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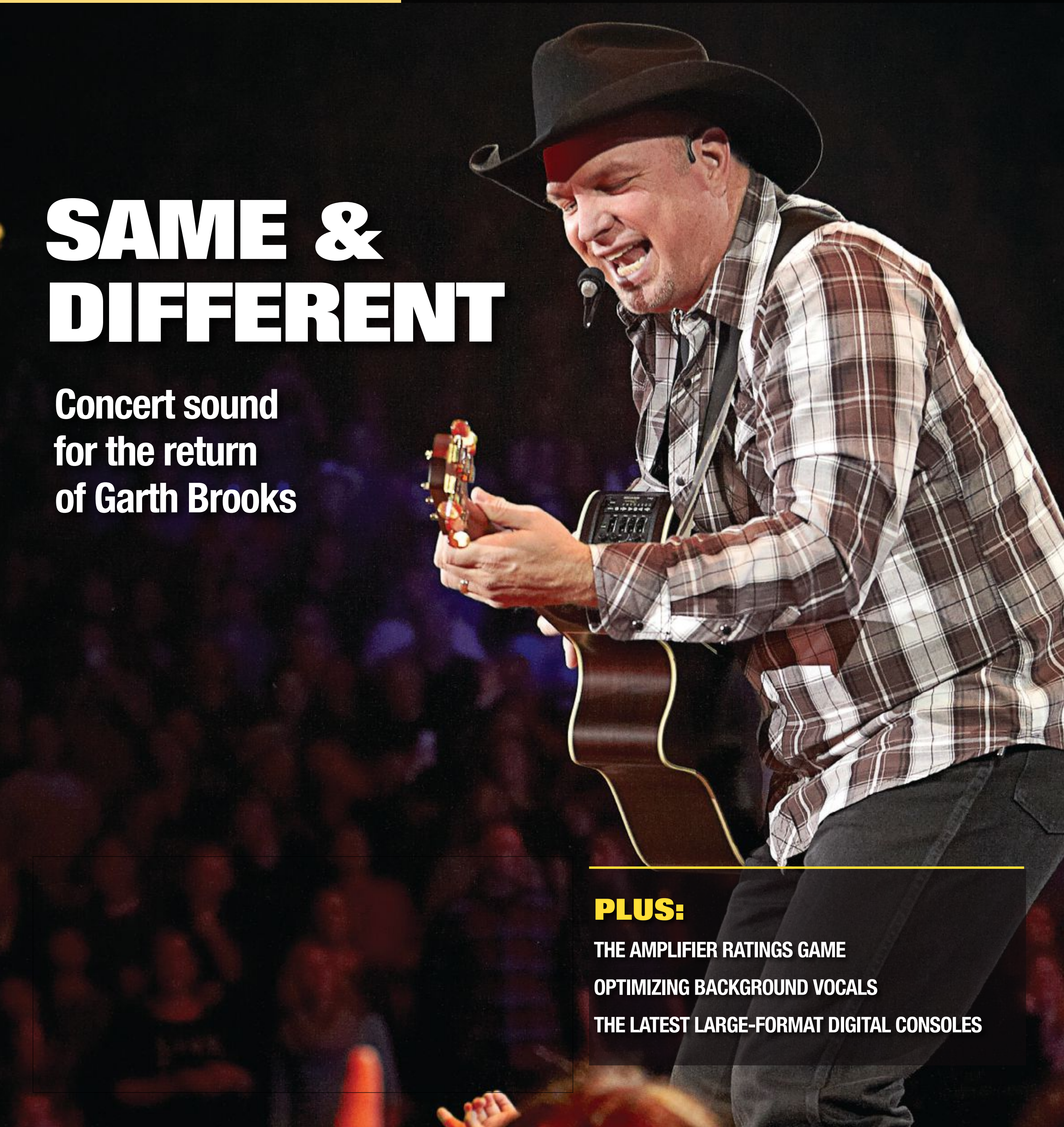

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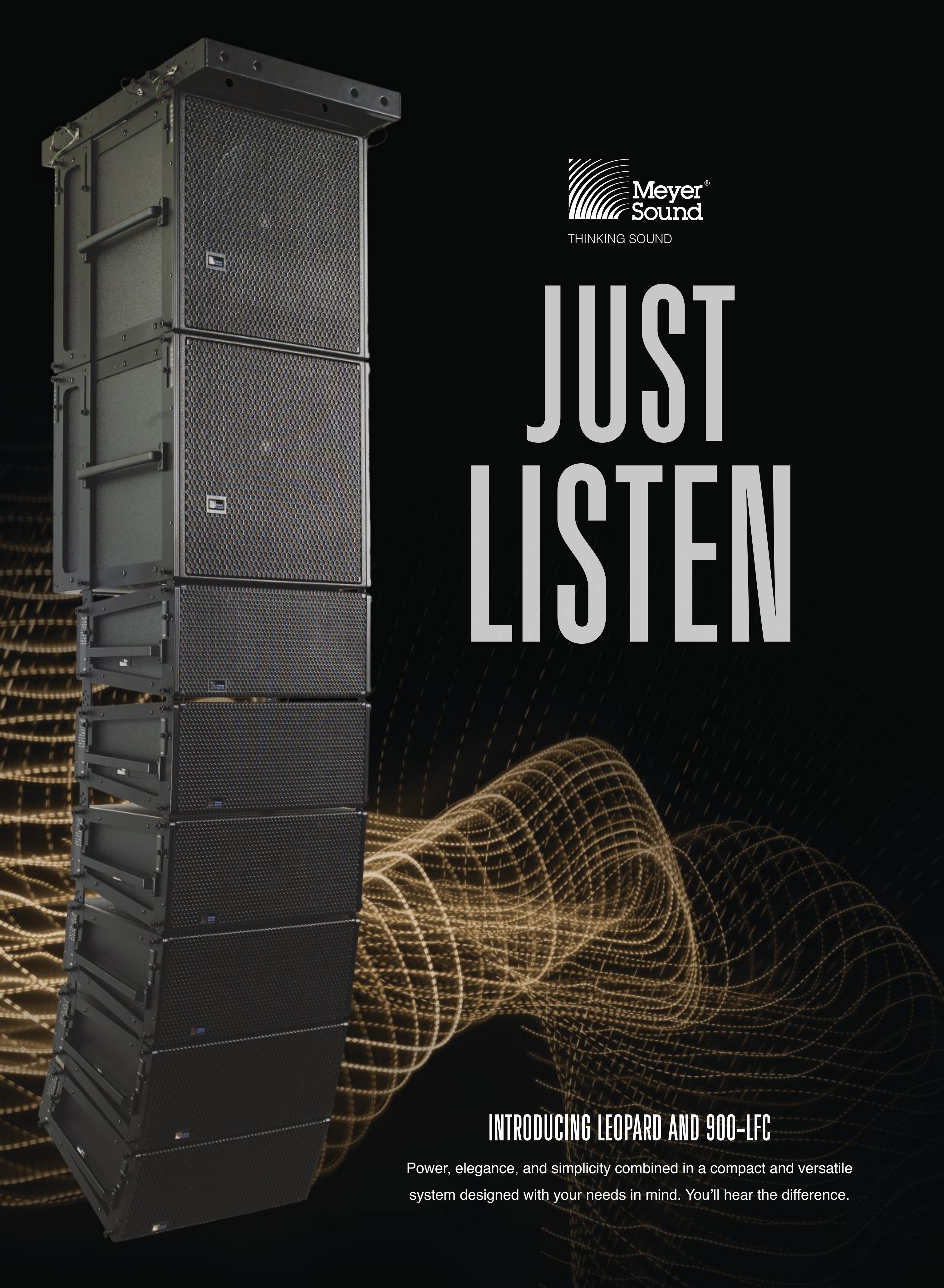
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FEATURES

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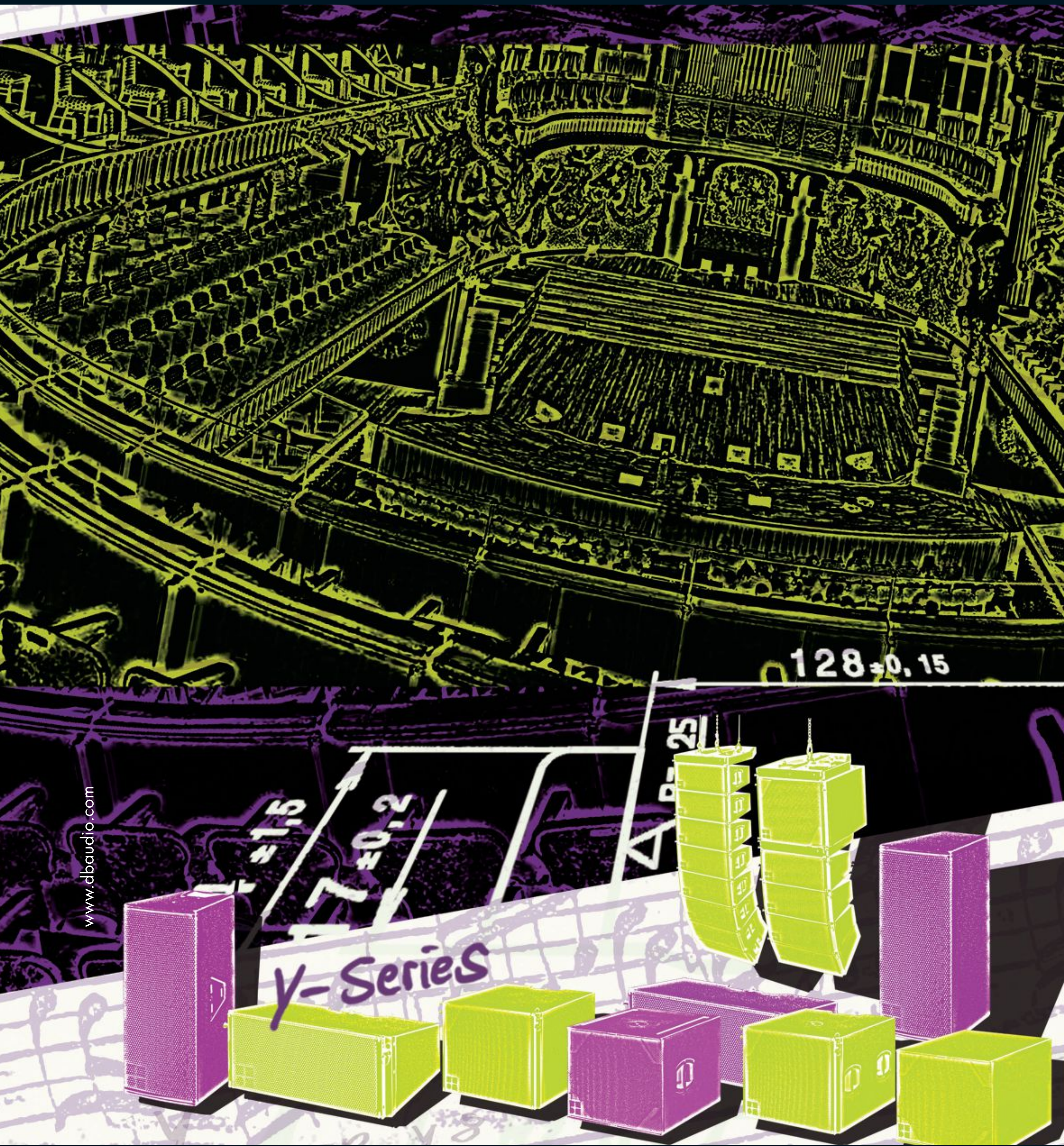

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From the Editor's Desk…

Well, this was a fun one to put together. A highlight of my particular vocation is being afforded the opportunity to work with some truly outstanding contributors.

Note that word "contributors." I use it purposefully, because the people who write for *Live Sound International* do much more than write. Their unique knowledge and invaluable experience go beyond words on a page to convey important, relevant and useful ideas and information that help further the science and art of sound reinforcement (and related facets).

For example, Bob McCarthy provides an in-depth discussion on what really occurs when a line of loudspeakers is "broken" or tapered in some way. It's fantastic to be able to understand what's actually happening as Bob translates subjective descriptions into facts, backed up by his own decades of work in the field.

Meanwhile, Joe Brusi contributes another piece of the puzzle in looking at loudspeaker measurement – how it's changed, and further, how current approaches can be understood and perhaps improved. He also offers a brief side discussion on the term "point source."

Pat Brown takes on an issue that's been the source of much discussion since at least the 1970s: amplifier power. He goes in-depth about what power ratings actually mean and offers a solution in obtaining a better idea of an amplifier's true performance.

Also in this issue, new senior editor Erik Matlock catches up with the mix engineers working with Garth Brooks on his triumphant return to concert touring, and Craig Leerman steps up with two informative pieces about best practices with background vocals and sound check.

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

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

ON THE COVER: Garth Brooks performing in his return to concert touring after a long hiatus. ***(Photo by Steven Wolf)***

LIVE SOUND INTERNATIONAL

VOLUME 24 | NUMBER 6

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Live Sound Customer Service
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Phone: 800-375-8015, ext 294
(Outside the U.S.: 508.663.1500 x294)
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A passive 2-way stage monitor that can also be pole-mounted on a subwoofer to serve as a small full-range system. The 15-inch neodymium cone driver and coaxially mounted 1.4-inch compression driver share the same magnet structure, fostering a compact design for the marine plywood enclosure. Frequency response is 55 Hz to 18 kHz, with a conical dispersion of 75 degrees. MAX2 can be driven by d&b amplifiers using the MAX2 preset or by other linear power amplifiers when integrated with existing stage monitors. **www.dbaudio.com**

Kaltman Creations RF-ResQ

An antenna signal processor designed to help with RF congestion and "spectrum squeeze" issues in working with analog and digital wireless systems in the 470 to 928 MHz range. Housed in a rugged 1RU package, it's installed between wireless antennas and receivers. Multiple military-spec, high-Q band-pass filters clean up the received RF spectrum, allowing for closer adjacent channel spacing, removing intermodulation effects, and improving reception. An onboard 8-channel distribution amplifier with RF router allows the filters to adapt to various receiver and distribution configurations. This provides one filter/frequency per receiver channel. Users can also select "combined signal routing" to feed integrated receiver/distribution systems. RF-ResQ assigns a single, frequency-cleaned-up, band-pass filtered feed for each transmitter's frequency, and its eight amplifiers can maximize individual RF signals up to 10 dB. DiverseQ antenna diversity technology pre-filters the antenna A/B signals in the IF stage for fast, accurate and quiet antenna switching. Frequency, gain and routing assignments are performed via a LAN or USB/laptop connection to the rack unit. **www.KaltmanCreationsLLC.com**

Soundcraft Si Impact

A 40-input digital console with ViSi iPad control and built-in Stagebox connectivity for I/O expansion. It provides 32 mic/line inputs, 32 mono inputs, 4 stereo channels/returns, and 31 output buses (all with DSP processing and GEQ) with 20 sub-group aux buses and 4 mono/stereo matrix buses. Eight combi-jacks are available for line inputs and instruments, while a 4-band parametric EQ is available for each channel and bus. Also onboard are effects and dynamics from BSS, Lexicon, and dbx. The console supports up to 8 VCA masters and 8 mute groups, plus 26 motorized input faders and LR/mono (motorized with 4 customizable fader layers). A 5-inch color touch screen display provides access to show setup, patching, FX, and security. The motorized faders come with Soundcraft FaderGlow illumination and LCD channel displays. A 32-in/32-out USB recording/playback interface facilitates multi-track recording and playback directly from a DAW, with a free download of Ableton Live 9 Lite provided. **www.soundcraft.com**

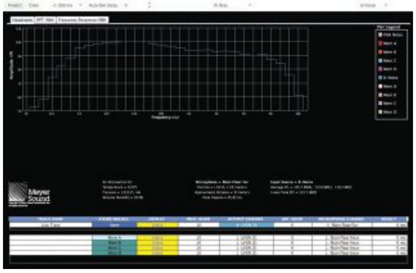
DiGiCo Orange Box

An audio format converter in a 2RU package with multiple options for DiGiCo Multichannel Interface (DMI) cards to create audio paths over a variety of interfaces. It's outfitted with 2 PSUs for redundancy and 2 slots to accommodate any of the 10 currently available interfaces, including Dante, Hydra2, BNC, Cat-5, Optocore, Aviom, ADC, AES, DAC and SoundGrid. For example, linking a device with MADI to one with Hydra2 requires just a DMI card with Hydra2 and a DMI card with MADI, with connection via the 2 slots on the Orange Box. **www.digico.biz**

Waves H-Reverb

A reverb plug-in based on a Finite Impulse Response (FIR) engine that allows shaping and customizing the reverb decay envelope beyond standard linear forms for gated, real reverse, and dense reverb tails. Advanced analog modeling along with a drive control delivers signal behavior found in high-end hardware reverb devices. The design also enables input drive. An EQ and dynamics module provides compression, ducking and de-essing. H-Reverb also includes a library of artist presets from leading mix engineers as well as presets inspired by vintage and modern classics. **www.waves.com**

Meyer Sound MAPP XT →



In addition to an updated user interface and additional features, MAPP XT shows the maximum levels to which Meyer Sound loudspeaker systems can be driven while maintaining linear performance. Headroom information can be viewed from the perspectives of 2 different input signals, including traditional broadband pink noise as well as B-Noise, a new input signal designed to better represent the most common input spectrum. In addition, the Auto-Splay feature allows users to select the starting and finishing angles and automatically calculates the optimum splay angles between cabinets. Factor of safety informs as to whether a proposed LEO or LYON array is within limits for the given configuration and angle to help users meet safety standards, including 5:1, 7:1, and BGV-C1. Trace store and recall provide fast comparisons of different configuration options.**www.meyersound.com**

Mackie DL Dante Expansion Card ↓

Provides 32 x 32 channels of network audio I/O that allow the company's DL32R wirelessly controlled digital mixer to be connected to any Dante-powered AV network. Flexible I/O patching in the DL32R fosters the routing of Dante signals to any channel input and sending any of the mixer's outputs to the network.

The card ships with a license for Dante Virtual Soundcard software, allowing direct recording/playback of up to 64 channels of audio from any Dante network to a PC or Mac computer. **www.mackie.com**

FaitalPRO 18XL1800 →

An 18-inch cone woofer designed to reinforce low frequencies in the 20 Hz to 30 Hz range. It provides mechanical excursion that is at least 1.5 times greater than the company's other 18-inch woofers. Rated power handling is 1,600 watts (3,200 watts maximum, AES standard), and sensitivity is stated as 95 dB. Two demodulation rings reduce unwanted harmonic distortion and modulation of inductance. The neodymium magnet assembly implements a "sectors" structure that substantially reduces the overall weight of the assembly without impacting performance. The 18XL1800 also offers a high-winding voice coil, very large spiders (over 230 mm) and a wider cone edge. The waterproof cone is made of cellulose pulp mixed with synthetic fiber, and it also has a large thermoformed and treated border as well as an efficient cooling system. Precise symmetry and strong suspension compliance (even with large signals) result in exceptional functional stability of the moving parts over a very extended excursion. **http://faitalpro.com**

dB Technologies DVX P Series ↓

A line of install-specific passive loudspeakers comprised initially of 2-way models with 8-, 10-, 12- and 15-inch woofers, as well as 15- and 18-inch subwoofers. The 2-way models have rear-cabinet angle cuts of 20 and 40 degrees for monitor applications, 12 M8-thread rigging points, 2 side-mount fly-pins for wall mounting, plus a pole-mount cup.

The subwoofers are outfitted with 2 lateral handles for transport, a standard M20 pole-mount plate, the predisposition for wheels on the bottom, and recesses on the top to facilitate the stacking of 2-way loudspeakers or another subwoofer. **www.dbtechnologies.com**

Pivitec CMx64io-AVB →

A network card designed to add AVB connectivity to pro audio equipment that supports the industry standard CM-1/CM-2 card format. It has a USB 2.0 interface and 10/100/1000 Ethernet. USB 2.0 allows the host device to act as a USB audio interface for Mac and PC with simultaneous Ethernet AVB audio streaming. It also facilitates "daisy-chaining" of multiple audio devices to a single Gigabit Ethernet connection. As a result, users can employ a simpler network topology. The card hosts audio and control I/O signals, which permits customization for a range of applications. The board supports 802.1Ba, 802.1AS, 802.1Q, 1722.1 and 1722 AVB standards. Able to manage up to 64 x 64 channels, the card is field firmware updateable and provides WiFi control and configuration. Sample rates range from 44.1 to 192 kHz. The CMx64io-AVB is available as part of the company's Ethernet AVB Hardware Reference Design Kit (XRDK). **www.pivitec.com**

RCF L-Pad 16CX USB & L-Pad 24CX

Two 4-bus mixers joining the L-Pad line, each with 4 stereo inputs and 4 group outputs, 4 aux sends per channel (2 pre, 2 post), 3-band sweepable EQ on the mic/line input channels (4-band fixed EQ on stereo line channels), dynamic compressors on select mic input channels, 24-bit DSP effects with 100 presets, 60 mm faders, USB interface onboard for computer software and monitoring, plus a slot for option cards (MP3 player/recorder or Bluetooth connection). The L-Pad 16CX USB offers 16 channels, 10 mic/line inputs with balanced XLR (6 with dynamic compression) or alternatively 8 mono line inputs with insert send/return, plus 4 channels of stereo inputs. The L-Pad 24 CX USB has 24 channels, 18 mic/line inputs with balanced XLR (14 with dynamic compression) or alternatively 16 mono line inputs with insert send/return, plus 4 channels of stereo inputs. A USB connection fosters 2-channel, 16-bit bi-directional audio for recording/playback via computer. **http://rcf-usa.com**

AirNetix AiRocks Pro

A multi-hop repeater system designed to transmit wireless audio to remote powered loudspeakers and amplifier racks, operating in the unlicensed 900 MHz band. It's particularly well suited for delay stacks in applications such as concerts, festivals, golf tournaments, parades, air shows, auto races, and more. The company states that the 900 MHz band feature penetrates walls, trees, people and other obstructions that often limit higher frequency devices that operate at 2.4 GHz. AiRocks Pro also includes built-in variable 500 ms delay, 158-398 mW of effective transmitting power, range of more than 1,000 feet, and XLR line-level audio input and output. It's housed in a weather-resistant aluminum outdoor enclosure. In addition, a network management system provides a toolbox of real-time monitoring and control functions, including the ability to monitor the receive signal at each remote unit from a central laptop, as well as to control output volume, add audio delay, and perform spectrum analysis at a remote site. **www.airnetix.com**

FBT VERTUS Series

A full-range active column loudspeaker (CLA406A) and active subwoofer (CLA118SA) for live and install applications. The CLA406A, with a stated frequency response of 65 Hz to 20 kHz, is outfitted with 4 custom 6.5-inch cone woofers and a 1.4-inch neodymium compression driver on a waveguide with dispersion of 100 by 25 degrees (h x v). The height of the waveguide fosters vertical control throughout the operating range of the driver. Via attachment points on the cabinets and optional hardware, 2 CLA406As can be coupled with vertical angles of 0, -10, and -20 degrees. The class D amplifier delivers 600 watts to the LF and 300 watts to the HF. DSP includes 8 presets. A control panel provides XLR input and link, volume, presets, high-pass filter, and ground-lift. The CLA118SA sub incorporates an 18-inch woofer (3-inch voice coil) in a bass-reflex design, with frequency response down to 33 Hz. It's also driven by a class D amplifier and outfitted with DSP with 8 presets, including cardioid configuration. Both models have birch plywood enclosures with internal metal bracing for added stability. **http://fbtusa.com**

JBL Professional VerTec DP-DA V5

Version 5 preset support is now available for DrivePack DP-DA input modules for all VerTec full-range and subwoofer models. V5 leverages the same OMNIDRIVEHD linear phase FIR processing capability of both Crown I-Tech HD DSP power amplifiers and DrivePack DP-DA input modules. V5 processing utilizes higher-order asymmetric filters and linear phase processing to improve horizontal coverage and tonal balance consistency. In addition, sound quality is improved through the use of arbitrary coefficient FIR phase linearization, also improving system response to equalization, far-field summation and throw, and inter-array interaction/summation and stereo imaging.

Recently released JBL HiQnet Performance Manager version 1.8.4 provides full support for deploying VerTec DrivePack DP-DA with V5 presets. **www.jblpro.com**

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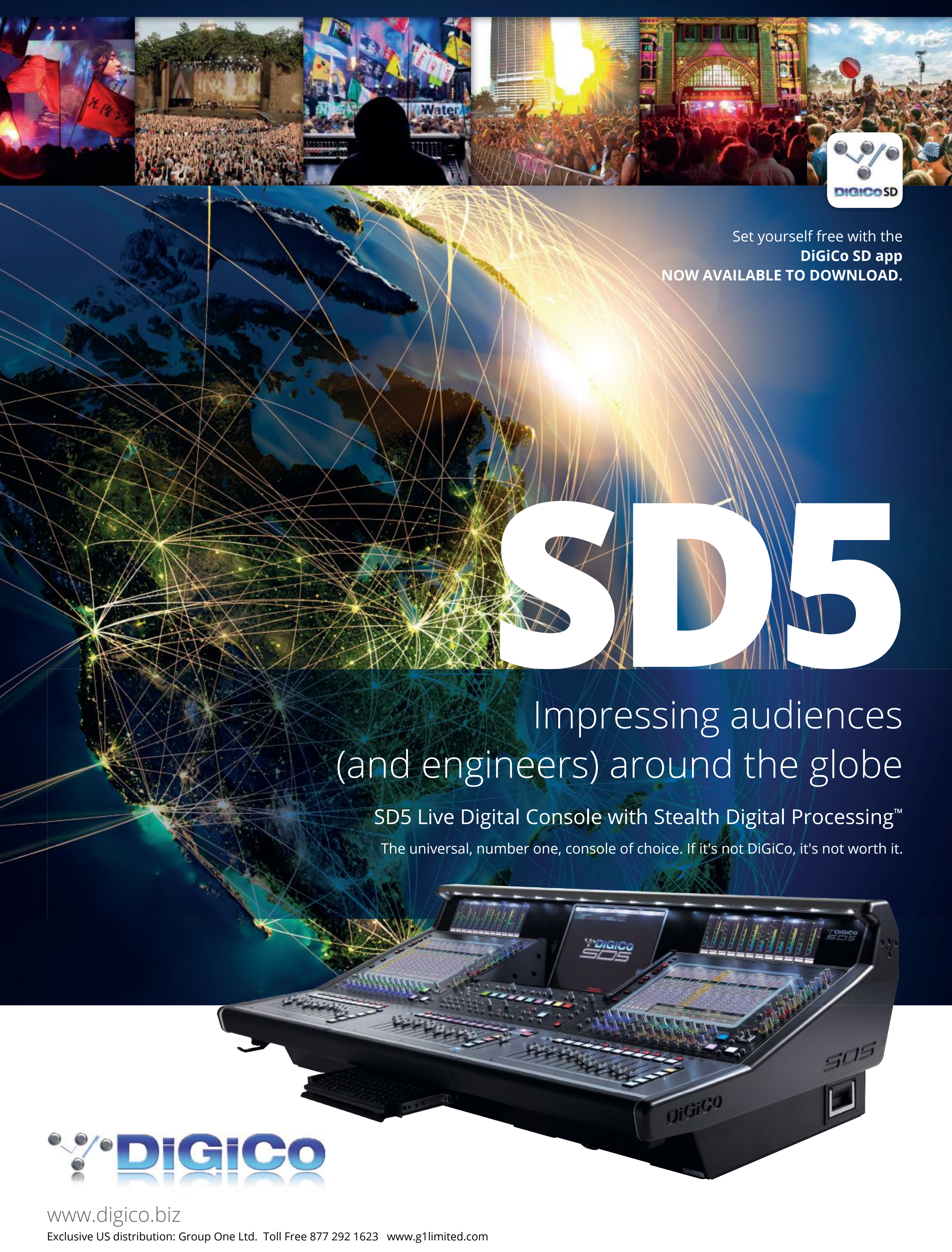
CADAC Waves Integration →

CDC six and CDC eight digital consoles include an integrated 64-channel I/O interface card providing connectivity to the Waves MultiRack SoundGrid via Cat-5e cable. Audio is streamed from the console to the SoundGrid server, processed, and then streamed back to the console. This allows up to 64 channels of Waves plug-ins to simultaneously run alongside the console's own native effects options, with the Waves GUI displayed on the touch screen for control. The interface can connect directly to a standard laptop computer, which, with the Waves SoundGrid Studio application installed, will connect to most types of DAW software, permitting the use of third-party plug-ins, multi-track recording and playback, and virtual sound check. The Waves processing is patched into the console in the same way as any CADAC I/O rack, and can be used as a send and return for effects processing, or alternatively, can be patched and used as inserts. **www.cadac-sound.com**

AKG D112 MKII →

A new version of the original kick drum microphone outfitted with an integrated flexible mount. The pivoting, rotating mounting system now enables the mic to be better positioned regardless of application. The cardioid dynamic design is specified as capable of handling more than 160 dB SPL, with a large diaphragm that has a very low resonance frequency delivering response below 100 Hz. Low end is complemented by a narrow-band presence boost at 4 kHz. The D112 MKII can also be used for miking bass cabinets and trombones. **www.akg.com**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Garth Brooks and his band live in concert this year after a long hiatus.

CREDIT: STEVEN WOLF

THE SAME AND DIFFERENT

Sound for the return of Garth Brooks to concert touring.

By M. Erik Matlock

GARTH BROOKS is back. Following a 13-year hiatus from touring, the Garth Brooks World Tour with Trisha Yearwood has been on the road for several months in support of his latest album, "Man Against Machine," continuing on an open-ended basis in selling out arenas across North America.

Brooks is certified by the RIAA as the number 1-selling solo artist in U.S. history with over 135 million albums. "Man Against Machine," his ninth studio album that was released last November, marks the 14th time in his career that one of his albums debuted at number 1 on the country charts. Meanwhile, co-bill Trisha Yearwood, who wed Brooks in 2005, is a top performer in her own right, with eight number 1 singles and twenty top 10 hits.

Brooks has also been a top draw on the concert circuit, noted for high-energy live performances of his unique traditional country infused at times with pop/rock flair. His recent return to touring picks up where he left off, greeted by packed houses at every stop, including numerous multi-show engagements.

TIMES HAVE CHANGED

As he has since 1989, Brooks' long-time front of house engineer Dan Heins is providing the artist and band PA mix, while monitor engineer Troy Milner signed on last year after more than a dozen years with Bruce Springsteen. Clair Global is the sound company for the tour.

Things have changed a bit in terms of sound reinforcement technology since the big tours of the 1990s, when both house and monitor desks for Brooks were ATI Paragon analog desks. Now, both Heins and Milner are utilizing DiGiCo SD7s – the first tour ever for Brooks mixed on digital consoles.

Additional modern technology touches can be found in wireless world. The band and Yearwood are outfitted with JH Audio JH16 in-ear monitors fed by Shure PSM 1000 personal monitoring systems. Milner's also using this combination for himself in the course of providing 16 separate stereo mixes in all.

Wireless microphone systems for all performers are Shure Axient, including transmitters, receivers and spectrum management working in tandem with the Shure Wireless Workbench 6 platform that also accounts for the wireless PSM 1000 monitoring systems. It makes coordinating the tour's dozens of frequencies in the increasingly challenging RF environment – particularly present today in metropolitan areas – a much more manageable task for Milner.

"I come in every day, set up the antennas, put in my numbers and hit

Another look at the stage layout on the current tour.

CREDIT: STEVEN WOLF

(Far left) Monitor engineer Troy Milner at his beach with a DiGiCo SD7 console and Shure wireless behind him. (Left) Front of house engineer Dan Heins at the SD7 that's replaced the ATI Paragon analog console he used on the previous tour.

CREDIT: STEVEN WOLF

deploy. The program pretty much does it all for me," he says. "I don't consider myself an RF guy by any means, but the system works absolutely great. We're very happy with it."

Operating in the UHF band, Axient is designed for advanced live concert and event situations, offering comprehensive remote control of all transmitter parameters. Audio is transmitted simultaneously on two independent frequencies, with interference detected in milliseconds. It also provides the ability to continuously monitor, prioritize, queue and assign compatible frequencies.

On stage is a veritable a sea of loudspeakers, including Clair CM22 wedges that Brooks prefers in a big way – to the tune of 44 of them. They're joined by Cohesion 8 side fills (12 per side) and CP218 subwoofers. "The CM22 is a beast. I'm always amazed at what comes out of that box and how great it sounds," Milner says.

Of course, keeping stage levels at the high volume Brooks prefers while maintaining control and delivering quality mixes is a balancing act. "It's essentially one common stereo mix for all 44 stage monitors, but it keeps me busy, constantly riding the faders to keep each zone hot as he moves around on the stage," he adds.

GETTING COMFORTABLE

A primary factor in the selection of the SD7 consoles is their capacity. For example, Milner is running 140 channels and 68 mixes between floor wedges, side fills, stereo in-ear monitors, and talkback systems. All processing, EQ and effects are handled by the console, joined by various plug-ins from Waves.

"I don't use a lot of effects or compression on the mixes," he notes. "We don't want the band playing off compression."

At front of house, Heins' outboard approach is relatively restrained, with a primary goal of keeping Brooks' vocal on top of the mix. An SPL Transient Designer 4 processor helps manage dynamics in the mix without affecting loudness, and an SPL De-Esser removes undesired sibilant frequencies without compromising the natural character of vocals. A Summit Audio DCL-200 hybrid compressor/limiter helps keep things clean while lending a touch of tube "warmth."

The selection of the consoles is also attributed to a newer reason. "The ability to put the DiGiRacks and consoles and everything on one fiber loop was huge for us," says Heins.

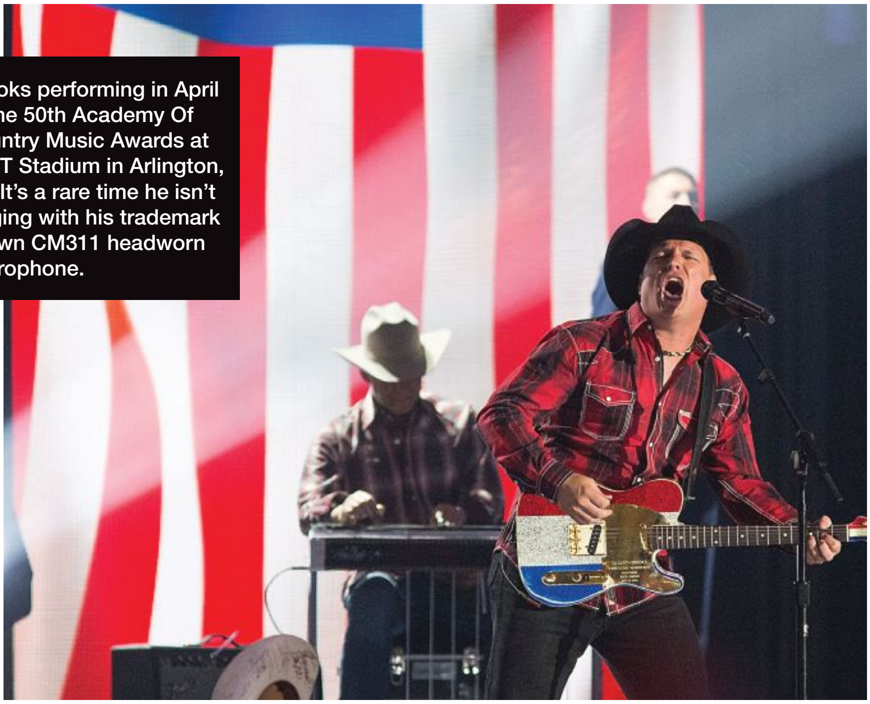
"I switched to the SD7 a few years ago because I needed more ins and outs, and I haven't looked back," Milner adds. "Monty Carlo and I moved to the SD7s on Springsteen's last big tour in 2009 since we had a lot of unknowns about band requirements when we were starting and knew we needed room to grow. I'm glad we made the switch because we added a lot of stuff and it worked out perfectly."

Reaper is implemented for recording for purposes of virtual sound checks. In addition, Pro Tools receives a multitrack feed in record shows at 96 kHz/24 bit into a set of hard drives that are sent for archiving.

"There have been some nights when we really paid attention to the recordings and got some amazing material," Heins notes. "There's also video being shot on the tour, so it only makes sense to multi-track along with that. It doesn't take any more time or effort to record, and since the cost is so low, it only makes sense to capture these shows in case something magical happens."

THE BEST FIT

Brooks is still using his trademark Crown CM311 headworn condenser microphone that he adopted in the 1990s. Linked to an Axient body pack transmitter, it provides both the vocal presence he and the engineers are seek-



Brooks performing in April at the 50th Academy Of Country Music Awards at AT&T Stadium in Arlington, TX. It's a rare time he isn't singing with his trademark Crown CM311 headworn microphone.

CREDIT: RICK KERN/WIREIMAGE

ing while also delivering the feedback rejection required for a show running at high volume.

"We just haven't found anything else that works for him the way the CM311 does," Milner explains. "With the stage volume and crowd noise, it's the best fit and it sounds good too."

Yearwood, meanwhile, sings through her preferred mic, a customized Shure SM58 on an Axient handheld transmitter. Background vocals are also primarily handled by SM58 capsules on Axient handhelds.

Kick drum sound is captured with the combination of a Telefunken M82 dynamic with end address design and a Shure BETA 91A half-cardioid condenser. A Telefunken duo of an M81 and M80 (both dynamics) are applied for snare, with Heil Sound PR28 dynamics on toms. Another Shure (KSM137 end address condenser) and Telefunken (M60 FET cardioid condenser) combo handle hi-hat. More M60 FETs are deployed left and right for overheads on the kit.

It's back to old school on guitar cabinets with Shure SM57s. They're also applied left-right on the Hammond B3's Leslie cabinet, joined by a BETA 91A for low frequencies. Bass is taken direct via a Countryman Type 85 DI, and more direct action happens with keyboard and piano with Radial Pro D8, JD6 and JDI boxes.

UNIQUE PERSPECTIVE

The tour is a beast that harkens back to the epic Brooks tours of a decade-plus ago. Out front is a large-scale rig capable of delivering 360-degree coverage that includes 40 i218M three-way line array elements and an additional 16 i218-LT long-throw elements and 48 i212 medium format line array cabs.

Thirty iS218 (dual 18-inch) subwoofers are flown, and if that's not enough low end, another 16 iS218s are on the floor to further shake the seats. Lab.gruppen PLM amplifiers (up to 80) with onboard Lake processing deliver the audio power to all loudspeakers.

The system can easily reach levels of 105 dB at front of house. "We're mixing for some very excited audiences," Heins states. "The crowd noise is often as loud or even louder than we're pushing."

Heins got his start as a free-lance engineer who first mixed Brooks in 1989. Also, while working with MD Systems, he was system engineer for acts like the Kentucky HeadHunters and Diamond Rio. During the hiatus, he stayed busy working for Clair Brothers Systems out of Nashville, which provides sales and installations.

As noted, Milner was long on the road with Bruce Springsteen (for 13 years, to be specific), and prior to that, he developed his skills with artists such as Backstreet Boys, Sugarland Amy Grant, Michael W. Smith, Sammy Hagar and the Charlie Daniels Band. Primarily a front of house engineer, his personality and skills made him a great fit for the hot seat on monitor beach. And he has a unique perspective having mixed monitors for Brooks and Springsteen, two of the biggest and most fan-beloved artists on the planet.

"They're both amazing artists to work with and they both keep me on my toes," he says. "Both feed off the crowd and it's all about the connection with their audiences, and neither one of them follow the set list. The big difference would be that one wears a hat and the other one doesn't. But on this tour, Garth's vocal is the show, period. When he plays four notes of a certain song on his guitar, the crowds go through the ceiling. It's pretty amazing." ■

M. ERIK MATLOCK is senior editor for Live Sound International and ProSoundWeb, and has worked in professional audio for more than 20 years in live, install, and recording.

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TECH TOPIC

The Ratings Game

Getting a better handle on power ratings

By Pat Brown

THIS ARTICLE IS PARTLY technical, and partly philosophical. It's about power, and has to do with how consumers use power ratings to buy products. It is not necessarily limited to audio, even though that is where it ends up.

There are two primary uses for power ratings:

1) To describe the rate of work performed by (or energy consumed by) a product, and

2) Because of the previous item, to determine whether or not a product can perform a specific task, prior to purchasing it.

Since item 2 will influence the sale of the product, one can see the potential for power ratings to be abused, exaggerated, implied, or sufficiently vague as to render them useless for item 2.

One can wander into the local Home Depot and see many examples. Try making a comparison between vacuum cleaners based on their power ratings. Visit the tool department and compare cordless drills. While you're there, compare some air compressors or welders. In each of these examples, the ratings have been sufficiently obfuscated to render them useless for making meaningful comparisons. The only recourse is to take advantage of the store's liberal return policy and try before you buy. If that is the case, then what is the power rating for?

ONE NUMBER RATINGS

There is something enticing about a single number rating. Since a single number is simple, we think we understand it and that we have the whole picture. The "one number" rating deceptively gives confidence that we have made an informed decision, and we lay our money on the table.

How does this apply to audio? Let's take the (arguably) simplest component in the signal chain – the power amplifier. It's a box with an input, an output, and voltage gain that can source the current flow demanded by the load impedance. If there exists an audio component with the potential for a one-number rating, this is the one. Voltage times current equals power. Amplifiers put out watts of power. More is better. End of discussion? No, it's just the beginning.

SOME HISTORY

Most things in life are cyclic. When you study the past you are studying the future. Each generation is destined to repeat the mistakes of the previous ones when they don't study history.

I'm old enough to remember the "peak power war" of the 1970s. This amounted to an "arms race" between power amplifier manufacturers to claim the biggest power rating. Like all con games, the amplifier ratings game was based on some indisputable facts of physics.

1) The output power of an amplifier can be calculated from the power equations, with the variables supplied by Ohm's Law (**Figure 1**).

2) The output voltage of a power amplifier is a waveform that varies with time, so quantifying it using a single number can be tricky.

Figure 2 shows the sinusoid waveform used to rate a power amp. Sine waves are easy to generate, easy to measure, and provide an objective means of describing the output capabilities of an amplifier. Since the power equation requires the signal voltage, and this voltage is a waveform that changes with time, one can legitimately ask "Which value do I pick?"

Good physics (and the Federal Trade Commission) says that you use the root mean square, or RMS voltage. RMS is a means of averaging a time-varying quantity to yield the power developed into a load. Synonyms for RMS voltage include the waveform's effective value, heating value, and equivalent DC value. In acoustics it is related to the perceived loudness of the waveform.

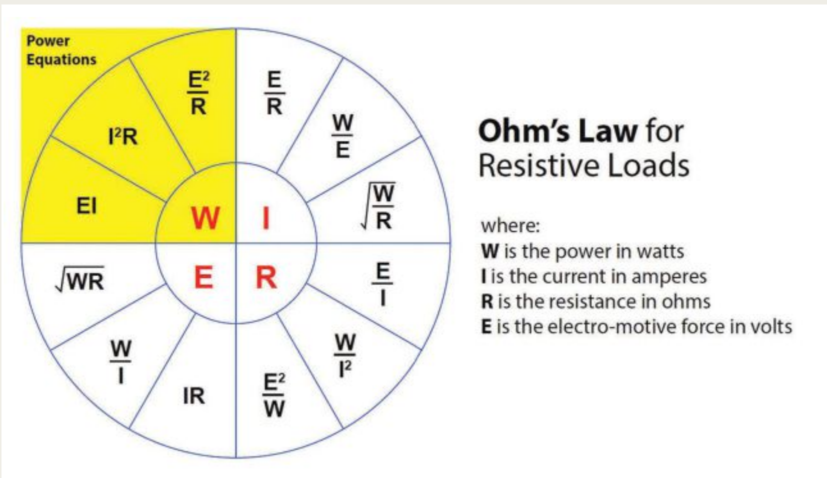

Figure 1

If you connect an amplifier to a load resistor and feed it a signal, the amplified signal voltage develops across the load and current flows as allowed by the load impedance. Since the current can only be varied by changing the applied voltage (the impedance is fixed), the voltage determines how much current flows and how hot the resistor gets. A more powerful amplifier will be able to produce a higher RMS voltage, and the resistor will get hotter. So while it is a power amplifier, the voltage is the variable that we can wiggle.

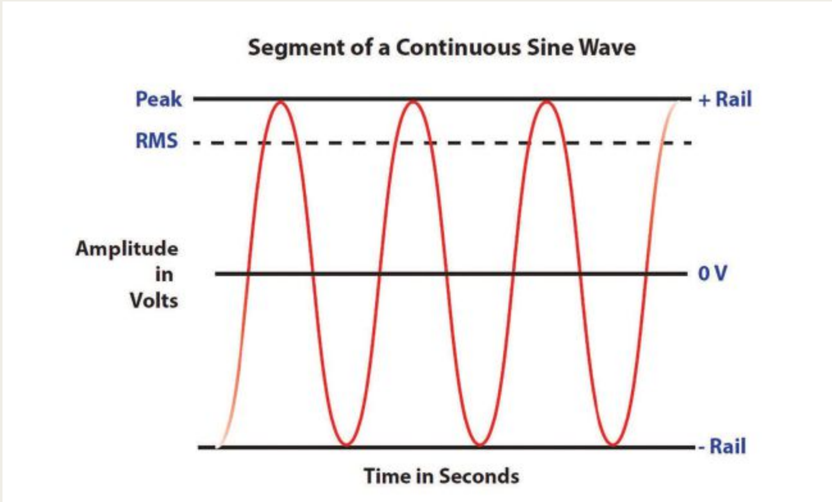

Figure 2

MANIPULATING THE NUMBERS

The marketeers of the day recognized that there are other ways to rate the waveform than the RMS voltage. The crest of the wave is its peak value, so plug that into the power equation and you automatically double your power rating (+3 dB). "Peak power" was born and many manufacturers jumped on the bandwagon, yielding to the temptation (and marketing pressure) to publish a higher, more marketable rating that would sell more amplifiers. Better yet, it required no change to the amplifier's design, just a change on the spec sheet. Putting "peak watts" on the spec sheet did not produce a hotter resistor or louder loudspeaker, it sold more amplifiers.

It should not be surprising that the fine print that stated that it was a peak rating was either omitted or ignored. The public didn't know to ask and the manufacturer didn't tell. Both wanted bigger numbers and "peak power" provided them.

Some went as far as to publish "peak-to-peak" power ratings, which are +3 dB (2x power) than a peak rating, and +6 dB (4x power) the RMS rating. So, a 50 watt amplifier (sine wave rating) could be rated at 100 watts peak, and 200 watts peak-to-peak. But no matter what you called it, the load resistor didn't get any hotter. That's when the FTC stepped in and forced the industry to clean up its specification practices.

FLASH FORWARD TO PRESENT

Most would like to think that in this enlightened, app-driven age where people can be seen sitting in coffee shops and at stop lights continuously absorbing information from their smart phones, that we have moved beyond playing games with rating methods to bolster product sales. Well, generations come and go, but people are still people. Consumers want "more" and manufacturers want to supply it. The ratings game has a modern incarnation that is no less deceptive than the peak power wars of the '70s.

THE BURST TEST

Chances are the largest (and maybe only) rating for a power amplifier that you will find in a brochure or specification sheet is the result of a burst test. Burst testing was developed for cell phones but found its way into the audio world.

In the '70s, the amplifier had to actually sustain the sine wave over time into the rated load resistance. The "game" was played with the voltage value used in the power equation. The burst test pulses the waveform, which greatly reduces the stress on the amplifier.

The test waveform for the CEA2006 burst test is shown in **Figure 3**. The sine wave is full-scale for 20 cycles, and then reduced by 20 dB for 480 cycles, and then continuously repeated. The test frequency is 1 kHz. There is also a 50 Hz version with a different pattern. An amplifier whose output voltage will drop with a continuous sine wave will be able to maintain the full amplitude of the burst waveform.

The amplifier rating is determined by the RMS voltage of the full-scale segment of the waveform, allowing it to masquerade as a continuous sine wave rating. The marketeers of the '70s failed to disclose that their power rating used the peak voltage. Today they fail to disclose that their amplifier's massive power rating is from a burst test.

The burst test has some indisputable merits. I actually like this test as part of an amplifier specification.

- It's more like music and speech than a sine wave.
- It generates far less heat into my dummy load than a sine wave, so

there is less danger of catching the lab on fire.

- It greatly reduces the stress on the amplifier, resulting in fewer blown fuses and thermal failures.
- I can test a "2000 watt" amplifier with a 200 watt dummy load.

Of the battery of tests that we run on power amps, the burst test yields the highest power rating, with the least stress on the amplifier. See the problem?

There are even ways to manipulate the burst test results. One is to simply choose a shorter burst. If 3 or 4 cycles are used instead of 20, then the amplifier will be able to hold its full output voltage into a lower load impedance. The result is a higher power rating.

Another way to "cook the books" is in how the load resistance is selected. You can coax a larger power rating from an amplifier by tweaking the load resistor to find a value that yields the highest burst test results. The problem is that the amplifier will never produce this much power in actual use, unless it is driving a resistive load of exactly this value, which will never happen. "Three kilowatts into 2.65 ohms" is not a useful metric. But, this "watts" rating still goes on the spec sheet and may even be used in the model number.

An additional variable is the amount of distortion tolerated in the measurement of the burst waveform. A 10 percent allowance will yield a much higher power rating than a 1 percent allowance. A device driven into distortion can yield a much higher RMS voltage.

Due to the many variables, meaningful comparisons of amplifiers from burst test ratings is practically impossible. There's always a way to get a bigger number, and people have taken greater license with the methods than our '70s predecessors.

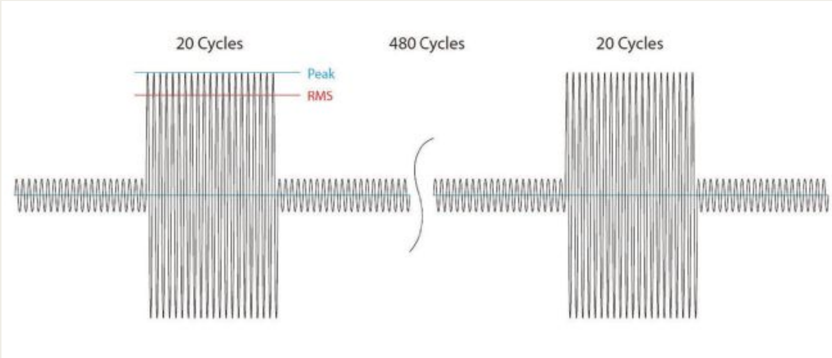

Figure 3

Audio Power

dB Crest	Hz Freq	% THD	Ohm Load	dBu In	Vrms Out	Load Effect	Watts Rated	Watts Meas.
Burst 1 kHz								
17	1k	2	8	4.7	42	0	**200**	220
17	1k	2	4	4.7	40	-0.4	**400**	400
17	1k	2	2.6	4.6	38	-0.9	**500**	555
Burst 50 Hz								
17	50	3	8	4.3	38	0		180
17	50	3	4	2.3	29	-2.3		210
17	50	3	2.6	1.0	23	-4.4		203
Sine Wave (1 kHz @ 15 s)								
3	1k	<1	8	4	39	0		191
3	1k	<1	4	4	37	-0.5		342
Sine Wave (50 Hz @ 15 s)								
3	50	<1	8	4	37	0		170
3	50	<1	4	4	34	-0.7		289

Figure 4

THE PHYSICS OF POWER

Power is the "rate" of doing work. This means that there is a time element involved. The dimension of power is energy divided by time. While "instantaneous power" has a physical meaning, it's just a stepping stone on the way to the actual power produced by a source. The integration of instantaneous power over time yields the continuous power. The continuous power rating describes the heat production capabilities of a furnace, and the work capabilities of an amplifier. Since continuous power is based on the RMS voltage, it is often (incorrectly) referred to as RMS power.

Today's "face value" power ratings have conveniently omitted the time metric, and power ratings, while correctly based on RMS voltage, are only for a blip, not a continuous waveform. The buying public is allowed to assume that the rating is for continuous output. If you leave out the time metric, you only have part of the story.

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Imagine what would happen if they did that in other fields. Examples could include:

- If the towing power of a truck were only based on what it can pull for a few seconds, there would be a lot of burned-out vehicles on our roadsides.
- If the horsepower rating of a jet engine were only sustainable for short flights, I would be reluctant to board a plane for a long flight.
- If the current output of a welder could only be sustained for a few seconds, you will be making a lot of pauses on a long bead.
- If the audio output power of an amplifier can only be sustained for a few cycles, what happens for continuous signals?

You get the idea. Time matters.

EFFICIENCY

To a physicist, a product draws power from a source and converts it to another type of power. For a vacuum cleaner, utility power becomes mechanical power. For amplifiers, utility power becomes audio power. The conversion efficiency cannot exceed 100 percent, for to do so your component would have to output more power than is input.

The only way to get more power from an amplifier than is drawn from the source is to redefine power, and the only reason for doing that is to increase sales. If the audio power rating is 2-3 times the power available from the electrical outlet, a red flag should raise.

"WATTS-ONLY" IS A BAD WAY TO RATE

An audio power amplifier is basically a voltage source that will try to deliver current that the load asks for. As such, current is "drawn" by the load, not "injected" by the source. This is why the impedance of the load is critical to amplifier performance, and why a power rating should always include the load impedance.

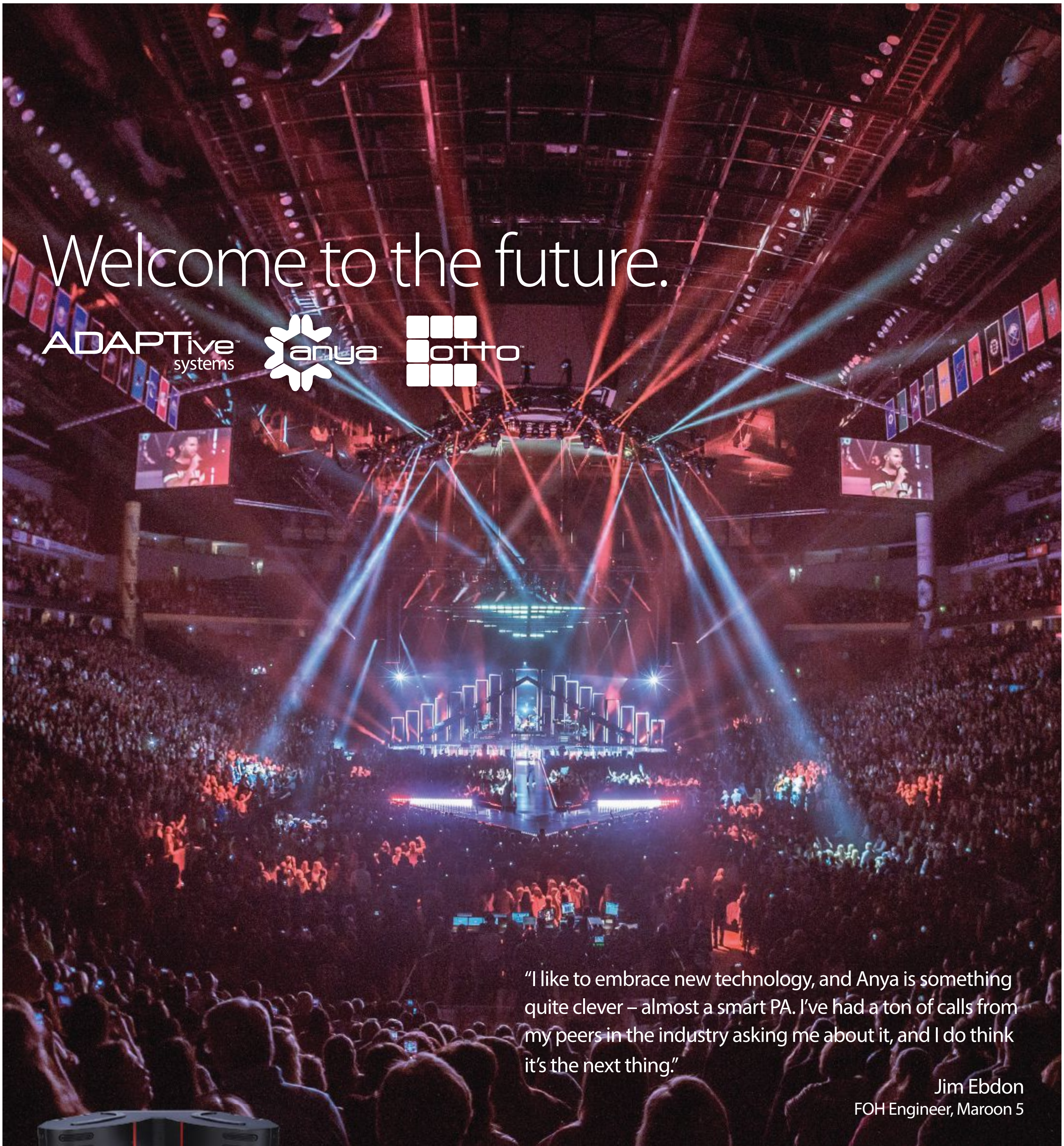
Current is cumbersome to measure, but the power equation lets us state the amplifier's output power using more easily measured quantities, like voltage and impedance. The impedance variable gets too complicated with real-world loudspeakers, so amplifiers are tested into resistive loads, which can be rated with one number. Eight ohms is eight ohms and unlike a loudspeaker's impedance it is independent of frequency.

So, the amplifier is connected to a dummy resistor, fed a test signal, and the RMS voltage across the load is measured. Square the voltage and divide it by the resistance of the load and you have a power rating. While that yields a "one number" rating for the amplifier, what is really needed to understand what an amplifier can do are the voltage and resistance values that produced the power rating. Only then can you see how the amplifier really behaves under different loading conditions, which can be completely masked by a wattage-only rating.

THE COMMON AMPLIFIER FORMAT

Meaningful amplifier selection (and comparisons) requires a complete picture of an amplifier's performance. The burst test is a piece of that pie, but there are at least two others.

The Common Amplifier Format (CAF) was created to address this issue. Rather than just giving a "one number" watts rating, it provides all of the input/output information needed to deploy a power amplifier. This includes the voltage and power performance for burst



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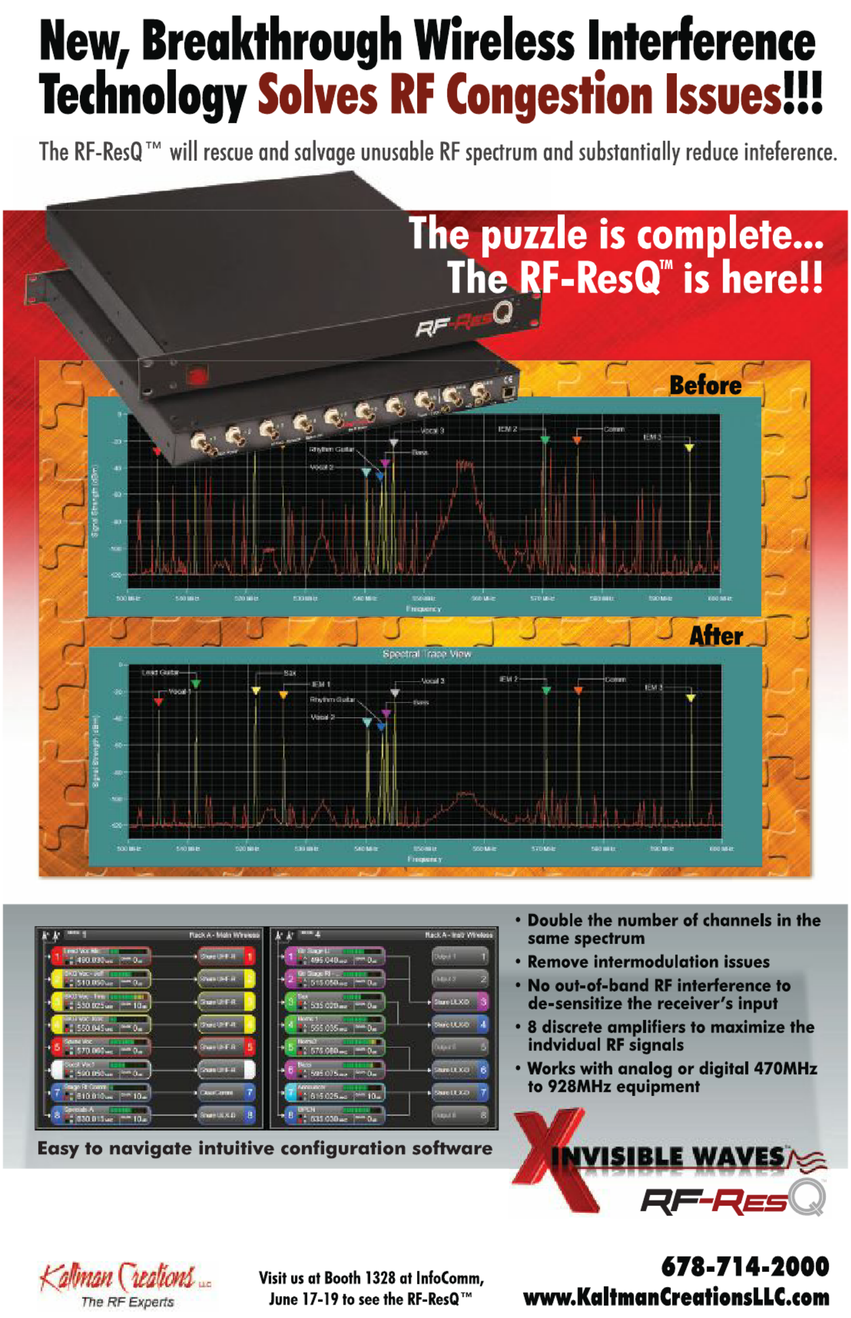
signals, continuous sine waves, and noise, all in a concise matrix.

Figure 4 shows a section of the Voltage, Power I/O Matrix. The details about the CAF are available from www.cafgroup.org. It is not a standard. It is a suggested, defensible way to specify the output capability of an audio power amplifier based on what sound system designers need to know. It is free for any manufacturer to use.

In addition to the traditional one-number watts rating, the CAF gives the output voltage of the amplifier for various signal types across various load resistances, allowing you to see how the wattage rating was determined. This can be an eye-opener.



THREE SIGNAL TYPES

The CAF includes the CEA2006 burst test for 1 kHz and 50 Hz, for the resistances that the amplifier is rated to drive. This is typically 8, 4, and sometimes 2 ohms. No, with the CAF you can't invent your own burst signal.

The continuous sine wave rating is still an important metric, if for no other reason than the fact that the amplifier may have to produce a full-scale sine wave for more than a few seconds to reproduce some types of music, signal tones, or tsunami warnings. If two competing amplifiers have similar burst ratings, then the continuous rating may differentiate between them.

The CAF includes a 15-second continuous sine wave test into the same resistances as the burst test. This is where many "2 ohm" ratings bite the dust. An amplifier that can sustain a sine wave into its minimum rated impedance does so because it was designed for it. That costs the manufacturer money, and it adds value to the product. Burst ratings alone do not give credit where it is due.

Noise ratings reveal the amplifier's performance for music-like and speech-like program. The CAF includes noise test results for 1/4-power and 1/8-power (ref. burst power). The waveform is BS EN 50332-1 noise, which is spectrally-shaped pink noise that is soft-clipped to a 6 dB crest factor. This is necessary to prevent the triggering of the protection or limiting circuits of the amplifier during the test.

One-fourth power simulates how an amplifier performs for heavily compressed program, such as in a touring rig. One-eighth power simulates slightly compressed typical program, and is the most appropriate stimulus for the majority of applications.

It should be noted that the 1/8-

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power noise test is not a measure of the amplifier's capability. It is not particularly stressful to the amplifier. This test reveals the current draw of the amplifier (for sizing circuit breakers) and the BTU output of the amplifier (for designing cooling systems), both of which are included in the complete CAF data matrix. Sizing either of these using burst or continuous sine wave ratings would result in over-designed auxiliary systems and wasted money.

The personality of an amplifier changes completely depending on the output configuration (e.g. mono, stereo, bridged, parallel). Change that and it's a different amplifier. The CAF requires a separate data matrix for each output configuration, so a manufacturer can't just publish the highest watts rating from all supported configurations.

The output of an amplifier also depends on how many channels are used, since in some designs they all share a common power supply. The CAF requires a separate data matrix for one channel driven and all channels driven.

CONCLUSION

As an industry, we have become sloppy in our technical practices. We have learned nothing from the '70s, other than that a different path was needed to exaggerate amplifier power ratings. "Peak power" was out, but now it is coming back in, and along with "burst power" is being used to sway consumers. In a world where more is perceived to be better, it is hard to resist the temptation to provide it, not by product design, but by playing the ratings game. ■

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It's not an absolute but it can be helpful to standardize on BGV mic models.

TCB With BGV

Approaches for optimizing background vocals

By Craig Leerman

MANY OF US spend a lot of time obsessing about the sound of kick drums and guitar amps. We audition different microphones with the lead vocalist during sound check, then spend tons of time getting the mic and effects dialed in for the show.

But unfortunately, when it comes to background vocals (BGV), we're too often content to simply put go-to ball mics on stands, roll off everything under 100 Hz, EQ out anything that starts to squeal, and move on. Which is kind of interesting when you really think about it, because the audience is usually focused more on vocals than the kick.

Presenting beautifully blended vocal harmonies through a PA, at least to me, is the icing on the cake of a quality mix. As a result I devote a decent amount of focus in pursuing and implementing best practices with BGV.

LIKE SUSPECTS

It starts with the right mics. We all have our favorites for vocals, and I'm no exception. My choices offer a tight cardioid pattern with good rejection at the rear, and low handling noise. Further, it's a good idea (although certainly not mandatory) to use the same mic model for all vocalists, with the exception of lead singers and drummers who sing (more on this a bit later).

For lead singers who don't bring their own mic to the gig or have a specific preference, select a few "likely suspects" based on their vocal characteristics and technique as a good starting point, and then have the singer quickly "audition" them. For band members that play an instrument, dynamic cardioids that offer a relatively flat frequency response or a slight rise in the mid frequencies tend to work well.

Unless a musician specifically requests a straight or a round-base stand, tripod stands with adjustable booms are a good choice because they offer more positioning options, along with good stability. Make sure the rubber tips on the legs are in place to help reduce stage noise being transmitted to the mics. Rubber mic clips can also help reduce noise transmission, and make sure each clip's swivel is tight enough so it won't slip during the show.

I'm also a big fan of external foam windscreens on mics, even indoors. They help in reducing pops and plosives, and also keep lipstick off of the grilles.

Drummers providing vocals is where I deviate from the "all the same mics" rule. While most other musicians can simply lean in or step up to a mic, drummers are pretty much locked into position, seated and with arms (and drum sticks) flailing about. While a standard mic on a boom can work, it gets in the way more often than not.

A good option is a headworn mic like the Crown CM311 "differoid" (differential cardioid) condenser, Shure SM10A-

There are a lot of ways to outfit a drummer with a vocal mic, including one on a boom stand with everything out of the way.

CN cardioid dynamic and Shure BETA 54 supercardioid condenser. The key here is that the cardioid pattern helps in rejecting the drum sound. Select a rugged headset style, and make sure it's adjusted (fitted) as well as possible to the head so it won't slip during performance.

Also take care to page the mic cable out of the way (flailing arms and sticks, remember?). Most of these headworn units include clothing clips, so the cable can be attached to the shirt and routed down the back, out of the way. Leave some slack in the cable so the drummer can reasonably move about and not pull the clips off.

For drummers who prefer wearing headphones for monitoring, there are models that also include an attached boom mic. While many of these units are designed for remote broadcasting, some, like the Audio-Technica BPHS1 stereo headset with cardioid dynamic mic element, also work well for singing musicians.

Drummers who prefer a mic on a stand can be outfitted with models with shorter bodies such as the Telefunken M80-SH dynamic supercardioid, Lewitt DTP 340 TT dynamic supercardioid, Audix D2 dynamic hypercardioid, and Heil Sound PR 31BW dynamic cardioid, so that there's less mic sticking out of the back of the clip. This can be further minimized by using an angled XLR connector on the cable. Another possible solution in the quest for a small footprint are mics like the Shure BETA 56A dynamic supercardioid with a built-in locking stand adapter and an integral XLR connector.

Some drummers prefer vocal mics be dropped in front of their faces via a boom. This can be accomplished by using a tall studio-style boom stand or a regular stage boom stand that's weighed down with sandbags so it won't tip over. A short gooseneck can then be used to drop the mic in front of the face.

MULTIPLE DUTIES

Some bands have dedicated backing vocalists. Sure, they may wave a tambourine or hit a cowbell now and then (and who doesn't want more cowbell?), but primarily their function is to sing. More often than not, they're positioned in a row, and frequently on a riser. Because risers aren't as solidly built as stages, they're very susceptible to transmitting noise into mics (especially if the singers like to dance), so it's wise to use large cast base stands with rubber feet.

Also if possible, provide vocalists with their own monitor wedges, even if they're all getting the same mix. That way each wedge can be positioned directly behind the null zone of its respective mic. Placing wedges between the mics opens up potential bleed problems. I make sure that all of the vocalists can be heard in each other's monitors at the outset of sound check, as well as some kick drum, keyboards and a little rhythm guitar. This provides a solid starting point.

BGVs can have one or more jobs over the course of a show – singing harmony with the lead, singing as a chorus in unison with or without the lead, solos, and duets with the lead or each other. As noted, my own preference is to have all BGVs on the same mic model unless one of the singers has a vastly different style or range (like a dedicated bass singer who "eats" the mic).

In sound check, I start by tailoring the EQ for each singer, usually rolling off the bottom end around 80 to 100 Hz for males unless they sing bass, where it's lowered to the 60 to 80 Hz range. For females, the roll off is usually at 100 to 140 Hz. The goal is keeping stage rumble and wayward bass frequencies out of the mix.

Lewitt DTP 340 TT dynamic mic, offering a shorter body.

I may also "thin out" male vocals a bit in the 160 to 400 Hz range when they're supporting a male lead vocal to give them a distinct place in the mix. The same goes in the 200 to 500 Hz range with female BGVs supporting a female lead vocal.

Some of the highs may be rolled off as well to help the lead vocal stand out. The idea is to get all vocals fitting right without overshadowing the lead. BGVs usually need to sound like a tight and cohesive unit that's a complement to the lead, rather than a group of distinct individuals.

Another thing that can help is keeping lead vocal focused a bit more center, with background vocals somewhat more to the sides. A trick I picked up from a recording engineer is to pan backing vocals off to one side and apply a short slap-back echo (no more than 20 to 30 ms or so), with that echo panned over to the other side. This can deliver an overall fuller vocal sound without things being too busy in the center where the lead resides.

SEAMLESS BLEND

I'm not a fan of gates on vocals (or gates in general), and also tend to use minimal channel compression, only applying a bit when necessary to keep levels in check. Typically, I route vocals to sub-groups

Careful on wedge location or there can be too much bleed into BGV mics.

on the console and use one VCA/DCA for control of all BGVs and another VCA/DCA for control of the lead. A bit of compression can be applied to these sub-groups if needed but it's important not to "squash" anything.

Vocal or room-type reverb can add a bit of life to the lead, but care needs to be taken when adding reverb to background vocals. Rolling off the lows on the reverb return is usually a good idea to keep the overall vocal mix clean while fostering intelligibility of the lead vocal. A longer reverb on BGVs, such as large room or big hall, can further help in this regard.

A few times I've worked with bands where some pre-recorded backing vocals are triggered (by a musician in the band or crew member). When this is the case, it's important to make sure that any click tracks are routed to the correct monitors. Also spend some time listening to the tracks to make sure they all sound similar and that the levels are consistent. If necessary, place a compressor on the tracks to keep levels in check.

Most of these tracks are simply a stereo mix, but if the playback unit offers multi-track outputs, treat each track as a separate vocalist. And apply the same panning approach noted earlier, with the tracks a bit more to the sides. The trick is to seamlessly blend the recorded tracks – sonically and visually – with the live singers on stage. ■

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


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THE MIGHTY AUX

Six superhero uses for console aux sends. *by Chris Huff*

FORGET THE FADERS, knobs, and digital displays – flip the console around and show off your superhero skills. Those mind-mannered auxiliary sends are for more than monitor mixes. Aux sends, controllable at both channel and master levels, provide solutions to six unique challenges that range from communication to improvising a stage subwoofer.

1: MIX A BETTER RECORDING

A single-track house mix recording never sounds the same when compared to the live event – fact of life. When multi-track recording isn't available, aux sends provide the next best thing because they provide channel-level control.

Plug an aux send into a recording device. Next, plug headphones into the recording device. See where this is headed? Use the channel-level aux volume control to mix the recording.

Additionally, the aux mix method works for one-off recordings such as when your friend says, "You owe me a favor – can you record my band performing this one song for my girlfriend?" So much for the glamorous side of audio production.

2: BETTER AUDIO FOR VIDEO

Yes, I said it: the "v" word. Videographers have a tough job and getting quality audio is part of it. Give them an aux send and they'll become a friend for life.

Be it an informal event, high-end wedding or corporate gig, a videographer will ask for a dedicated line from the console to tie into their camera. Use the aux send, like in the aux recording section, to provide a nice audio mix.

Video cameras can take an assortment of audio jacks and what the videographer has and what is required aren't always the same. A TS-to-3.5mm stereo cable and a TS-to-RCA cable should do the trick. Yes, they're mono-sends but that shouldn't be a problem.

Aux sends can be pre-fader or post-fader so set them accordingly. A good rule of thumb is pre-fade for monitors and post-fade for everything else.

3: SEND AUDIO ANYWHERE

Secondary locations include hallways, overflow rooms, outdoor patios, and anywhere else people want to hear the mix from the main venue. Most of the time a straight house mix suffices, but there are a few unique situations when an aux send comes to the rescue.

Some people don't want to hear everything broadcast during an event. Maybe they only want to hear the person speaking at an event, not the band. Or maybe they want the band and not the emcee's blathering pronouncements. Whatever the case, pull the source from an aux send mix instead of the house mix.

Note secondary location loudspeakers may or not be powered, therefore don't assume the output routing change is as simple as swapping a few cables. Draw up a system schematic if there isn't one and observe how to alter the routing.

4: CAN I GET A SUB?

Ever walk into a venue to mix a band and notice a lack of subwoofers? The bassist is about to be your best friend – fake it if you must, desperate times and all that.

Send the kick drum and bass channels out through an aux send which goes out to the bassist's amp. The musician gets the bass and the kick, keeping them happy, and the venue now has a sub sitting on stage.

Who said audio production solutions have to be perfect or involve buying more gear? Many times solutions are possible with existing gear – we just have to be creative in how we use it.

5: VOCAL EFFECTS GROUPING

This was my first exposure to aux send mastery. I was subbing for a front of house tech who had a vocal effects channel on his Yamaha analog console.

At the time, I thought he was a genius.

Route the vocal channels to an aux send, take the aux output to a reverb unit, then run the reverb's output to a console channel. All vocals now have reverb and there's one more benefit. Reverb on the singer is good but a ton of reverb when the singer is giving a song's backstory makes for an amateur mix. Use the channel fader to fade the reverb in and out as appropriate.

Compression as an effect is a great tool for improving a drum's presence. Use the same process to route drum skins (toms, snare, and kick) to a compression aux and then feed the compressor into a console channel.

6: TWO-WAY COMMUNICATION

Stage-to-booth communication can be handled through an aux send. Yes, wireless intercoms are the way to go but in some venues, we must use what we have available.

Route a microphone located in the booth to a muted channel, and then route that channel to an aux send. This send can be routed to a backstage loudspeaker or headphone jack. Likewise, route a backstage mic into a standard stage jack and place it into a muted channel routed to an aux send that goes to a set of headphones.

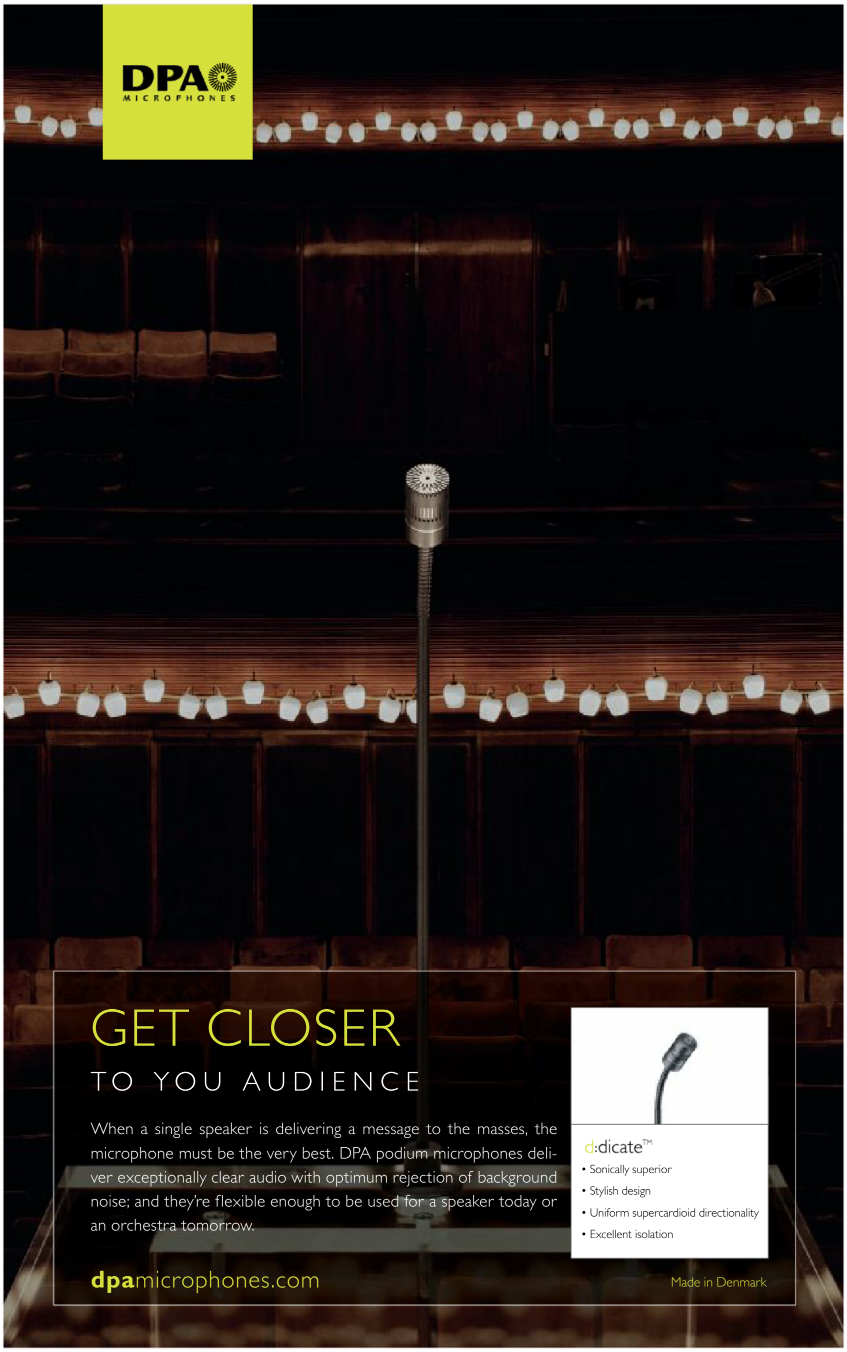
Many times solutions are possible with existing gear— we just have to be creative in how we use it

This isn't an optimal communications scenario, but for venues where you need two-way communication, it's effective. And let's be honest, there are some people we don't want to have our cell phone numbers.

Digital consoles provide some solutions like multi-track recording and effects groups. Squeeze every bit of functionality out of those consoles. However, there are times when a digital console can't solve every problem and the audio tech should know the back of the console is equally as powerful as its surface.

The power of a mixing console isn't limited to what can be done with faders and knobs or digital displays. The mastery comes when it's used for everything it's meant and then a little more. What superhero uses can you imagine? ■

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.*



BREAKING THE LINE

The actual impact of breaking and tapering a line of loudspeakers.

by Bob McCarthy

A friend of mine who's not an audio engineer went to a show and later told me it sounded too loud and unintelligible. Plausible? Sure. Now try this: another friend said it sounded like the line of the loudspeakers had been broken. Still seem plausible? No?

I figure everyone knows what a broken line sounds like since it's such a big issue for discussion in the audio community. Every time I tune a system or hold a seminar, someone tells me all the things I can't do because I'll break the line: "Whatever you do don't break the line because it will sound like..." Help me out here – sound like what?

DEAF & BLIND TEST

I invite you to blindfold me and roll me around the venue. We can listen to pink noise or music and I'll use my trained ears to tell you when it gets louder or softer, more reverberant or less, and brighter or duller as we move around the venue. I can tell you specific peaks and comb filter areas, and even identify transitions between elements of the sound system or the timing and strength of echoes. This is not because I'm special – any audio engineer with a trained pair of ears can do this.

Conversely, remove the blindfold, cover my ears, and put me in a remote room in front of an FFT analyzer. I can still identify these same features as a measurement microphone is moved around a room. Again, I'm not special – any audio engineer with a trained pair of eyes can do this, as long as they know how to read the analyzer and recognize the relationship of the traces to what they hear.

It's admittedly an advanced skill, but one required of anyone tuning systems. The reason is these are objective, verifiable, audible characteristics of a sound system in a space. A 6 dB level difference between two locations is not a theory – it's true or not true. It can be directly experienced and measured.

THE SOUND OF BREAKING THE LINE

Here's what I cannot tell you by either of the above methods: whether or not the line array theory has been violated and the line is broken. When the line breaks, do we hear a snap? Does the frequency response show tear marks? These are absurd questions but please tell me, what are the tell-tale signs? Why is there such widespread fear of breaking the line? My theory is that there's as much or more fear of breaking the party line as the acoustical one.

There are two principal manifestations of the "don't break the line" strategy. The first is the prohibition of level tapering within a multi-element array. So if it's 6 dB too loud in the front area (a verifiable fact we can hear and measure) we should not solve this with level tapering because we'll break the line (a theoretical construct we cannot characterize sonically or measure).

The second is the prohibition of spatial separation between sections of an array. Balconies are the prime movers in this discussion. Should we cover the upper and lower levels with one array or split it into two? The answer should be whichever of these can achieve the highest uniformity (something we can measure). Line length absolutists will vote to keep the array together even when this results in reduced uniformity in order to preserve the line.

Let's take a moment to consider exactly what line length does. We begin by noting that the beamwidth of a single loudspeaker approaches 90 degrees when the transmitted wavelength (λ) equals the radiating cone size. Lower frequencies are wider than 90 degrees and higher ones are narrower until the wavelengths are so small that lobing scatters the pattern. A coupled line source also has a size-related directivity milestone: coverage is 72 degrees when the transmitted λ equals the line length, e.g., an array 2 meters long will be 72 degrees at 170 Hz. The coverage widens below and narrows proportionally above. The narrowing is also frequency/wavelength-limited by lobing, in this case by the combing interaction caused by the element spacing.

Some notable conditions are attached to the 1λ= 72 degrees standard coverage. Line length equations assume omnidirectional sources, so the coverage numbers will shrink with directional elements. We also need enough quantity for the line shape to be fairly full of loudspeakers. Think of the 1l length as a spread of 360 degrees of phase offset over the elements. With just two elements, the sources are a full λ apart, causing a huge side lobe (0 degrees + 360 degrees). Three elements within the same length spaces them at 180-degree intervals, which should set off alarm bells. Five elements brings the spacing to 90-degree intervals, which can achieve the orderly chaos needed to cancel the side radiation.

Beyond this point the quantity has less impact on the cut-off frequency. Line lengths of 3 meters will have the same cut-off frequency (113 Hz) whether we fill it with 8, 12 or 16 elements. Line lengths of 1.35 meters, 2.7 meters, and 5.4 meters correspond to cut-off frequencies of 250 Hz, 125 Hz, and 63 Hz, respectively.

Primary features of line length:

- Standard attributes apply to omnidirectional sources.
- Nominal coverage is 72 degrees when line length = 1λ, 36 degrees when length = 2λ, and so on.
- Directional elements reduce the nominal coverage proportionally
- Line must be effectively continuous (sufficient density to maintain coupling)

Angular reduction is proportional at frequencies above the 1λ length, i.e., we get 36 degrees when we double the length, or double the frequency. The narrowing continues with successive doublings. Simple proportionality does not hold below the 1λ frequency as the coverage approaches 360 degrees.

It's easy to exaggerate the ramifications of line length as also often happens with coverage angle. The 1l frequency is a convenient reference point but not something that stands out like a misaligned crossover. It's not as if 72 degrees is the world's greatest coverage angle. The frequencies below this marker widen gradually, not suddenly, so don't be disappointed when adding that 16th box reduces the coverage all the way to 68 degrees.

Line length is decisive only in the very low end of these arrays, where omnidirectional elements are otherwise un-steered. As frequency rises and elements become directional, the line length extends the uncoupled/undefined (near field) area of the parallel pyramid. The extension varies with frequency, becoming steadily longer in the HF. This results in widespread spectral variance over angle and over distance.

It's often believed that line length is a property exclusively relevant to the coupled line source (all boxes at 0 degrees). The coupled point source also has length, albeit in curved form. A few degrees of splay between boxes is negligible to omnidirectional elements, so the difference between a line and point source in the relevant range is not worth spending any time to analyze.

We can summarize the line length behavior by the Goldilocks method: too big, too small and "just right" (assuming 72 degrees is your target coverage angle). We have less than full control when λ > line length, extra control when λ < line length and just right control when λ = line length (Figure 1).

LEVEL TAPERING

Let's start with level tapering, considering a basic arena shape in the vertical plane: longest throw to the top – shortest to the bottom. Is the relationship of level taper to line breaking a digital phenomenon? (i.e., 1 or 0? Broken or unbroken?)

It's assured that our line is unbroken if we have a line of 16 identical boxes at the same level. If one box is turned down 0.1 dB, is the line broken or just slightly bent? If you think it's broken already, here's some bad news: You've never heard an unbroken line because manufacturing tolerances aren't that good. How about variations of 1 dB? Again, nobody can deliver 16 boxes that are within a single dB.

Let's step it up: Turn the bottom box down 6 dB. This makes it effectively half a box. The loss in the lowest frequency range (where directionality is so low and

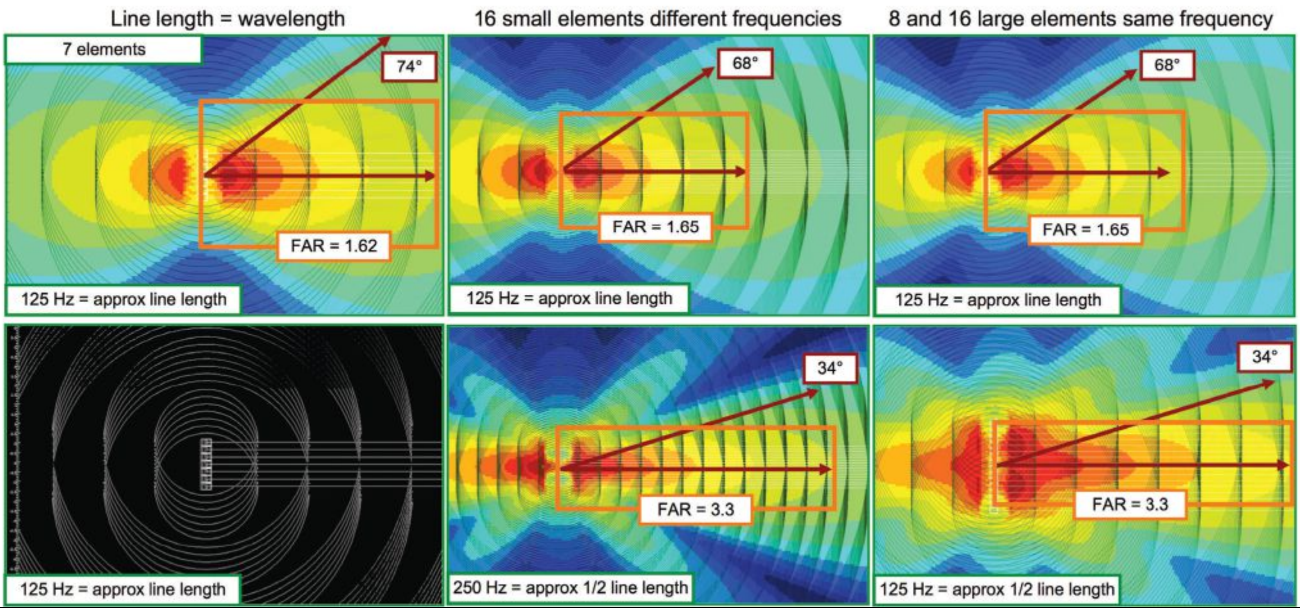


Figure 1: The effects of line length in the coupled line source; length is a 2.76 meter (1λ at 125 Hz). The circles are sized to the wavelength of the frequency shown (lower left). Notice the front lobe follows the confluence of the circles and cancellation occurs in places where the circles diverge. The coverage angle is around 70 degrees regardless of quantity when the line length equals the wavelength (upper panels). The coverage angle is reduced by half when λ equals half the line length (lower center and right panels).

wavelengths so large that they sum well) that the combined response is reduced by less than 0.25 dB. This does not seem so scary in the big picture, does it?

Let's go further and reduce 3 boxes by 6 dB. The lowest frequencies now lose 1 dB. The high frequencies, meanwhile, are substantially reduced in the area where the bottom three boxes are pointed – such as the early rows of seats. This is a tangible benefit (up to 6 dB of HF control) for a minimal cost (1 dB loss in overall LF power). Is the line broken now? If so, how can I tell? Is it broken everywhere or just at one place? Is it broken at all frequencies?

It should be easy to find the break point at 10 kHz, since there's enough directional control to hear the isolated areas on either side of our fault line. You could find it blind-folded or measure it with an analyzer. Do you think you can identify a break point at 100 Hz? Good luck. These large wavelengths don't turn on a dime. When people talk fretfully of breaking the line is it the VHF range they are worried about? Not from what I hear. The concerns I hear are much more about the lows than the highs.

A practical taper should be more gradual, with 1-3 dB steps being more typical – and which makes it difficult to locate the transitions in the space. If we tapered the bottom 5 boxes at 3-3-3-6-6 dB, we would get an overall LF loss of 1 dB and gain substantial HF steering. The same price would be paid for a taper of 1-1-1-1-2-2-4-4 dB on the bottom 8 boxes of a 16-element array. These are just two of a myriad of options that can be employed to help tailor the response of our bent/broken system to the shape of the audience.

The trade-off for the potential problems this tapering might cause also includes additional benefits: besides increasing front-back level uniformity in the HF range, it also improves uniformity (to a lesser extent) as frequency falls – all the way down. The dB that we lose in the low end is offset by the fact that the beam center is steered upward off the floor and therefore spread more evenly front to back.

GAIN TAPER RISKS VS BENEFITS

There are tangible risks to offsetting levels and EQ within a coupled loudspeaker array. Gentle slopes and gradual level transitions will reduce the risk, while steep slopes and transitions will increase it. There's potential for phase offset between elements that have different EQ curves, but this should be minimal as long as reasonable EQ is applied.

It's the effect on dynamic range that merits detailed discussion. Asymmetric EQ and level taper will reduce the dynamic range and potentially cause dynamic shifting, i.e., some parts of the system may reach limiting before others. This can be a classic TANSTAAFL (There ain't no such thing as a free lunch) trade-off between maximum uniformity, stability, and headroom. Detectible instability in the frequency and spatial response are unacceptable outcomes. In short, dynamic uniformity is a vital part of level and spatial uniformity.

Level taper affects on dynamic range (Figure 2) can be used as a reference point. This is one area where the program material and usage are extremely relevant. If the system is to be operated at its absolute maximum at all times (e.g., a heavy metal music festival), then front/back level uniformity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8 speakers @ various levels	A	B	C	D	E	F	G	H	SUM	Log gain	Taper loss
8 @ 0 dB	1.00	1.00	1.00	1.00	1.00	1.00	1.00	1.00	8	+18.1 dB	-0.0 dB
7 @ 0 dB, -1dB	1.00	1.00	1.00	1.00	1.00	1.00	1.00	0.90	7.9	+18.0 dB	-0.1 dB
6 @ 0 dB, -1, 2 dB	1.00	1.00	1.00	1.00	1.00	1.00	0.90	0.80	7.7	+17.7 dB	-0.4 dB
5 @ 0 dB, -1,2,3 dB	1.00	1.00	1.00	1.00	1.00	0.90	0.80	0.70	7.4	+17.4 dB	-0.7 dB
4 @ 0 dB, -1,2,3,4 dB	1.00	1.00	1.00	1.00	0.90	0.80	0.70	0.63	7.03	+16.9 dB	-1.2 dB
3 @ 0 dB, -1,2,3,4,5 dB	1.00	1.00	1.00	0.90	0.80	0.70	0.63	0.55	6.58	+16.4 dB	-1.7 dB
2 @ 0 dB, -1,2,3,4,5,6 dB	1.00	1.00	0.90	0.80	0.70	0.63	0.55	0.50	6.08	+15.7 dB	-2.4 dB

Figure 2: Log/linear decoder example for combined levels of eight loudspeakers (neglecting phase). Tapering in successive 1 dB increments is shown. The loss from the maximum possible summation is shown in the last column.

must yield to maximum dynamic range and stability (level tapering and asymmetric EQ must be absolutely minimized).

By contrast, if the system will be operated with ample headroom under controlled conditions (e.g., musical theater), then asymmetric EQ and level tapering become viable options to maximize uniformity. This decision must be evaluated in the field based on risk and return.

My personal approach is generally to compromise, using minimal asymmetric EQ and leveling to reduce, but not necessarily fully eliminate the level increase in the front of the room. Risks are relatively low if the asymmetry can be kept low through a large majority of the array. Increased asymmetry at the bottom has more potential benefit and is less likely to cause audible instability to the majority shareholders above them.

Bear in mind, however, that much of my work is in applications that highly prioritize uniformity and fidelity, which leaves me with more room for asymmetry than those who optimize systems that absolutely must prioritize for maximum stability while delivering bone-crushing power.

Real-world loudspeakers will have variable amounts of overlap over frequency. An array of constant beamwidth models will have an upper range with

constant overlap and a lower range with increasing (and variable) overlap. The effectiveness of asymmetric EQ techniques therefore varies over frequency, with the HF range enjoying the lowest side affects and risks.

An array of proportional beamwidth models has proportional overlap, i.e., it rises gradually and steadily with wavelength. Combined EQ can be approached here by segmenting the spectral range into isolated (HF) and overlapped (LF) regions. EQ and level setting for the isolated HF is carried out by the ABC approach (separate EQ for A, B and C as soloists followed by a mutual EQ to compensate for the combined effects).

By contrast, the LF region is approached as a single block, with all EQ and level settings matched (i.e., symmetric). This approach will likely yield less uniformity than the fully asymmetric approach but eliminates the dynamic risks inherent in asymmetric EQ.

We all strive for front/back level uniformity, yet many fear the perils of breaking the line. There's a limit to how much we can squeeze these systems by splay angle asymmetry alone. The cluster is hung and isn't coming back down, so we need to look at how much level disparity is still left. If it's too loud in front, I vote we break the line and turn down the lower boxes, until or unless we know we are reaching the point of dynamic instability. If you have a better measurable and verifiable method, I'd love to hear it.

THEORY WORLD

One funny historical aspect related to this subject: 30 years ago when we started utilizing FFT analyzers to tune sound systems, many engineers would deride us. They resisted our use of an analyzer to help make decisions, saying we lived in "theory world" and that they lived in the "real world."

Nowadays some engineers still want to limit the actions we perform based on the analyzer, but the problem has been flipped on it's head: we're the ones living in the real world and they're living in theory world. Funny that one, eh? ■

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.*


The scene at Major Rager as viewed from behind the stage.

MAJOR RAGER

Top concert production values at Masters Week

By Kevin Young

THERE WERE SEVERAL records set at this year's Masters Golf Tournament at Augusta National in Georgia in April, many of them by 22-year-old winner Jordan Spieth. For Rob Boggs, owner and system designer at Quest Sound and Productions of nearby Grovetown, GA, however, it was business as usual, which meant that his growing company was extremely busy providing production for a variety of events and concerts held in affiliation with the tournament.

One that stands out is the Major Rager, which is held at the downtown Augusta Common. Sponsored by local charity organization Friends With Benefits Fund, it's grown from local acts to a full-fledged concert event that was inaugurated last year with a headline performance by Umphrey's McGee. This year, southern rock jam band Gov't Mule topped a bill that also included Lettuce, The Revivalists, Phillip Lee Jr., and Michael Baideme.

Boggs, who notes that his father was a greenskeeper and that he virtually grew up on a golf course, enjoys the hustle of Masters Week. "It's really exciting for us, but we were so booked with multiple shows I didn't get to follow it," he notes. A portion of the proceeds from Major Rager goes to a local charity, which this year was Turn Back the Block, which focuses on the revitalization of older neighborhoods in Augusta.

The show attracted about 3,000, with Quest Sound and Productions deploying a concert-scale sound reinforcement system headed by Renkus-Heinz ST Series line arrays. The same system also served another show a few days prior that featured country rap artist Colt Ford.

STARTING EARLY

Boggs began advancing the second annual Major Rager last November and was in touch with the show's talent by January of this year, acquiring essential details as well as making sure the proposed rig had the support of the headliner.

"I use Renkus-Heinz for a variety of applications – for everything I can, in fact," he says. "Our CFX101LAs (compact arrayable loudspeakers) serve as speakers on sticks for speaking events, and the (larger) line arrays handle concerts and bigger events. The products cover everything we serve, and the company's customer support is probably the best I've ever had."


Rob Boggs of Quest Sound and Productions at a Yamaha PM5D console.

The Quest Sound and Productions team set up the system and provided support throughout its use, with all bands supplying their own mix engineers. The coverage area at Augusta Common is rectangular, roughly the size of two football fields, with no obstructions.

The main arrays were made up of five Renkus-Heinz STXLA/9 cabinets per side, flown from wings extending from the roof above the 32-foot-wide stage. The top cabinet in each array flew about 20 feet above the ground, with the first four boxes from the top down offering 90-degree horizontal dispersion while the bottom box delivered wider 120-degree coverage for the near field.

"With the Renkus boxes, there's very little EQ needed to get them where they need to be, which makes it an easier job," Boggs states. "The clarity is unbelievable. For Colt Ford we didn't even use front fills because the arrays covered that (front) area so well."

Low-frequency reinforcement came from Martin Audio Blackline S218+ dual 18-inch direct radiating subwoofers, ground stacked six per side below the line arrays. "I cut them off at 99 Hz so the response was nice and tight," he adds.

Quest Sound and Productions utilized Lake LM26 processing for management of the house system. All loudspeakers were powered by Crown Audio XTi amplifiers, a combination of XTi 6000s and XTi2 6002s, with eight of each rack-mounted per side. "Basically it was a 6002 or 6000 per link, and then we ran one for the mids and one for the highs," Boggs notes, adding that XTi was one of the amp models specified on the Gov't Mule technical rider.

It's also an amplifier series that he prefers for gigs in the southern U.S., where humidity and moisture can be a problem: "I've never, ever, had issues with Crown, and I try to stick with the companies and products that don't give me any problems."

He returns to the topic of loudspeakers. "About five years ago, the local Renkus representative let me test out some of the company's smaller arrays. When they came out with the CFX101LA cabinets, I added 20 of them right off the bat because I loved the sound," he says. "And when it was time to move up to the larger format boxes, Renkus provided a demo system. They've been very good to us, and as a result, we use them on as many shows as we can."

The stage and arrays deployed at Augusta Commons.

FLEXIBLE PACKAGES

The headliners for both concerts brought in their own consoles and monitor rigs. Colt Ford carries a Yamaha CL5 console for the house and an in-ear monitoring (IEM) rig, with everything pre-wired.

Gov't Mule, which doesn't use IEMs, deployed a portion of its touring rig, which is supplied by Clair Global. In addition to a DiGiCo SD8 console for the house and an SD9 for monitors, the stage was outfitted with seven Clair 12AM-II monitor wedges, as well as an assortment of Clair boxes for side fill, drum sub and front fill. The band also brought its own mic locker.

Meanwhile, most of the other bands supplied their own IEMs and wireless mic systems. Lettuce was the only band that used the Midas PRO2 digital board, which was supplied by event partner Gattis Pro Audio (Lexington, SC), for both monitors and house duties. "The PRO2 is just a solid console," Boggs says, noting that it was linked to its stage box at the stage via Cat-5.

Gattis Pro Audio also delivered the mic package used by the other bands, including Shure BETA 91 and BETA 52 for kick in/out, joined by BETA 57s on snare top and bottom as well as toms. Shure SM57s were available for guitar amps, BETA 58s for vocals, with plenty of Radial DIs on hand for those preferring to go direct.

As is their usual practice, Boggs and his team went back over details in doing a thorough post mortem of this year's Major Rager to isolate what went well as well as what can be improved next year. "Everyone was happy, and there were no complaints," he says. "I think the biggest challenge we faced this year was making sure the bands were comfortable with the change-overs."

GETTING IT DONE

With the exception of some specials carried by Gov't Mule, Quest Sound and Productions also supplied all lights as well as trussing. "Over the past three years we've developed ourselves as a one-stop shop, carrying enough now to do four complete stages," Boggs says. "We're trying to gear ourselves more towards festivals."

The southeastern U.S., particularly the territory Quest services – from all of Florida to the northern part of Virginia – has a fair number of festivals, he adds, and it's helped lead to a significant growth in business. In handling that growth, especially at a time when so much is happening in one area in a short amount of time, as is the case during Masters Week, Quest vice president of production and lighting John Berret, as well as Boggs' wife Lisa, who serves as vice president of operations, are key to the company's ability to seamlessly serve multiple projects simultaneously.

"People ask us, 'How do you do this? You've got to be getting two hours sleep a night.' But it has to be done," he says.

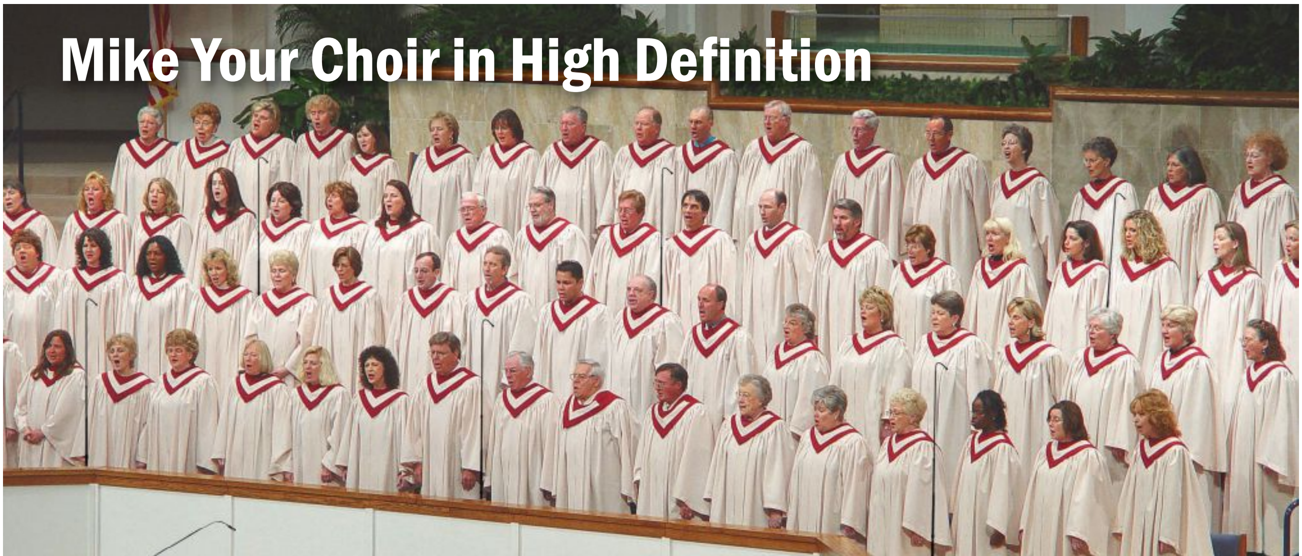
A main array comprised of Renkus-Heinz STXLA/9 cabinets.

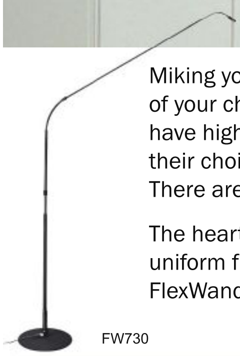
"During Masters Week, we look at the light at the end of the tunnel, waiting for Sunday evening when the last ball goes into the hole and the gig is done."

It fits with a bright future for an enterprise that's finding its legs after being a long time in the making. "I've been around bands for 30-some years," Boggs concludes. "I couldn't play, but my best friend then was like, 'If I'm going to play in your band, Rob's got to be a part of it.' So I became a sound engineer. When I moved down here 20 years ago, I wasn't doing audio, but a local band found out that I had worked as an engineer and asked me to do sound for them.

"Soon I started getting phone calls and I'd mix the band somewhere and the venue people would say, 'We need a house guy. Would you consider it?' The next thing I knew, I had more work. One thing led to the other, and I decided to form Quest about 11 years ago. You go out, do a good job, and if people like you and your work, you grow." ■

Based in Toronto, KEVIN YOUNG is a freelance music and tech writer, professional musician and composer.





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Check One, Two

A primer on sound check and a whole lot more.

By Craig Leermanx

FOR MANY ENGINEERS, the term "sound check" refers to the time the band hits the stage before the gig, plays a few numbers, has their monitor levels adjusted and then heads off to relax before the show. For others, including me, the term encompasses an entire procedure from system tuning through line checks, not stopping until both the performers and sound crew are happy. This is especially important because my company handles a lot of one-offs where we've never previously worked with the acts.

For us it starts with loudspeaker positioning, making sure they're going to cover the entire audience without too much spill on to the stage. A frequent problem is too much bass spill, so we arrange the subwoofers in cardioid or end fire configurations to reduce the problem.

In tuning the main system, we start with the mains and make sure each box or driver band is working by sending them noise or a song. Next we test any front or side fills, as well as delays (if applicable). Once we've verified that every cabinet is working correctly, a test track is played with the system EQ flat and we critically listen to each zone (mains, fills, delays) independently to make sure they're sounding OK. At this point we're not worried about the interaction of the system and room, just that the loudspeakers are performing properly.

If we're using an installed house system or a portable system not our own, it's fed a mono track that's panned left and right to make sure the speakers sound the same on each side. Quite frequently we encounter installed systems that sound vastly different from side to side.

RINGING IT OUT

We're certified "old school" so don't just rely on software-based analyzers for tuning, but they're a valuable tool, particularly in evaluating higher frequencies that our (older) ears don't hear quite as well.

The process starts with a stand-mounted cardioid microphone placed near the front edge of the stage with the capsule pointing straight up. Meanwhile the measurement mic for the analyzer is placed an equal distance from left and right main loudspeakers. (Think of a triangle with the mic at the point and each main stack at the corners.)

Before making any noise, we warn everyone in the room (via the board mic) that some feedback is coming soon. They can clear the area or put in ear plugs (which we offer).

We start with the mains and bring up the level until there's a single ring of feedback. We make a note of the frequency and pull it out via the system graphic EQ, then push the level up until another ring is heard. This continues until there are multiple rings at the same time. Note: If you're not good at picking out frequencies, then a system analyzer's display can be valuable in showing what frequencies are being excited. This process is then repeated with fills and delays playing. And if drastic cuts are called for, a parametric EQ is utilized for a more "surgical" approach.

Next, the mic on stage is moved where the primary performer will be located and faced the correct way, and then we evaluate the sound of the system and how much gain is available. A few test tracks confirm how the system sounds, and at this point, balance is set between the mains, fills and delays. I walk the room listening to tracks, judging tonality and balance, and making any adjustments by ear. I also talk into a vocal mic and listen.

ONE AT A TIME

Now it's time to wire up the stage. One of the keys to an easy show is to label everything in case of a problem. We label all input cables and sub-snakes to trace a signal path quickly or for a quick, accurate re-patch when changes happen (and they always do). When working corporate shows, we also run a few backup lines for important channels like the podium, wireless feeds and video sends.

For bands, the console is arranged in a typical festival style, kick in channel 1, then snare, hat, toms, overheads, percussion, bass guitar, guitars, keys, other instruments like horns, and then vocals. On corporates, the podium mics are the first channels, followed by wireless, playbacks, audience Q & A mics, VOG ("voice of god") mic and board talkback mic.

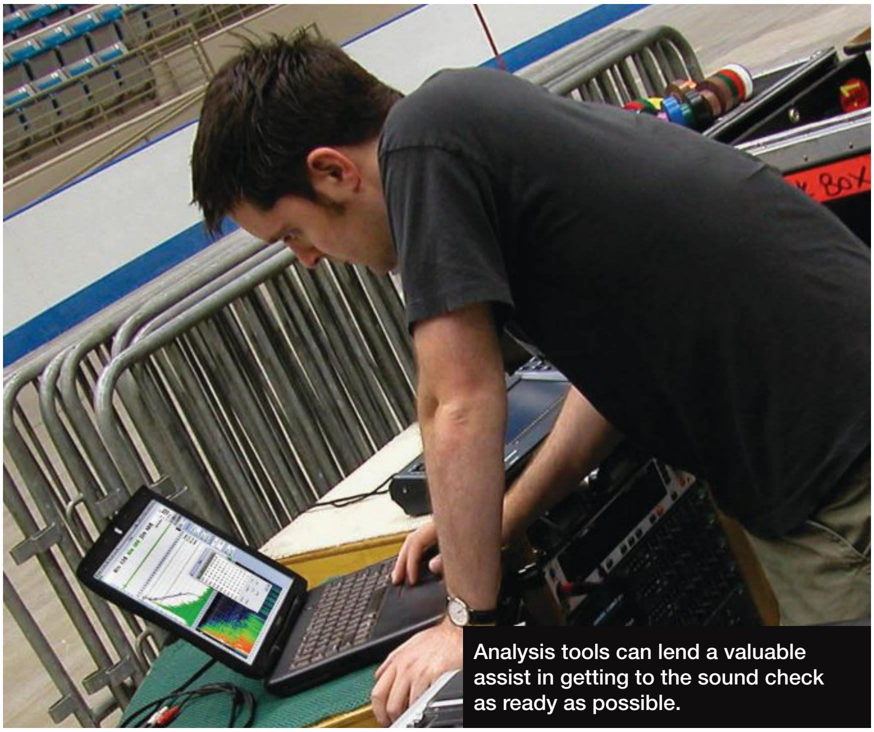
Now it's time for a line check, starting with making sure every input is working and then testing every mic and DI. If we have enough crew at the gig, this may take place simultaneously while the stage is being wired. Once the mics and DIs are placed and tested, it's time to start dialing in the channels before the band arrives. For example, we roll off the bottom end of the vocal mics and drum overheads to keep any stage rumble from bleeding into the PA. The input trims are set to a good starting position and the vocal mics are double-checked. Groups and VCAs can be assigned now, along with setting up a few standard effects like snare and vocal verbs and delays.

When the band arrives we give them time to noodle around a bit and get things positioned to their liking. Then we begin sound check by going through the instruments one at a time. Kick drum is the usual starting point, with drummers asked to strike as hard as they do during a show. The gain setting is checked to make sure there's no clipping, and then the tone is dialed in with the channel EQ. Compression and gating are also set.

Once kick is done, the channel stays live as we move on to snare. When it's sounding right, the snare verb (if requested) is dialed in, and then the drummer is asked to play both kick and snare so the balance can be evaluated. Then it's on to hi-hat and toms, played individually first and then with the other parts of the kit.

Overheads come next, with the drummer asked to play all around the kit while we listen on headphones soloed to the overhead mics. The goal is to pick up mostly cymbals and not so much snare. Depending on the show, a mic might also be employed for ride cymbal because it can get lost in the mix when using just overheads.

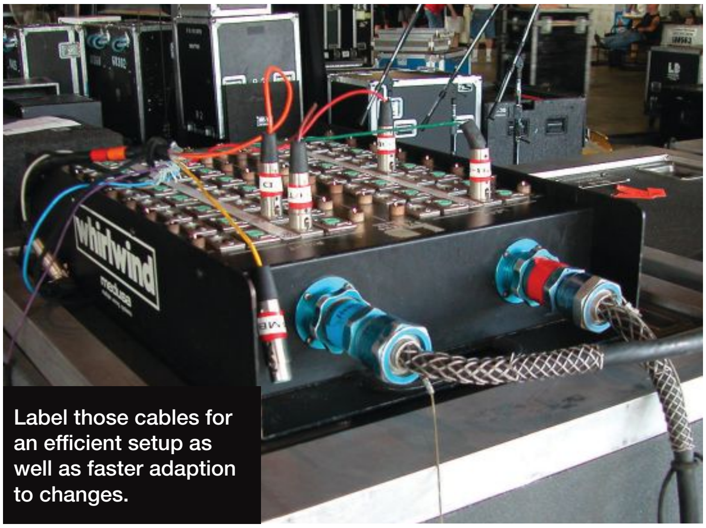
If the PA is stereo, drums are panned a bit to add some depth. Specifically, kick and snare remain centered with hi-hat panned slightly to one side and toms to the other side.


Analysis tools can lend a valuable assist in getting to the sound check as ready as possible.

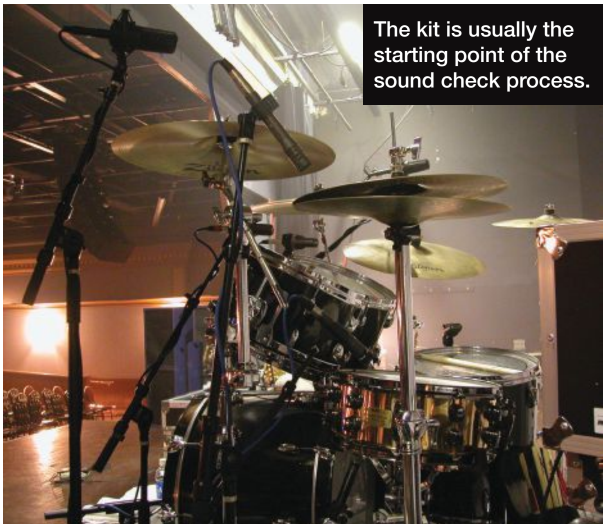
SOLID FOUNDATION

The monitor engineer does sound check simultaneously with front of house. But if we're also running monitors at front of house, now's the time to get the kit sounding good for the drummer. We ask them what they want in their monitor mix, it's provided, and once they're happy, we run a quick check to make sure the monitor and house sound are playing nice, and that's that.

Percussion follows, subject to the



Label those cables for an efficient setup as well as faster adaption to changes.


The kit is usually the starting point of the sound check process.

same methodical approach as the kit, with instruments first followed by "toys" like shakers. Panning percussion left or right (or even to both sides) can create a nice soundstage for the rest of the band.

Moving along to bass guitar, it's usually on a DI, sometimes with an additional channel for a mic on the bass amp. When that's dialed in, the bass player and drummer are asked to play a groove, which is used to set and check the balance. In stereo, the bass is usually centered with the kick. Then the percussionist (if there is one) is asked to join the groove so that all three elements can be evaluated. To me, drums, bass and percussion form the foundation of a good mix.

All stage mics are kept live as additional instruments are dialed in to make sure they aren't picking up anything they're not supposed. If so, time to move the offending mic and/or source. For performers playing an instrument and singing, the instrument is dialed in first, but with the mic on in order to hear any interactions between the two elements. Keyboard players are asked to change patches to check for drastic level and/or tonal changes between the different sounds.

Background vocalists come before the lead. More often than not, lead singers are the "stars" of the band who want an inordinate amount of time in sound check, and once they're done, they want to move on to playing a few songs.

With everyone happy with their monitors and front of house dialed in, we move along to full band songs where the mix can be fine-tuned, with every instrument heard appropriately and panned correctly. This is also the time to adjust gain settings to account for the fact that bands are usually louder in the performance than for sound check.

Finally, we talk with the band to make sure we're all on the same page, to see if they have any concerns, to talk about where they want the vocals to sit in the house mix, and so on. It's also a time to inquire about specific cues and get a set list (if one's not already been provided).

ADDITIONAL VARIABLES

In the corporate event world, sound checks work differently. Some events provide an entire day to set up and tweak the PA, others only have a couple of hours available. There may be rehearsals that provide the chance to dial in settings for individual presenters, or we may first meet presenters when clipping a mic on them right before they walk on stage.

Typically, there's enough time to set up and ring out the system. It's also best to make sure there's a ton of gain before feedback with the podium mics. Recently we've been using a cardioid mic as the main together with an omni mic for recording and as a backup. The omni has come in handy in better capturing the voices of presenters who move the mics down and away from the pre-set position.

Since presenters are hard to pin down before the actual event, a crew member is outfitted with lavalier or headworn mics and sent on stage to read some copy. This allows us to dial them in pretty well, and without having to hear "test" and/or "check" over and over.

Our biggest concern at corporate events is wireless Q & A mics in the audience. They need to be carefully placed (i.e., not in front of the PA), and a lot of time is spent removing threats of feedback. Of course, they're wireless, so they can travel anywhere... But that's the point of the whole setup and sound check process – doing our committed best to optimize everything as fully as possible while being absolutely vigilant in eliminating potential problems. ■

Senior contributing editor CRAIG LEERMAN is the owner of Tech Works, a production company based in Las Vegas.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Pac-Man Is Back

Advantages of 3D polar loudspeaker measurements. *by Joe Brusi*

WHAT'S PAC-MAN got to do with loudspeakers? Well, if you ever see him on a prediction coverage map, chances are the directivity balloon of the loudspeaker was not fully measured.

For a long time in pro audio, loudspeaker directivity was measured horizontally and vertically, with this data shown as polar plots that were most commonly seen on product specification sheets (**Figure 1**). The loudspeaker would be placed upright on a turntable with horizontal data measured, and then turned on its side for vertical measurement.

With the advent of modeling software, the need for 3D measurements (that is, measurements taken all around a sphere surrounding the device under test) became apparent (**Figure 2**). However, a system that was able to measure and store all that was complicated, slow and expensive, so most often interpolation was performed to fill in the data for the angles in between. For a typical horn, say 90 x 60 degrees, the error would be up to 2-3 dB on the front hemisphere, but many are content to live with these limitations.

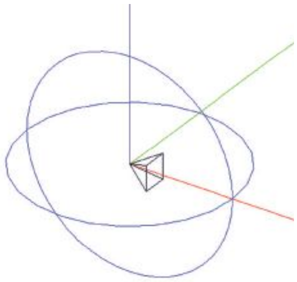


Figure 1

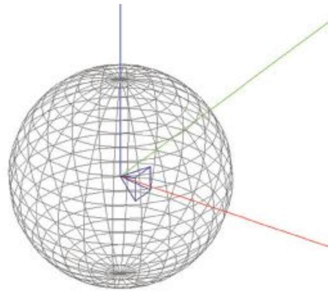


Figure 2

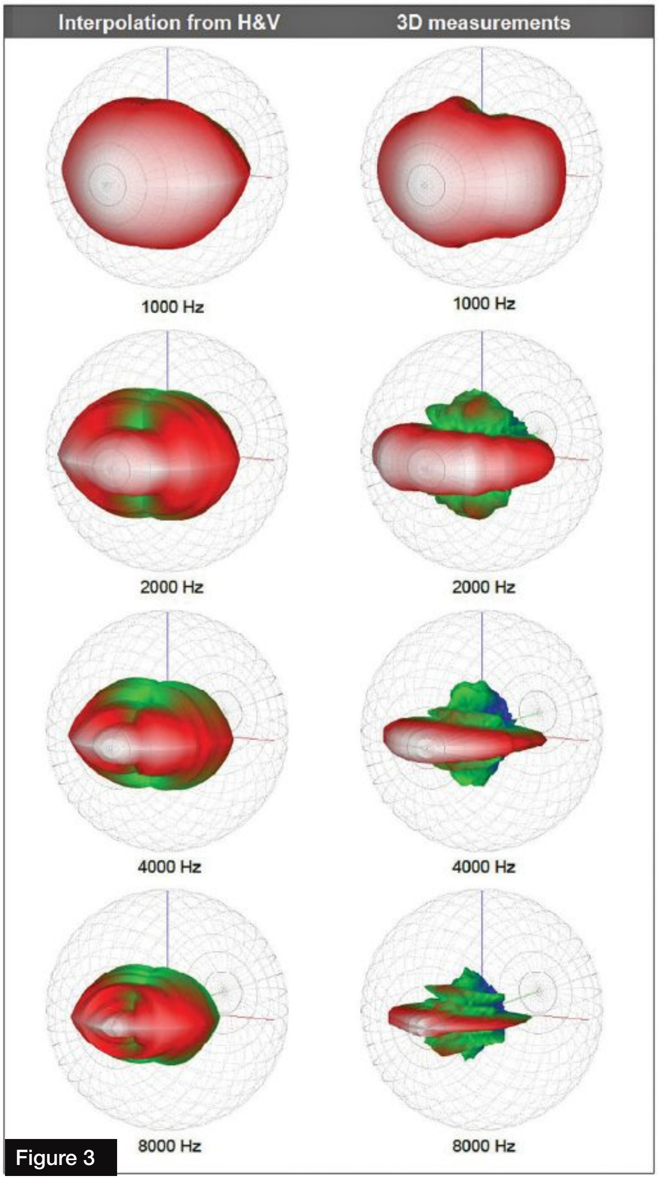


Figure 3

MUCH TO BE DESIRED

The emergence of line arrays (call them line source arrays or use any other term to your liking) for concert applications has brought the limitations to further light. These vertically articulated array modules most often use compression drivers that are invariably attached to a wave shaping guide or ribbon emulator of some sort, and this produces very narrow coverage angles at high frequencies (passive column loudspeakers commonly also have much narrower vertical dispersion as compared to horizontal, which arises from the stacking of drivers). The 3D representation of the directivity of a high frequency band looks like a balloon that's squashed vertically. Unfortunately, this means that interpolation leaves much to be desired.

Users of line arrays expect the manufacturers to provide some sort of simulation software in order to figure out the size of the required array, as well as the location and rigging configuration, ahead of time. Most earlier line array-specific prediction software calculated only vertical coverage, so the lack of good data for the missing oblique angles was not a problem, but with 3D modeling software for line arrays (that maps the results for complete listening planes) becoming much more common, the issue can't be avoided anymore. Of course, perfect coverage when seen in a simple vertical plane can show uncovered areas when a complete 3D room mapping is performed, and sub-systems such as center fills and out fills can now be included in the prediction, which means that the user will tend to request that the manufacturer provides 3D simulation software.

THE EVIDENCE: BALLOONS

To be able to compare full spherical measurements with those using interpolation, I ran 3D directivity, fully anechoic measurements on a commercial line array module from one of my customers. A resolution of 5 degrees was used, which means a whopping 2,664 (automated, naturally) measurements, a lot more than seen on the simplified drawing above. The full data set was read and directivity balloons were plotted.

I then read just the horizontal and vertical parts of the very same data (that's just 144 points, or about 5 percent of the data) and generated the information for the missing 95 percent angles via interpolation. **Figure 3** shows the balloons obtained from interpolation (left) with the ones obtained from full spherical measurements (right). For the 1 kHz, 1/3-octave band, vertical and horizontal directivity is similar, and hence the interpolation works relatively well — the resulting balloons differ but are still reasonably close. However, at 2 kHz, 4 kHz and 8 kHz, the result gets increasingly disastrous.

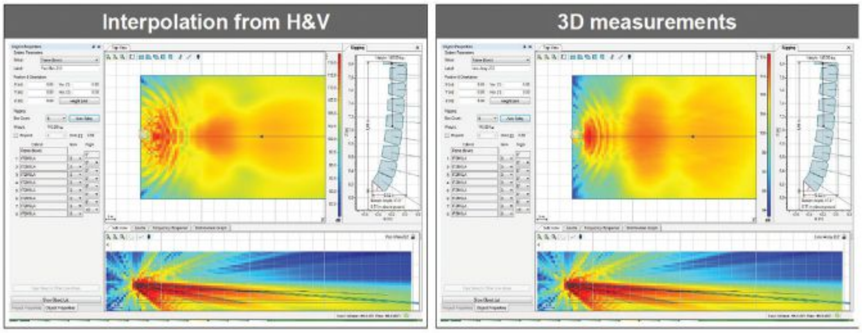


Figure 4

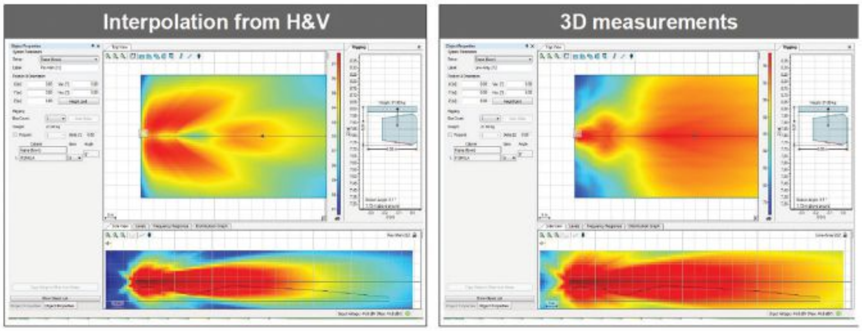


Figure 5

MAKING A DIFFERENCE

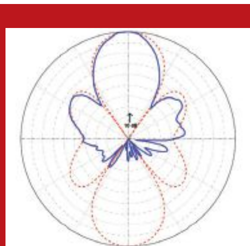
Now that we've seen how lacking directivity balloons look when only horizontal and vertical polars are measured for a line array module, let's see what happens when we model it with 3D prediction software.

Figure 4 shows the result of modeling the area below a single box, pointing forward. A 1-octave band centered at 4 kHz was used, and all settings are exactly the same. At left we see the result of using interpolation from horizontal and vertical measurements (Pac Man returns!), while at right we see the 3D measurements. All settings are exactly the same. Big difference.

Because these loudspeakers are most often used in multiples (that's why they're called arrays), let's now run some coverage predictions for an 8-module, arced array (**Figure 5**). Now the differences are not as obvious as for the single box, but they're still significant. Results from complete spherical measurements show wider more uniform coverage, a cleaner nearfield, and about 1 dB more sound pressure.

For line arrays or any other loudspeakers where the vertical directivity is significantly different than the horizontal (such as column loudspeakers), 3D polar measurements are required because interpolation from vertical and horizontal polars can result in a high degree of error. ■

Did you know that although a cone loudspeaker is modeled and often referred to as a "point source," it's far from that? This is probably not surprising, since it's a radiating surface, and as such, it shows array-like characteristics. When measuring a polar plot for a single frequency (here we used 2413 Hz on a single 18-inch subwoofer), this becomes very obvious, with all of its notching glory (we can model the front part of the polar using Bessel functions for a somewhat smaller piston, which you can see on the dashed curve). Oh, and you'll never get a perfectly symmetrical polar plot from a paper cone.



JOE BRUSI runs Brusi Acoustics (www.brusi.com), an electro-acoustic consulting firm based in Valencia, Spain that provides a range of services, including loudspeaker directivity measurement.

Radial Engineering Firefly

Evaluating a recently released tube direct box.

By M. Erik Matlock

DURING MY YEARS on the road, working with hundreds of performers, Radial products appeared regularly. I honestly never saw a single one fail, and every Radial product I ever used provided clean, solid signal.

When the Firefly tube direct box arrived for my evaluation, I was greeted with a larger package than expected. It was also surprising to see it painted about the same color as a Caterpillar bulldozer, immediately giving the impression of something powerful. The box feels indestructible, and the controls are logical and well labeled.

IN, OUT & AROUND

The Firefly is a high-output DI that combines the company's zero-negative feedback class-A front end with a 12AX7 tube drive circuit and signature transformer coupled output. A and B inputs with easy access controls enable switching between two different instruments while delivering a consistent signal to the PA and onstage amp. Switching can be done via the front panel or with the optional JR2 remote control that connects to the Firefly using a balanced cable. It also adds a mute function for quiet on-stage adjustments.

The rear panel offers the two instrument inputs (1/4-inch) as well as a THRU output for the onstage amp. THRU provides a choice of pre or post tube configuration, with a TRS insert jack enabling external effects to be shared between both inputs. A buffered "always on" tuner output can be used to split the instrument signal or it can be dedicated to an electronic tuner. The tuner output can be used for silent tuning when the JR2 remote is connected.

Housed in a 14-gauge steel enclosure, Firefly measures 1.7 x 5.7 x 8.2 inches (h x w x d), with height increasing to 2.7 inches with the handle. Weight is just under 4 pounds. It can also be rack-mounted alone or in pairs with an optional kit.

The front panel offers two inputs with individual trim controls for adjusting the input level. There's also a "drag" control for passive pickups, adjusting the load settings. And the low-cut knob is a nice touch for cleaning up subharmonics and tightening everything up.

The front panel includes a level knob to control the output, while the rear panel features recessed buttons for phase inversion, ground lift and pre/post for sending the aux out processed or direct.

THROUGH THE PACES

My expectations for the Firefly were high. Being a moderately lousy bassist, I already knew what I was going to do with it. Retired from the studio and no longer working live shows for a living, I don't have million dollar rigs and world-class gear at my disposal, and thus used pretty much the same gear as the average bar band to test the performance of the Firefly.

The first thing I did was drop the manual back in the box. Had to. I love to go as far as possible before cracking the manual open, to see how user friendly and intuitive a piece of gear is. I'm happy to say this one is pretty straightforward.

A look at the front and back panels.

The Radial Engineering Firefly tube direct box.

My bass, an older Fender Squire MB5, has been used on several recording sessions and live shows over the years. Not the best out there, but it has nice action and tone. I inserted the Firefly between the bass and an older VOX amp used for practice. It's no powerhouse but fills up a decent-sized room.

With expectations peaking, I fired up this little rig, and it worked fine but nothing dramatic. I saturated the 12AX7 tube and tried again. Better but not over the top.

I moved along using the Firefly with my small PA, a powered mixer with 15-inch loudspeakers. And wow, things got down to business. I immediately felt the change. The bass was now fat and rich, with the tone thickening up substantially. It felt like I 'd just switched over to a Fender Precision studio bass. The sound was well defined and overall better than ever.

Time for the final test, using the Firefly with my favorite recording studio remnants, a pair of vintage Pioneer HPM-100 four-way reference loudspeakers that I nicknamed the "lie detectors." They were my secret weapon for impressing clients, using them as far-field monitors, mounted on the front wall of my control room. They sound spectacular and can get wicked loud. I find that they regularly expose details and weirdness in tracks that I've been listening to for years.

Well, the old Pioneers approved. The tone and definition were truly impressive. The bass came alive with warmth and detail that I've never able to capture when recording through it. Sold. Radial has raised the bar and impressed me all over again.

Firefly offers plenty of useful features for both live and studio applications. The flexibility in setup, the rich tone and warmth – along with the road-worthy construction – make it a winner. For anyone trying to find that signature sound, this may be just the ticket.

U.S. MSRP: $599.99

M. ERIK MATLOCK is senior editor for Live Sound International and ProSoundWeb, and has worked in professional audio for more than 20 years in live, install, and recording.

DPA Microphones d:dicate 2011C

Evaluating a twin diaphragm capsule and compact preamp.

By Craig Leerman

The DPA d:dicate 2011C.

THE DPA MICROPHONES d:dictate 2011C is outfitted with a twin-diaphragm cardioid capsule designed to handle a range of miniature microphone duties. A member of the company's d:dicate reference standard series, it includes two opposite-facing miniature capsules that are custom re-built into a double-diaphragm/one-capsule composition.

The capsules, which also incorporate DPA's proprietary Interference Tube technology, are loaded to the MMP-C compact preamplifier body. The twin diaphragms combine the advantages of small capsules (fast impulse response and large frequency bandwidth) with lower inherent noise achieved from a larger diaphragm area. The Interference Tube assists in controlling both directivity and low-frequency response.

The 2011C can also be mounted in any of the company's other MMP preamp bodies, including the standard MMP-A and smaller MMP-B with low-cut and high-boost filters. The MMP-C preamp evaluated here offers a subtle character alteration comparison with the MMP-A, providing a bit more "body" for lower frequencies in addition to an easier-to-position size.

For even further flexibility, the capsules can be mounted on the available modular active cable assemblies (offered with rear or side cable entry) so that they can be used as hanging or low-profile table and instrument mics. And, modular active booms foster positioning the capsule on a podium or above performers. Other capsules in the series include single and twin diaphragm models in a variety of sizes and patterns, including compact and shotgun configurations. They're also available in kits and stereo matched pairs.

NEW FAVORITE

The 2011C is a pre-polarized condenser type offering a stated frequency response of 50 Hz to 17 kHz with 3 dB soft boost at 12 kHz (± 2 dB) when used at roughly 12 inches. The sensitivity rating is -40 dB and maximum (peak) SPL is 146 dB before clipping. The dynamic range of the MMP-C preamp only is specified as 136 dB, with a low-cut frequency of 15 Hz (-3 dB).

Out of the box I was really surprised at just how small the capsules and preamp actually are. When connected together they measure just over 3.5 inches long and weigh about 2.3 ounces. A nice metal base mic clip was included with the U.S. standard 5/8-27 thread size, along with an adapter for the smaller 3/8-16 European standard. Optional mounting accessories include shock mounts, stereo bars with clips or shock mounts, and the aforementioned cables and booms.

On the bench I plugged the mic into my test PA system and supplied it with the required phantom power. It sounds great and retains the same tonality off to the sides of the pattern. There's also excellent rejection to the rear.

Since I had a small console on the bench that recorded directly to USB, I grabbed a thumb drive and made a room recording. The mic exhibits a very low noise floor and sounds very natural. Next I recorded an acoustic guitar, positioning the mic a few different ways. No matter where I placed it, the guitar sounded quite good, with some positions sounding stellar. As a result, this may be my new favorite acoustic guitar mic.

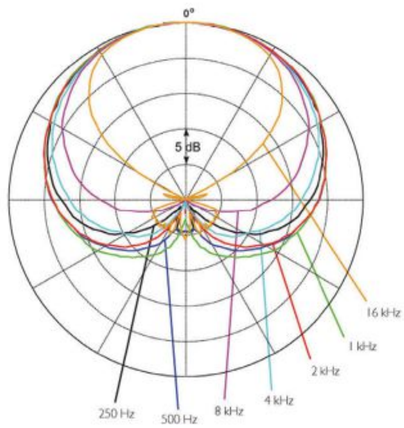
DIFFERENT USES

Assured that everything was working properly, I took the 2011C out to a couple of gigs. First up was a band in a large hotel ballroom. They were playing over the course of two days, so I decided to apply the mic to two different instruments.

Day one it pulled conga duty with the percussionist, placed between the drums about a foot away and above the rims. The congas sounded great, and as a result I even had to fight the urge to sit the congas a little hotter in the mix.

Day two, I put the 2011C on a clarinet, swapping it in for a large diaphragm mic. When the clarinet player arrived, he noticed the change, and I let him know we could switch back to the other mic if he didn't like this new one. But after just a few minutes, he told me how much better it sounded, and at front of house, I noticed that it was requiring far less EQ than the previous mic.

Next up was a corporate event with an announcer providing the introductions for a company's award winners and making general show announcements. Being an international company, some of the names were extremely difficult to pronounce, so to avoid any mistakes during the show he wanted to record files of the names of the award winners, along with other info, and then have me play them back as cues during the event.



Directional characteristics of the 2011C (normalized).

So I set up the 2011C on a desk stand and had the announcer speak from about a foot away from the mic. At that distance, there were no problems with plosives, and the recordings sounded very natural.

Because he usually speaks closely into a standard dynamic mic, he was accustomed to hearing the bass proximity effect on his voice and thought the recordings sounded a little "thin." So I added a bit of low-end boost on playback to match the tone of the standard dynamic announce mic, and he was very happy. Of course, so were the award recipients who all had their names pronounced correctly, never knowing that some names took more than 20 takes to get right!

So if you're looking for a flexible microphone system for live and recording applications, the MMC2011 capsule with MMP-C preamp is definitely recommended. In fact, the entire d:dicate line should make the list -- with eight capsules, three preamp options and plenty of mounts, cabling and mounting options, there's at least one configuration to meet every need.

U.S. MSRP: $829.95 ■

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

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CELESTION

MAKING A DIFFERENCE

Recent developments deployed for a variety of applications. *by Live Sound Staff*

MULTIPLE STAGES AT SXSW

Nomad Sound (Austin) deployed NEXO STM arrays and Yamaha digital consoles at Austin Music Hall for SXSW, which hosted performances by The Ting Tings, Chance the Rapper, Tove Lo, Clean Bandit, Elle King, the Mowglis, and a several dozen others.

Specifically, arrays of nine M46 main modules per side were joined by nine B112 bass modules, with 20 S118 sub modules on the ground and four M28 compact modules for front fill. On stage, the monitor package consisted of eight 45N-12 wedges downstage, with an RS18 Ray Sub/45N-12 combo for drum monitoring. Seven NUAR amp racks powered the loudspeakers.

"The S118 subs are true and very musical in their response," says Stefan Bouts, system tech for Nomad Sound. The M46/B112 combination continues to amaze me in their performance, particularly for me, as primarily a system technician, the speed at which we can deploy or strike such a large and great sounding array, thanks to the PistonRig system."

Behind a NEXO STM array and subs at Austin Music Hall at SXSW.

A Yamaha CL5 digital console was used for front of house mixing with another CL5 for monitors. Rio3224-D and 1608-D input/output boxes were combined, daisy-chain style, for interface purposes. Meanwhile, at South by San Jose, one of the unofficial SXSW side parties, Nomad Sound deployed a NEXO rig consisting of three per side M28s ground stacked, joined by two per side B112s sitting on four per side STM subs. A six-pack of NEXO PS15s, along with a Yamaha QL5 digital console, worked together in the monitoring system. The company also outfitted several other stages with NEXO/Yamaha combos.

King Charles monitor engineer Ross Rothero-Bourge mixing on a laptop tied into Allen & Heath iLive.

HOUSE & MONITOR MIXING FOR KING CHARLES

A recent UK tour by British singer-songwriter King Charles carried an Allen & Heath iLive digital mixing system to manage both front of house and monitors. The mix system included an iDR-32 and iDR0 MixRacks, with an iLive-80 surface for front of house and a laptop and iPad for monitor control.

"On a concentrated tour like this with 28 gigs in a row, being self-contained is really important. The fact that iLive is so compact but so powerful is really beneficial," explains front of house engineer Jonathan Lewis. "The band is a pretty standard setup of drums, bass, keys, two guitars, lead vocals and three backing vocals. We could have managed just with the iDR-32, but adding the iDR0 means we didn't have to share PFL or scenes, and it gave us another 64 channels to cater for all the in-ear splits."

Monitor engineer Ross Rothero-Bourge adds, "The venues ranged from 200 to 2,000 capacity, so space was often a tight squeeze, but as I used a laptop or iPad instead of a surface, I could tuck myself away while being free to move close and keep eye contact with the band.

"iLive is so simple to use and set up," he concludes. "We can put the surface out front but keep the mixer on stage with me and just run Cat-5 between them."

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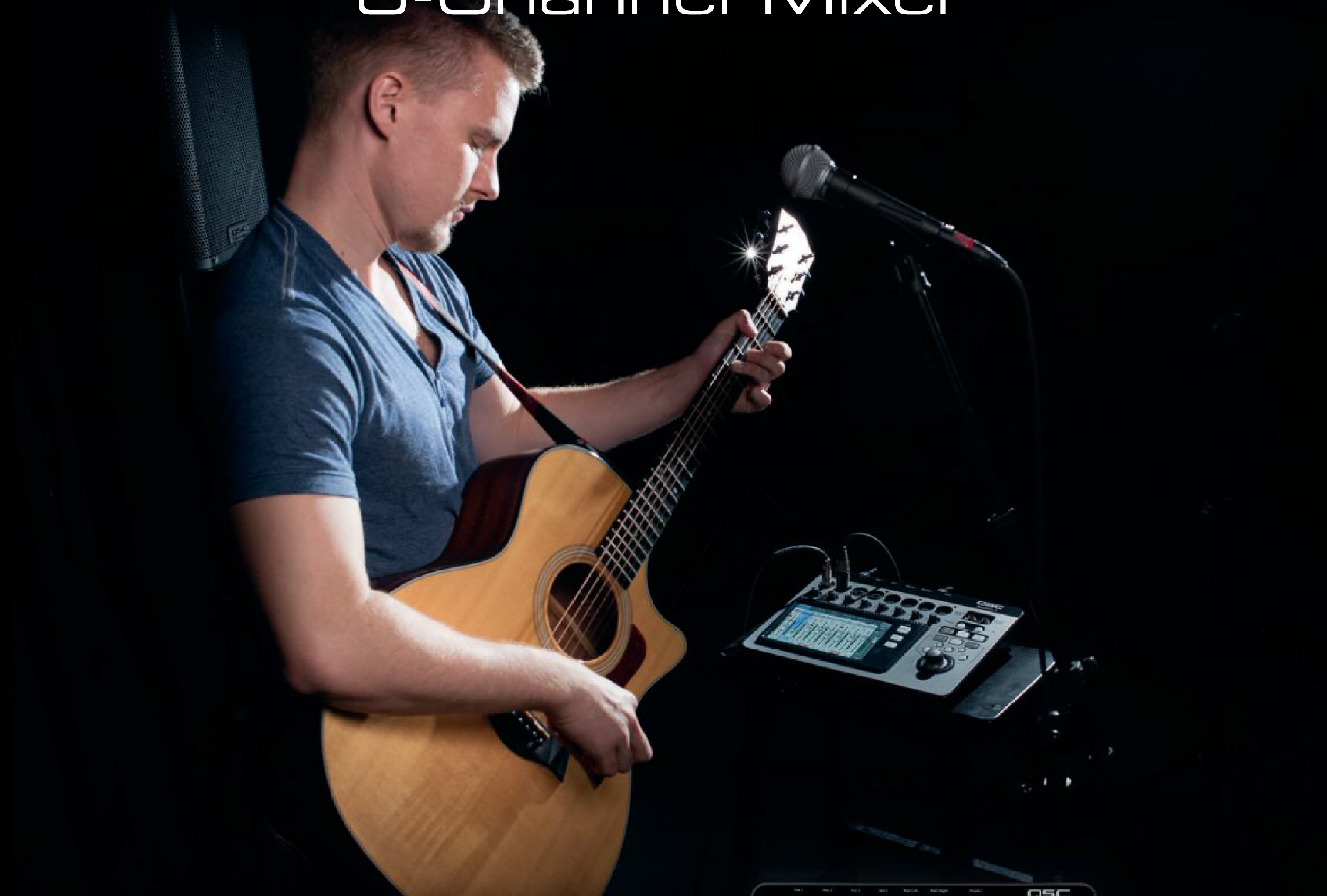
8-Channel Mixer



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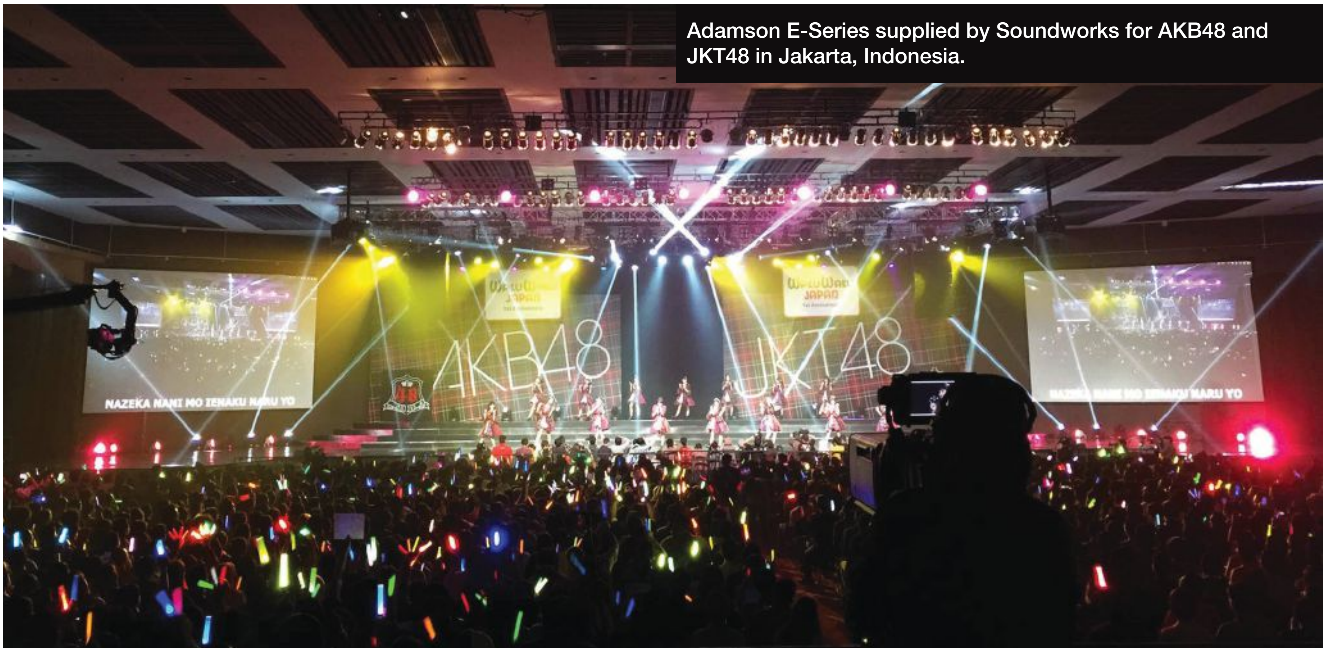


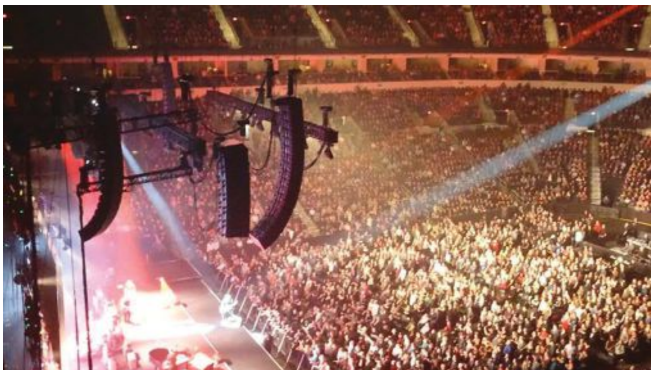
Adamson E-Series supplied by Soundworks for AKB48 and JKT48 in Jakarta, Indonesia.

MAKING AN IMPACT IN JAKARTA

Soundworks (Jakarta, Indonesia) recently joined the Adamson Systems E-Series Network, adding 24 E-Series E15 line array enclosures and 16 T21 subwoofers that were immediately put to work at a concert by female "idol" groups AKB48 (Japan) and JKT48 (Indonesia) at The Kasablanka, also in Jakarta.

"We were looking for a large-format line array to complete our arsenal when we saw the Linkin Park concert in Jakarta with the new E-Series loudspeakers, and it was the best sounding concert we had ever heard," states Rio Felani, senior sound engineer with Soundworks, which has been providing the region with production services since 2012.

The new rig has also been used in support of concerts and tours by Peter Cetera, David Foster and Tommy Page as well as at the 2014 Indonesian Music Awards and Yahoo OMG Awards.



d&b audiotechnik arrays and monitors in the arena with Bryan Adams.

Cowboy Keith Thompson, front of house engineer for Peter Cetera, notes, "I was happy to see the Adamson E15 rig when I walked into the venue in Jakarta. The local crew had hung and tuned the PA well. The coverage was solid from the front row, all the way to the back of the theater. The high mids were tight and had a great warm tone for Peter's vocals."

FLEXIBLE COVERAGE FOR BRYAN ADAMS ON TOUR

Gearforce (Coquitlam, Vancouver) is supplying a d&b audiotechnik main system and monitors for the current tour by Bryan Adams. Gearforce owner Rob Nevalainen invested in d&b V-Series last year, noting, "We used it almost immediately on a Bryan Adams solo tour. The V system brought many benefits, in particular better coverage with the use of V8 and V12 combinations, and the weight issue was also a key factor: we could fly it virtually anywhere."

This time, the Gearforce V-Series system is out with a full-size arena production for Adams, traveling across Canada before moving on to Europe. "Our first full band tour, we have d&b J-Series for left/right mains which I've sub-hired from EDS, and our own V-Series for the side hangs, with Y-Series for the rear in the bigger venues where we play 270 or 360 degrees, all driven by D80s. It sounds great," he says.

Nevalainen, who has also served as Adams' monitor engineer for more than a decade, is pleased with his monitor as well: "Although all the band are on IEMs I have side fills composed rather unconventionally of J-INFRA with Y10 on top. The lower octave reach of the INFRAs is ideal for building

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the kick drum sound on the stage apron; it's big, loud and rich, and all the band feel it.

"The Y-TOPs enable me to feed guitars to stage left, and vocals to both sides with remarkable clarity and power for such a small cabinet," he concludes. "The vocals in particular are very useful for the many guest artists that turn up on Bryan's shows; the tightness of the dispersion putting sound right where I want it."

UNDER THE TENTS AT COACHELLA

Rat Sound (Camarillo, CA), which has been providing sound reinforcement for the Coachella festival in Indio, CA since 2001, this year chose to deploy EAW Anya line arrays at the Gobi and Yuma tents. The Gobi tent, covering approximately 18,000 square feet, hosted artists this year that included Kimbra, Lights, FKA Twigs, Cahsmere Cat and others. Coverage was delivered by two Anya arrays, each comprised of eight modules, flown to the left and right of the stage.

The Yuma tent – complete with wooden floors and air conditioning – presents EDM to an oval-shaped area measuring roughly 135 by 200 feet, with a large, elevated DJ booth in the center of one of the longer sides of the oval. The sound design included four stacked Anya arrays, each made up of five modules with an inner column (three modules) and an outer column (two modules). Two of these arrays were located to the left and right of the DJ booth, with the other two placed on the opposite wall. All were on five-foot platforms.

"The visiting engineers responses to both systems were good," notes Dave Rat, owner of Rat Sound. "Most of them had heard of Anya but never mixed on it, so they were intrigued with trying it out. The system came up and sounded great — which is what an engineer wants for a live show."

EAW Anya in the Gobi Tent at Coachella 2015.

Martin Audio MLA and MLX subwoofers support The Prodigy in Berlin.

POUNDING THE EDM BEAT IN GERMANY

Complete Audio Berlin (Germany) recently supplied a Martin Audio MLA system for The Prodigy at Berlin's 12,000-capacity Velodrom. Mixed by Jon Burton, the electronic music group is known for heavy breakbeats and chest-thumping basslines.

Complete Audio fielded two main left-right hangs of 11 MLA elements each, with two MLD Downfills. Side hangs consisted of two drops of eight MLA Compact. Because The Prodigy is all about the bass, the production company deployed 24 MLX enclosures, each housing two hybrid-loaded 18-inch drivers. The crew spread 18 along the stage apron, leaving gaps for other in fill subs, with a further three MLX per side flown between the main PA and MLA Compact out fills.

"We made the design using Martin Audio's MLX sub-arc calculator and our main goal was to get a pretty even dispersion of the sub energy throughout the venue," explains Complete Audio managing director André Rauhut. "Jon is quite clear about what he wants and asked for a broadside sub-array, which in any case is what we would do in that kind of venue."

While Burton mixed on a Midas XL3 analog desk, head system tech Nils Uhthoff, assisted by Benny Franke and Martin Eckert, helped in optimizing the acoustics using Martin Audio's Display 2 software, including use of the program's "hard avoid" function to keep energy away from surfaces with no audience in order to prevent unwanted reflections. ■

Design Meets Intuition

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Big, Bigger & Biggest

The latest large-format digital consoles. *By Mark Frink*

BEFORE THE ADVENT of stereo in-ear monitors and double-miking instruments, long before dedicated effects and "sweetening" tracks (and "click"), way before the B-stage and the guest inputs, consoles with 32 channels and 16 auxiliaries covered any band and most festivals. But today, 32 x 16 desks are consigned to clubs and support acts, with today's largest shows often requiring 100 channels and 50 mixes or more.

Left to right, engineers Dickie Chappell, Dee Miller and Ben Findlay with an SSL Live.L500 large-format console on Peter Gabriel's Front to Back tour.

The digital train keeps a-rollin': the advantages of digital consoles are so numerous we may take many for granted. And Neal Young is right: once considered a frivolous extravagance, today's largest consoles operate at 96 kHz.

From "copy and paste" to preset libraries for EQ, effects, dynamics, entire channels and even sets of channels (like an entire drum kit), there's a lot to work with. Presets can even be made of an auxiliary send, making monitor mix presets possible.

Control groups are even larger on the biggest desks. Engineers used to worry they wouldn't have enough faders on digital consoles to manage high channel count shows, but the new digital workflow allows engineers to mix from DCAs, "spilling" contributing channels to the next fader bank where they can be tweaked. New designs improve workflow with custom fader layers, but control group "spill" onto adjacent fader banks is the ultimate custom layer.

Size and weight are obvious digital advantages. Analog consoles and their associated outboard racks, splitters and multi-core interconnect snakes are becoming a thing of the past. Large digital control surfaces weigh one or two hundred pounds and take far less room than analog equivalents for smaller footprints in theaters and at festivals.

Digital snakes are eliminating the hums and buzzes of yesterday, and networking a digital console's preamps on stage shortens the path from mic to preamp. Gain-sharing means consoles don't need a splitter, so mix engineers don't have to listen to their inputs through transformers.

"In-the-box" mixing, where mix processing is performed entirely within a digital console, means engineers can save shows and open them on identical or similar consoles. They can email a file to the next gig's console vendor, "cc-ing" themselves a copy as backup.

Remote control of a console from an iPad has become a standard feature and a powerful tool, allowing front of house engineers check their listening area while monitor engineers can stand beside performers – both with controls in hand.

MADI has enabled manufacturers to provide multi-track I/O for recording and playback to laptop-based DAWs providing affordable virtual sound check. This simple innovation allows engineers to easily check a previous show, test sound systems with a previous performance, practice mixing or tweak a show file, teach others to mix, and afterwards, easily mix a show down for distribution using the same console.

Another advent is simple 2-track recording and playback using USB "thumb drives," allowing engineers to walk away from a console with a board mix they can easily listen to and then e-mail to others if they like. It also provides simple and foolproof playback of walk-in and intro music with no moving parts.

Enjoy our Real World Gear tour of today's big, bigger and biggest digital consoles.

MARK FRINK is an independent engineer, consultant, author, and a presenter at Summer NAMM's new TEC Tracks.

Hey Marseilles. Nectar Lounge. Seattle, WA. 09.17.2014

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01 **Hardware:** Flexible, professional I/O in an incredibly compact 3U rackmount design

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

05 **Master Fader:** Intuitive wireless control over everything, proven at more than 2 million live mixes

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Roland Pro AV M-5000

proav.roland.com

Based on the O.H.R.C.A. platform, the M-5000 live mixing console is built on a core philosophy of "Open," "High Resolution," and "Configurable Architecture." It allows for 128 freely assignable audio paths, support for Dante, Waves, MADI, native 96 kHz audio quality throughout the system, and introduces liberating user definable workflows that make operation fast, accurate and friendly.

In the rapidly changing world of live sound, the ability to adapt on the fly to meet the needs of the application is sometimes the difference between a good show and a great show. The power of O.H.C.R.A. in the new M-5000 combines flexibility, usability and incredible sound quality in an approachable console ready to meet the current and future demands of audio professionals regardless of application.

The M-5000 console is currently on tour with Manhattan Transfer and Take 6.

OF NOTE

Up to 300 inputs and 296 outputs (and more at 48 kHz) are managed in separate patchbays. Any input can be patched to any or multiple outputs, including control of gain and phantom power, without having to be patched through a mixing channel.

KEY SPECIFICATIONS

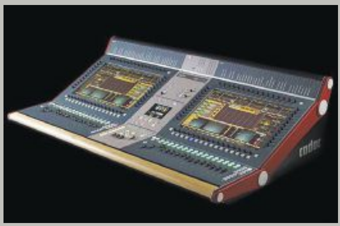
Faders: 28
Total Mix Paths (Inputs/Aux/Group/Matrix): 128
FX: 8 stereo
DCA: 24
GEQ: 32
App: M-5000 Remote (iPad)
Screen: 12-inch touch screen
Local I/O: 16 + 16 analog XLR, 2 + 2 AES/EBU
Stage Boxes: S-4000-3208, S-2416, S-1608, S-0808
Card Options: REAC, Dante, MADI, Waves
Also: iPad dock, M-48 personal mixers
Physical: 37 x 29 x 14 inches, 79 pounds
Additional Models: M-480

Yamaha PM10 RIVAGE » www.yamahaca.com

Faders: 36 + 2
Mix Inputs: 144
Aux/Group: 72
Matrix: 36
FX: 45 types, up to 384 instances
DCA: 24
GEQ: 48
App: StageMix (iPad)
Screen: Two 15-inch touch screens
Local I/O: 8 + 8 analog XLR, 4 + 4 AES/EBU
Stage Box: RPio622
MY Cards: Dante, MADI, Aviom, many others
Also: Neve inserts, TC and Eventide FX, SILK mic preamp
Physical: 61 x 33 x 17 inches, 187 pounds
Additional Model: CL5

DiGiCo SD5 » www.digico.biz

Faders: 36 + 1
Mix Inputs: 124
Aux/Group: 56
Matrix: 24 x 24
FX: 24 stereo
DCA: 24
GEQ: 32
App: DiGiCo SD (iPad)
Screen: Three 15-inch touch screens
Local I/O: 8 + 8 analog XLR, 4 + 4 AES/EBU
Stage Boxes: SD Rack, SD-MINI, SD-NANO, D2-Rack
Options: UB MADI adapter, Waves SoundGrid, Aviom
Also: 40 Smart Key macro buttons, Optocore fiber loop
Physical: 58 x 33 x 18 inches, 256 pounds
Additional Models: SD7, SD10



CADAC CDC eight

www.cadac-sound.com

The CDC eight features 128 input channels and 64 buses – 56 of which are user assignable – accessed by Cadac's intuitive "high agility" gesture navigation user interface, displayed on 24-inch touch screens. With Cadac's innovative approach to the user interface, which is as far less menu dependent than traditional digital consoles, it leads to an instinctive use of "touch and swipe" of the encoders and screen.

The CDC eight's exceptional audio quality is due to Cadac's legendary mic pre, combined with a time-aligned, phase-coherent mix bus architecture, resulting in sub 0.4 millisecond latency from analog inputs on stage, through the console, to analog outputs on stage.

The CDC eight comes preconfigured with an integrated 64 x 64 Waves interface for direct multi-track recording to a laptop and connection to Waves MultiRack SoundGrid server.

OF NOTE

Audio transport employs Cadac's propriety MegaCOMMS protocol which provides 128 bi-directional channels of 96 kHz / 24-bit audio. With the addition of the CDC MC Router, the audio network can be expanded up to 3,072 channels and integrated into third-party audio protocols via CDC MC MADI and CDC MC Dante network bridges.

KEY SPECIFICATIONS

Model: CDC eight-32
Faders: 32
Mix Inputs: 128
Aux/Group/Matrix: 56
Matrix: Up to 32 x 32
FX: 16 stereo
DCA: 16
GEQ: 56
App: Cadac Remote (iPad)
Screen: Two 24-inch and one 6-inch touch screen
Local I/O: Up to 64 AES/EBU, analog XLR or TRS
Stage Boxes: CDC I/O 6448, CDC I/O 3216
Options: MC MADI bridge, MC Dante bridge
Also: SAM theatrical automation software
Physical: 64 x 35 x 17 inches, 245 pounds
Additional Models: CDC eight-16, CDC eight-16S

Allen & Heath GLD 112 Chrome Edition »

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

Faders: 28
Mix Inputs: 48
Aux/Group/Matrix: 20
FX: 8 stereo
DCA: 16
GEQ: 20 (4 types)
App: GLD Remote and OneMix (iPad)
Screen: 8.4-inch color touch screen
Local I/O: 4 + 4 analog XLR, 4 + 2 analog RCA, 1 + 1 AES/SPDIF
Stage Boxes: AR2412, AR 84, AB168
Card Options: Dante, MADI, Waves, EtherSound, ACE, ADAT/Aviom
Also: ME-1 personal mixers, automatic mic mixing
Physical: 47 x 29 x 11 inches, 60 pounds
Additional Models: GLD 80 w/20 faders

Avid S6L-32D w/E6L-192 » www.avid.com

Faders: 32 + 2
Mix Inputs: 192
Aux/Group: 96
Matrix: 24 x 24
Plug-in Slots: 200
DCA: 24
GEQ: 32
Screen: Four 12-inch touch screens
Local I/O: 8 + 8 analog XLR, 4 + 4 AES/EBU
Stage Boxes: Stage 64
Card Options: Ethernet AVB, Dante, MADI, Aviom, Thunderbolt
Also: HDX-powered 64-bit AAX plug-in DSP, True Gain sharing
Physical: 51 x 31 x 15 inches, 154 pounds
Additional Models: S6L-24D, S6L-24, E6L-144

Lawo mc² 36
www.lawo.com

The Lawo mc²36 is an all-in-one mixing desk with a comprehensive feature set that covers theater, house of worship, live, install and broadcast applications. Along with uncompromised sound quality and Lawo-grade micpreamps, it offers unprecedented value.

Available with 16, 24 and 40 faders and integrated I/O, it's ideally suited to permanent installations with limited space. The intuitive layer concept with its reveal feature fosters convenient handling of large productions without the need for dozens of faders.

The powerful DSP micro-core with internal 512 x 512 port audio matrix is natively equipped with RAVENNA/AES67 technology, allowing the mc²36 to integrate seamlessly into IP infrastructures. In addition to Lawo's internal processing, the seamless integration of Waves SoundGrid provides convenient access to the world of Waves plug-ins.

OF NOTE
The mc²36 can supply up to 496 channels, and the integrated RAVENNA/AES67 audio-over-IP interfacing makes the console a future-proof investment that can grow with demand.

KEY SPECIFICATIONS
Faders: 24
Total Mix Paths: 192 @ 48 kHz, 48 @ 96 kHz
Aux/Group: 32
GEQ/FX: Waves SoundGrid
DCA: 128
App: Remote Desktop (PC)
Screen: Two 21-inch touch screens
Local I/O: 32 + 32 analog XLR, 4 + 4 AES/EBU
Stage Boxes: mc² Compact I/O
Also: MADI port, RAVENNA/AES67 audio-over-IP, remote control of Neumann AES42 digital mics
Physical: 42 x 35 x 16 inches, 106 pounds
Additional Models: 16- and 40-fader versions of mc² 36

Soundcraft Vi7000 » www.soundcraft.com

Faders: 40 + 4
Mix Inputs: 128 @ 48 kHz, 64 @ 96 kHz
Aux/Group/Matrix: 32 stereo
Matrix: Up to 16
FX: 8
DCA: 16
GEQ: 35
App: ViSi Remote (iPad)
Screen: Five 12-inch Vistonics touch screens
Local Rack I/O: 16 + 16 analog XLR, 8 + 8 AES/EBU plus 3 XLR mic
Stage Boxes: Vi Stagebox, Compact Stagebox
Card Options: EtherSound, CobraNet, Dante, MADI, ADAT, Aviom
Also: UA real time plug-ins, RF mic monitoring, 16 BSS DPR 901
Physical: 68 x 29 x 13 inches, 139 pounds
Additional Models: Vi5000, Vi3000

Solid State Logic Live.L500 » www.solidstatelogic.com

Faders: 36 + 2
Total Mix Paths (Inputs/Aux/Stems/Masters): 208 full processing paths + 48 insertable dry paths
Matrix: 32 x 36, can be split into up to 4 matrices
FX: 20 stereo
DCA: 36
GEQ: 96
Also: Optical FX loop for Waves SoundGrid
Screens: 19-inch touch screen, 7.5-inch touch screen
Local I/O: 32 + 32 analog XLR, 8 + 8 AES/EBU
Stage Boxes: ML 32.32, ML I.32, D 32.32
Options: MADI-Dante Bridge, Live-Recorder SSD PC
Physical: 47 x 35 x 17 inches, 194 pounds
Additional Models: L300

Studer Vista 5SR » www.studer.ch

Faders: 30 + 2
Mix Inputs: 80 mono/20 stereo
Group: 8 stereo (FOH configuration)
Aux: 10 mono/10 stereo (FOH)
Matrix: 5 mono/5 stereo (FOH)
FX: 12 stereo (w/3 optional Vista FX units)
DCA: 16
App: Virtual Vista (PC)
Screens: Three Vistonics touch screens
Local I/O: 16 + 48 analog XLR, 16 + 16 AES/EBU
Stage Boxes: Compact Stagebox, D21m, D23m
Options: EtherSound, CobraNet, Dante, MADI, ADAT, Aviom
Physical: 59 x 30 x 15 inches, 149 pounds
Additional Model: Vista 1

StageTec AURUS Platinum » www.stagetec.com

Faders: 36 as shown (up to 96)
Mix Inputs: 300
Aux/Group: 128
Matrix: 32 x 32
DCA: 24
GEQ: 32
Screens: Five 19-inch metering and one 10-inch touch screen
Stage Boxes: NEXUS Base Device modular frames
Also: 32-bit TrueMatch A/D mic preamp conversion
Physical: 54 x 42 x 41 inches, 90 pounds
Additional Model: CRESCENDO

Midas Pro X » www.midasconsoles.com

Faders: 26 + 4 +1
Mix Inputs: 144
Aux/Group: 72
Matrix: 24
FX: Up to 48
DCA: 10
GEQ: 36
App: MixTender 2 (iPad), PalmMix (iOS)
Screens: Two 15-inch touch screens
Local I/O: 8 + 8 analog XLR, 4 + 4 AES/EBU
Stage Boxes: DL151 through 155, DL251&252, DL351, DL451
Options: DN9650 & DN9652 MADI/Dante/CobraNet/ES bridges
Also: 3-way KVM switch
Physical: 54 x 37 x 17 inches, 213 pounds
Additional Models: PRO9, PRO6, PRO3

Yamaha CL5 » www.yamahaca.com

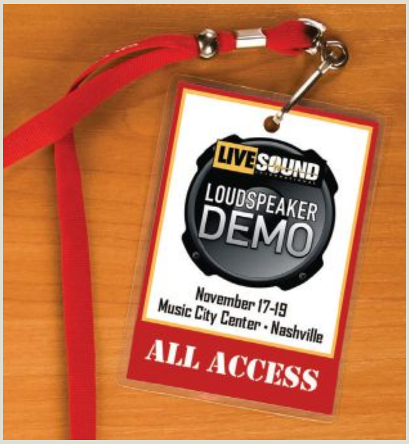
Faders: 32 + 2
Mix Inputs: 72 + 8 stereo
Aux/Group: 24
Matrix: 8
FX: 8 + 8 Premium
DCA: 16
GEQ: 16
App: StageMix (iPad)
Screens: 10-inch touch screens
Local I/O: 8+8 analog XLR, 1 AES/EBU output
Stage Boxes: Rio3224-D, Rio1608-D, Ri8-D, Ro8-D
MY Cards: Lake, Dante, MADI, Aviom, many others
Also: Gain Compensation, Premium Rack processors
Physical: 42 x 26 x 12 inches, 79 pounds
Additional Models: CL3, CL1

Mark Your Calendars: LSI Loudspeaker Demo Coming To Nashville

The next Live Sound International Loudspeaker Demo is slated for November 17-19 in Nashville at the Music City Center. All pro audio and church sound personnel are invited to attend this free event.

The demo, which has been presented to thousands of attendees at previous WFX (Worship Facilities) Expos in Dallas and Atlanta, continues to grow, and this year is no exception. Dedicated demo sessions focusing on portable loudspeaker systems will be presented for the first time, joining the popular compact system sessions.

In addition, a bonus demo session has been scheduled on the evening of November 17, created for audio professionals who may not be able to attend the daytime sessions. And, the demo will also be offering free educational



classes focusing on system networking and wireless/RF best practices.

The event presents a unique controlled environment offering side-by-side listening opportunities to evaluate top loudspeaker systems, in addition to getting further technical details and pricing information from qualified representatives of each company.

More than a dozen top loudspeaker manufacturers have already signed on to participate this year, with more to be announced soon. The list includes:

- Alcons Audio
- Bose Professional
- Cerwin-Vega
- Danley Sound
- EAW
- Electro-Voice
- L-Acoustics
- Martin Audio
- RCF USA
- Renkus-Heinz
- Tannoy
- Turbosound
- VUE Audiotechnik
- WorxAudio/PreSonus

We'll be providing further updates as soon as they're available, both in upcoming issues of LSI and on ProSoundWeb. Mark your calendars now to make sure you don't miss this one-of-a-kind opportunity.

PEOPLE

Roland Pro AV is adding staff and support in support of the pro audio market and the new M-5000 digital console, which is now shipping. Specifically, current sales engineer **Doug Schouten** has been joined by newly hired sales engineer **Brian Belcher**, pictured here.

Based in Nashville, Belcher is noted for his work as a mix engineer and production manager with a long list of touring credits, including Gary Allan, Joe Nichols, Geoff Moore & the Distance, and the production of GodWhy.com. He's also the broadcast mix engineer for Vanderbilt University sports airing on ESPN.

Netherlands-based **Jasper Ravesteijn** has been named sales manager for Europe for **Adamson Systems**. He is working with **Jochen Sommer**, director of European operations for Adamson, to manage sales and support for new and existing customers in Europe, Russia and Africa.

Prior to joining Adamson, Ravesteijn worked in sales for Audiopro BV, where he was responsible for pro audio sales as well as project system design for customers. He is also the owner of Pro AV Educatie, an educational company that organizes seminars and training programs for audio engineers.

Frank Loyko has been named vice president of North American sales for **EAW**. He began his career at the company, and was instrumental in establishing the EAW brand before moving on to serve as vice president of worldwide sales for LOUD Technologies.

Loyko has also worked in sales management capacities for TC Group Americas and Avid.

American Music & Sound named **Gilbert Perales** as its Southern California sales representative for **Allen & Heath**, where he is providing technical sales support and training for dealers, installers, and end users. He has worked in pro audio for 15 years and studied audio production with an emphasis in live sound at The Art Institute of Seattle.

Perales has held senior positions in technical support, marketing and development with Mackie and several other pro audio/MI companies.

RCF USA presented **Quest Marketing** with the Arturo Vicari Award of Excellence as its sales rep firm of the year. Quest serves the Southeast region of the U.S. The award, named on behalf of RCF

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Group CEO **Arturo Vicari**, is presented annually to the rep firm that meets/exceeds sales goals while representing the company's products with high standards of professionalism.

Munich-based live engineer **Björn Seeländer** has been appointed as live sound product specialist for **Waves Audio**, where he is communicating with other live sound engineers across Europe, demonstrating Waves plug-ins, hardware and workflows. He offers considerable experience in both studio and live sound, in particular as a front of house, monitor and system engineer.

Seeländer's credits include artists such as Mark Knopfler (monitor support/stage), Prince (monitor/Pro Tools live recording), Justin Bieber (front of house), Usher (system/front of house support), Taylor Swift (system/front of house support), and Rod Stewart (system/front of house support), along with many others.

L-Acoustics appointed **Tim McCall** as sales manager for the Latin American market. With five years of experience as sales manager for Europe, Africa and Oceania with L-Acoustics, and 25 years of experience in the pro audio industry to his credit, McCall is now developing the brand in the growing territory.

Fulcrum Acoustic named **Paul Carelli** as its new Northeastern U.S. regional sales manager, where he is handling sales and support for customers in New York, New England, and Northern New Jersey. A veteran sound engineer with more than 40 years in pro audio, he comes to Fulcrum after several years as field systems specialist for the communications division at Electro-Voice/Bosch.

Carelli's career also includes more than a decade with EAW, where he worked as a product specialist and market manager. He began his career as a freelance audio engineer touring North America and Europe.

Lectrosonics announced the addition of **Audio Related Technologies (ART)** to the company's list of authorized factory service centers for the European Union. Located in Chesham, Buckinghamshire in the southeast of England, ART was founded in 2000 by **Simon Griffett**, **Duncan Crimmins** and **Murray Harris** and is an approved service agent for several brands in professional audio and video.

Bright Norway and **Sonalyst** in the UK are the first companies to add the new LEOPARD linear line array system

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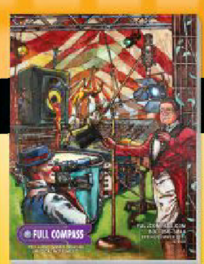

and accompanying 900-LFC low-frequency control element, the latest members of the Meyer Sound LEO family.

Bright Norway indicates that the new rig will help simplify its rental operations. “In a perfect world, audio companies would love to rely on one single model to bring consistency to all its projects, but that’s not possible,” states **Asle Nilsen**, head of audio at Bright Norway. “Now with LEOPARD adding to our arsenal of LYON and LEO systems, we have a much more streamlined inventory to bring highly accurate audio reproduction to every project that comes our way.”

For Sonalyst, a leading production supplier for comedy and theatre shows, LEOPARD’s power-to-size ratio and experience with the LEO products were a significant influence. “Just when we thought that quality in audio production had reached its peak, Meyer Sound released a product representing a giant step forward for the industry,” says **Rory Madden**, owner of Sonalyst. “We were astonished when we first heard LEO and LYON and we couldn’t be happier to become the first company to bring LEOPARD to the UK.”

Sonalyst purchased 48 LEOPARD loudspeakers and 24 900-LFC elements, while Bright Norway’s order includes 96 LEOPARD loudspeakers and 24 900-LFC elements.

Pictured here, left to right, are Scott Gledhill, Helen Meyer and John Meyer of Meyer Sound with Asle Nilsen, head of audio for Bright Norway, at Prolight + Sound 2015 in Frankfurt.

MUSIC Group has acquired **TC Group**, which includes brands such as **Tannoy**, **Lab.gruppen**, and **Lake**. Founder and CEO **Uli Behringer** states, “MUSIC Group stands for relentless focus on innovation, business transformation and overall IP creation. Since the acquisition of **Midas**, **Klark Teknik** and **Turbosound**, we have been continuously pursuing brands that complement the mixing console, processing and loudspeaker excellence offered by these historic brands.

“Throughout our search, TC Group has clearly stood out as the ideal match because of their world-class brands, impressive intellectual property, sterling reputation and first-class team of people. I’m very proud to welcome the TC Group team into our family.” ■

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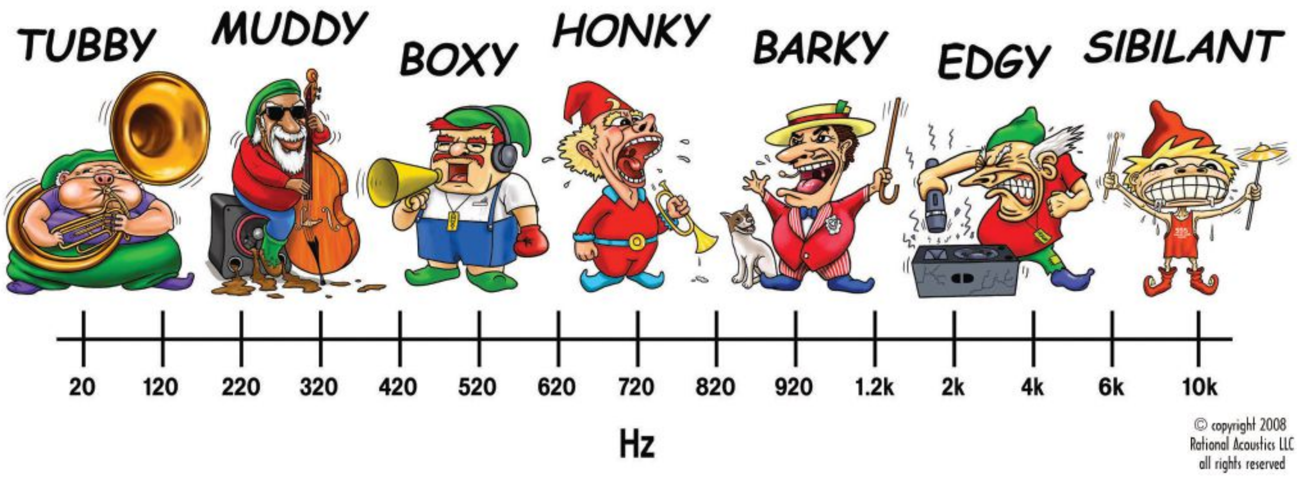
The Seven Bad System Dwarfs

Don't invite any of these guys to your next gig. **By Rational Acoustics**



These seven little gremlins like nothing more than to give your sound system their own distinctive voice. Sure, they're friendly enough when encountered as a balanced group, but hanging out with any one of them for too long can drive you to distraction – or worse. Don't be fooled by their diminutive stature and (sometimes) cute appearance – these little offenders have a long rap sheet filled with everything from simple charges of disturbing the peace to more flagrant offenses like system hijacking and mix vandalism. Let's meet them:

Tubby. Low, slow and rarely on the go, but a formidable ally for getting a house rockin' – when Tubby gets out of control, he has a bad habit of rolling around the bottom of your system and squashing everything.

Muddy. Sure, you can hear him...but WHAT is he saying? And while it is true that he can be warm and embracing when you first meet him, Muddy is also the dwarf that always seems to overstay his welcome.

Boxy. This guy certainly doesn't suffer from claustrophobia. Boxy loves the congestion of enclosed spaces and is often criticized for his lack of openness. Given the chance, Boxy would remove all letters from your system's alphabet besides the vowels "a," "o" and "u."

Honky. A long-time supporter and admirer of Charlie Brown's teacher, this little loudmouth likes to lend even the most well behaved system that lovely "paging horn" sound. Honky's one ambition in life is to show off your system's horns – even when you would rather he didn't.

Barky. Never shy about giving you a piece of his mind, Barky can be aggressive about getting his point across. And mind his bite – you definitely don't want this little guy to "sit" and "stay" in your system.

Edgy. A known associate of lead guitarists and harmonica-wielding frontmen, Edgy has been known to display some serious "anger management issues." Spend too much time with this sharp, irritable little fellow and your ears just may need a valium.

Sibilant. Known to frequent clubs and discos with his pal Tubby, Sibilant is a great lover of all that "sizzles." Unfortunately too much is often never enough for this high-strung dwarf. To be honest, we don't even know if this little guy is from planet Earth. Seriously, just look at him! ■

Thanks to our friends at RATIONAL ACOUSTICS (Smaart, 10EaZy and more) for this little gem. By the way, you can download and/or print out the image from the "About Us" section of www.rationalacoustics.com.

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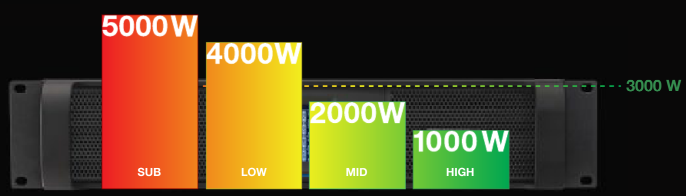

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