed what was already programmed.

## CAREFUL ARRANGEMENT

Antenna placement is key – in particular, it's really important to keep transmit (Tx) and receive (Rx) antennas separated. This is because a Tx antenna in close proximity to an Rx antenna will desensitize it. This works in much the same way as the human eye or ear: if you're trying to look at a dim object like a distant star, and there's a bright light shining in your eyes, they're desensitized by the bright light. So an IEM antenna right next to a mic antenna is going to make that antenna less sensitive to a mic on the far side of the stage.

What to do? Assuming you're using cardioid directional antennas (a.k.a., "paddles," "shark fins," helical, circular-polar, etc.), place the Rx antennas behind the Tx antennas. This seems counterintuitive in that the receive antennas are being placed farther away (slightly) from the transmitters, but it puts them in the "null" area of the Tx antennas and therefore the area of least interference.

## POWER FOCUS

Time and again I see techs use the "high power" setting on any transmitter that happens to have one. But with RF, we usually don't benefit from having more power than needed. All that's needed is enough. It's analogous to digital audio in that going over the maximum input level (0 dBFS) does not do anything positive, just harsh clipping and overloading.

What more power does get us is more intermod products and more desensitized antennas, which leads to more interference and drop-outs as well as more reasons to think more power is needed to make the system

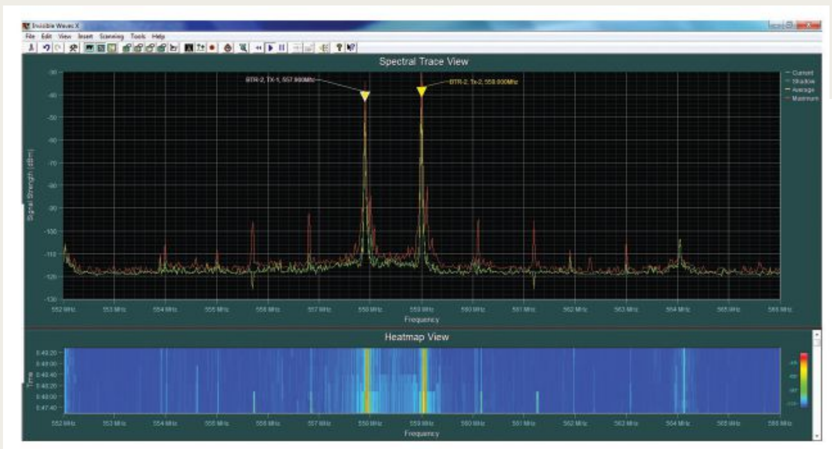



Figure 3

work. In my experience, the only time the high power setting on a transmitter is needed is when it's known for sure that the transmitter is going to be a long distance from the antennas. An example is an anthem mic set up in the middle of a football field.

**Figure 3** shows the before and after effects of having a Telex BTR-800 base station at "high" and "low" ("norm") power. Note the red lines showing the intermod products.

Note also that this is a from a unit that has been modified to have separate antenna outputs for Tx-A and Tx-B, and that these have in turn been put through an antenna combiner (which suppresses some of the IM products). A stock unit would generate much more IM in the high power mode, and more than what is shown here in the normal power mode.

## NOTHING TO GAIN

The same holds true for antenna gain, which exists solely to overcome signal losses in cables. Using higher gain on an antenna does not help to "pull" Tx signal out of the air. What it does do is make the antenna more sensitive to everything, including off-air DTV and low-level intermod products generated by the normal interaction between wireless transmitters. In other words, it raises the noise floor.

To avoid having to use excess antenna gain, utilize quality cable (this is a whole separate topic), keep cable runs as short as possible, and use a cable loss calculator to determine exactly how much gain is needed and then choose the closest gain setting on the antenna.

In summation:

- Improper handling and placement of transmitters can generate a considerable amount of IM products and raise the overall noise floor. This wastes precious bandwidth.
- Antenna placement, especially in respect to orientation of Rx and Tx antennas,  is critical for successful outcomes.
- High Tx power and high Rx gain settings are not helpful unless the specific situation absolutely calls for one or both. ■

IKE ZIMBEL *has worked in pro audio for 35-plus years, and during that time he has served as a wireless technician and coordinator, live engineer, studio technician, audio supervisor for TV broadcasts, and has also managed manufacturing and production companies. He runs Zimbel Audio Productions (zimbelaudio.com) in Toronto, specializing in wireless frequency coordination and equipment repair/modifications.*


A perspective of Wire, with Bose Pro RoomMatch arrays flying left and right.

# WIRED IN

## Music is the focus at a unique Chicagoland live venue.

***by Keith Clark, photos by David Kindler***

**The poet** Henry Wadsworth Longfellow famously said, "Music is the universal language of mankind." It's certainly a dialect shared by four lifelong Chicago-area musicians who have come together to create Wire, a venue for live music performance, education and production carved out of an old Teamsters hall in near-west suburban Berwyn, IL.

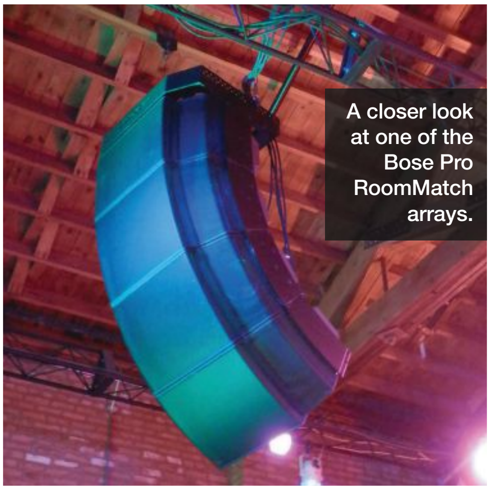
The four musicians include Chris Neville of Tributosaurus, which faithfully reproduces the recordings of classic rock groups (a different one each month); Paul Bolger of popular jam band Mr. Blotto; Tracy Dear of alt country group Waco Brothers, and Jon Smith, a noted recording engineer. A collective statement from the partners presents their vision: "Wire was an idea that came about several years ago, an idea that intends to reclaim music's heritage as a method of communicating – not something done in isolation, but something shared with other musicians, the surrounding community, and the world at large."

Wire's stage hosts a steady stream of live performances both diverse and eclectic: rock, blues, reggae, acoustic, jazz – and those that escape easy definitions. Space behind the stage has been carved out as the classrooms for Rock University, where students learn both music and production, and above that is the makings of a soon-to-be implemented recording studio. The venue is an important part of a thriving transformation of the arts community along Roosevelt Road in Berwyn, just a 20-minute drive from downtown Chicago and already the home of venerable FitzGerald's Nightclub, a staple of the music circuit celebrating its 35th year.

It's not surprising given the collective decades of performance experience that the partners are well-informed on the subject of sound reinforcement. Bolger, for example, notes that Mr. Blotto has owned its own PA for two decades, made up of premium components. Equally unsurprising is that they turned to fellow musicians for design and installation of Wire's house and monitor systems.

Neville's initial call went to TC Furlong, head of the sound company based in Lake Forest, IL since 1973 that bears his name and also a long-time player on the region's music circuit. Furlong in turn brought Brian O'Connell of his staff into the loop to serve as project manager, and he too is a veteran player on the Chicago scene, noting, "I've known Paul (Bolger) for a long time. Our bands have opened for each other over the years."

Wire partner Paul Bolger in the booth at the venue's Yamaha CL5 console that handles the house, monitors, and also serves as a teaching tool.



A closer look at one of the Bose Pro RoomMatch arrays.

## Developing The Space

The building that houses Wire actually began life as the Oak (and later the Oakwyn) Theater in 1934, largely presenting motion pictures. Later it was acquired by the local chapter of the International Brotherhood of Teamsters, which transformed it from a theater into a two-story office building. It remained that way, even after the Teamsters vacated it, until being purchased by the Wire partners.

The interior was essentially "gutted," revealing a beautiful, open space framed by exposed beige brick. After decades of inattention, the brick was restored to its original beauty, topped by a natural wood ceiling that adds to a warm, clean aesthetic. A large stage (27 feet wide and 15 feet deep) dominates the front of the room, and looking out from there, one sees a large audience area capable of accommodating about 400, with a bar behind that topped by a VIP area comprised of a couple of the old offices that were left in place.

Right away, it was obvious that the room's numerous hard, parallel surfaces (the floor is concrete) presented a primary demand of the house loudspeakers: pattern control. At the same time, the partners wanted a system with a highly musical signature. During the evaluation process, O'Connell tipped off Bolger and Neville to a demonstration being presented by Bose Professional of its RoomMatch loudspeaker arrays at another local theatre. They liked what they heard, and a direction was set.

"RoomMatch has a beautiful musical signature," Bolger says. "You can hear all of the subtleties, the true tonality of acoustic instruments, the 'whisper' of guitar strings. It's beautiful, clean, and intelligible."

O'Connell notes that a key aspect of the RoomMatch approach, with models available in numerous dispersion patterns, also met the absolute need for output that could be focused on the audience while keeping stray energy off of the room's many reflective surfaces. Further, with the TC Furlong team new to RoomMatch, Bose Professional stepped up to provide design support, particularly in terms of modeling via the company's Modeler software.

## Main Configuration

What resulted are flown left and right main full-range arrays flanking the Wire stage, each comprised (bottom to top) of one RM12020 module, two RM9020 modules, and a RM7010 module. The model numbers reveal the coverage patterns of the modules, with horizontal coverage wider at the bottom (120 by 20 degrees) and then tapering to more narrow (90 by 20 and then 70 by 20 degrees) as you move up the array. The result, according to Bolger, is uniform coverage of the room both front to back and side to side.

"Dispersion is very even, and there's distinct clarity in fully revealing the subtle detail of the music," he states. "You can walk anywhere in the room and the hi-hat doesn't go away, the vocals don't go away, the real character of the music doesn't go away."

The open nature of the space and a truss grid above the stage translated to easy selection of optimum locations to fly the arrays. Meanwhile, there wasn't enough height to fly the subwoofers with the arrays, so to keep the floor clear of obstructions, the system's two RMS218 (dual-18-inch) subs are positioned together beneath the center/front of the stage, and they're in a cardioid configuration topped by a judicious use of digital processing to help keep their output under control.


The dual Bose Pro RMS218 subwoofers are under the stage in a cardioid configuration.


Brian O'Connell at another Yamaha CL console in the TC Furlong shop.

All main system loudspeakers and the subs are driven by four Bose Professional PowerMatch PM8500N networked amplifiers and managed by a ControlSpace ESP-00 digital processor using four ESP I/O cards. These components are located backstage.

## Mix & I/O

The other primary component in the system is a Yamaha Commercial Audio CL5 digital console that's capable of handling both house and monitors. Fronting the custom sound booth constructed in one of the rear corners of the room, it provides plenty of capabilities for guest engineers. The console is linked via (Audinate) Dante networking to two Rio stage boxes, one to each side of the stage, that accommodate up to 64 inputs, with another eight inputs available on the console.

"The CL5 is a great choice for this application," O'Connell notes. "The technology is proven, engineers like to mix on it, there's plenty of capability, plus we'll be able to link it via Dante to the recording studio system when it's ready."

"This is an excellent console both sonically and operation-wise, with plenty of onboard facilities in terms of effects," Bolger adds. "Plus it's great for teaching our students. We like to get them behind the board and show them the cause and effect of what they're doing on stage as it relates to what's happening with sound in the house."

Currently a Dante run from the console feeds a multitrack recorder for capturing live performances, and Cat-5 is also in place to facilitate a Dante link to the main system rack in the near future. (That feed is analog for the time being.)

There really weren't many options with respect to the booth location – placing it centrally would have impeded crowd traffic flow while also occupying too much prime listening real estate. However, this difficulty is largely alleviated with the use of the console's StageMix app with an iPad that's kept at front of house, allowing engineers to dial in the mix from anywhere in the room.

## At The Stage

This applies to monitors as well, with the CL5 accommodating up to 16 mixes on stage. Several QSC KW122 (12-inch) active 2-way loudspeakers are available for artists, placed horizontally on their cabinet's monitor position, with a 15-inch KW152 provided for drum fill. They're also outfitted with EQ modes for switching between optimized settings with the press of a button on the back panel.

"I really like these boxes as monitors, both the 12s and 15s get it done," Bolger says. "I also like that you can adjust them if you want. It's incredible what compact loudspeakers can do these days."

"Actually, my band played at Wire a few months ago, and as a musician, I found the stage monitoring situation to be great," O'Connell adds.

The microphone package offers a variety of models from Shure (SM57 and SM58, several BETA mics, and three KSM137 condensers). Direct needs are met with Radial Engineering ProDi boxes as well as a ProD8 providing eight channels in a single rack-mount box.

All of the equipment hadn't arrived as the venue's grand opening date approached, so TC Furlong supplied loaner gear from its extensive rental stock to handle the situation until everything was delivered and installed. The company also provided training on the console and a technician for opening night.

QSC KW122 active loudspeakers serving as monitors on the Wire stage.

### Desired Result

Wire is convenient to expressways and public transportation, seeing notable success in attracting patrons from all over the Chicagoland area for live music several nights a week. It's already garnered a reputation among musicians as a place to play and be heard, and among patrons who appreciate the care that's gone into creating the venue.

"It sounds more like a concert hall than a club," says O'Connell. "This system sounds fantastic, to us and to the club's owners and to everyone who's been there."

"We set out to create a place where we'd want to play as artists, and that's led to a great result and what we see as a valuable addition to the area music scene," Bolger concludes. "We've worked carefully in how we've allocated our resources, and those choices have paid off for everyone involved." ■

KEITH CLARK *is editor in chief of* Live Sound International *and* ProSoundWeb.

# PHASE WAVELENGTHS

The end fire cardioid subwoofer array made visible.

***by Bob McCarthy***

THE END FIRE ARRAY is one of the most talked about – even if not the most often implemented – of the cardioid subwoofer arrays. It can be a challenge to wrap our heads around how we get the loudspeakers to play leap-frog in the forward direction and demolition derby on the back side.

Looking at the loudspeakers from a coverage angle point of view is a non-starter. They are omnidirectional. How do you add 360 degrees and 360 degrees?

## DRIVEN BY PHASE

The answer to the end fire behavior is not in the amplitude domain. All of the loudspeakers face the same way, and they overlap by a factor of 100 percent. The spatial picture of the level is only a small factor in the upper range of our interest, 125 Hz, where the loudspeaker has become somewhat directional.

End fire defined: *"A line array of emitters (in our case: loudspeakers) that are spaced and time-sequenced to provide in-phase addition on the forward side and out-of-phase rejection on the rear. The timing is set to compensate for the displacement between the sources in the forward direction. The most forward element is delayed the most, and sequentially less as we approach the last element."*

In our discussion here, we'll use four elements (more or less can be used – more makes it more directional) and space them 1 meter apart. The delay required will be multiples of 2.9 milliseconds to sync them in front. The physical setup is found in **Figure 1**.

The next thing to view is the individual radiation character of a single element in our frequency range of interest. The Meyer Sound MAPP plots are 1/12-octave, which might seem severe for an omnidirectional loudspeaker, but we must use high resolution to see the driving action of phase as we progress.

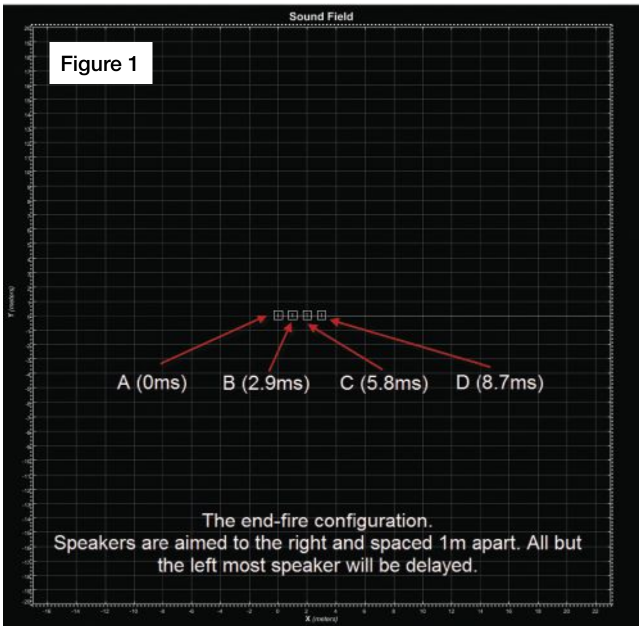



Figure 1

The end-fire configuration. Speakers are aimed to the right and spaced 1m apart. All but the left most speaker will be delayed.

Think about the fact that an octave resolution plot incorporates a 2:1 range of wavelength. In order for us to clearly see the driving effect of phase, we can't have a 2:1 slope factor in the data. What you see in **Figures 2**, **3** and **4** is the decreasing omnidirectional nature as we rise in frequency. This means that as frequency rises, we'll have both level and phase steering controls. At the bottom, only the phase lever will be operational.

## THE UNFINISHED PRODUCT

Next we look at what could have been. What would the response be if we spaced the elements in a 1-meter line (facing

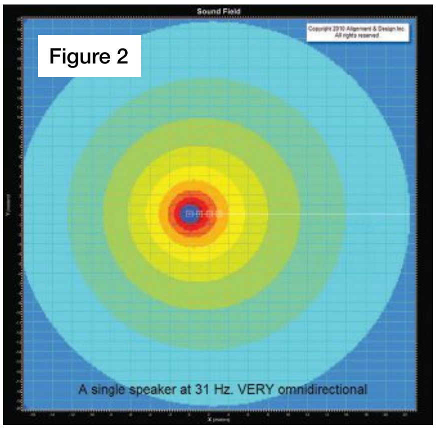



Figure 2

A single speaker at 31 Hz. VERY omnidirectional

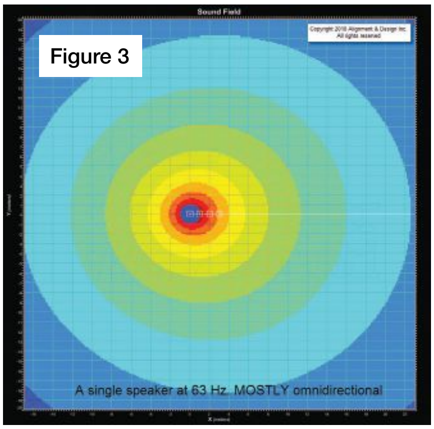



Figure 3

A single speaker at 63 Hz. MOSTLY omnidirectional

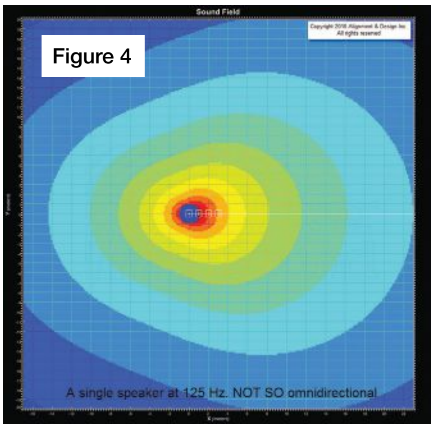



Figure 4

A single speaker at 125 Hz. NOT SO omnidirectional to the right) without the sequential delay taps? We could call it the end-no-fire array or the begin-fire. (You choose.)

The reason to do this is to see where the amplitude goes. The answer: it follows the phase. Let's look now at the 31 Hz response in **Figure 5a**. We see the phase wavelengths laid on to the empty MAPP plot. If the loudspeakers are 100 percent omnidirectional, this is all we need to know to see where the sound will go.

The location where the lines cross is where they're in phase. The fronts of the loudspeakers are pointed to the right, but by sleight of phase, we've magically moved the main lobe up and down.

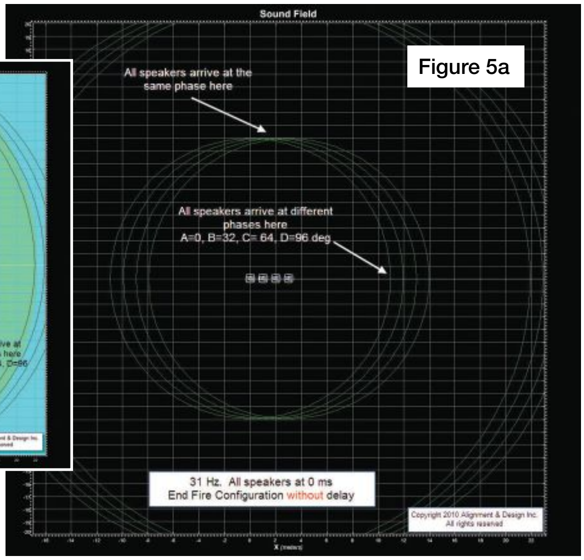


Figure 5a

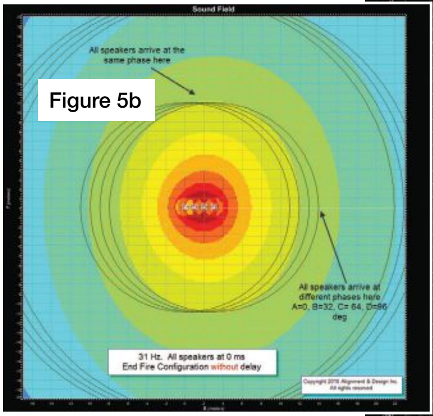


Figure 5b

**Figure 5b** shows the combined response of the four loudspeakers, and indeed the strongest sound is heading north and south. However, the steering is not extreme. Why? The answer, again, is in the phase. The loudspeakers are sequentially only 32 degrees apart (2.9 ms and 31 Hz). The response in the left and right directions doesn't fall all the way out of phase – no 180-degree type of differentials. Therefore, the relationship between the elements is more like a lack of cooperation than a serious fight.

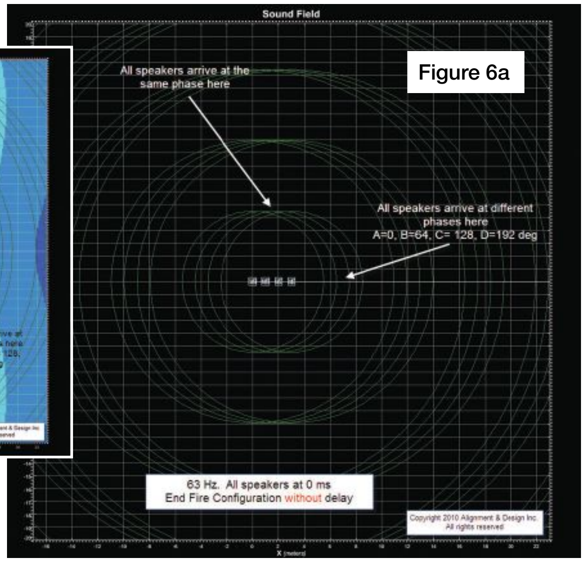


Figure 6a

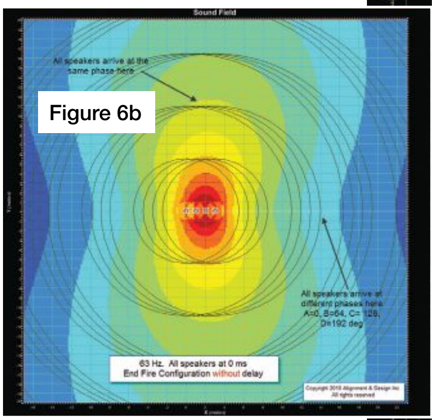


Figure 6b

As we rise is frequency to 63 Hz (**Figures 6a** and **6b**), the wavelength is cut in half. The displacement (1 m) is still the same but the phase shift is now 64 degrees per element. By the fourth element we've reached 192 degrees of phase shift. The first and fourth elements are in full conflict.

The result can be seen in the squeezing of the sides in favor of up and down, where all four elements are 100 percent in phase. As we move around the circle (from the top) we can see the lines gradually moving apart. This coincides with the gradual loss of level as we move to the sides.

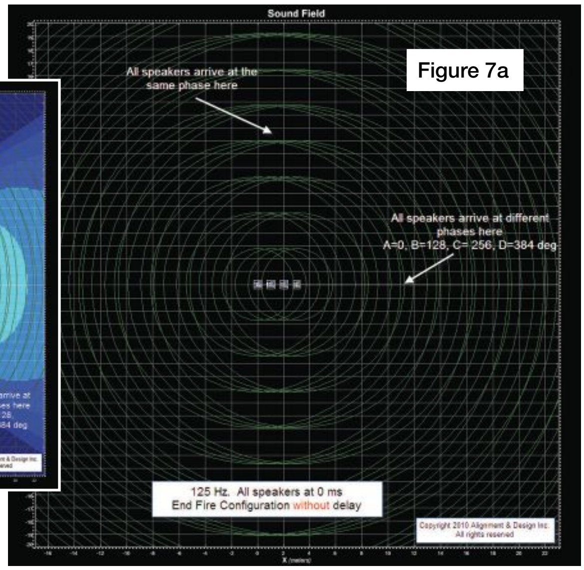


Figure 7a

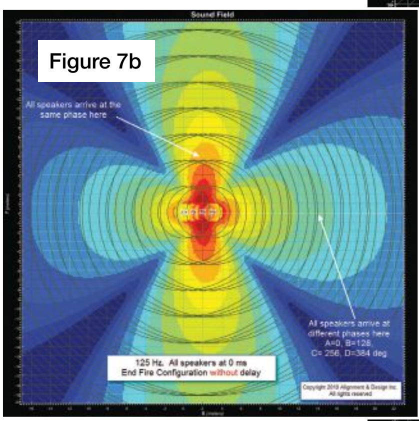
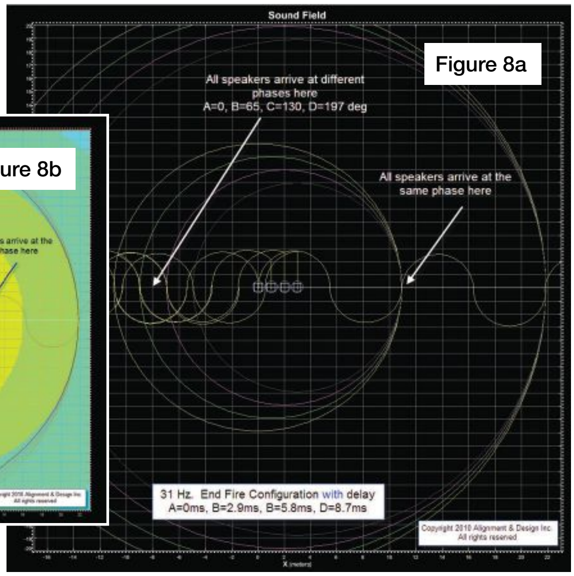
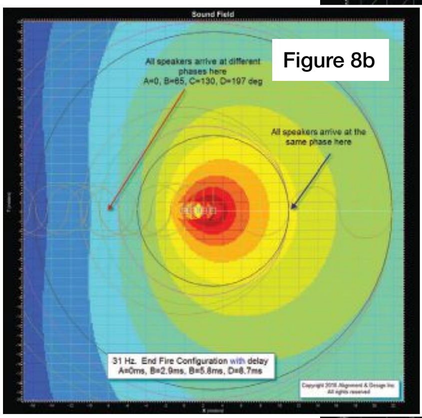


Figure 7b

Next up is 125 Hz (**Figures 7a** and **7b**). Once again the wavelengths shrink in half. Now we find ourselves with the four loudspeakers lapping each other on the sides and spreading out evenly in the corners. The full laps create addition on the sides – mixed with the loudspeakers that are not in phase – creating a push/pull situation. This is how side lobes are built. On the diagonals we see the deepest cancellations, due to four evenly spread arrivals.



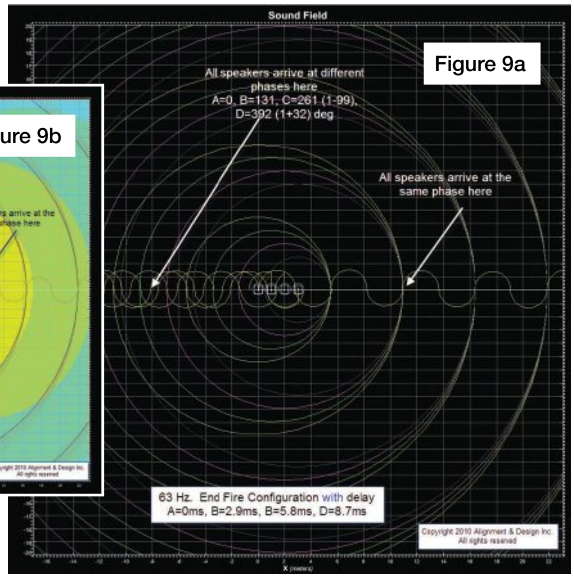



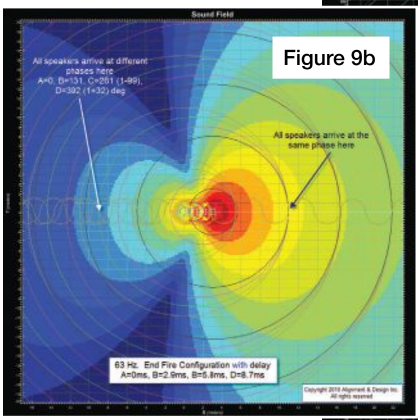

## WITH DELAY

Now let's add delay to the array. What happens is that part of the cycle elapses inside the electronics (the delay) and this means that the cycle completes its first turn at a shorter distance from the loudspeaker. From then on it turns again at the normal distance relative to its wavelength.

In our first look (**Figures 8a** and **8b**) we will see 31 Hz. The four loudspeakers all arrive in phase at the right side (in front of the loudspeakers). Each travels a different distance, but each has a different electronic head start. The result is that they all finish their first lap at the same spot and then go forward from there.

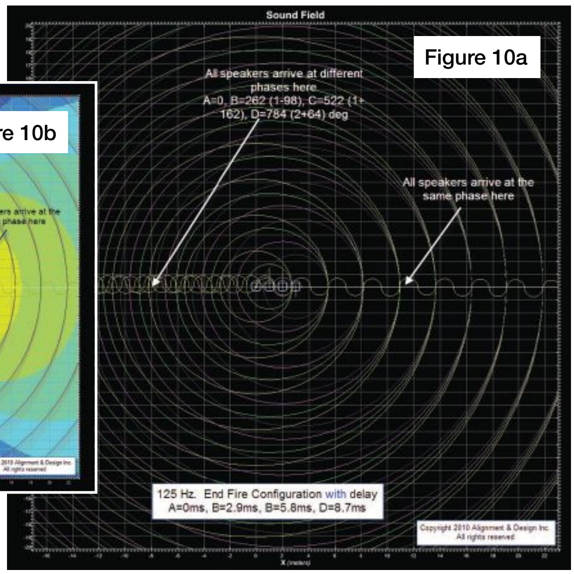
On the back side, the electronic head start still applies – but the physical head start is reversed. The result now is that the phase responses fall more quickly apart, such that loudspeakers A and D are 197 degrees apart. Big time cancellation.

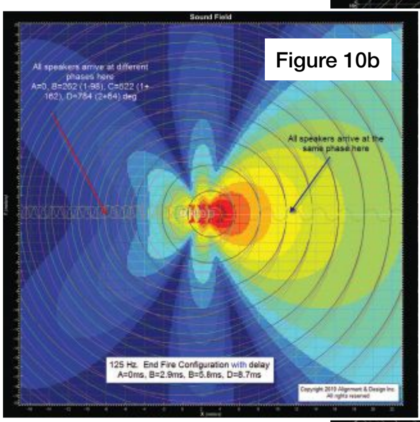



**Figures 9a** and **9b** shows 63 Hz. The same thing happens in front, but now the back side is spread by more than a full lap. The sides (top and bottom of the screen) gradually fall apart as we move from front to back, creating the incremental steering that concentrated energy forward and rejects it rearward. The mechanism is laid bare here – where the lines converge is where we see the energy, and where they spread we see blue.

By the time we reach 125 Hz (**Figures 10a** and **10b**), we're turning multiple laps on the back side and even on the sides (hence the side lobes). There's also a small component of directionality of the loudspeakers here.




Hopefully this graphics-based presentation helps clarify some of the mysteries of the end fire sub array. ■

BOB MCCARTHY *has been designing and tuning sound systems for over 30 years. His book* Sound Systems: Design and Optimization *is available at Focal Press (www.focalpress.com). He lives in NYC and is the director of system optimization for Meyer Sound.*

Allen & Heath Qu-Pac

# THE INS & OUTS

## Detailing the latest digital console I/O.

*by Craig Leerman*

WHEN SHOPPING TO ADD a new console to my company's inventory, I look at three primary aspects: reliability, sonic quality, and routing options. Most consoles from the major manufacturers are reliable and sound great, so the choice often comes down to the inputs, outputs, networking and overall routing capabilities.

There needs to be the required number of inputs necessary to handle most/all of the gigs my company works, and further, we want extensive routing options and outputs to both send feeds everywhere and patch things easily. The corporate events that are our specialty require a multitude of output sends, including feeds for recording, video world, podium and backstage monitors, dressing rooms and show intercoms, overflow and/or breakout rooms, and of course, feeds to the main PA, delay zones, front fills and possibly tie-ins to the house and lobby loudspeakers.

On smaller budgeted shows, the front of house console might also have to supply monitor and stage fill mixes for the entertainers onstage. Live remotes add a few more sends that could include splitting off certain inputs or crafting sub mixes to send to a broadcast truck. It's surprising that even small gigs can have a lot of routing requirements, including stage and in-ear monitors, aux fed subs, a feed to a videographer, and again, the main system.

And it's just not outputs that need to go everywhere. Inputs can be located on stage, off stage, backstage, in another room, in a remote truck, or right at the console. The ability to patch inputs has taken on increasing importance, even when a gig doesn't appear to be all that complicated at first glance. Another concern can be a console's networking ability, and how it can connect to various digital network protocols as well as analog systems.

My days of running large, heavy analog multi-core splitter snakes are coming to an end, replaced by networked audio over coax, fiber or Cat cables. Even many smaller consoles feature the increasing capability to interface with a variety of network protocols like Dante, MADI and/or the manufacturers' own protocol. Networking lets us to locate stage boxes anywhere without having to run obtrusive copper snakes, and it also allows the console to interface with additional consoles and recording devices.

Remote live access is another item that has been added to many smaller consoles/mixers. Being able to walk around a venue and mix the show on an iPad or stand onstage next to a performer during sound check and dial in their wedges has made my job easier. Another benefit of wirelessly accessing the console lets us place it at the stage and not even run a snake at all.

An increasing number of models also allow wireless access via tablet or phone so that performers can adjust their own stage mixes. This can save a ton of money for a band or church that can't afford a stand-alone personal monitoring system, and it can also save engineers and techs a lot of time and hassle. With so many new models and technologies hitting the market recently, let's take a look connectivity capabilities.

Compact rack-mount consoles/mixers are becoming quite popular and **Allen & Heath** unveiled a new one at last month's NAMM show. Qu-Pac can be used on a tabletop or mounted in a rack, with full control of the mixer from the front-mounted touch screen as well as with wireless iPad and iPhone remotes. The 16 onboard XLR mic and TRS line inputs can be expanded up to 38 inputs via the Allen & Heath dSNAKE network, and the unit also offers a 32 × 32 USB audio interface, 2 stereo TRS input channels, 12 mix buses and AES output. A cool feature is the ducking circuit, which can come in handy for making announcements over music or allowing one person's mic to override others at a meeting.

Also new at NAMM was the **Soundcraft** Ui Series of compact mixers that can be placed anywhere, like a stage box. (The larger unit can also be rack-mounted.) Both models have built-in Wi-Fi and the ability to be controlled by any connected device via a standard web browser, including iOS, Android, Windows, Mac OS, and Linux devices. The smaller Ui 12 has 4 XLR inputs, 4 XLR/TRS combo inputs, dual 1/4-inch line inputs, stereo RCA line inputs, 2 XLR aux outputs and a pair of main XLR and 1/4-inch outputs. The larger Ui 16 adds 8 XLR/TRS combo jacks to 4 XLR inputs and 4 aux XLR outputs.

Staying in the rack-mount genre, **PreSonus** recently added a couple of new compact models to its StudioLive mixer lineup. The RM16AI and the RM32AI offer the same features, with the only differences being the number of channel inputs and mix outputs. Both are operated using UC Surface control software for Mac, Windows, and iPad. The RM16AI has 16 XLR inputs, 8 XLR aux buses, and 3 XLR main outputs for LCR. Meanwhile the RM32AI offers 32 XLR inputs, 16 XLR buses and 3 XLR main outputs. Both also provide stereo RCA inputs and an option card slot. A S/PDIF digital output option card, as well as FireWire S800 and Ethernet cards, are available now, with Thunderbolt, Dante and

Soundcraft Ui 16

AVB cards announced. Using the same one-click recording software as the rest of the StudioLive line, these new units foster easy virtual sound checks.

Let's shift gears to larger models. **Yamaha Commercial Audio** just introduced a new flagship console, the RIVAGE PM10. The control surface offers 8 analog inputs and outputs, 4 AES inputs and outputs, and a pair of MY option card slots. The DSP-R10 engine provides an additional 2 MY slots along with 4 HY slots (TWINLANe and Dante). The main connectivity is via RPio622 stage boxes, which can provide up to 96 mic preamps per rack along with 2 HY slots. I'm also anxious to hear the newly-developed RY16-ML-SILK hybrid mic preamp with digital modeling based on Rupert Neve Designs transformer circuitry, with users able to

PreSonus RM32AI

choose between a transparent audio path or one enhanced by the Silk processing, which provides a bit of color and character to each input.

The new mc²36 is an "all-in-one" console from **Lawo**, offering 32 mic/line inputs, 32 line outputs, 8 digital AES3 inputs, 8 digital AES3 outputs, MADI and 3 RAVENNA network ports that can provide additional connectivity for up to 384 external inputs and outputs. Up to three mc² compact I/O stage boxes can be connected, each with 32 mic/line inputs, 32 line outputs, 8 AES inputs, 8 AES outs, and MADI. Also of note, the console has 21.5-inch HD

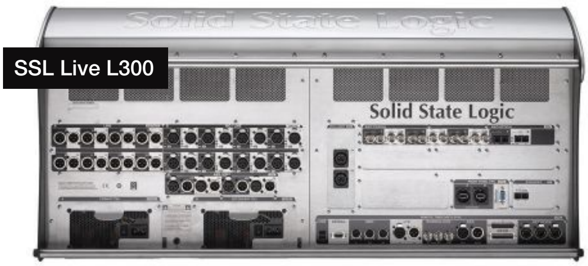
Yamaha RIVAGE PM10

touch screens and touch-sensitive color-illuminated rotary encoders, helping the operator keep track of mix functions.

The new **Roland Professional A/V** M-5000 console offers the company's proprietary Open High Resolution Configurable Architecture (OHRCA), which basically means that the internal mix architecture is not fixed and can be freely defined within a range of up to 128 input/output channels/buses. Onboard connections include 16 XLR inputs, 16 XLR outputs, 2 AES inputs, 2 AES outputs and REAC networking ports. Two option card slots allow interfacing with Dante, MADI, additional REAC streams and Waves SoundGrid. Additional connectivity can be had via Roland digital snake boxes, including the latest, the S-2416, offering 24 XLR inputs and 16 XLR outputs. A highly readable, well-defined 12-inch color control screen, color changing encoders, and wireless iPad control are very operatory friendly.

**DiGiCo** has a new version of its popular compact mixer, the SD11i, which can be used as a desktop unit or rack-mounted. All 32 mix channels are now Flexi channels (configurable as stereo or mono). The SD11i offers 16 XLR inputs, 8 XLR outputs, AES input and output, 3 USB ports, MADI and a D-Rack interface so stage boxes can be interfaced to gain additional inputs. Waves integration is also an option. In addition, DiGiCo recently released the SD app for iPad that can control all of the mix parameters for the entire SD line.

Just a couple of months ago, a second model joined the **Solid State Logic** Live Series, the more compact Live L300. While smaller than the original L500, it still offers 128 mix paths (96 fully processed, 32 dry) and up to 568 inputs and outputs. Onboard there are 16 XLR inputs, 16 XLR outputs, 4 pairs of AES ports and 8 MADI ports (4 redundant pairs). Optional stage boxes can be used to increase the input count via analog, AES or MADI inputs. SSL has a nifty network called Blacklight II, a high bandwidth multiplexed MADI approach that can be used to reduce the number of interconnecting cables. Blacklight II carries 256 (at 96 kHz) audio signals, equivalent to 8 MADI connections, bi-directionally down a single multimode fiber optic cable.



SSL Live L300

**Midas** recently added the 40-input M32R desktop/rack mixer to the M Series. It has 16 XLR inputs, 8 XLR bus outputs, 6 (six) 1/4-inch aux inputs and outputs (and a pair of RCA jacks also on aux 5/6), AES50 and Ultranet network ports, and USB interface port. Additional input counts and routing can be had by adding stage boxes, including the newer DL32 (32 XLR inputs and 16 XLR outputs) and DL16 (16 XLR inputs and 8 XLR outputs). Of note is that the DL16 offers dual AES50 ports so you can cascade a pair of them with no merger or router unit required.

Roland Professional A/V M-5000

**Mackie** just came out with the DL32R rack-mount mixer that's wirelessly controlled via iPad and iPhone. The unit has 24 XLR inputs, 8 XLR/TRS combo inputs, 14 XLR bus outputs, and a stereo AES output. There's an option card slot, and Mackie has just announced that a Dante card is now available. A neat feature is that USB ports are also provided for feeding audio to a computer DAW or directly into a hard drive, with the Master Fader software providing the recording and playback controls. *(For more about the DL32R, see my Road Test review on page 52 of this issue.)*

**QSC** recently entered the mixer market with the TouchMix line, comprised of two very compact models. The TouchMix-16 has 12 XLR and 4 XLR/TRS combo inputs, 2 stereo TRS inputs, 6 aux XLR outputs, 2 stereo aux TRS outputs, and main LR XLR outputs. The TouchMix-8 (now shipping, by the way) offers 4 XLR and 4 XLR combo inputs, 2 stereo inputs, 4 aux XLR outputs, 1 stereo aux TRS, and main XLR outputs. Both models sport a color touch screen for control instead of faders, and they can also be operated wirelessly via an iPad or iPhone. A new firmware update (2.0) provides added functionality, including support for password-protected, multi-level security access, expanded Wi-Fi options (including wired connection to an infrastructure router), and programmability of user buttons.

**Behringer** just added a desktop-style and three compact stage box-style mixers to the X AIR digital lineup. The new X AIR X18, XR18, XR16 and XR12 incorporate an integrated Wi-Fi module and are controlled using X AIR software for both iPad and Android tablets. The XR18 has 16 XLR/TRS combo inputs and 8 XLR bus outputs, while the XR16 offers 8 XLR combo inputs, 8 (eight) 1/4-inch line inputs and 6 XLR bus outputs. The XR12 provides 4 XLR combo connectors and 8 (eight) 1/4-inch line inputs, plus 1/4-inch aux outputs and main output XLRs. All models have a USB connector for file storage or uncompressed stereo WAV recording and playback. A future planned firmware update will add Dugan auto mixing to the feature set. ■

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

# STREAMLINED EFFICIENCY

## Getting started with Smaart v.7 Di analysis software.

***by Mark Frink***

RELEASED LAST YEAR, Rational Acoustics Smaart v.7 Di (dual-channel interface) is a streamlined (and less expensive) 2-channel version of its industry-standard Smaart v.7 multichannel analysis software. Yet while v.7 Di is a simplified version, it still uses the same powerful and efficient Spectrum and Transfer Function measurement engines as v.7.

Smaart is a Fast Fourier transform (FFT) measurement system that transforms "Time Domain" audio signals into frequency domain "Spectrograph" response displays when making single-channel measurements – great as a visual reference of a measurement microphone or the output of a cue bus being monitored.

The power of FFT is in its 2-channel measurements, typically comparing a signal sent to a loudspeaker with the sound coming out of it, in order to show its frequency response. Further, you can make these measurements using almost any source material that is "dense" enough for the signal, not just pink noise.

### SEEING IT

My "light bulb moment" with Smaart came in 1996 at a multi-band festival. Placing my Pentium PC laptop on the drive rack and simply taking a board feed and putting up a measurement mic, I could quickly and easily identify and tame a few rough spots by tweaking the main graphic EQ while the band before mine was playing on stage. No "hoo-hoo, ha-ha, test one-two" into an SM58 board mic needed.

As a monitor engineer, I could line up all of the wedges-du-jour and quickly take a picture (capture a trace) of each one's frequency and phase response to find the problem child in the bunch. There's almost always one bad apple: someone replaced a driver and either used a substitute model or got red and black confused.

At front of house, essential daily 2-channel measurements include timing the front fills and delays with the main loudspeakers by subtracting their arrival times, as well as aligning subwoofers to the mains.

### REMOVING COMPLEXITY

The Smaart v.7 Di user interface (UI) is functionally similar to SmaartLive, a previous version on which many of today's power users first learned over a decade ago. For sound techs embarking down the path of FFT measurement, v.7 Di's simpler user experience can flatten the learning curve and save money – and can later be upgraded to a full v.7 license.

New users get to learn the same controls and commands without the added complexity of the multichannel interface. Both v.7 versions use the same Spectrum and Transfer Function data formats, and both share the same on-screen controls and short-cut keystrokes, so moving back and forth between versions is easy.

All critical Spectrum and Transfer Function controls are visible simultaneously on the v.7 Di UI. Data is presented in what Rational Acoustics calls a "concise, single measurement pair" configuration that's accessible from top level of the UI. This includes two Live Spectrum engines (one for each input channel), two channels of broad-band metering (one for each input channel), and one Live Transfer Function engine.

v.7 Di doesn't include the full version's Acoustic Tools or a separate impulse response (IR) mode. Time domain measurement capabilities simply reside in the transfer function's Live

IR display. And since it's streamlined, v.7 Di has the ability to only measure one transfer function at a time.

## CAPTURE POINT

Besides the software, the minimum needed for making 2-channel acoustical measurements with Smaart is a computer interface and a measurement mic, plus cables to connect computer, mic and interface to each other and a console. Most professional audio systems employ balanced XLR connections, though a TRS adaptor can help occasionally. FireWire has faded and today most interfaces use USB, so a USB cable and a couple of XLRs is all that's usually needed, cable-wise. Most USB interfaces get power from the computer, so you won't even need the wall-wart.

The Rational Acoustics RTA-420 is an effective yet inexpensive measurement mic, similar to the RTA-M sold by dbx as an accessory for the DriveRack PA and 260 processors. You might not use it in a laboratory or a studio, but it's a quality unit that's "plenty good enough for rock 'n' roll."

As you advance in your study and practice of measurement, there might be a desire to upgrade, with plenty of measurement mics available that range from a few hundred to a couple thousand dollars. I own an Audix TM1 and an ISEMcon EMX-7150, which are both rugged workhorse tour standards and cost about $300.

## RANGE OF INTERFACES

The interface is a computer peripheral device that acquires and converts analog audio signals to digital for a PC or Mac, and most work with both, though PCs require a "driver file" that Macs do not. A basic interface for 2-channel measurements needs at least one XLR-input mic preamp with phantom power and one line level input, though an interface with two XLR "combo" inputs that also accept TRS connections helps avoid adaptors.

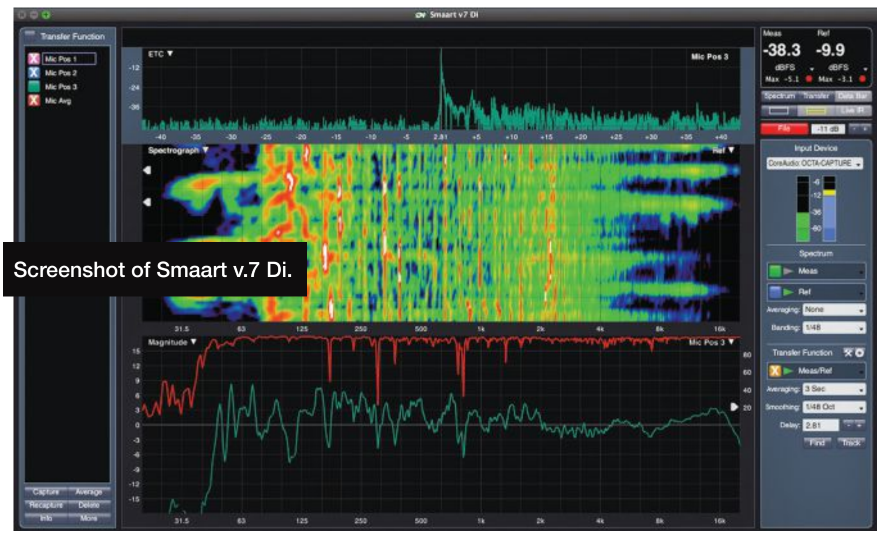
If you also want to use your interface for recording when not using Smaart, think about getting an interface with two mic preamps. In addition, having two line inputs allows recording stereo room mics plus a stereo board feed at the same time.

Don't let indecision about a computer interface stop you from getting started. There are a wide variety of units available, starting at a price of about a hundred bucks. Almost everyone makes one. Just be sure to check for compatibility. (And for solid advice and recommendations, stop by the Rational Acoustics and/or ProSoundWeb forums.)

A PreSonus AudioBox USB interface (2 in/2 out) can be had for about $100, or for a few dollars more, there's the Focusrite Scarlett 2i2 interface (also 2 in/2 out) that's become popular on many tours. *(There's one hiding on the cover photo of the September 2014 issue of* LSI. *– Editor)*


Rational Acoustics RTA-420 measurement mic.

For power users, the Rational Acoustics Smaart I-O is a measurement grade 2-channel USB interface designed and built specifically for use with Smaart v7. It provides SPL calibration as well as software-controlled gain in 1 dB steps. And multiple Smaart I-Os can be combined for multichannel measurements.



Screenshot of Smaart v.7 Di.

## PLENTY OF OPTIONS

Smaart v.7 runs under Windows XP or newer (including 8.1) and Mac OS X 10.5 and newer, with a 2 GHz dual-core computer with 2 GB of RAM recommended. But because v.7 Di only runs two transfer engines, I've found that it works with some older computers, like my first generation single-core black MacBook.

v.7 Di is available at two-thirds the cost ($595) of v.7 ($895). Both provide two installations ("seats") per license, and can be purchased at the Rational Acoustics online store (as can the RTA-420 mic, Smaart I-O interface, and many other accessories). Registered v.7 users can also purchase a special single-install license of v.7 Di at half price. Similarly, v.7 Di owners can buy a special single-install license of v.7 for half price, using the new "license add-on" purchase feature within their online license management accounts. For those still using v.6 who prefer a 2-channel to


PreSonus AudioBox USB, Focusrite Scarlett 2i2, and Smaart I-O interfaces.

a multichannel system, upgrading to v.7 Di provides an increase in responsiveness and features at a cost of $300.

Rational Acoustics offers a 1-day Smaart Basics class for $250 that provides an introduction to the v.7 Di user interface and covers basic RTA, Spectrograph and Transfer Function measurements. It also offers attendees a 25 percent discount for a Smaart v.7 Di license. In addition, a free 30-day, one-time demo version with minor functionality obstructions is available for download if you want to try out your new computer interface and measurement mic to see how they work with your computer.

Smaart (short for Sound Measurement Acoustical Analysis Real Time, by the way) was inducted into the TEC Hall of Fame at last month's NAMM show at the relatively young age of 20, having won it's first TEC Award in 1998. Its introduction at the New York AES show in 1995 has been called the dawn of the modern era of live sound measurement. It's an industry standard and the one software tool that every pro audio technician should own. ■

MARK FRINK *(livesound@markfrink.com) learned how to use Smaart on his first kd lang tour when Jamie Anderson of Rational Acoustics was the system engineer. (Thanks Jamie.)*

# NOT THE SAME

Factors in capturing and presenting snare drum. *by Chris Huff*

I CREATED A FROG. It wasn't intentional. Naturally, I'm not talking about a real frog, but look at the photo that opens this article. You'll never read a mixing book that says, "Make the snare's EQ curve look like a frog in water." (If you do, stop immediately and back away.)

Seriously, when it comes to snare mixing, the last place (literally) you want to be is behind the mixer. With that in mind, here are three primary factors I've identified in getting a good snare drum sound.

## 1: THE DRUM

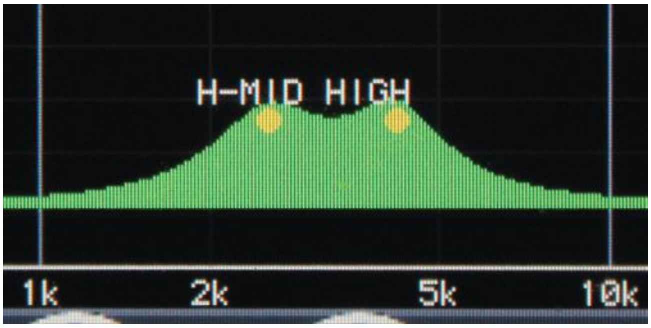
Snare drums don't all sound the same, just like all acoustic guitars don't sound the same. Even with a house drum kit, a drummer might bring his own snare because of it's sound. Know that each snare has a unique sound. This is the baseline sound for the mix. Use the same mic and the same EQ settings with two different snares, and you'll get two different results.

Consider these three different snares shown in **Figure 1**. Just by looking at them, you can almost hear the tonal differences. Material composite, drum size, drum head skin, all of these are factors. Even tuning makes a difference. Snares can be tuned to match whatever the drum tuner decides is to his liking. To generalize, there can be a low or high tuning (true for any drum).

Stand near the drum kit while the drummer plays the snare. This is the sound you'll be mixing with – not against. Don't try making it sound like something its not.

## 2: THE MICROPHONE(S)

A mic should be paired with an instrument and so it is with miking the snare. The Shure SM57 pairs great with a snare drum because of it's polar pattern and frequency response.




I polled some techs and their pairings include the Telefunken M80, Heil PR22, Heil PR28, DPA 4099, and the Granelli Audio Labs G5790, a modified Shure SM57 designed for tight spots. And don't think mic designs are the same (**Figure 2**).

Photos are nice but let's get real – we need to look at specifics. They have different polar patterns, different sensitivity, and they don't have to all be dynamic mics. For example, a Heil PR31 BW is a dynamic while the DPA 4099 is a condenser.

While several characteristics can make a big difference in how a mic treats sound, frequency response is a factor never to be overlooked. It alters the tonal characteristics of the snare drum. Take just one snare drum from above, like the Pearl Chad Smith Signature, and apply three different mics – the result is three different sounds. And we haven't even touched the EQ.

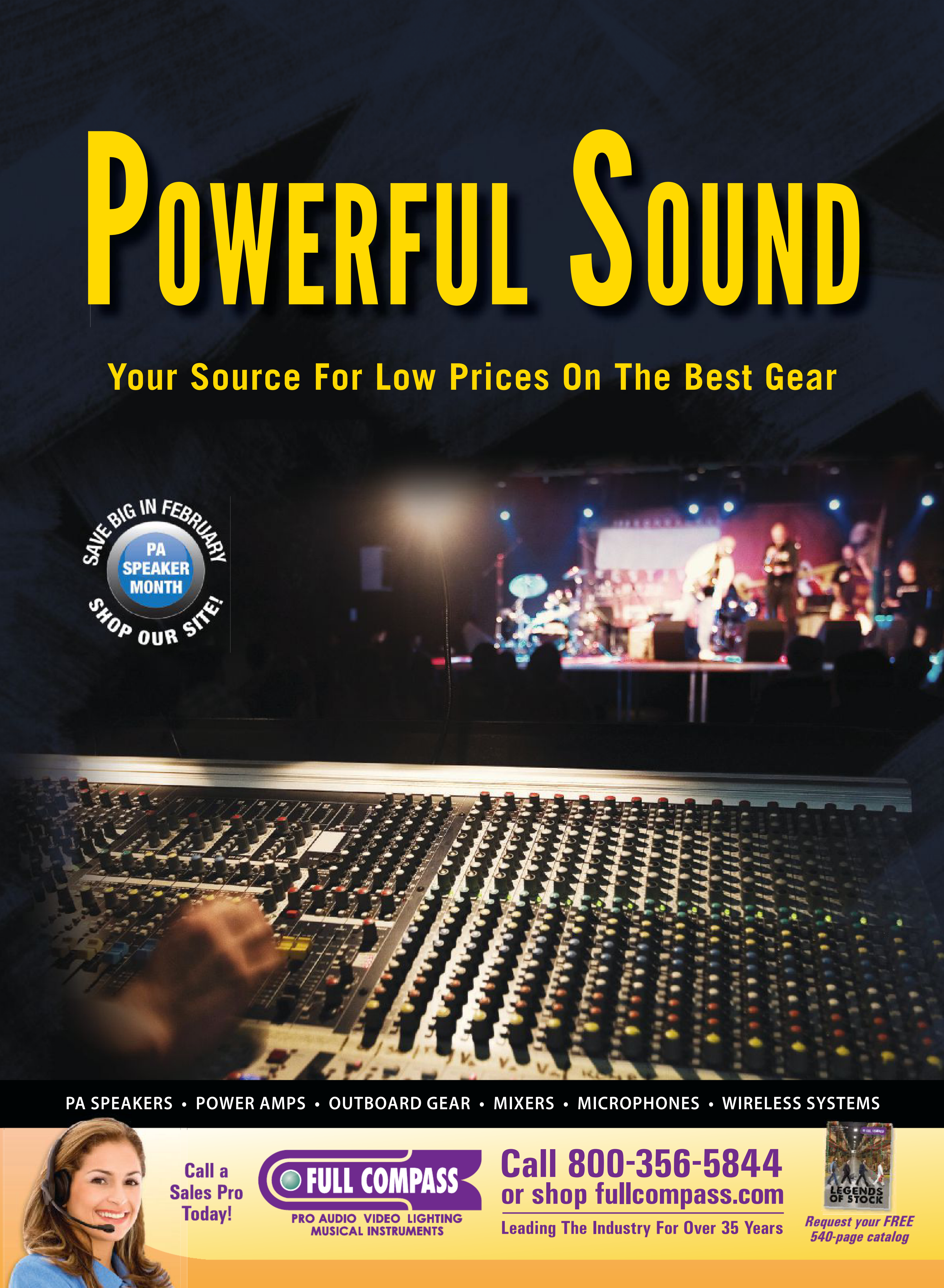
For comparison, **Figure 3** offers the frequency response charts for the SM57, PR31 BW, and 4099. (Note the charts with multiple lines are showing the differing frequency responses when not on-axis with the sound source).

In addition, snares can be (and are often) miked both over and under the drum. Here are a few combinations my tech friends recently sent me: Shure BETA57 over, Shure SM81 under; Audix i5s both over and under; Sennheiser MD421 over, Heil PR31 BW under; Audix i5 over, SM57 under; Heil PR22 over, Sennheiser e904 under.

## 3: THE EQ

Equalization should only happen after we listen to the natural tone of the snare and consider the mic(s) we're pairing with

Figure 1: Left to right, PDP Blackout Maple, PDP LTD Classic Wood Hoop, and Pearl Chad Smith Signature snare drums.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it. Here's an example: take a snare tuned high and pair it with a mic that has a large high-end boost. Want to cut the highs in the mix? It's not going to be easy as you're mixing against what is being sent, not mixing with it.

At this point, you can be as simple or as creative as you want. How do you want to mix a dual-miked snare? How do you want the snare to sound for the song?

It's not a matter of "how do I use the equipment?" but rather a matter of "what would sound right and how do I get there?" By having the right snare and mic combination, you've got the hard part out of the way. (I know this isn't always within your control.)

I like a single-miked snare. That's not to say I won't fall in love with a dual mic setup next week. But a single mic setup is a good place to start. By establishing a good single-mic sound, when moving into two mics, you already know how to get a good sound from one. Make sense?

## SUM OF THE PARTS

The ideas that follow are based on my experiences and can serve as a starting point in capturing and mixing a snare. All of the sounds of the drum kit (and the whole band for that matter) have to be considered. The right sound for the snare for a particular song might be really flat on its own.

*High-pass filtering*. A mic like the SM57 has the low-end rolled off, and I'll roll off a bit more if I notice a positive impact on the sound. I'm not going to roll off more just to then flatten the snare sound. If you're running an analog board, hit the HPF switch and listen for a difference.

It's a good idea to remove low-end frequencies from all mics that aren't focused on a low-end instrument. For instance, use an HPF on vocal mics. Snare and cymbal mics, which aren't focused on low-end kit pieces, are another good place for applying an HPF.


Figure 2: Top, Granelli G5790. Left to right, Heil PR31 BW, Telefunken M80, and DPA 4099.

*Gating*. I've never been quite happy with the results of gating snares, at least as it applies to general snare sound. I've gated the snare for a song to get a specific sound, but for all-around mixing, I tend to skip it. (Your mileage may vary.)

*Out with the bad*. I sweep the mid-range with a 6 dB cut and find the area of offending frequencies. You know, that area where

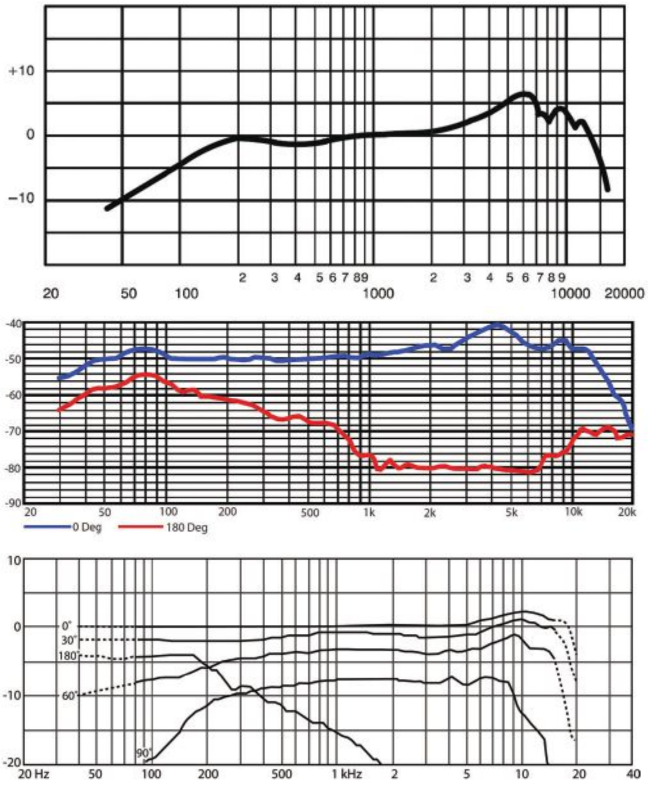


Figure 3: Top to bottom, published frequency response charts for the Shure SM57, Heil PR31 BW, and DPA 4099.

you make the cut and suddenly think, "now that sounds much better." In the case of my "frog EQ," I didn't find that spot but rather found a huge boost was needed. Some days it's like that.

## SCULPT TO FIT

It's of real benefit to have a sound in your head that you want the snare to match. It's that internal reference sound. You know what sounds good, now make it a reality. Is there too much snap? Not enough? Perfect the way it is? (This is where that snare/mic pairing pays off.)

There's no magic formula for exactly what to boost or cut and where to do it. It all depends on the snare and the mic and the drummer and the room and…eh, you get the point. That said, here are a couple of places to start:

- *Snap and presence, 3 kHz to 12 kHz*. The higher you go, the less presence and more snap.
- *Body, sub-500 Hz*. If more substance is needed.

The take away is to do the homework: know the tone of the instrument, pair it with the right mic, and then step behind the mixer. A good snare sound is the very likely result. ■

CHRIS HUFF *is a long-time practitioner of church sound and writes at Behind The Mixer (www.behindthemixer.com), covering topics ranging from audio fundamentals to dealing with musicians – and everything in between.*

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Mark Knopfler
Mark Tremonti
Maroon 5
Marty Stuart
Matchbox 20
Meatloaf
Megadeth
Melissa Etheridge
Metallica
Michael Bublé
Mötley Crüe
Motörhead
Mumford & Sons
Muse
Nathan East
Neil Young
Nelly Furtado
Nickelback
Nine Inch Nails
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R H Chili Peppers
Rick Wakeman
Rickie Lee Jones
Rihanna
Ringo Starr
Robert Plant
Robert Randolph
Rod Stewart
Roger Waters
Rolling Stones
Rush
Ryan Adams
Sam Roberts
Sammy Hagar
Santana
Sarah McLachlan
Seal
Sevendust
Shakira
Shania Twain
Sheryl Crow
Shinedown
Sigur Rós
Skunk Anansie
Slash
Slayer
Sleigh Bells
Slipknot
Snow Patrol
Sonic Youth
Soundgarden
Spice Girls
Steely Dan
Stereophonics
Steve Earle
Steve Lukather
Steve Miller
Steve Morse
Steve Stevens
Steve Vai
Steve Winwood
The Stills
Sting
Stone Temple Pilots
Styx
System of a Down
Taylor Swift
The Black Keys
the National
Three Doors Down
Timbaland
Tom Waits
Tommy Emmanuel
Tommy Lee
Tony Bennett
Tony Levin
Tony Maserati
Toots & The Maytals
Tragically Hip
U2
Usher
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FACTORY DIRECT

# Yamaha MG Series

Detailing a new compact mixer line.

***by John Schauer***

**WORK ON THE NEW** Yamaha MG series of compact mixers started almost immediately after the launch of the previous models, an effort with several guiding principles, including increased fidelity, reduced noise and distortion, and taking advantage of advancements in component and circuit designs. By upgrading every single aspect of these units and incorporating features that the company developed for other mixer lines, our engineers have developed workhorse products that will provide clean audio and reliable performance night after night.

Construction is based around a solid metal chassis that stands up to road use while rejecting outside interference. An interesting feature incorporated from the CL console series is a step between the I/O panel and the control panel, which segregates the mixer into functional zones. A special forming process that bends the surface after printing creates a seamless appearance. Channel numbers are printed on the step's riser for enhanced visual recognition.

In addition, mixer controls are color-coded to indicate their function at a glance and use primary colors for maximum visibility in dark environments. This color coding is the same as on all Yamaha mixers.

Single-knob compression, incorporated from the larger format MGP mixer line, is included on eight of the new MG models. Available on both the standard and XU models, it's useful in taming an unruly vocal or for bringing out a bass guitar in the mix. Without it, users need to purchase an outboard device and learn to set many controls, including attack, ratio, release and output gain. We found that the complexity of these devices led to poor sound.


**Yamaha MG20XU, the largest and most sophisticated model in the new MG Series.**

## PROVEN EFFECTS

SPX digital effects, offered on five of the new models (MG06X and the XU line), represent another technology that trickles down from higher-end models. As pioneers in LSI (large scale integration) chip design, our engineers developed one of the most popular digital effects processors ever, the SPX90. Used on countless recordings and tours, its ability to provide natural reverbs, delays and other sound effects is universally recognized.

We created 24 different useable effects to help add ambiance to the mix, whether live or via the USB I/O for recording. Users also benefit from the low noise floor and can find the right effect pretty easily. Since these mixers are also operated by performers, a footswitch input is included to mute the effect between songs.

With the growth and popularity of powered loudspeakers, the use of XLR balanced outputs allows for long cable runs while rejecting noise and interference. High-pass filters eliminate unwanted low-frequency noise and a 3-band channel EQ provides tone


**Back panel capabilities included on the MG20XU.**

sculpting capability. MG models also include up to four aux sends to incorporate additional effects, access external devices or feed stage-monitoring systems. Return level controls for the aux and stereo buses provide seamless integration with external gear. LED level metering allows output levels to be monitored accurately in either darkness or daylight.

All models come with Yamaha Class-A D-PRE mic preamps, which use an inverted Darlington circuit topography to provide added power, deliver lower impedance, and supply a wide frequency range able to handle signal from any source without coloration. The mic pre, which amplifies the quiet, noise-susceptible signals up to workable levels inside the console, has the toughest job in the entire audio chain. It's important that this gain not add "color" or distortion to the original signal.

Though some manufacturers employ an IC (integrated circuit) for this job, we instead chose to utilize discrete components specifically oriented on the circuit board to reduce interference while keeping the signal clean and undistorted. It's a more costly way approach, but the result is worth it.

## NUMEROUS OPTIONS

We also developed a new integrated circuit, the MG01 op-amp, to enhance the line level areas inside the mixers. Working directly with the circuit manufacturer, we were able to develop a design that focuses on sound rather than just electrical design. These ICs incorporate high-end silicon wafers and oxygen free copper wire to enhance resolution.

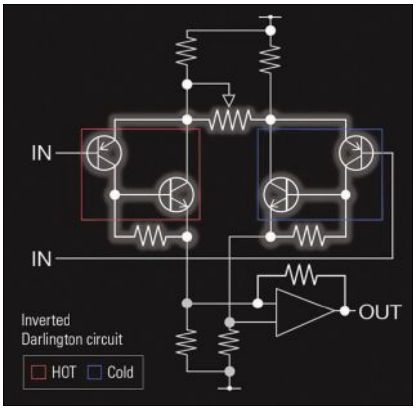
The physical design is contoured for effective convection cooling, with the internal layout separating the power supply from the analog circuitry for added noise reduction while also further extending the life of the components. Knob and control placement above the surface of the chassis diverts impact and pressure away from the circuit board or components underneath. Models with 12 channels and above offer a universal AC power supply mounted internally with the familiar IEC removable power cord, allowing for operation worldwide.

The new models come in three categories: two 6-channel compact models, four standard models, and four XU versions that feature computer connectivity. The MG06 and MG06X offer six line and two XLR balanced outputs from the stereo bus, and they also include a PAD switch on the mono inputs and 48-volt phantom power. The MG06X adds six high-grade SPX effects.

On standard models, the MG10 offers three stereo inputs and four microphone inputs (two with single-knob compression). The MG12 comes with six microphone inputs (four with single-knob compression) and the MG16 provides 10 microphone inputs and three stereo inputs. Finally, the MG20 includes 16 microphone inputs (eight with single-knob compression) and two stereo inputs.

Versions in the XU line, which include USB I/O and digital effects, start with the MG10XU, offering four microphone inputs and three stereo inputs. Next is the MG12XU, with six microphone inputs (four with single-knob compression). The MG16XU steps up to 10 microphone inputs (eight with single-knob compression) and three stereo inputs. The MG20XU includes 16 microphone inputs, (eight with single-knob compression) and two stereo inputs.


The more compact MG10, an option without the digital effects package.




The inverted Darlington circuit topography utilized in the Class-A D-PRE mic preamps.

Steinberg Cubase AI recording software is supplied with MG mixers. Playback of digital content from a PC or Mac, along with recording of the mixer output using DAW software, is handled by a USB 2.0 audio interface capable of 24-bit/192 kHz sound quality. iPhone and iPad connectivity is available with Apple's Camera Connection Kit or Lightning-to-USB Camera Adapter. A recently released free app for recording and playback, MG Rec & Play, available at the iTunes app store, improves on this feature.

Finally, Yamaha builds MG mixers in its own factories, allowing engineers to take full advantage of production and quality control. Prices range from \$129 to \$929 (U.S. MSRP). ■

JOHN SCHAUER *is product manager at Yamaha Corporation of America.*

# LET'S GO SMALL

## Inside the application of miniature microphones.

***by Craig Leerman***

WHAT ARE GENERALLY categorized as "miniature" microphones come in three basic configurations: lavalier, headworn and suspended. As someone who does a lot of corporate shows and events, I've got quite a bit experience with all three types.

Lavalier mics ("lavs") can be attached to clothing (usually via a clip) or hidden in costumes, hats and even hair (usually for theatrical performances) to pick up vocals without being visually distracting. They come in wired versions but are far more commonly are plugged into wireless transmitters that allow the person talking or actor/singer freedom of movement.

Broadcasters typically prefer lavs with an omnidirectional pattern, but in live audio we tend to prefer a more directional pattern (cardioid and hypercardioid) to help keep feedback at bay. Lavs used to be more limited in frequency range, optimized for speech and not capable of handling very high sound pressure levels, but many modern models handle wide frequency ranges and high SPL, making them a viable choice for loud singers (think opera) as well as certain instruments.

For speech, it's great to easily clip a lav to a presenter, but there can be drawbacks, particularly in terms of positioning. Even if the mic is secured in the mid-center of the person talking (i.e., attached to a tie), there can be off-axis issues when turning the head. This effect is compounded even more if the mic is positioned on one side (i.e., on a lapel). Deploying a model with a wider pattern can help, but it will be more prone to feedback. Positioning the mic farther away from the head to widen the pickup area can also help, but there's potential to lose too much gain (and again, increased potential for feedback if you boost too much). It's a tricky balancing act to get it right, and no two people are the same.

Another issue can be clothing noise. Some fabrics can bunch up and rub on the mic as the wearer moves about. Other problems can be caused when a layer of clothing (like a scarf) covers the mic element, generating noise and muffling the voice. I've found that the biggest key is educating and directing wearers. Emphasize that they turn their body (not just their head) in the direction in which they want to face, and after checking for garments that rustle or obscure the mic, let them know about it as well. Also be sure there's some slack in the cable so if they twist or turn, it won't yank the mic from its position.

Theatrical users have almost eliminated these issues by placing lavs on performers' heads – at the hairline pointing down at the mouth, on the sides near the ear pointed toward the mouth, and even hidden in beards. Sometimes lavs can be attached to a pair of glasses or a hat that doesn't get removed during the scene. All of these options ensure the mic is always in the same relative position to the mouth, meaning that pickup stays consistent.

Audio-Technica BP896 MicroPoint lavalier mics in black and beige.

### MAKING HEADWAY

Unfortunately a lot of other applications don't afford us with enough time to hide a lav on the heads of our presenters, and besides, most of them don't want us attaching things to their hair or face. And that's where purpose-designed headworn mics come into the picture.

While a lav hidden in an actor's hair is certainly headworn, the term actually refers to a type of mic that is positioned from the ear. If the mic mounts around a single ear, it's called an earset or ear-worn model, and it if uses both ears for support, it's referred to as a headset or dual-ear model. Both types use a small boom to position the mic element in proximity to the mouth. Boom lengths vary; some place the element very close to the mouth while others put it a couple/few inches away from the ear. In general, the closer the element is to the source, the better (and more consistent) the gain will be.

Mounting issues are pretty much eliminated, and there no clothing noise generated. Many manufacturers offer models in a few colors to better blend in with particular skin tones.

The main issues I run into with headworn mics have to do with the fit and the cable run. Most models allow for ample adjustment and flexible booms that can position the mic optimally, but it's important take the time to actually fit and adjust the support system to the wearer's head or the mic can slip out of position during use. A minute of attention on a snug fit can save a lot of headaches later.

Cables typically exit on the side and include small clips to guide it to the back of the garment, where it can then run down to a wireless transmitter. Problems arise when there's not enough slack between the mic and the clips. The mic can get pulled out of position when the wearer turns his/her head, or worse, the cable can get pulled from the mic. Some models come with replaceable cables because damage is common when dealing with these relatively fragile wires, but the best solution is to always leave enough slack while verifying wearers have freedom of movement before they hit the stage.

Point Source Audio SERIES8 available in both earset and headset styles.

## SOLVING PROBLEMS

Suspended mics (a.k.a., choir or chorus mics) have gotten a lot smaller in recent years, to the point of being almost invisible when suspended above a performance area. They can also be hidden in scenery and props to provide additional pick-up in dead coverage areas on a stage. While primarily for vocal ensembles, they can also be effective with orchestras, hung over the top for additional capture when the stage is too crowded for additional mics.

Speaking of instruments, miniature mics can come in very handy, especially when you don't want to see the mic or it would get in the way. A simple approach I use at times is to clip a lav to the musician with the element pointed at the instrument. This has worked well for me with hand percussion, a bongo player, and even a violinist who didn't want a mic clipped to her expensive fiddle while also refusing to stay near stand-mounted mic. I outfitted her with headset mic and pointed it at the violin. It sounded quite good and there was plenty of pickup.

I've also used this method with flute players, including one with poor mic technique and lots of wind noise getting into the mic. With the element pointed at the head end of the flute, most of the noise was eliminated while a nice tone was captured.

Miniature mics can be directly attached to some instruments if the element will handle the SPL. I've mounted lavs on guitar, upright bass, cello, autoharp, and even a metal shaker for a percussionist. Use easy release tape like blue painters tape or board tape, and be sure to ask the permission of the owner before attaching anything to an instrument.

A few years ago I worked with a cool traditional bluegrass band that wanted the look of a single mic typical of performances in that genre, but they weren't getting the desired sonic result. The solution was attaching lavs to the mandolin, upright bass, and the shirts of the guitar and fiddle players. They attained a much better sound, maintained the "single mic" look, and as a bonus I had more control of the mix.

With older vintage mics that look great but aren't sonically usable anymore, I remove the components and mount rubber suspension for attaching lavs inside. It works surprisingly

The DPA d:screet necklace microphone is a very fast and flexible miniature option.


The new Countryman I2 is a lav-type mic specifically designed for instruments.

well, and provides the ability to swap in different pickup patterns. In fact, I've rented some of these mics for movie production, and the dialog captured during the filming is used in the final product, not replaced by additional dialog recording (ADR) in post-production.

Lavs also make great back-up mics. One of my favorite tricks is to tape a lav just below the head of a podium mic, making sure we're still going to receive audio even if the primary mic fails. Plus, it's a more attractive solution than deploying two podium mics. Sometimes I use a lav with a wider pickup pattern for added flexibility, depending on who's at the podium. It can also be used as the recording feed mic.

Another approach is attaching lavs as back-ups on singers. One "diva" I worked with a while back had horrible mic technique, and she wasn't about to change. (Thus the diva reference.) However, I was able to talk her into also wearing a lav, so I could utilize the two sources together to get a more consistent signal. Not perfect, but way better than what we started with. I've also placed lavs on singers and used them just for recording in case there's a glitch with the main handheld wireless.

A colleague owns a studio and sometimes tapes a lav onto the booth window, using it as a recording boundary mic. I've heard the results and it works well for certain sessions.

Recently my company handled production for a meeting here in Las Vegas and one of the participants was stuck at his office back East because of snow. As a result, he needed to call in for his portion of the presentation, and while the hotel property had a conference unit, it didn't have an audio output that could be used with the PA. So I placed a lav on the unit's built-in loudspeaker, applied some EQ, and it ended up sounding a whole lot better than a lousy phone line while also being invisible.

There's a lot of great technology when it comes to modern miniature mics, so don't hesitate to "think small" the next you've got a problem to solve. ■

*Senior contributing editor* CRAIG LEERMAN *is the owner of Tech Works, a production company based in Las Vegas.*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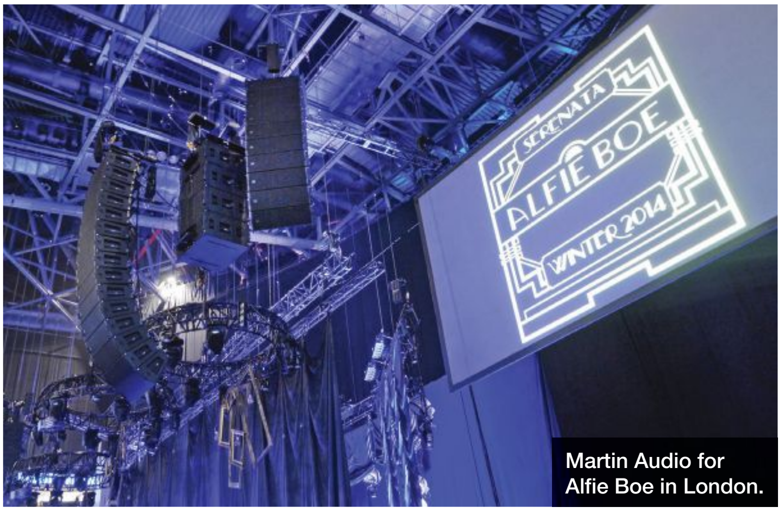
# AROUND THE GLOBE

Notable applications of sound reinforcement technology.

***by Live Sound Staff***



Martin Audio for Alfie Boe in London.

THERE'S BEEN A LOT of interesting applications happening of late with touring sound, as well as sound at big-time events, around the world. Let's have a look.

## PHARRELL WILLIAMS IN EUROPE

UK-based Wigwam, working with Sound Image (Escondido, CA), provided the systems and support for the recent European leg of The Dear GIRL Tour by singer-songwriter Pharrell Williams, including Adamson Systems Energia line arrays and subwoofers.

The tour, which visited arenas, was supported by a main PA with left and right line arrays each made up of 15 (fifteen) E15 and 3 E12 enclosures. Out fill was handled with two more line arrays, comprised of a dozen E12s followed by 4 Adamson SpekTrix enclosures. Twenty T21 (dual 21-inch-loaded) subwoofers (eight stacked under each array and four in the the center) delivered the low-end, also hosting small SpekTrix enclosures for front fill.

"The E15 sounds exceptional – all of the nuances in the midrange are extremely clear," states Kyle Hamilton, front of house engineer for Williams. "I hear actual notation, not just a lot of rumbling. The clarity is incredible – it's like being able to see everything in a picture. The Energia system has incredible depth.


Pharrell Williams in Europe with Adamson.

"In addition, the T21 subs have a lot of horsepower," Hamilton continues. "I was amazed. Once I got them dialed in they were terrific and provided a very musical sounding low end."

The Williams team is currently gearing up for North American dates, which will also utilize Adamson components in the main system.

## SERENATA AT THE O2

Alfie Boe's Serenata tour recently visited the O2 Arena in London. where the popular tenor and actor was supported by a 5-piece band, a 16-piece orchestra, the New Zealand musical trio Sol3 Mio, and Martin Audio MLA provided by Capital Sound (London).

Veteran producer/engineer Matteo Cifelli (owner of Fastermaster Studio) handled the front of house mix, supported by system tech Joseph Pearce. With 86 inputs at the console (including 80 mic channels), there was plenty to challenge the sound team as Boe worked through an Italian-based repertoire, assisted in one instance by Rick Wakeman on keyboards.

The main hangs incorporated 14 MLA elements (plus two MLD Downfills) on each side, with side hangs of 10 MLA. Although the show was virtually all acoustic, and hardly dependent on LF overkill, Pearce set four MLX along the front—left, right and a split pair in the center – while flying a further eight (four each side), with the top and bottom enclosures in each hang rear facing.

A challenge had been to shape the voice to deliver warmth and presence via the EQ in the face of loud stage monitoring. The vocal then had to nestle in the midst of a conventional band but with the addition of a 100-year-old Dulcitone and accordion, as well as the orchestra when it came in.

"With the changes we have made the PA now sounds absolutely great, and the subs are also impressive," Cifelli notes. "It's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Freedom from FOH – mix from anywhere!
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02 Wireless: From mic pre gain to control over multi-track recording and playback
03 Recording/Playback: Complete wireless control over multi-track direct-to-drive recording and playback
04 DSP: Powerful processing on all inputs and outputs that replaces racks of outboard gear
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Free yourself from the confines of FOH. With the DL32R, you get 32-channels of powerful digital mixing that's completely controlled wirelessly — MIX FREE.
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Wireless router and iPad required for operation (not included).

now completely silent behind the hangs. One of the best qualities of the PA is that I can get the sound I'm looking for straight away. It reacts very well to the way I EQ instruments and I find it is an extremely musical PA that throws huge distances without losing any detail. The clarity at 200 to 260 feet is fantastic."

The show was driven digitally all the way to the loudspeaker boxes via AES3 from Cifelli's Avid D-Show console and sidecar, with all five DSP card slots fully populated, rendering the signal path noiseless.

## NEW YEAR'S EVE IN BERLIN

The celebration of the new year in Berlin attracts well over a million visitors annually to a street party in front of the Brandenburg Gate, with national and international performers on three stages putting on a spectacular show. This year, regular audio provider auvisign (Berlin), which traditionally utilizes NEXO PA for the event, deployed new STM M28 modular line arrays.

NEXO at the Brandenburg Gate in Berlin.

PHOTOGRAPHY COURTESY OF BEN KRIEMANN

Specifically, Johannes Raack at auvisign specified arrays comprised of 18 STM M28 omni modules flown beneath three STM B112 bass cabinets to serve the principal stage. On the ground, 32 STM S118s provided the sub bass. In addition, NEXO GEO S8 loudspeakers handled center and near fill.

"Without a doubt, the M28 did a great job," says Raack. "The variation we used showed unexpectedly high headroom in the low end; that's remarkable for such a thin column with a total weight of less than 950 kilograms (about 2,000 pounds). But what most impressed me was the handling of the M28 as well as the full system.

"For instance," he continues, "we had some positions where it was not possible to rig the system directly below the hoist points. There were positions, such as the delay towers, where we had to place the cabinets out of line below the flying point – and there was no problem at all."

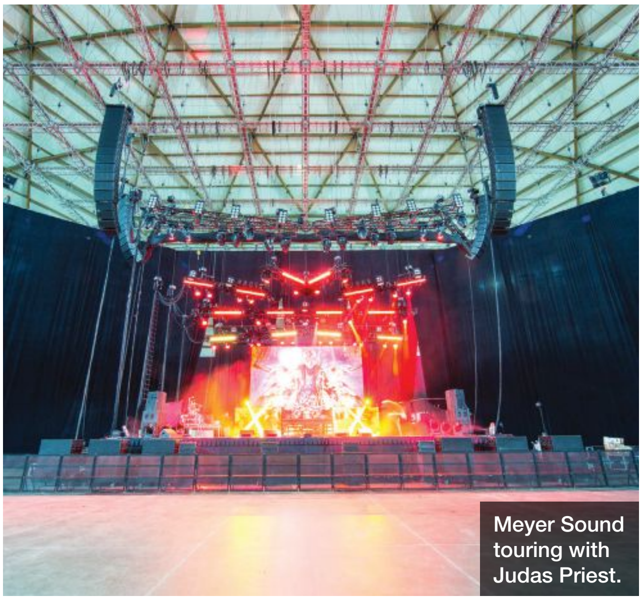
The PA was set up on an (Audinate) Dante network, with a separate EtherSound network for the delay lines. Power and processing was supplied by NEXO's NUAR (Universal Amp Racks) and NXAMP4x4s. Under the control of network operator Peter Sobisiak, the inputs for the mixer were implemented via Yamaha AD8HR converters and EtherSound mini-YGDAI cards over another EtherSound network. All of the different networks were consolidated into one Gigabit network, which used fiber links to cover distances of up to 400 meters (more than 1,300 feet) between devices.

## JUDAS PRIEST AROUND THE WORLD

Front of house engineer and production manager Martin Walker selected Meyer Sound LEO-M and LYON linear line array loudspeakers, supplied by UK-based Major Tom Ltd., for the latest world tour by metal legends Judas Priest.

"The response of the system is pretty seamless, whether going from main LEO arrays to LYON side arrays, or going from a main LEO hang in an arena to a LYON hang in a theatre," Walker says. "You turn on the PA and there's really no difference to worry about. The headroom is there, the weight saving is there, and the clarity and fidelity are there. It's a great system with either box out front."

The main and side arrays for the tour are assembled from the tour's full inventory of 28 LEO-M and 32 LYON loudspeakers and 18 1100-LFC low-frequency control elements. Twelve M'elodie line array loudspeakers are deployed as fills, and a Galileo Callisto loudspeaker management system with one Galileo 616 AES and six Galileo Callisto 616 array processors handles drive and optimization.



Meyer Sound touring with Judas Priest.

"I try to recreate a total experience of the band," adds Walker, who mixes behind a DiGiCo SD7 digital console. "I want a loud sound with full bandwidth, warmth, and depth, a sound that really brings the band to life. With LEO and LYON, I can realize that, every night."

### MUSIC & ARTS IN HONG KONG

The recent Clockenflap music and arts festival in Hong Kong offered six stages outfitted with K-array Firenze and Concert Series loudspeakers that offer some of the company's latest technologies, joined by a brand-new K-array loudspeaker development not even announced yet.

Held at the West Kowloon Cultural District Waterfront Promenade, this seventh annual iteration of Clockenflap saw Sennheiser Hong Kong providing the systems and tech support for the 3-day festival. The main stage, which hosted artists such as Tencious D, Kool & The Gang and The Flaming Lips, was outfitted with 24 new K-array KH8 full-range, self-powered line array modules joined by 16 new KS8 subwoofers, each loaded with dual 21-inch custom-engineered transducers.

In addition, eight K-array KH15 loudspeakers on either side of the stage provided front fill and side fill, joined by four K-array KH4 and four KO70. Up to 14 K-array KF12MT could be arranged on stage for monitoring, with two KO40 for drum fill.

K-array deployed at Clockenflap in Hong Kong.

"The Clockenflap organizers are always looking to push the limits of sound, art and music," states James Mak, sales manager for Sennheiser Hong Kong. "They have big expectations of the equipment. We're always confident when we suggest K-array; they love the flexibility of the Firenze and Concert Series. It allows them to reach their ambitions." ■


# Mackie DL32R

Evaluating a new wirelessly controlled digital mixer.

*by Craig Leerman*


Mackie DL32R mixer and iPad with Master Fader app.

**EVER SINCE MACKIE** debuted the DL806 and DL1608 digital live mixers a few years ago I've eagerly waited to see what would follow, and it certainly does not disappoint. The new DL32R is a 32-channel digital mixing system completely controlled wirelessly – including DSP and recording/playback – from Mackie's Master Fader iPad app.

Specifically, the package offers 32 inputs by 14 outputs, with new Onyx+ pre-amps, flexible routing, three effects units, and direct to hard drive or computer multitrack recording, all in a compact 3RU format. Using any size iPad (user supplied) with the free Master Fader control app gives the operator(s) total wireless control of every function (except headphone volume) of the mixer, and performers onstage can access their own monitor mixes via the My Fader app for iPhone iPod touch.

All inputs and outputs are on the front of the mixer. Channels 1-24 have XLR connectors and channels 25-32 offer combo TRS/XLR jacks. In addition to the inputs, there are 14 XLR outputs, a stereo AES output, two 1/4-inch monitor outputs, and a 1/4-inch headphone jack with volume control. Two LED lights indicate power and network connection.

While the front panel is three rack spaces high, the rear of the unit is only about half that tall, providing a "shelf space" for mounting a Wi-Fi router or optional hard drive that can be secured by the provided "hook and loop" straps and built-in strap holders. (Nylon cable ties could also be used.)

Rear-panel connections include a Type A USB connector for the hard drive, a Type B USB for connection to a computer DAW, a removable expansion slot (loaded with a card for Wi-Fi, and a new Dante expansion card is also now available), two fans, a power switch and IEC power cord socket. At 15.6 inches deep and weighing only 18 pounds, the unit is very compact and could be racked up with other gear (like wireless microphone receivers and IEM transmitters).

## FAST SETUP

Out of the box, I was struck that the unit looks like a remote stage input box – the footprint is that miniscule. I really like the rear shelf and the included straps/attachment points for securing my router rather than accidentally leaving it at the shop. I grabbed my iPad and visited the app store to make sure I had the latest version of the Master Fader software (I did). Then I pulled a router from the shelf and used a short section of Cat cable to plug it into the mixer. Without cracking open the manual I soon had a mic and stereo playback from an iPhone wafting through the shop.

I'm a big fan of the Master Fader app. It's great looking and easy to get around on, and the same software can control all three DL mixers – it scales itself to whichever mixer it's working with. And tons of features have been added for the DL32R.

Each channel now has wireless control of its Onyx+ pre-amp. There's also a gain control in the channel strip as well as individual controls for 48-volt phantom power and phase. A source button offers a choice between the mic pre or USB 1 and the playback of multi-tracks for mix down, or more likely, a virtual sound check. Instead of six aux sends for the DL1608, there's now access to 14 assignable buses.

The DL32R's three effects units

include two reverb and a delay. The reverbs sound nice and offer choices that include plate, spring, ambience, as well as small through cathedral-sized rooms. Choose from existing presets and/or store your own creations. Delay options are mono, stereo, tape echo, ping-pong, and multi-tap, along with presets and user storage. There are tap tempo buttons right on the screen, or use sliders to change delay times. Each channel also has a gate and compressor that offer presets, user storage, and a lot of adjustments. Outputs each have a compressor, 31-band graphic EQ, 4-band EQ with HPF and LPF, 300 ms alignment delay, and a channel assignment screen.

## A TOUCH AWAY

Users can choose between a modern-looking GUI (swipe and move) and a vintage faceplate of older gear where you can turn knobs, push buttons, and watch a meter swing back and forth. I like it! (They also have their own algorithms and functions.) Getting around on the app is easy. You can use the VCAs to control things, but there's also a feature called View Groups where the user can set up just the input and output channels needed for a particular song or event.

For example, on a corporate gig where I might need a podium mic and lavalier that are patched in to channels 1 and 2, as well as some video playback patched into 29-32, I could just assign all of those channels into a single View Group and they alone would show up on the screen, along with the master fader. This is perfect for restricting access to certain channels and/or for grouping important channels.

Another new software feature accommodates multitrack recording to hard drive or computer. The recording and playback screen has two sections. The right side controls the recording/playback of the multitrack unit, offering selection of bit depth and containing the transport controls. The left side handles play back of files, and it can also play back click tracks or songs simultaneously, along with the multitrack recording. So you can play back tracks for an artist while recording – nice!

The patch screen allows all inputs and outputs to be configured as needed. Inputs include the onboard connectors, USB and Dante when used with the just-released Dante expansion card. Being able to integrate the mixer into a Dante network takes it to the next level, in my view.

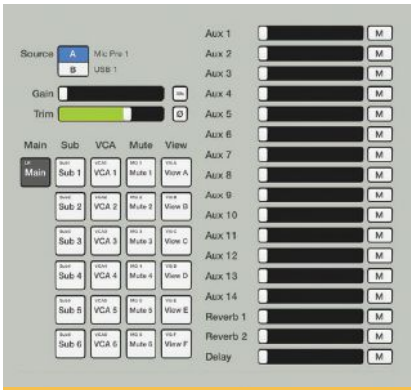
The new features don't make the software seem crowded and confusing. Everything is easy to see and configure, and mixing is effortlessly accomplished. I also like that more than one user can access the mixer; for example, mains and monitors can be handled by two people. Performers can access just their own mixes while not messing with any others. Personal monitor systems can cost quite a bit, so this mixer and the My Fader app can save a bundle.

## DIVERSE APPLICATIONS

My company utilized the DL32R at a few shows. The first was a multi-day festival where, because of its extensive routing capabilities, we used the DL32R to send signals to different zones all around the grounds. Setting up the delay times was easy as the main bus, auxes and matrix outputs all offer delay. The delay setting shows milliseconds, meters and feet, and there's a slider for setting the ambient temperature in Fahrenheit or Celsius. I assigned a few matrix outputs with delay to feed the various PA zones, and everything worked out well. Truth be told, the DL32R would have handled the main stage as well as the consoles we had there that are priced far higher.

Next up was an event stage hosting live musicians, singers with tracks, comedians and raffle drawings. Setup took just minutes, and it was great being able to walk around the room and stand onstage

The My Fader app that provides mix control via iPhone or iPod touch.




Channel view of input routing on the Master Fader app.

to ring out the PA and monitors.

Around this time, a friend who's in a local band mentioned that they're tired of dealing with venue-supplied stuff that's usually unmanned, so they're seeking solutions. I took the DL32R over to their rehearsal studio for a look, and they immediately noted how they could package everything needed (wireless receivers and IEM transmitters) into one small rack together with the mixer.

At their next gig we tested the concept. They loved having control of their monitor mixes via the app, and were really impressed with how easy it was to mix the mains from the stage. The keyboard player became the "FOH engineer" since he doesn't sing and understands a bit of audio. I routed the left main mix output to the venue's mono PA and the right main mix output to his in-ear mix so any changes to the PA made a change in his mix. After a few songs he got a pretty good house mix happening.

The DL32R is an excellent choice for a variety of gigs and applications. The preamps, effects and processing sound great, and the unit provides tremendous flexibility, including extensive routing options in about the smallest package possible. And the Master Fader app is simply one of the best remote control software products available today. This mixer is highly recommended. ■

*U.S. MSRP: $2,499.99*

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

# Most Valuable Players

The latest on 2-way compact loudspeakers. ***by Live Sound Staff***

AVAILABLE IN A RANGE OF SIZES and cabinet shapes, compact 2-way loudspeakers are an MVP (Most Valuable Player) in the live sound world. They serve as mains, monitors, side fills, center fills, near fills, front fills and delays, and can quickly be ground stacked, flown, or placed on a stand. In addition, they can almost always be carried by one person.

Many (most) of these boxes are trapezoidal in shape, with many offering an enclosure angle for stage monitoring. Smaller 2-way models incorporate a single 8- or 10-inch woofer, but the most popular models offer 12- or 15-inch woofers for additional low-end performance.

Usually the woofers are accompanied by a compression driver on a horn or waveguide for mid and high frequencies. Also don't overlook coaxial models where the individual driver units radiate sound from the same point/axis, which, when designed properly, can offer enhanced coherence.

When evaluating 2-way loudspeakers, start by defining the right box for the job – size, scale, mounting, portability, and so on. It all depends on the requirements of the application(s).

Also note that most of the models we're featuring here represent an entire series offered by their respective manufacturers. Further, many of them are available in either passive or active versions with (usually) class D amplification and DSP. Some models are now even networkable.

When making "apples to apples" comparisons, basic factors to consider include dispersion, passive or active, maximum SPL, and mounting options. But again, these just scratch the surface, so please use the information as a basis for your own further investigation. Enjoy this Real World Gear look at 2-way loudspeakers.

## EAW VFR159i » www.eaw.com

**Frequency Response:** 54 Hz – 20 kHz (-10 dB)
**Dispersion (h x v):** 90 x 60 or 90 x 50 degrees
**Drivers:** 1-in compression driver on a rotatable horn; 15-in woofer
**Max SPL:** 132 dB (peak)
**Power Handling:** 600 watts (8 ohms)
**Cabinet:** Baltic birch
**Size (h x w x d):** 28.3 x 20.1 x 18.3 inches
**Weight:** 60 pounds
**Mounting:** Rigging points, pole mount, optional U-Bracket

## Adamson Systems Point 15 | www.adamsonsystems.com

**Frequency Response:** 55 Hz - 18 kHz (±3 dB)
**Dispersion (h x v):** 60 x 40 or 40 x 20 degrees (rotatable)
**Drivers:** 1.4-in-exit compression driver on rotatable waveguide; 15-in Kevlar/neodymium woofer
**Max SPL:** 136 dB/134.8 dB (active/passive)
**Power Handling:** Self-powered; onboard DSP (passive version also available)
**Cabinet:** Baltic birch
**Size (h x w x d):** 28.6 x 17.5 x 13.1 inches
**Weight:** 54.5/58.5 pounds (active/passive)
**Mounting:** Rigging points, pole mount

## Martin Audio DD12 | www.martin-audio.com

**Frequency Response:** 65 Hz - 18 kHz (+/- 3 dB)
**Dispersion (h x v):** Proprietary Differential Dispersion technology
**Drivers:** 1.7-in voice coil neodymium compression driver; 12-in neodymium woofer
**Max SPL:** 131 dB
**Power Handling:** Self-powered; onboard DSP, networkable
**Cabinet:** Multi-laminate birch ply
**Size (h x w x d):** 22.5 x 14.2 x 15.1 inches
**Weight:** 57.2 pounds
**Mounting:** Pole mount, rigging points

## QSC KW152 | www.qsc.com

**Frequency Response:** 44 Hz - 20 kHz (- 10 dB)
**Dispersion (h x v):** 60 degrees axisymmetric
**Drivers:** 1.75-in compression driver; 15-in woofer
**Max SPL:** 133 dB
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** 15 mm birch
**Size (h x w x d):** 32.1 x 17.5 x 15.2 inches
**Weight:** 64 pounds
**Mounting:** Pole mount, rigging points

## Cerwin-Vega P-Series

www.cerwinvega.com

The award-winning Cerwin-Vega P-Series is a powerful, portable loudspeaker line which combines rugged construction, extreme clarity, and unmatched bass performance in an easy-to-use and versatile package.

The P-Series active loudspeaker family includes three models: the full-range 1,500-watt, 15-inch P1500X that delivers up to 134 dB; the full-range 1,000-watt, 10-inch P1000X that provides up to 128 dB; and the 2,000-watt, 18-inch P18000SX subwoofer that produces up to 136 dB of tight, deep and dry low end.

Each model is designed for a variety of demanding sound reinforcement applications and offer flexible input and output options, feature Vega-Bass and exceptional dispersion for superior coverage.

**TECHNOLOGY FOCUS:** Both full-range models are outfitted with a proprietary hemi-conical horn that fosters extreme clarity over an even and wide coverage area. A built-in mixer with I/O connections provides EQ, Vega-Bass boost and high-pass filters for enhanced performance.

**OF NOTE:** The pole mount socket can be angle adjusted to set the loudspeaker in a normal flat position or a 7.5 degree downward tilt for improved audience coverage from a raised location.

**KEY SPECIFICATIONS**

**Model:** P1500X
**Frequency Response:** 50 Hz - 23 kHz (+/- 10 dB)
**Dispersion (h x v):** 90 x 65 degrees
**Drivers:** 1.75-in compression driver; 15-in woofer
**Max SPL:** 134 dB (peak)
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** Heavy duty polymer
**Size (h x w x d):** 27.5 x 17 x 13.5 inches
**Weight:** 53 pounds
**Mounting:** Rigging points, dual-angle pole mount

### L-Acoustics ARCS II » www.l-acoustics.com

**Frequency Response:** 50 Hz - 20 kHz
**Dispersion (h x v):** 22.5 degrees symmetric; Vertical: - 60 degrees asymmetric (-20/+40 degrees)
**Drivers:** 3-in diaphragm compression driver; 15-in cone woofer
**Max SPL:** 140 dB
**Power Handling:** LF - 600 watts; HF -100 watts
**Cabinet:** Baltic birch
**Size (h x w x d):** 32.3 x 17.3 x 25.7 inches
**Weight:** 110 pounds
**Mounting:** Rigging points

### d&b audiotechnik Q7 » www.dbaudio.com

**Frequency Response:** 60 Hz - 17 kHz
**Dispersion (h x v):** 75 x 40 degrees
**Drivers:** 1.3-in compression driver on rotatable horn; 2 x 10-in woofer
**Max SPL:** 138 dB (w/d&b D12 amplifier)
**Power Handling:** 400 watts (RMS), 1600 watts (peak)
**Cabinet:** Marine plywood
**Size (h x w x d):** 22.8 x 12.1 x 16.1 inches
**Weight:** 49 pounds
**Mounting:** Rigging points, optional stand/swivel mounts

### Renkus-Heinz PN151 » www.renkus-heinz.com

**Frequency Response:** 40 Hz - 18 kHz
**Dispersion (h x v):** 90 x 40, 60 x 40 and 40 x 40 degrees (choice)
**Drivers:** 2-in compression driver on proprietary waveguide; 15-in woofer
**Max SPL:** 128 dB
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** Multi-ply hardwood
**Size (h x w x d):** 29.5 x 19 x 18.5 inches
**Weight:** 88 pounds
**Mounting:** Pole mount, optional rigging, optional mounting bracket

### JBL Professional SRX815P » www.jblpro.com

**Frequency Response:** 36 Hz - 21 kHz (-10 dB)
**Dispersion (h x v):** 90 x 50 degrees
**Drivers:** 2432H 3-in voice coil neodymium compression driver; Differential Drive 15-in ferrite woofer
**Max SPL:** 137 dB
**Power Handling:** Self-powered; onboard DSP, networkable
**Cabinet:** Baltic birch
**Size (h x w x d):** 27.2 x 18 x 18.8 inches
**Weight:** 63 pounds
**Mounting:** Pole mount, rigging points

## VUE Audiotechnik

www.vueaudio.com

The new h-5 is an ultra-compact, full-range loudspeaker designed to cover stage lip, underbalcony and other hard-to-reach areas that are often left uncovered by the main system, as well as for other situations where space is at a premium.

The h-5 combines two precision-engineered 5-inch low-frequency transducers with a 1-inch exit compression driver outfitted with a proprietary Pure Truextent beryllium diaphragm at its core that significantly improves high-frequency extension and response linearity. The beryllium compression driver is mounted to a new, Mike Adams-designed precision horn with wide 120-degree by 40-degree (h x v) coverage.

Dual-channel, high-efficiency amplifiers deliver power to the transducers while eliminating the need for noisy cooling fans. And, the h-5 incorporates a 64-bit digital processor providing sophisticated networking capabilities and DSP.

**TECHNOLOGY FOCUS:** The DSP is programmed to address every individual element in the h-5, handling EQ, time alignment, crossover management, and loudspeaker protection, as well as SystemVUE network control and monitoring functions. A fully protected switch-mode power supply with auto voltage detection provides worldwide operation.

**OF NOTE:** The robust enclosure includes integrated M10 hanging points, and optional yoke mount and rigging hardware make easy deployment as a main system, on lighting grids, or in a variety of support configurations.

**KEY SPECIFICATIONS**

**Frequency Response:** 80 Hz - 21 kHz (+ 2.5 dB)
**Dispersion (h x v):** 120 x 40 degrees
**Drivers:** 1-inch-exit beryllium compression driver; 2 x 5-in woofer
**Max SPL:** 114 dB (long-term)
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** Birch
**Size (h x w x d):** 7.6 x 18 x 11.7 inches
**Weight:** 28 pounds
**Mounting:** Rigging points, pole mount

### PreSonus StudioLive 315AI » www.presonus.com

**Frequency Response:** 46 Hz - 23 kHz (-10 dB)
**Dispersion (h x v):** 90 x 60 degrees
**Drivers:** 1.75-in titanium compression driver coaxially mounted to an 8-in cone midrange; 1 x 15-in woofer
**Max SPL:** 131 dB (peak)
**Power Handling:** Self-powered; onboard DSP, networkable
**Cabinet:** Chemline polyurethane over Baltic birch plywood
**Size (h x w x d):** 26 x 24 x 21.8 inches
**Weight:** 71 pounds
**Mounting:** Rigging points, pole mount

### NEXO GEO S12 » www.yamahaca.com

**Frequency Response:** 53 Hz – 19 kHz (+/- 3 dB)
**Dispersion:** Non-coupled MF/HF coverage adjustable at 80 or 120 degrees
**Drivers:** 1 x 3-in voice coil compression driver; 1 x 12-in woofer
**Sensitivity:** 103 dB SPL
**Crossover:** 1.1 kHz passive or active (internally configurable)
**Power Handling:** LF - 1750 to 3100 watts; HF - 875 to 1550 watts (3 cabinets, 4 ohms)
**Cabinet:** Baltic birch
**Size (h x w x d):** 13.5 x 26.5 x 14.8 inches
**Weight:** 61.8 pounds
**Mounting:** Rigging points, pole mount with optional U bracket

### Meyer Sound UPA-1P » www.meyersound.com

**Frequency Response:** 80 Hz - 17 kHz (+/- 4 dB)
**Dispersion (h x v):** 100 x 40 degrees
**Drivers:** 3-in-diaphragm compression driver on constant-Q horn; 12-in woofer
**Max SPL:** 133 dB
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** Birch plywood
**Size (h x w x d):** 22.4 x 14.5 x 14.3 inches
**Weight:** 77 pounds
**Mounting:** Rigging points, pole mount option

### D.A.S. Audio Vantec 15A » www.dasaudio.com

**Frequency Response:** 45 Hz - 20 kHz (- 10 dB)
**Dispersion (h x v):** 90 x 50 degrees
**Drivers:** 1-in compression driver; 15-in woofer
**Max SPL:** 135 dB (peak)
**Power Handling:** Self-powered; onboard DSP (passive version also available)
**Cabinet:** Birch plywood
**Size (h x w x d):** 28 x 17.5 x 14.8 inches
**Weight:** 52.8 pounds
**Mounting:** Rigging points, pole mount

## Yamaha DXR Series

http://usa.yamaha.com

DXR active loudspeakers are the fruit of collaboration with sister company NEXO. Their detailed analysis of the transducers and cabinet designs allowed the engineering team to take output performance farther than previous designs.

By employing proprietary 48-bit signal processing we are able to produce accurate sound while carefully controlling the amplifiers to maximize SPL. FIR filtering provides remarkable phase accuracy, allowing better clarity and imaging. DXR loudspeakers offer settings for both FOH and monitor use. Not just EQ circuitry, this intelligent dynamic control adjusts as output increases.

Incorporating separate 24-bit A/D and D/A convertors allows higher signal-to-noise ratio and reduces self-noise. Comprehensive protection circuitry monitors all aspects of the power supply, amplifiers and transducers to ensure reliable operation.

**TECHNOLOGY FOCUS:** All full range models offer proprietary FIR-X tuning utilizing linear phase FIR filters for the crossover network. FIR-X tuning simultaneously optimizes frequency and phase response while adjusting the time alignment between the HF and LF transducers. This creates a very smooth response around the crossover point, providing much better clarity and imaging than what is possible with typical crossovers.

**OF NOTE:** In addition to optimal limiting, DXR loudspeakers employ many of the same protection functions used in Yamaha's top-class TXn Series power amplifiers. A microprocessor and high-power DSP monitor the status of the power supply, power amplifiers, transducers and ongoing signals to protect all aspects of each component.

### KEY SPECIFICATIONS

**Model:** DXR15
**Frequency Response:** 49 Hz - 20 kHz (-10 dB)
**Dispersion (h x v):** 90 x 60 degrees
**Drivers:** 1.4-in diaphragm compression driver; 15-in woofer
**Max SPL:** 133 dB (continuous)
**Power Handling:** Self-powered; also onboard DSP
**Cabinet:** ABS
**Size (h x w x d):** 27.5 x 17.5 x 15 inches
**Weight:** 49.6 pounds
**Mounting:** Rigging points, dual-angle pole mount

### Mackie SRM650 » www.mackie.com

**Frequency Response:** 39 Hz - 20 kHz (-10 dB)
**Dispersion (h x v):** 90 x 50 degrees
**Drivers:** 1-in voice coil ferrite compression driver; 15-in ferrite woofer
**Max SPL:** 133 dB
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** 15 mm poplar
**Size (h x w x d):** 26.7 x 17.5 x 17.4 inches
**Weight:** 46 pounds
**Mounting:** Pole mount, rigging points

### Grund Audio GP-12A » www.grundaudio.com

**Frequency Response:** 47 Hz - 20 kHz
**Dispersion (h x v):** 90 x 60 degrees
**Drivers:** 1-in compression driver; 12-in woofer
**Max SPL:** N/A
**Power Handling:** Self-powered; built-in sensitivity & 2-band EQ
**Cabinet:** Ribbed, two-piece molded, low-flex
**Size (h x w x d):** 23.5 x 16.5 x 12.7 inches
**Weight:** 37.5 pounds
**Mounting:** Pole mount, rigging points

### FBT MITUS 114A » www.fbt.it

**Frequency Response:** 46 Hz - 20 kHz
**Dispersion (h x v):** 70 x 50 degrees (rotatable)
**Drivers:** 1.4-in-exit neodymium compression driver on rotatable waveguide; 14-in neodymium woofer
**Max SPL:** 135.5 dB
**Power Handling:** Self-powered; onboard DSP (passive version also available)
**Cabinet:** Baltic birch
**Size (h x w x d):** 27.7 x 15 x 15 inches
**Weight:** 54.8 pounds
**Mounting:** Rigging points, pole mount

### Electro-Voice EKX-15P » www.electrovoice.com

**Frequency Response:** 45 Hz - 20 kHz (- 10 dB)
**Dispersion (h x v):** 90 x 60 degrees
**Drivers:** DH-1M 1-in titanium compression driver; EVS-15M 15-in woofer
**Max SPL:** 134 dB
**Power Handling:** Self-powered; onboard DSP (passive version also available)
**Cabinet:** 15 mm wood
**Size (h x w x d):** 27 x 17 x 16.9 inches
**Weight:** 53.8 pounds
**Mounting:** Rigging points, pole mount

## BASSBOSS DV8

www.bassboss.com

The DV8 is an elegant and extremely versatile loudspeaker that delivers a combination of flat frequency response, high output, broad coverage and long throw from a very compact, lightweight and visually attractive loudspeaker. Frequency response is +/-1.5 dB from 50 to 18,000 Hz.

Self-powered by a 1,500-watt amplifier, the DV8 is capable of extremely high output, with a full and rich presence. It can run on mains voltage from 90 to 250 volts, and is compatible with all power supplies worldwide.

Designed for the professional who moves gear on a regular basis, the ergonomic handles incorporated in the sides, top and bottom of the cabinet extend past the amplifier to protect it in transit. Providing remarkable bass output for a speaker of its size, the DV8 can replace much larger, heavier cabinets.

**TECHNOLOGY FOCUS:** Extremely wide coverage is made possible through use of a line array waveguide, which along with the vertical alignment of the drivers, helps to constrain vertical coverage to a nominal 15 degrees. The tight vertical pattern provides more consistent coverage over greater distances and minimizes reflections to enhance intelligibility.

**OF NOTE:** The cabinet incorporates a 4-degree down-angle when deployed on a pole stand or resting on a flat surface. The down-angle ensures even coverage over the additional distance afforded by the tight vertical coverage pattern and the line array waveguide.

### KEY SPECIFICATIONS

**Frequency Response:** 50 Hz - 18 kHz (+/-3 dB)
**Dispersion (h x v):** 120 x 15 degrees
**Drivers:** 1.7-in diaphragm compression driver on isophasic waveguide; 2 x 8-in neodymium woofers
**Max SPL:** 123 dB (short-term)
**Power Handling:** Self-powered; also onboard DSP
**Cabinet:** Baltic birch
**Size (h x w x d):** 22.5 x 18 x 10.7 inches
Weight: 41 pounds
Mounting: Rigging points, pole mount

### RCF HD 32-A » www.rcf-usa.com

**Frequency Response:** 45 Hz - 20 kHz (-3 dB)
**Dispersion (h x v):** 90 x 60 degrees
**Drivers:** 2-in compression driver; 12-in woofer
**Max SPL:** 131 dB (peak)
**Power Handling:** Self-powered, onboard DSP
**Cabinet:** Composite material
**Size (h x w x d):** 25.5 x 15 x 15 inches
**Weight:** 41 pounds
**Mounting:** Rigging points, pole mount

### Danley Sound Labs SM80
www.danleysoundlabs.com

**Frequency Response:** 110 Hz - 22 kHz (- 10 dB)
**Dispersion (h x v):** 80 degrees conical
**Drivers:** 1.4-in coaxial compression driver; 12-in woofer
**Max SPL:** 134 dB (peak)
**Power Handling:** 400 watts continuous
**Cabinet:** Polyurea coated Baltic birch plywood
**Size (h x w x d):** 25.5 x 24 x 12.7 inches
**Weight:** 50 pounds
**Mounting:** Rigging points, pole mount

### WorxAudio Technologies MAX 1.5A
www.worxaudio.com

**Frequency Response:** 30 Hz - 18 kHz (-10 dB)
**Dispersion (h x v):** 60 x 60 degrees
**Drivers:** 1.4-in compression driver on waveguide; 12-in woofer
**Sensitivity/Max SPL:** 109 dB/127 dB (peak)
**Power Handling:** 400 watts LF/100 watts HF
**Cabinet:** Baltic birch
**Size (h x w x d):** 25.2 x 18 x 15 inches
**Weight:** 81 pounds
**Mounting:** Rigging points

### Tannoy VP 12 » www.tannoy.com

**Frequency Response:** 70 Hz – 25 kHz (-3 dB)
**Dispersion (h x v):** 90 degrees conical
**Drivers:** 12-in driver (proprietary Dual Concentric)
**Max SPL:** 127 dB (peak)
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** Birch plywood
**Size (h x w x d):** 19.1 x 14.6 x 14.1 inches
**Weight:** 41.9 pounds
**Mounting:** Rigging points, optional pole mount

## Outline DVS 12 SP » www.outlinearray.com

**Frequency Response:** 44 Hz - 20 kHz (-10 dB)
**Dispersion (h x v):** 77 x 30 degrees
**Drivers:** 1.4-in compression driver on horn; 12-in woofer
**Max SPL:** 129 dB (peak)
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** Birch plywood
**Size (h x w x d):** 22.4 x 13.8 x 13.8 inches
**Weight:** 48.5 pounds
**Mounting:** Rigging points, pole mount

## Ramsdell Pro Audio 15-X-SS
http://ramsdellproaudio.com

**Frequency Response:** 15 Hz - 18 kHz (+/- 3 dB)
**Dispersion (h x v):** 60 x 40 degrees
**Drivers:** 2-in compression driver on horn; 15-in woofer
**Max SPL:** 132 dB (peak)
**Power Handling:** 1200 watts program (LF), 150 watts program (HF)
**Cabinet:** DuraBlast polyurea coated birch plywood
**Size (h x w x d):** 33 x 17 x 19 inches
**Weight:** 80 pounds
**Mounting:** Pole mount, optional rigging points

## Montarbo Fire 15A » www.montarbo.com

**Frequency Response:** 45 Hz - 20 kHz (+/-10 dB)
**Dispersion (h x v):** 50 x 40 degrees
**Drivers:** 1.7-in compression driver; 15-in woofer
**Max SPL:** 130 dB (peak)
**Power Handling:** Self-powered
**Cabinet:** High-impact painted birch plywood
**Size (h x w x d):** 30.7 x 17.2 x 17.2 inches
**Weight:** 59 pounds
**Mounting:** Pole mount, optional rigging points

## Alcons Audio VR12 » www.alconsaudio.com

**Frequency Response:** 50 Hz - 20 kHz (-10 dB)
**Dispersion (h x v):** 90 x 40 degrees
**Drivers:** 6-in ribbon driver on rotatable waveguide; 12-in woofer
**Sensitivity/Max SPL:** 100 dB/132 dB (peak)
**Power Handling:** 450 watts (RMS), 1500 watts (peak)
**Cabinet:** N/A
**Size (h x w x d):** 26.5 x 13.8 x 13.6 inches
**Weight:** 39.6 pounds
**Mounting:** Rigging points, pole mount, optional U bracket

## Carvin TRX2115A » www.carvinaudio.com

**Frequency Response:** 54 Hz - 20 kHz (- 10 dB)
**Dispersion (h x v):** 80 x 50 degrees
**Drivers:** 1.4-in-exit compression driver; 15-in woofer
**Max SPL:** 133 dB (peak)
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** 15 mm Baltic birch
**Size (h x w x d):** 25.7 x 18.5 x 14.5 inches
**Weight:** 55 pounds
**Mounting:** Rigging points, pole mount

## dB Technologies DVX P12 » www.dbtechnologies.com

**Frequency Response:** 70 Hz - 19 kHz (+/- 3 dB)
**Dispersion (h x v):** 90 x 40 degrees
**Drivers:** 1.4-in compression driver on rotatable horn; 12-in woofer
**Max SPL:** 135 dB (peak)
**Power Handling:** 500 watts RMS, 1,000 watts program
**Cabinet:** Multiplex birch plywood
**Size (h x w x d):** 24.6 x 14.6 x 13.5 inches
**Weight:** 51.1 pounds
**Mounting:** Rigging points

## KV2 Audio EX12 » www.kv2audio.com

**Frequency Response:** 45 Hz - 30 kHz (-10 dB)
**Dispersion (h x v):** 80 x 40 degrees
**Drivers:** 3-in-diaphragm neodymium compression driver; 12-in neodymium woofer
**Max SPL:** 130 dB
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** Baltic birch
**Size (h x w x d):** 23.5 x 14.5 x 14.5 inches
**Weight:** 63.8 pounds
**Mounting:** Pole mount, rigging points

## Yorkville élite EF500P » www.yorkville.com

**Frequency Response:** 50 Hz - 16 kHz (+/- 3 dB)
**Dispersion (h x v):** 60 x 40 degrees
**Drivers:** 2-in-exit titanium compression driver; 15-in ceramic magnet woofer
**Max SPL:** 133 dB
**Power Handling:** Self-powered; onboard DSP
**Cabinet:** Baltic birch
**Size (h x w x d):** 28.2 x 18.6 x 14.5 inches
**Weight:** 37.6 pounds
**Mounting:** Rigging points, pole mount

# NEWSBYTES

## :: The latest news from ProSoundWeb.com ::

### PEOPLE

**PreSonus** has appointed **Blaine Wilkins** as inside sales coordinator for the company's **WorxAudio** division, where he is responsible for interfacing with prospective customers and providing support for the company's dealers and sales representatives. Previously he served as an audio engineer for Advanced Audio & Stage Lighting of Denham Springs, LA, where he was directly involved in system design, integration, programming, and installation.

**Adamson Systems** has expanded its support team with the addition of **Jeremiah Karni** as an applications engineer, where he is assisting customers with system design and providing product support and training. Before joining Adamson, he worked freelance in a variety of roles, including front of house and monitor engineer, as well as system tech, for acts such as The Black Eyed Peas, Broken Social Scene, Timber Timbre and Brendan Perry (Dead Can Dance), among others.

**Harman Professional** announced the appointment of **David McKinney** as vice president and general manager of the Mixer Business Unit, where he will lead the global operations of the **Soundcraft** and **Studer** brands, based in Potters Bar, England. In total, McKinney has worked for Harman Pro for 11 years in four different country offices in contributing to the company's growth. Filling McKinney's previous position of senior director and general manager of China operations for Harman Pro will be **Frank Xiao**, who brings more than 15 years experience in the pro AV industry to his new role.

In addition, Harman Pro announced a new leadership team for **AKG's** professional business, naming **Karam Kaul** as director of marketing and **Eric Boyer** as vice president of worldwide sales. Both will be based out of Harman Pro's Northridge, CA headquarters and will work closely with the AKG team in Austria.

**Group One Ltd.**, U.S. distributor for **DiGiCo**, has appointed **Sal "Chip" Sciacca** to the post of technical support—DiGiCo. He will be based at the distributor's Farmingdale, NY office, providing national technical support for the manufacturer in conjunction with DiGiCo U.S. technical manager **Taidus Vallandi**. Prior to joining Group One, Sciacca most recently served as systems technician and audio engineer for the Grammy Award-winning Tedeschi Trucks Band, which tours with a pair of DiGiCo consoles (SD10 at front of house and SD8 for monitors).

**Grundorf Corporation**, parent company of **Grund Audio Design**, has announced the appointment of **Bob Tjarks** to the newly created position of account manager. Based at the company's Council Bluff, IA headquarters, he is responsible for maintaining existing relationships with North America retail dealers and contractors, identifying new sales opportunities, establishing new dealers, and developing social network resources. Tjarks brings a proven track record in both retail sales and facilities management to the position, previously serving as assistant retail manager/pro audio department coordinator for New Berlin, WI-based Cascio Interstate Music.

**Meyer Sound** has promoted **Michael Creason** to the newly created position of product manager—system applications and training, while **Ashley Hanson** moves to design services manager. Creason joins the growing product management group to expand the company's technical content and training resources.

In her new management role, Hanson leads a design team that works with audio consultants and integrators to create system designs for pro audio, cinema, and other commercial markets. Both Creason and Hanson are based at the company's Berkeley, CA headquarters.

## COMPANIES

The **Solid State Logic** Live console partner network has gained a new partner with the appointment of **Morris Light & Sound** of Nashville. For its first act of business with SSL, Morris Light & Sound will supply an SSL Live L500 for Kenny Chesney's upcoming The Big Revival Tour, which kicks off in late March.

**FE Live Audio**, a growing sound company based in Glasgow, Scotland, recently added a **d&b audiotechnik** V-Series rig to its inventory, acquired from local d&b distributor **The Warehouse Sound Services**. FE Live Audio is headed up by **Andrew McMillan** and **Ryan McIlravey**, who have been partners in enterprise since their school days. Still both only 25, they've recently appointed a new co-director, **Andrew Baillie**, who brings system tech expertise.

**L&M Sound & Light**, based in Staten Island, NY, has been named the exclusive distributor of K-array Firenze and Concert Series products as they make their official debut in the U.S. L&M Sound & Light is headed up by **Lou Mannarino** (audio engineer for The New York Philharmonic) and has been utilizing K-array loudspeakers for several years in the tri-state area, including installations at The Metropolitan Museum of Art, Lincoln Center for the Arts, Columbia University, American Museum of Natural History and MOMA. ■

# What Is Equalization?

Striving for clarity amidst the terminology.

***by Pat Brown***

AS I WAS PREPARING for our recent OptEQ workshop in Dallas, a few things crystallized for me, the most fundamental being "what is equalization?" If you ask 10 audio practitioners this simple question, you're likely to get 10 different answers.

In the most literal sense, equalization means to "make equal." But make *what* equal? Here are some thoughts for consideration.

## LITERAL DEFINITION

In the strictest use of the term, equalization applies to the correction of minimum phase aberrations in the transfer function of a device (usually electro-acoustic) using minimum phase filters. This could be called "corrective" equalization, because the anomaly is removed, as though it never existed.

This is an "audio world" definition. Unfortunately, we don't have exclusive rights to the term. Even so, some practitioners want to limit the term "equalization" to this use of filters only.

## EQUALIZERS

The filter banks, analog or digital, that precede the power amplifiers are called "equalizers." Graphic or parametric, they're used for spectral shaping of the response. These filters may be used with no regard as to whether they're "corrective."

## PROGRAM EQUALIZATION

The channel strip of a mixer includes filters for "program equalization." These can be technical, such as a high-pass filter used to band-limit a microphone, and they can also be artistic, used to change the sound of a vocal or instrument microphone. This is always done by ear, and it is subjective, yet "program equalization" is a very important part of the operation of a sound system. It gives the system operator some control over the way things sound without touching the "house EQ," which should only be used to establish a neutral system response.

## ADAPTIVE EQUALIZATION

Some equalizers have intelligence and can establish their own response curve based on an ongoing measurement. "Adaptive equalization" is very much a part of modern communications, and is used to compensate (not correct) for the dispersion (time smearing) of signals in a communications channel. This may be the most common form of equalization that most of us encounter, given the proliferation of cell phones and digital communication links.

## ROOM EQ

Dr. C. Paul Boner applied notch filters to compensate for room resonances. A resonance is a frequency that a room stores longer than others, providing some natural amplification. This in turn produces tonal coloration for the listener. The use of room resonance compensation filters is probably the origin of the term "room EQ."

Of course, these filters are not corrective, and those that see equalization as only corrective are greatly distressed by the suggestion of electronic room equalization. Since the term "room EQ" isn't going away, expanding the definition of equalization to include "compensation" makes the term work.

## GENERAL DEFINITION

In light of these facts, a general definition is in order. Audio equalization is the application of filters, either analog or digital, to the audio signal.

"Equalization" can be made more specific with a descriptor, such as corrective, program, or adaptive equalization. So, there are many forms of equalization. This is one case where Wikipedia gets it mostly right: *"Equalization is the process of adjusting the balance between frequency components within an electronic signal."* This "balancing" can include magnitude, phase, or both. The target response may be flat or it may be a curve, such as the "EQ" setting on a media player.

To limit the term equalization to any one of these specific applications causes confusion, as we are then required to create new terms for the others.

SynAudCon recognizes the "application of filters" as equalization, and filter banks as "equalizers." We teach the specific uses of equalizers in our seminars, and we encourage audio practitioners to make their intended meaning of the term clear by the context in which they use it.

*Editor's Note: We present this discussion both as a means of offering clarity on a much-discussed topic as well as a lead-in for a report on the OptEQ workshop by Kent Margraves in the next issue of* LSI.

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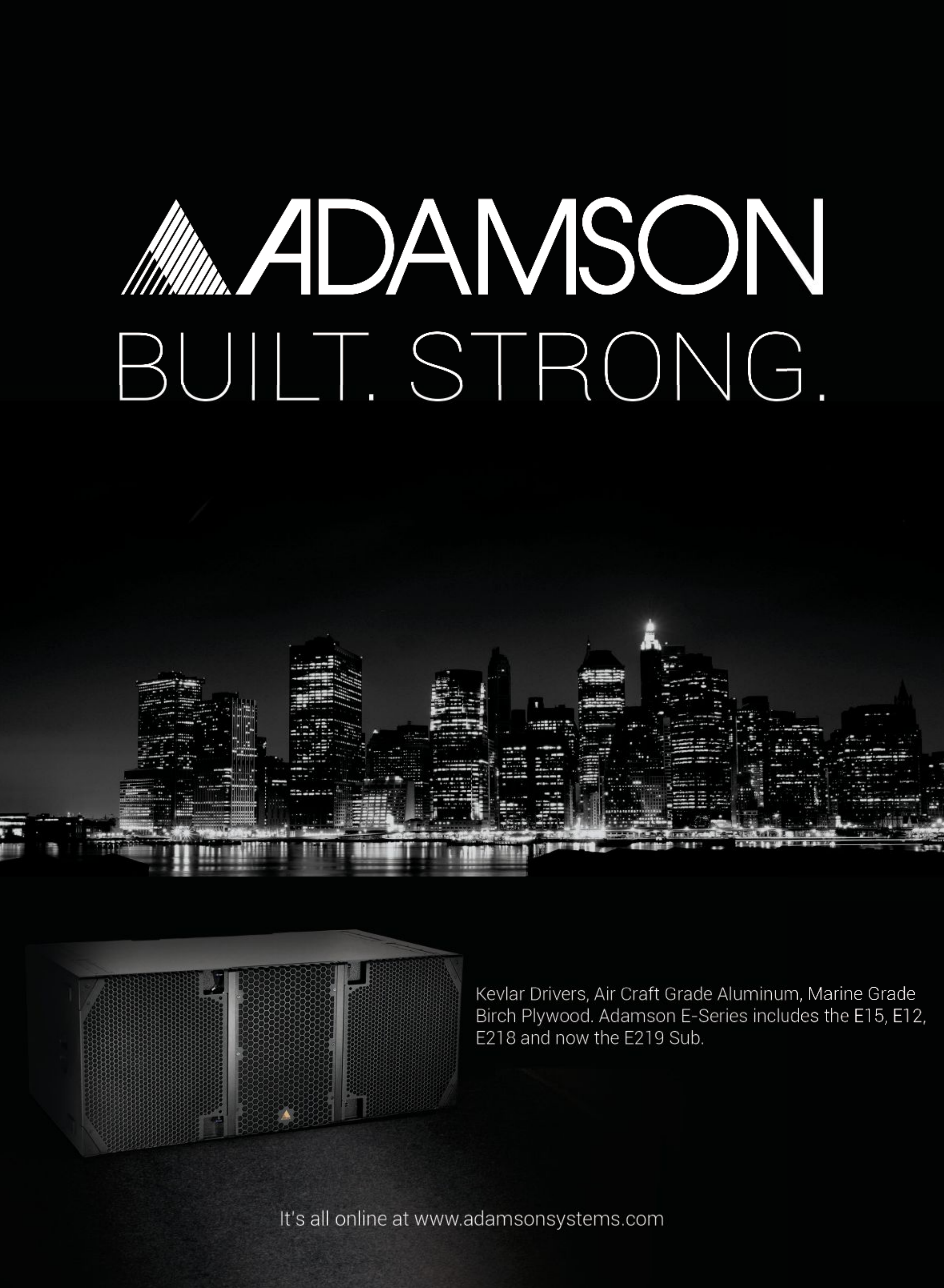
PAT & BRENDA BROWN *lead SynAudCon, conducting audio seminars and workshops online and around the world. For more information go to www.prosoundtraining.com.*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