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

LIVE SOUND INTERNATIONAL

INSIDE

TOURING WITH ABANDON

No gig too big, no hall too small for Cheap Trick

PRESENTING THE DIRECT BOX (DI) MASTER CLASS

EVOLUTION, FACTS & THEORIES ABOUT SUBWOOFERS

REAL WORLD GEAR: VOCAL MICROPHONES

25th ANNIVERSARY


On
A
12
Y8 Line
B
Y8 Line
Slope 1
Gain
12dB/oct
-6.0dB
C
Y-SUB
Slope 2
24dB/oct
1414Hz
D
Y-SUB
D20


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Total portability: take your mixing console anywhere
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Gen 5
IC²
IC Live Install
ICONYX
IC Live Mobile

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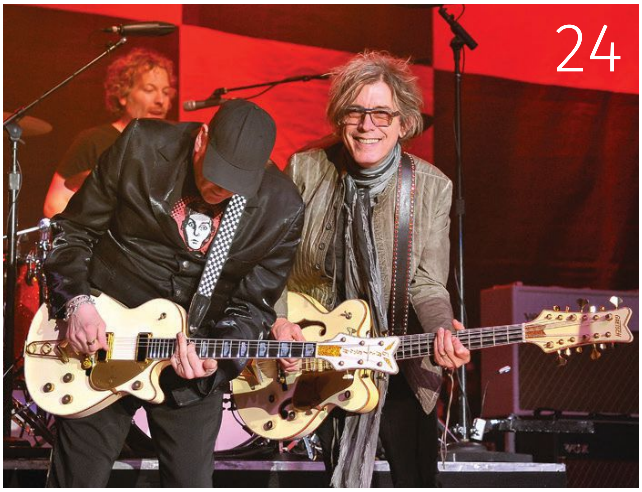
25th ANNIVERSARY
LIVE SOUND INTERNATIONAL



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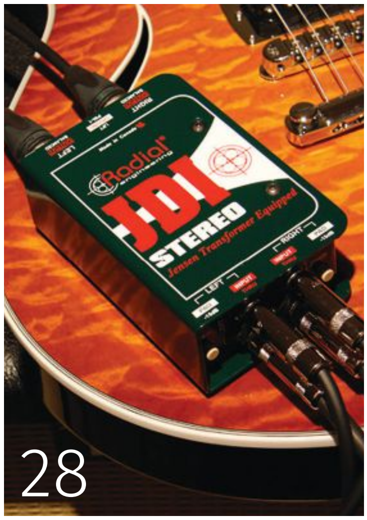

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From the Editor's Desk

THE AUTHOR OF THIS ISSUE'S COVER STORY, Greg DeTogne, had even more fun than usual in detailing the audio approach for Cheap Trick on tour. It's a band many of us of a "certain age" grew up with, part of the soundtrack of formative experiences that are increasingly wrapped in a warm haze as time goes by.

The great part is that these unique musicians remain as busy and committed to their craft as ever in entertaining audiences around the world, old and new alike, backed by an accomplished sound team. In short, the piece is great reading, accompanied by the always-superb photography work of Steve Jennings.

Elsewhere in the issue, Mark Frink steps up with a thorough and highly useful look at direct boxes, those ubiquitous little devices that are often out of sight but rarely out of mind when it comes to wiring the stage. Andy Coules treats us to an interesting discussion about the emergence of the lower end of the frequency spectrum in sound reinforcement, and Craig Leerman provides a wealth of handy tips when it comes to stage monitoring at mid-/smaller-scale festivals and events.

Also be sure to check out Steve Harvey's account of the sheer amount of hard work, coordination and infrastructure it took to pull off simultaneous concert-level audio at three different venues, and Mike Sokol details important findings in working with AC power. Shifting gears, Nicholas Radina offers encouraging advice regarding building a career in pro audio.

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

Keith Clark
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ON THE COVER: Iconic rock band Cheap Trick playing a recent concert date. *(Photo by Steve Jennings)*

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RCF HDL50-A

An active three-way line array equipped with two 12-inch woofers, four symmetrical 6.5-inch mid-range transducers, and two 2-inch compression drivers, all driven by a built-in digital amplifier capable of delivering up to 4,400 watts. A proprietary waveguide is designed to provide accurate coverage of 90 x 10 degrees (h x v) while also furthering linear HF response. The compression drivers have 3-inch voice coils with a crossover point of 800 Hz, so they handle almost all of the vocal range. Symmetrical cabinet design produces identical left and right coverage. Component positioning and DSP crossover filtering enhance constant directivity without spots of break-up or attenuation. A new DSP platform handles all processing, with settings accessed by an encoder on the cabinet rear or via integrated RDNet remote control. Cabinets are made of plywood and polypropylene, and weigh 106 pounds. Integrated rigging supports the building of J-shaped and spiral arrays with 0.5-degree resolution. HDL50-A arrays can be accompanied by the new dual 21-inch-loaded SUB 9007-AS active subwoofer. ***http://rcf-usa.com***

Mackie DC16

A control surface for the DL32R 32-channel digital mixer designed to deliver a scalable system with the benefits of digital and wireless mixing. The DC16 includes 32 remote-controllable Onyx+ mic preamps and 16 outputs paired with built-in DSP. Dante networking provides the communication between the DL32R and DC16, enabling additional networking capability. Visual feedback is provided by full-color channel displays. Up to three iPads can be docked in the SmartBridge, with "smart sensing" that knows when an iPad is in place. ***www.mackie.com***

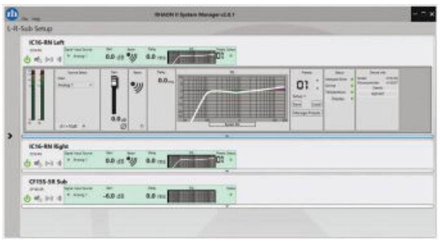
Shure KSM8 Dualdyne

A dual-diaphragm dynamic handheld microphone designed for live sound performances where vocal clarity and sound quality are critical. The patented cartridge design has two ultra-thin diaphragms (one active and one passive) and an inverted airflow system. A high degree of control of proximity effect increases working distance without on-axis coloration. More accurate frequency response due to the controlled proximity effect allows for more neutral mid- and high-frequency reproduction. Off-axis attenuation is flat and broad, boosting rejection of unwanted sound sources for increased vocal clarity. An advanced Pneumatic Shock Mount is designed to tightly reject handling noise without loss of LF response. The hardened carbon-steel grille design is lined with hydrophobic woven fabric that provides enhanced plosive and wind protection while also enhancing waterproof protection. The aluminum handle is offered in brushed nickel or black finishes. The KSM8 is also available in select Shure wireless systems and is compatible with all interchangeable Shure handheld wireless transmitters. ***www.shure.com***

Audio-Technica Artist Elite AE2300

A dynamic cardioid instrument microphone incorporating the company's proprietary double-dome diaphragm design to enhance high-frequency and transient response. With rugged metal construction and low-profile design, combined with the ability to handle high SPLs, the AE2300 is designed for guitar amps, brass and woodwinds, drums and percussion instruments. The double-dome diaphragm maintains directionality across the entire frequency range, with little off-axis coloration (frequency response is nearly identical at 0, 90 and 180 degrees). A switchable low-pass filter is also included. Accessories include an isolation clamp for 5/8"-27 threaded stands, 5/8"-27 to 3/8"-16 threaded adapter, and soft protective pouch. ***www.audio-technica.com***

Renkus-Heinz RHAON II Version 2.0.1



An update of RHAON II (Renkus-Heinz Audio Operations Network) that expands control to all current R-H products, including Iconyx Gen5 systems, IC Live, IC2, VARIAi modular point source arrays, and CF Series loudspeakers. In addition, the Iconyx Gen5 Series now incorporates Dante networking, including for Iconyx and IC Live (ICL-F-RN). Dual RJ45 connections are provided for fully redundant operation, and a single cable carries both Dante and RHAON II control data for streamlined connectivity. Dante-enabled models will be designated with a suffix of RD (e.g., IC16-RD). ***www.renkus-heinz.com***

QSC E Series

Four passive loudspeakers incorporating proprietary DMT (Directivity Matched Transition), designed to deliver uniform frequency response across the entire coverage area. Performance is further enhanced when used with the company's PLD and GXD amplifiers and TouchMix digital mixers offering advanced DSP settings. The E10 has a 10-inch woofer housed in an enclosure that can also serve as a stage monitor. The E12 and E15 are both trapezoidal and use a 12-inch or a 15-inch woofer, respectively. All three have a dual-angle 35 mm socket for pole mounting, and this can be angled down by 10 degrees. Other mounting options include yoke and M8 eyebolts. Input connections are a choice of dual NL4 or screw terminal receptacles. The E18SW single 18-inch subwoofer has a threaded M20 pole socket for use with any of the two-way loudspeakers. All four models are constructed of plywood and coated in black, textured paint. ***http://qsc.com***

Klark Teknik KT-USB & KT-AES50

Two network modules that are compatible with the DN9650 and DN9652 network bridges as well as the Midas NEUTRON-NB expansion module. The KT-USB is class-compliant USB 2.0, requiring a single USB cable to connect to either a Mac or PC running any industry standard DAW software. Forty-eight (48) bidirectional channels are available at 48 kHz, and 24 bidirectional channels are available 96 kHz. The KT-AES50 provides a dual-port AES50 digital audio interface that operates from the incoming AES50 clock or an external clock. It provides up to 48 bidirectional channels and operates at 96 kHz and 48 kHz sample rates. ***www.music-group.com/brand/klarkteknik/home***

Radial Engineering DiNET DAN-TX & DAN-RX

The DAN-TX stereo direct box is equipped with 1/4-inch, RCA, and stereo 3.5 mm input jacks, allowing users to connect instruments or line level sources directly to networked audio systems using the Dante protocol. The DAN-TX has 24-bit/96 kHz analog-to-digital conversion, and a local 3.5mm headphone output provides the means to quickly test audio. The DAN-RX is a 24-bit/96 kHz digital-to-analog endpoint that allows users to output audio from a Dante network to stereo systems. It is equipped with left and right balanced XLR outputs with level control that allows for connection to microphone inputs and up to +4 dBu line level systems. A local 3.5 mm headphone output is provided to test or monitor audio before connection to a PA system. ***www.radialeng.com***

d&b audiotechnik DS10

A network bridge designed specifically for d&b amplifiers that provides 16 AES3 output channels via the Dante protocol. Using the DS10, multiple channels can be sent from the console to the amplifiers via a single network cable. At the amplifiers, the DS10 distributes the audio signals to the AES3 inputs within the amplifiers. It sends meta data information, such as Dante channel labels and cabling information, via the AES3 channel stream for simplified routing and troubleshooting. In addition, four AES3 input channels enable use as a Dante break-in from a console. The DS10 also offers a "bypass" mode for operation as an AES3 splitter, directly distributing the four AES3 inputs to all AES3 outputs. An integrated 5-port switch provides options for a primary and redundant network as well as further connectivity for external devices such as a laptop to access the d&b Remote network and control amplifiers using the d&b R1 remote control software. The DS10 can transport both audio and remote control data via the same cable. ***www.dbaudio.com***

Celestion CF1840H

An 18-inch multi-purpose LF driver for use in reflex, horn-loaded, and band-pass subwoofers as well as serving as the bass unit in large multi-way systems. It combines a 4-inch/100 mm multi-layer voice coil with a ferrite magnet structure and cast aluminum chassis. Available with 4-ohm impedance, the CF1840H is rated to handle 1,000 watts rms (AES standard), with 98 dB sensitivity. Proprietary BAV (Balanced Airflow Venting) provides enhanced magnet assembly cooling, improving performance by minimizing thermal compression. CF Series transducers are designed using specialist FEA (Finite Element Analysis) modeling techniques. ***http://celestion.com***

Yamaha Tio1608-D & NY64-D

Both for use with TF Series digital mixers, the Tio1608-D is a Dante networking equipped I/O rack while the NY64-D is a Dante I/O expansion card. Offering a QuickConfig mode that auto patches inputs and outputs without requiring a computer, engineers can network the Tio1608-D from the stage to a TF mixer for low-latency audio transfer with 16 mic/line inputs and eight line outputs. Up to three units can be daisy-chained to offer 48 inputs and 24 outputs. The NY64-D increases the capability of TF mixers to route audio over a Dante network, facilitating the transmission and reception of up to 64 input and 64 output channels of uncompressed 48 kHz/24-bit digital audio. In addition, new v2.0 TF Firmware will include feature enhancements requested by users plus additional QuickPro presets for microphones and loudspeakers. ***www.yamaha.com***

DiGiCo Stealth Core2

Stealth Core2 is an upgrade to the company's existing Stealth Digital Processing, delivering more processing power and providing a new screen graphical interface. Upgraded FPGA power moves the company's complete range of consoles

to new levels of processing. Specifics include full dynamic EQ on every channel and bus; an increase in the SD9 channel count from 48 to 96 channels of full processing at 96 kHz; an increase in the SD7's total number of processing strips to 600, all at 96 kHz, along with several additional features; and moving the SD5EX to a higher level, matching that of the current SD7 in terms of connectivity and audio processing. The upgrade will be available as an option to all existing users. ***www.digico.biz***

Waves Audio eMotion LV1

Based on SoundGrid technology, a software mixer that provides real-time audio mixing for front of house, monitor, and broadcast applications. Each of the mixer's channels has its own plugin rack capable of running up to eight Waves and third-party plugins. All plugin presets and chains saved in eMotion LV1 can be shared with the Waves MultiRack and StudioRack plugin hosts. The mixer's channel strip (standard EQ, filters and dynamics processing) is provided by Waves eMo plugins: eMo D5 Dynamics, eMo F2 Filter and eMo Q4 Equalizer. eMotion LV1 comes in three configurations: 64, 32 or 16 stereo/mono input channels. The mixer can be controlled by hardware control surfaces and multi-touch devices ranging from four touchscreens to a single laptop or tablet. ***www.waves.com***

PreSonus ULT-series

The ULT12 (12-inch) and ULT15 (15-inch) are active two-way loudspeakers that combine very wide horizontal dispersion (110 degrees) with focused vertical dispersion (50 degrees) for longer throw. Both have a proprietary rotatable Pivot X110 horn, a cabinet angle for stage monitoring applications, and are biamped, driven by a 1,300-watt (peak) Class D amplifier. Cabinets are equipped with two combo XLR and 1/4-inch TRS inputs: a mic/line input with PreSonus XMAX mic preamp and a line-level-only input. The inputs have independent level control, allowing up to two audio sources to be mixed internally and summed to a balanced XLR output for "daisy-chaining" multiple units. In addition, there's a separate direct output for the line-input channel. The series also includes the ULT18, an 18-inch (4-inch voice coil) direct radiating subwoofer housed in a ported enclosure and driven by 2,000-watt (peak) Class D amplifier. A variable low-pass filter is provided to set the crossover transition between the sub and full-range loudspeakers. All models have Baltic birch enclosures with handles, locking IEC power connector, and a defeatable front-panel power-indicator LED. ***www.presonus.com***

Grund Audio LC Series Subs

Three subwoofers to accompany the company's LC Series column array loudspeakers, all equipped with a 2.1 amplifier (class D) designed to drive both the sub and stereo mains. Model LC-2SA has a 12-inch woofer and a 1,200-watt amplifier, model LC-5SA incorporates a 15-inch woofer and a 1,600-watt amplifier, and model LC-8SA offers an 18-inch woofer and a 1,800-watt amplifier. All three also offer onboard DSP capabilities that provide programmable loudspeaker management settings to optimize the column array/subwoofer combination for a variety of environments. Cabinets are constructed of 13-ply Baltic birch. ***www.grundaudio.com***


HDL50-A
BIG BOX. BIG SOUND.
Powerful 3-way line array
2 x12" + 4 x 6.5" + 2 x 2"
4400 watts
140 dB max SPL
32-bit DSP
RDNet on board
www.rcf-usa.com
sound culture
RCF

CAD Audio CADLive 4000 & 3000

Two wireless systems with frequency agility. The 4000 digital system operates in the 902 and 928 MHz bands, while the 3000 system operates in the 500 and 600 MHz bands. Both include proprietary CAD Scan-Link technology with infrared transmitter setup to scan, select, and link to the optimum channel within the specific RF environment. True Diversity operation minimizes multipath interference, joined by proprietary CADLock Automatic Tone Encoded Squelch to eliminate unauthorized transmissions in the signal path. Receivers and transmitters have high-definition LCD displays and RF, AF, battery life, mic sensitivity and RF power metering. Handheld transmitters have a CADLive D90 dynamic supercardioid vocal mic element, while body pack transmitters are supplied with Equitek E19 ear-worn and E29 lavalier mics. Both transmitters also include SoftTouch multi-function on/off/mute switches and selectable power (10, 30, and 50 mW). ***http://cadaudio.com***

TELEFUNKEN DC7 & DC6

Two drum microphone packages, both with three unique voices of the Dynamic Series microphone line in combination with two matched small-diaphragm FET condensers. The DC6 includes six mics, while the DC7 offers seven. Models include the M82 for kick drum, M80-SHB for snare, M81-SH for toms, and M60 FET cardioid for cymbals and/or overheads. The M60 FET is the first non-vacuum tube, FET-based solid-state condenser model produced by the company. Both packs ship in heavy-duty hard shell HC87 flight cases and include multiple drum mounts (M782, M784 and M785), as well as SGMC-5R five-meter XLR cables with right-angle female connectors for each mic in the set. ***www.t-funk.com***

Electro-Voice ND Series

Four vocal and four instrument microphones with a new large-diaphragm capsule design. All models have dent-resistant Memraflex grilles, humbucking coils to guard against EMF noise, and shock-mounted capsules to minimize handling noise. Vocal mics: the ND76 and ND76S (with on/off switch), designed for excellent all-around vocal performance with a cardioid polar pattern and pronounced upper-mid-range presence; the ND86, designed for large concert and festival-sized venues with a controlled supercardioid polar pattern and reduced sensitivity to specific off-axis frequencies (particularly stage noise); and the ND96, designed for loud stages with supercardioid pattern, tailored frequency curve and squared-off grille that allows singers to get as close to the capsule as possible. Instrument choices: the ND44 clip-on mic for tom and snare drums; the larger ND46 for drums and general instrument miking, with unique locking pivot mechanism; the ND68 for kick drum; and the ND66, a condenser design with specialized filters and pads optimized for performance with drum overheads, hi-hat, close-miked drums, acoustic guitar, piano, and other applications. The ND66 also has a locking pivot mechanism. ***www.electrovoice.com***

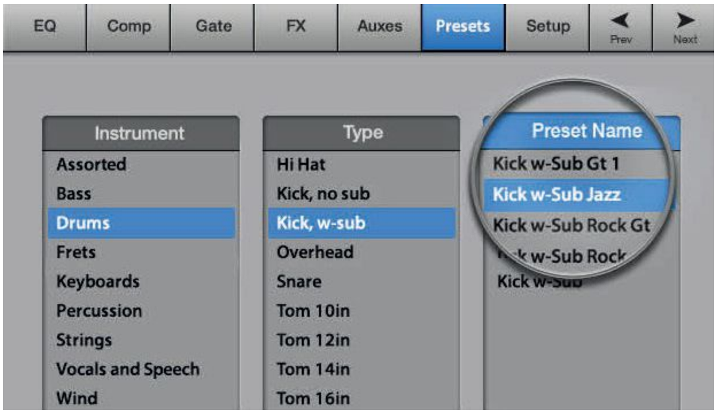
AKG MicroLite Series

Miniature wearable reference microphones, including lavalier, ear-hook and head-worn options. The LC81 MD is a cardioid lavalier with a diameter of 4.8 mm, length of 10 mm, and weight of 2 grams. It's available in four colors. The LC82 MD omnidirectional lavalier has a diameter of 3 mm and a length of 6.5 mm. It's also available in four colors. The EC81 MD (cardioid) and EC82 MD (omnidirectional) ear-hook mics are engineered for accurate, specific placement. Each has a flexible ear-hook and adjustable boom length, with the EC82 MD offering a moisture-resistant design. Both are available in two colors. The HC81 MD (cardioid) and HC82 MC (omni) headworn mics are also equipped with flexible ear-hooks and an adjustable boom, and are available in two colors. The HC82 MC offers a moisture-resistant design. All have a cable with a Microdot connector that is compatible with a variety of different connections as well as with all major wireless systems. A variety of accessories are also available, including wire-mesh protection caps, foam windscreens, phantom power adapters, clips, and perspiration and makeup protectors. ***www.akg.com***

Hosa Technology PWN-200 & PCN-200

Two series of power cords for gear outfitted with Neutrik powerCON connectors. PWN-200 cords are designed to carry power from a normal wall outlet to a powerCON-equipped device. For distribution boxes equipped with powerCON outputs, PCN-200 cords offer powerCON connectors at both ends and are also well-suited for daisy-chaining multiple devices. Both series utilize 12 AWG, Oxygen-Free Copper (OFC) conductors. PCN-200 cords have a maximum operating voltage of 250 volts while PWN-200 cords are rated up to 125 volts. ***http://hosatech.com***

POINT-AND-SHOOT **MIXING**



With TouchMix you don't need to be an experienced sound engineer to sound like one.

In the hands of an experienced photographer, a modern DSLR camera can create stunning images in any number of shooting conditions. But see the presets for portraits, landscapes, close-ups and more? The camera is pre-programmed by experienced professionals so that anyone can get a quality image in just about any condition. Just point and shoot. The camera knows what it needs to do to give you a great shot.

Same with TouchMix. It's one of the most advanced digital mixers ever made, with features that rival consoles costing thousands more. But its real genius is how the Presets, Wizards and Simple Modes put all that power to work easily and seamlessly to deliver you an amazing mix that will have everyone convinced that you are a professional sound engineer. How? Just like the DSLR, our own team of pony-tailed professionals* put everything they learned over decades of mixing live sound into TouchMix so that whether you're a pro or not, you'll get great results quickly, easily, and on your very first gig. No other mixer can make this claim and that's why we say that TouchMix is Simply Genius.

TouchMix™ Series

Compact Digital Mixers

TouchMix™ 8 I TouchMix™ 16

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*Our research indicates that professional sound engineers have, per capita, more ponytails than any other profession. We're still investigating the cause of this phenomenon.

QSC™

dbx VENU360D & VENU360B

Versions of the company's loudspeaker management system offering Dante (VENU360D) and BLU link (VENU360B) networking capability. They also incorporate the latest advancements in proprietary AutoEQ and Advanced Feedback Suppression (AFS) algorithms, additional input channels, and Ethernet control via an Android, iOS, Mac, or Windows device. With an RTA mic "listening" to the room, the updated DriveRack VENU360 AutoEQ algorithm sets loudspeaker levels and room EQ automatically in a matter of seconds using sine sweeps. As a result, room adjustments can be made very quickly without subjecting an audience to lengthy broadcasts of pink noise. ***www.dbxpro.com***

JBL Professional VTX M Series

Two stage monitors utilizing the company's Neodymium Differential Drive woofers and D2 Dual Diaphragm high-frequency drivers. Specifically, model M20 has dual 10-inch woofers while model M22 offers dual 12-inch woofers. The HF section of both utilizes the D2430 dual 3-inch compression driver, which is also found in JBL's VTX V25-II line array. The D2 driver is coupled with a newly developed 60- x 60-degree waveguide that has a unique shape specifically tailored to stage monitoring applications. It's based on the proprietary Image Control Waveguide originally developed for the M2 Master Reference studio monitor. The bass-reflex enclosures have a large LF port. Both dual-channel (bi-amp) and single-channel (passive) operating modes can be set via accessible selection switches. The M Series is powered by Crown iTech HD DSP-based amplifiers. In single-channel mode, a single 4x3500HD amplifier can power up to eight monitors (in four independent monitor mixes). ***www.jblpro.com***

DPA Microphones d:vote Mounts

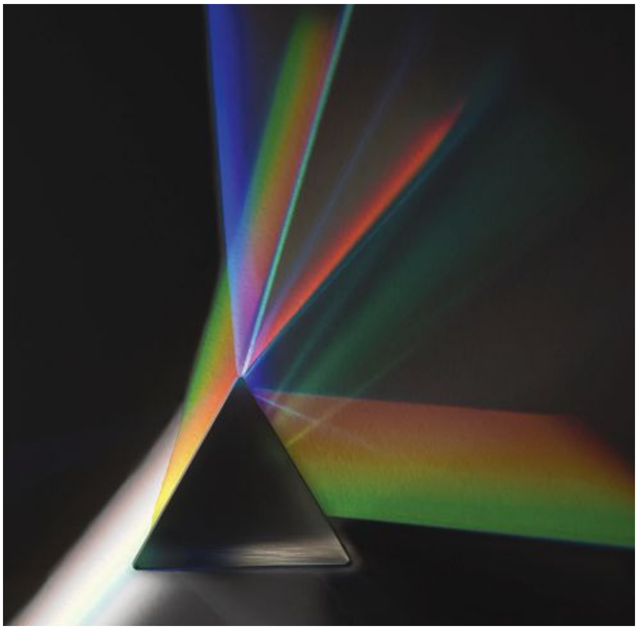
The line of d:vote instrument microphones expands with the addition of three mount options, including a clamp mount, mic stand mount, and accordion clip. The AC4099 clip for accordion attaches directly on the instrument. The musician can choose to either secure the clip permanently to the instrument using screws or temporarily using adhesive. The SM4099 stand mount makes it possible to mount a d:vote mic on a traditional mic stand, while the CM4099 clamp mount is a clip that is designed as an all-round approach for a wide range of instruments. Four different d:vote touring kits are available with quantities of four or 10 microphones packed in a pencil case as well as 10 or 25 mounts, respectively. ***www.dpamicrophones.com***

Neutrik USA XIRIUM PRO

A wireless digital audio system intended as a cable replacement system in providing low-latency, compression-free audio signal to and from devices without cable runs. Central to the system is what the company calls DiWA (Digital Wireless Audio) technology, which offers compression-free, FCC license-free wireless transmission within the 5 GHz band. Transmitted audio is specified as full-bandwidth (20 Hz to 20 kHz), with latency equal to 3 milliseconds regardless of how many XIRIUM PRO devices are used. All signals are sample accurate with no “slippage” in signal timing. The system’s modular format accommodates flexibility of signal type, with both input and output modules available for analog, AES, and Dante. Transmission distance is stated as greater than half a mile (>1 Km, line-of-sight). The system also works with mixed signal types, including analog TX (transmission) to AES RX (receive), Dante TX to analog RX, and so forth. The system can provide RF output of up to two watts when used with a high-gain antenna. Configurations support as many as 10 discrete channels of audio via five transmission units. ***www.neutrikusa.com***

THE JUDGMENT TRIANGLE

You may be right, I may be crazy. *by Jonah Altrove*



As a rule, I don't join in conversations about politics or religion. Conversations become debates, passions get involved, and it usually doesn't end well. There's a sort of primal defense mechanism that engages when strongly-held beliefs are challenged.

Much as I try to avoid these situations, they arise frequently enough in audio. Just visit the online forums and you'll see no shortage of heated debates over sampling rates, clock jitter, dithering versus truncation, audibility of phase shift, array shading…

Before engaging in, uh, "passionate discussion," it's critical to ask oneself a single question: "Am I willing to allow for the possibility that I might be wrong?" If the answer is yes, debate away, but if not, it might be better to move on to something else. There's no point in two people hollering (or typing) at each other if neither party is willing to even consider the other's position.

EXPANDING UNIVERSE

Our field is young, highly technical, and quickly evolving. This isn't stone masonry; less than a century ago our jobs did not even exist. So everything we know about our craft is relatively recent, and our best practices keep "bettering."

Honestly, who expects to see a gigantic ground stack at an arena show these days? I cringe at the thought (mostly on behalf of the folks in the front row) but not too long ago, it was the best method available to do the job. Magnetic tape wasn't chosen for recording because of its vintage sonic characteristics, it was the best thing they had at the time. Today "tape sound" is a creative choice, as it should be, not something forced upon us for lack of a better option.

There's continuous redefinition of what's possible. I remember when Celemony released Melodyne DNA, claiming that it enabled an engineer to somehow reach inside a stereo recording and change single notes played by individual instruments in the mix. Everyone knew that was impossible, and I thought it was a hoax until I actually used the software. I stood corrected.

Similarly, years ago I was researching compact-format array modules for an installation project when I discovered a product that, on paper, seemed to be better than anything in its class while also being significantly cheaper. I was skeptical right up until we powered up the demo rig in our shop. I spec'd the rig for the venue, and years later, visiting engineers are still impressed with it. This forced me to admit that I held a preconceived notion, and that notion was wrong. Again, I stood corrected.

This happens a lot as technology marches on – think of a certain digital desk that turned the 32-channel console market on its head. I judged it, then was proven wrong, and then bought one.

SHIFTING PERCEPTIONS

Humans, on the whole, are pretty suggestible creatures. A friend won his local science fair by showing that people had difficulty identifying Kool-Aid flavors when they either couldn't see the color of the liquid or were served a dyed version. A similar test gave samples of white and red wine to experts for review. The testers reported vastly different flavors, despite the fact that the red was just a food-colored version of the same white wine. Blindfolded, they would have reported identical flavors.

There are a number of psychoacoustic effects that can influence the way our brain perceives audio, on top of (and in addition to) our general suggestible nature. A basic example would be the Equal Loudness Contours (or Fletcher-Munson Curves), which show us how our tonal perceptions can shift with simple level changes. In a more extreme case, I read an article by a gentleman who claimed that he could hear a difference between two bit-for-bit identical copies of the same audio file. (No comment.)

When we need to make a call on something, it pays to approach the situation with a dose of humility, as well as awareness that our preconceived notions can conflict with what's really happening. Rather than form one corner of what I call the "judgment triangle," I try to base my decisions upon my knowledge and experience, and check that against theoreti-

cal predictions (second corner) and empirical evidence (third corner). If all three agree, I'm on the right track, and if there's disparity, I might be missing something.

For example, if the claim is "96 kHz audio sounds better than 48 kHz audio," I'd first think of the experiences I've had that support or contradict that claim. It's also important to be specific here: What do we mean by better?

The question from the theoretical corner would be something like, "What does sampling theory predict would be the result of a doubling sample rate?" This is the part of the process where we live up to the titles of "engineer" and "technician."

The empirical corner might ask, "In a scientifically valid environment, can listeners accurately identify 96 kHz and 48 kHz samples? If so, how consistently? 100 percent of the time? 50 percent?"

I'm not getting into the answer to these questions, because we're talking about the process here, not this specific example. Dr. Floyd E. Toole's "Sound Reproduction" (Focal Press, 2013) does describe a number of these experiments, but more relevant to our discussion here, he also discusses a number of the powerful psychoacoustic effects that can influence our perception. Since none of us is immune to them, it would be prudent to be aware of their existence.

I know I'm not the only one who's made an EQ adjustment until I heard the change I wanted, then looked down and realized I was operating a different channel.

THE WIN-WIN

What's my point? Our brains are powerful instruments but not infallible. I know I'm not the only one who's made an EQ adjustment until I heard the change I wanted, then looked down and realized I was operating a different channel. I would have sworn I'd heard a change!

I know now that in similar situations, there's a chance that I might be mistaken, and that's OK. That's how we learn. *If* I can learn, I'll be better at my job, which is good for everyone.

So I say, go ahead – dazzle me. I'd like to learn something, just don't get too upset with me if I don't believe you right away. LSI

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

IT'S ALL ABOUT THE **BASS**

The evolution, facts, and theories about the low end of the spectrum.

by Andy Coules

OVER THE LAST few years I've been fortunate enough to use some of the best sound systems currently available from many of the leading manufacturers, and one of the things that struck me is how far we've come in terms of delivering the bottom end of the mix. Subwoofer technology and placement techniques have developed to the point where we have a lot more control over how this important band of frequencies is deployed. So I thought it would be good to take a look at how we got to this point and explore how we can deliver a better bottom end.

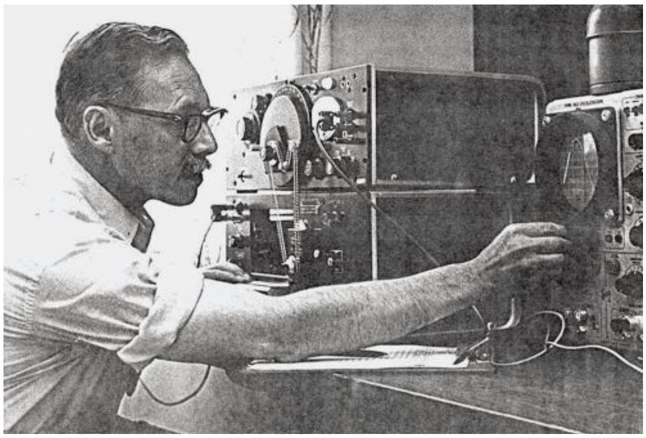
When talking about bass frequencies we're typically referring to those below about 250 Hz, but I want to focus on the lowest of the low; in other words, sub bass – frequencies typically between 20 and 100 Hz. The nature of the way in which we hear means we don't perceive all frequencies equally (this also changes with volume). But generally speaking, humans are much less sensitive to frequencies below 100 Hz, which is why this region requires special attention.

Looking at standard PA loudspeakers or a pair of studio monitors, most of them extend down to a reasonable frequency for reference purposes, but both will always benefit from the addition of subwoofers to enhance the bottom end. Amplifiers designed to power subs tend to be more powerful than those used for mid-range or high frequencies, and the loudspeakers themselves are always larger (usually 15 or 18 inches in diameter) than their higher-frequency counterparts.

A common misconception is that large loudspeakers are needed to produce low-frequency sounds, but reality says otherwise – a simple proof of this is ear buds and headphones. The issue here is the medium through which the sound travels. Air is a relatively inefficient conductor of vibrations, and bass vibrations are relatively slow, so much more power is required to produce bass frequencies that are perceived to be the same level as the corresponding higher frequencies. For instance, if we all lived underwater, a relatively low-powered 2-inch woofer would be quite capable of producing pleasing levels down to about 20 Hz.

EARLY RUMBLES

The first acoustic suspension woofer was invented by Edgar Villchur in 1954: the AR1 debuted at the New York Audio Fair and quickly went into production for his newly founded company, Acoustic Research. The design utilized the elastic cushion of air within a sealed enclosure to ensure a linear restoring force for the woofer's diaphragm, thus producing louder and cleaner bass frequencies.



CREDIT: AES.ORG

Edgar Villchur, developer of the first acoustic suspension woofer, in the lab circa 1965.

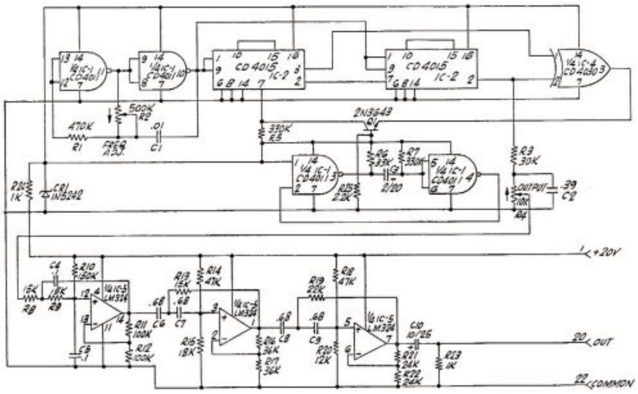


Diagram of the Sensurround pseudorandom generator circuit to create low-frequency rumble.

Loudspeakers capable of producing frequencies below 100 Hz with adequate volume and minimal distortion become more common in the 1960s and 70s but weren't commonly applied in the way that they are now (i.e., at concerts, in recording studios and for home hi-fi). The main use was in cinema to enhance the movie-going experience.

One such approach was called Sensurround, a process developed by Cerwin-Vega in conjunction with Universal Studios that used multiple subs powered by racks of 500-watt amplifiers triggered via control tones printed on the audio track of the film to produce energy between 17 and 120 Hz. The most famous application of Sensurround was to add "realism" to the 1974 film *Earthquake*. The low-frequency entertainment method was credited with making the film a box office success and generated a lot of publicity for subwoofers. (And the film won an Academy Award in the Best Sound category.)

One of the main reasons subs weren't commonly used in the home was due to the limitations of the primary playback medium of that time (vinyl records). Loud and deep bass was difficult because it affected the stylus's ability to track the groove; even a moderate amount of bass (by modern standards) would cause the needle to oscillate excessively until it jumped out of the groove. Truly accurate and deep bass wasn't really possible until the compact cassette tape became a common medium, followed by the compact disc (CD).

MEETING EXPECTATIONS

In the early days of concert sound, there wasn't really an overriding need for a full and deep bottom end. To illustrate, let's take a look at the two instruments most commonly found in the subs in traditional guitar-based rock music: kick drum and bass guitar.

If you stand next to an unamplified drum kit while it's being played, you might notice that the kick drum isn't actually that bass heavy. It moves a certain amount of air, and has the most bass of all the drums in the kit, but it doesn't have that sound that "hits you in chest" like it does when standing in front of the PA at a gig. That sound is actually an artificial construction whereby the drum is close-miked with microphones that are tuned to bring out the bass frequencies – often in a disproportionate way.

The standard electric bass guitar has four strings that are tuned an octave down from the bottom four strings of the guitar. These oscillate at about 41, 55, 73 and 98 Hz, with the specific sound produced depending largely on the bass amp. (Very few bass amps are able to generate 41 Hz; in fact, most struggle to go below 80 Hz with sufficient level.) Again we use techniques such as direct injection (DI) or close miking the bass amp cabinet to help bring out those lower frequencies.

The point I'm making is that routinely, we artificially enhance the bottom end of key instruments to create the sound we've come to desire and expect. This sound is the product of an evolutionary process pushed forward by a combination of production techniques, advances in amplifier and loudspeaker technology, and the expectations of the musicians and their audiences.

But why do we prefer a full and deep bass sound in our music? According to recent research by Laurel Trainor at the McMaster Institute for Music and The Mind in Ontario, Canada, our ears are much better at discerning subtle timing differences in low frequencies than high frequencies. The study suggests that this effect arises within the physiological mechanism of the ear and not in the perceptual center of the brain. This tells us that we rely much more on the low-frequency content of music to help us lock into the rhythm, and further, it shows how important it is for live sound engineers to get the bottom end right, particularly for dance music.

A Diatone loudspeaker from Mitsubishi; the first ones were built in the spring of 1945 for the Japanese radio station NHK.

Iconic Electro-Voice MTL-4 subs (each with four 18-inch woofers), a staple of the touring circuit in the early to mid 1990s, flying with their mid/high companions.

There's one more important factor that contributes to our enjoyment of low frequencies: they're felt as much as heard. The sheer amount of air movement at low frequencies resonates in our chests and adds a visceral element to our enjoyment of the sound. Much like that early Sensurround system, it helps put us "in" the sound and adds to our enjoyment of the music.

VARIOUS APPROACHES

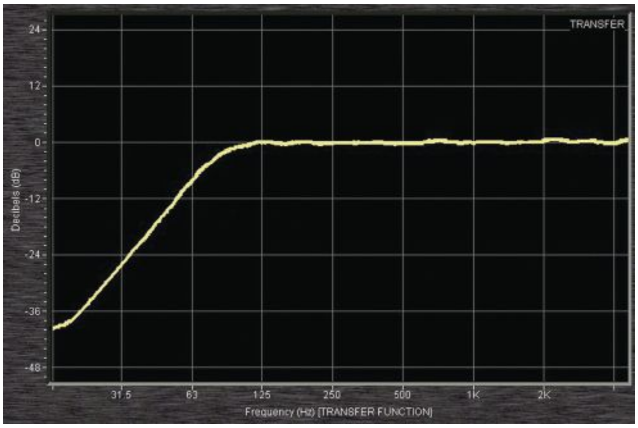
So how do we get the best out of modern sub bass systems? The first thing to consider is that more is not necessarily better – just because we now have prodigious power at our fingertips does not mean that we need to wield all of it. There's a fine line between levels that induce stomach jiggling excitement and those that induce nausea. Bass coverage is rarely completely even throughout a venue, so it's always a good idea to be familiar with the region where it's at it's maximum and then to set levels accordingly, also keeping in mind that the levels at front of house don't always represent the whole.

One of the biggest causes of messy, muddy mixes is bass energy getting into the subs that shouldn't be there. This can be avoided in a number of ways. First and foremost is microphone choice and positioning – putting the right mic in the right position is the best way to ensure getting just the desired sound from the source. Also be aware of how directional mics exhibit the proximity effect; knowing when to exploit this and when to obviate it can help a lot in delivering a crisp and clean bottom end from multiple miked sources.

I'm also a big fan of the artful use of high-pass filtering, which serves two very important purposes. It can help reduce the amount of low-frequency leakage, which is inevitable when multiple mics are in close proximity to each other, and it can also help reduce frequency masking, which always occurs in a downwards direction, frequency-wise (i.e., guitars are more likely to mask the bass than the other way round). And if that doesn't help reduce the amount of unwanted bass in the subs, then the aux fed sub method provides further control to ensure only the desired signals are present in the subs.

Sometimes, particularly in small venues, the issue is not too much bass but rather not enough of it. This is where a little knowledge of psychoacoustics can help. There's an interesting phenomenon called the Missing Fundamental where we hear the overtones of a sound but not the fundamental frequency (usually because the sound system doesn't go that low). Under the right conditions, our brains process the information present in the overtones and fill the gap left by the fundamental frequency so that we actually perceive it even though it's not present.

I've often exploited this principal when the PA has a less-than-stellar bottom end. The key is not just to boost the upper reaches of the bass sound but also to reduce the low frequency content, which has the added bonus of stopping the PA from wasting energy trying to produce frequencies it can't.



CREDIT: BINKSTERNET

A 75 Hz high-pass filter from an input channel of a Mackie 1402 mixer as measured in transfer function mode on Smaart software. The filter has a slope of 18 dB per octave.

Hopefully this discussion provides some new insight into a familiar but sometimes overlooked band of frequencies. After all, if you're planning to build a sturdy mix, it always helps to start with a solid foundation. LSI

Andy Coules *(andycoules.co.uk) is a sound engineer and audio educator who has toured the world with a diverse array of acts in a wide range of genres.*

Cover Story

TOURING WITH ABANDON

After 5,000 shows, huge vocals, thundering bass, pounding drums, and acres of guitar are still the order of the day for Cheap Trick. "We're too dumb to quit," guitarist Rick Nielsen was overheard saying not long after learning that the foursome will be inducted into the Rock and Roll Hall of Fame later this year.

THERE'S AN OLD ADAGE still floating about that claims if you remember the '70s, you probably weren't there. I was, and while I do have to admit that there are many memories that I just didn't make during that decade, one that endures is of seeing Cheap Trick at Sammy G's Circus, a bar in Kenosha, WI.

This was well before the seminal live album *Cheap Trick at Budokan*, light years prior to the announcement last December that they will soon be inducted into the Rock and Roll Hall of Fame. The band was just starting out, these guys from Rockford, IL, and I can still see them performing on a cramped little stage one night in '74, cutting through the bad vapors and smoky miasma that poisoned the place with potent original material as well as hardcore covers they made their own. My girlfriend threw-up and passed-out on the bar shortly after midnight. Her loss. Cheap Trick continued to rock with abandon till closing.

On another night they were playing, someone stole the cash box containing the take from the door and ran off on foot down Sheridan Road, which stretches parallel to Lake Michigan out front. "I remember that," guitarist Rick Nielsen told me once backstage at the House of Blues Anaheim while we were reminiscing. "The owner lit out after the guy with a .45-caliber semiautomatic pistol in hand. Came back with the cash box in short order too…"

And that's sort of what happened to Cheap Trick next, they took off with the

No gig too big, no hall too small for Cheap Trick on the road.

by Gregory A. DeTogne, photos by Steve Jennings

cash box, only got away and just kept running. With their popularity reaching rabid, Beatlemania proportions in Japan towards the end of the decade, success followed in the U.S. along with critical acclaim. Today they're credited with inspiring countless others, ranging from Slash and Pearl Jam to the Red Hot Chili Peppers.

GOING AND GOING

The band has toured almost continuously for over four decades, playing more than 5,000 shows. At this point, it's probably easier to list the sound companies they haven't worked with as those they have. Having opened for Aerosmith on the 2012 Global Warming Tour and taken to the road with Peter Frampton last year, the tireless quartet is set to go again this spring after a short break early this year.

"We were with dB Sound out of Chicago for some time," relates Bill Kozy, who has been at front of house for 13 years with the band. "Then with Sound Image a bit, Thunder Audio, and lately we've been using a company out of Detroit called Burst. Our credo is 'no gig too big and no hall too small,' and that's how a list of our annual dates breaks down. Over the years, PAs have gotten more consistent, so we don't run into problems in these diverse environments like we used to. The varied nature of these gigs still presents its fair share of challenges, however."

Back when he started with the band, Kozy was on a Midas XL4 console. When they weren't carrying a full front-of-house package, whoever supplied audio on any given night provided the basic gates, comps, and other electronic staples. Over the course of the past couple years, he's been using an Avid VENUE Profile as it's a board that's always available and can be easily found virtually anywhere. Additionally backed by a deal with Waves, he notes this combo makes it easier to be consistent than when he was analog and is more relevant within the world we live in today.

While there may be a digital board at front of house, Cheap Trick hasn't changed its basic stage formula one bit over the years. It's still a loud, ripping rock band with a real backline. Wedges and guitar cabinets stacked about are the norm, and that's not going to change.

Known for an extensive collection of vintage, custom, and downright unusual guitars and basses, the band utilizes a considerable number of wireless packs that facilitate the regular switching of instruments. The need for tech-talent is correspondingly high, with stage manager Mark Messner also pulling double-duty as frontman Robin Zander's guitar tech and bassist Tom Petersson's bass tech. Larry Melero serves as Rick Nielsen's guitar tech, while Todd Trent is the drum tech orchestrating events for drummer Daxx Nielsen, Rick's son.

KEEPING IT SIMPLE

Twenty-six inputs arrive at the FOH console from the stage. "Cheap Trick has long been referred to as one of the progenitors of power pop," Kozy says. "In my estimation, that's a fairly accurate description of who they are. I'm working with clear vocals, economical arrangements, and prominent guitar riffs. Everyone onstage still sounds amazing, it's straight ahead, and the songs are great. On a base level here, you can't fail if you simply remember to just not screw that up."

His mix starts with big, clean drums, generally snare-up. While that isn't always

"There isn't much I don't like, as long as it's well maintained," FOH engineer Bill Kozy says of gear options these days. Avid Profile is his current console choice.

Monitor engineer Steve Funke at the helm of a Yamaha PM5D. "When it comes to digital," he says, "this is the one for frontman Robin Zander."

the way things work with newer bands, with these songs it tends to be just fine. His operative strategy is to keep things clean and separate, with the vocal on top.

"There are a handful of plugins I go to regularly from my Waves bundle and use judiciously here-and-there," he reveals. "I use a C6 multiband compressor on Robin's vocals to nail things down that might get a little woofy on the low-end, or too harsh in the mid-highs. It just opens things up and insures that he remains sounding as great as he is."

Going down Kozy's list of plug-in favorites, next you'll find an L2 Ultramaximizer and an SSL G-Master Buss Compressor, both of which are on his mains and used very subtly to "just like in the studio, push everything forward as needed all at once without making a mess." For parallel compression on drums, there's a PuigChild, Waves' modeling of a classic Fairchild. MaxxVolume is used on Zander's vocals in a configuration with a pair of thresholds: One for softer passages and another for louder ones that both bring up the sibilant content and make things more personal. An SSL G-Channel Strip with gated compression is on toms.

He doesn't use any automation at all, preferring instead to rely upon his own experience and grey matter to orchestrate mixing events for the 200 songs the band may pull from on any given night. Effects are spare as well, limited to essentially a bare minimum of reverb, slapback delay on numbers like "California Man," and a slight delay just underneath to make things sound bigger.

"The set list will always contain the core songs," he says. "But from there you never can tell what will happen from night to night. There are benefits to my taking a hands-on approach. Sometimes it works to pull the guitars down and let the vocal take over, mostly in the verses, on tunes like "Top of the World" and "I Can't Take It". "Auf Wiedersehen", on the other hand, is a good example of a song where everything's big at the same time. Like my friend, longtime monitor mixer for Motorhead Ian 'Eagle' Dobbie says: 'Everything louder than everything else'. When I'm faced with a strict dB cap, "Auf Wiedersehen" is the song that's the most challenging. These are all elements that make Cheap Trick what it is. No one can change that, and shouldn't."

HAPPENINGS ON STAGE

Kozy's cohort onstage handling monitors is Steve Funke. Working from behind a Yamaha PM5D, he takes a similar straightforward approach, eschewing automation and avoiding the use of effects and compression entirely. Inputs arriving at the PM5D number 29, including a guest guitar channel plus FOH talkback; there are 16 outs for a mix of wedges and side fills, a pair of stereo in-ear mixes for the crew, and one guest in-ear mix.

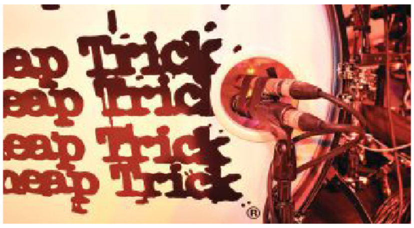
Zander's vocal microphone is a wireless Shure UR4S unit with a BETA 58A capsule, while the rest of the vocal posi-

The 1968 Orange London 50W combo 212 at left was purchased by Nielsen on one of his first trips to the UK. The checkerboard wall of light and sound at right features a pair of 12-inch Celestion Vintage 30s miked with a Shure SM7 (inset).

On the road, guitar tech Larry Melero keeps a close eye on a pair of vaults containing Rick Nielsen's guitars, which include an infamous five-neck Hamer (foreground), and "Uncle Dick", the custom, double-neck Hamer at right built in 1983 as a caricature of Nielsen himself.

tions use BETA 57As. Other input items of note include a drum kit miked at overheads with Shure large-diaphragm Shure KSM32s, an SM81 at hi-hat, SM98s on toms and more BETA 57As on snare.

Down on the kick drum, the standard BETA 52/SM91 combo you'd expect has been usurped by an SM57 and KSM137 that are taped together and mounted in front of the head so that both microphones' diaphragms are aligned. "Our previous monitor guy came up with the idea," Kozy explains, "after Paul Owen of Metallica fame from Thunder Audio loaned us a dual-element Audio-Technica mic one day. We tried it at kick and it worked, so



A homemade dual-element mic made by taping a Shure SM57 and KSM137 together captures the kick drum.

after we gave the mic back we just figured out a way to Frankenstein a similar version using our existing Shure stuff.

"We found that if you get far enough away from the beater head you'd still have body, but you wouldn't get the back-pressure from the front head that could potentially cause problems. Once we found the sweet spot, it worked well for everyone."

The only musician onstage fully on in-ears, Zander is equipped with a Sennheiser EM 2050 beltpack receiver and uses JH5 earbuds from JH Audio. Nielsen prefers to stick with just one JH Audio JH7 in his left ear, fed by a Shure PSM 1000 personal monitoring system. Electro-Voice Xw15s are Funke's favorite choice for wedges, who adds that he can "get just about anything to work" for stage fills.

FUTURE LOOKS BRIGHT

Along with the band, the instruments, guitar cabinets, and amp heads seen and heard on the Cheap Trick stage probably deserve to be inducted into the Hall of Fame at Brooklyn's Barclays Center this April 8 (be sure to check out the photos and captions for more on this gear).

In late January, the band announced the release of its first album in five years, *Bang Zoom Crazy...Hello*. Slated for availability on April Fool's Day, the album contains "No Direction Home," a single that debuted along with the announcement in January. Live dates will return to the band's schedule starting in early March in Florida and continue along the East Coast, through the Midwest, and into Canada before winding back down in late September in West Palm Beach, FL. LSI

***Gregory A. DeTogne** is a writer and editor who has served the pro audio industry for the past 33 years.*

TAKING THE DIRECT ROUTE

A "master class" on DI (direct box) design principles.

by Mark Frink

In audio there are inputs and outputs, sometimes called sources and destinations. Audio input sources break down into two basic types: microphones and direct inputs (DI), which also stands for "direct injection." We commonly refer to the devices that perform this function as a direct box, a DI box, or a DI for short.

The various kinds of microphones utilized in pro audio are acoustic-to-electric transducers that convert sound waves in the air into electrical signals. A DI, on the other hand, converts an existing electrical signal without taking it from the air.

Direct boxes have many advantages over microphones. If they're passive, they require no batteries or phantom power and can, in general, handle high transient levels without distortion. They also provide extremely predictable and consistent sound, as they're a closed circuit and immune from stage and venue changes from one performance to the next. Aside from musicians changing the volume on their instrument, DIs offer repeatable console gain and EQ settings.

The total isolation from ambient sounds make DI inputs impervious to adjacent sound sources, allowing engineers to turn them up without adding the bleed of nearby players that can "pollute" the signal. Unlike mics on stands, a DI's location doesn't affect its sound, so other than its switches, it can't be accidentally misadjusted. Finally, DI inputs often provide improved gain-before-feedback, allowing acoustic instruments to be mixed with loud sources, such as a string quartet with a rock band.

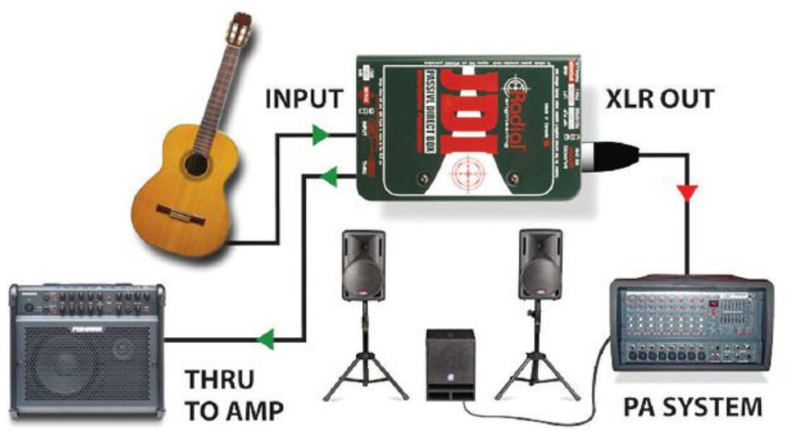
A direct box interfaces unbalanced musical instrument or consumer audio signals to balanced signals that can travel longer distances to be used with pro audio consoles, converting high impedance to low impedance, Transformer-equipped passive DIs have the benefit of providing isolation from the hum and buzz of ground loops.

TWO TYPES

There are two basic types of DI: passive and active. A simple rule of thumb is that if the source is active, use a passive DI, one that employs a transformer. If the source is passive, use an active DI, one that uses phantom power. For example, with an old Fender Precision bass (with magnetic pickups), an active DI will probably work best as its buffer will generate plenty of signal for the console. It also will not load the pickup or affect the sound on stage.

On the other hand, a high-output active bass employing its own 9-volt battery can be used in confidence with a passive DI, knowing that its transformer will easily handle the signal without distortion. That said, most active DIs have input pads to

CREDIT: RADIAL ENGINEERING



A typical DI application.

reduce signal by 15 to 30 dB.

The electrical symbol for transformers uses two squiggly lines representing coil windings separated by straight lines signifying the laminated steel or nickel core between them. Current passing through the primary winding creates a magnetic field in the transformer's core that produces a sympathetic current in the secondary winding, effectively passing a signal from one circuit to another without the two being physically connected.

The strength of the secondary signal relative to the first is determined by the ratio of the number of primary windings to the number of the secondary windings. The world's best transformer manufacturers, such as Jensen, have made models specifically designed for direct box applications for decades. Direct box transformers are typically 12:1.

At its "core" (pun intended), every passive DI is a transformer with an input jack for the instrument, an unbalanced thru connection for the stage amp, and a male XLR for the mixing board's balanced mic input pre-amp. In the golden age of audio, all technicians could solder, since most snakes, stage boxes and many cables were made by hand from scratch, along with the best passive DIs money could buy. Installing and wiring the connectors and transformer into a project box to make a passive DI was a rite of passage.

Honoring the tradition, Jensen recently released the JIK-DB1 Iso-Kit, developed specifically for pro audio educators and available only to their students for $125. Unlike hand-wired project DIs of old, it's full-featured, including a printed circuit board with beefy solder points for novices, plus Jensen's JT-DB-EPC transformer, a rugged ground, pad and polarity switches, connectors, and the full complement of resisters and capacitors.

Jensen makes a single DI transformer, the JT-DB-E (and its variant, the JT-DB-EPC), which is designed to be soldered to a PC board. Since a passive DI is no more than a transformer and its connectors when its switches are off, all direct boxes that use a Jensen transformer measure and sound identical.

Jensen transformers are designed to produce minimum overshoot and ringing, as well as flat group delay for best time alignment of a musical waveform's spectral components. This means that harmonics arrive at the same time as the fundamental, providing clear midrange and highs without the harsh, edgy sound of inexpensive DI transformers.

Jensen JT-DB-EPC transformers.

While there are several other high-end transformer manufacturers, Jensen is the one that audio engineers ask for by name. The JT-DB-E demonstrates excellent performance, providing nearly flat frequency response and minimal phase shift from 20 Hz to 20 kHz.

FURTHER DEVELOPMENTS

The 1980s saw a rise in the number of passive direct boxes due to the widespread addition of synthesizers and other electronics to live sound input lists. Inexpensive DIs employing low-cost transformers began to dominate sound company inventories.

In 1996, however, Radial Engineering introduced the first version of its JDI Jensen direct box, re-establishing a high standard for passive direct boxes that remains to this day. In 2001, the company changed to its innovative bookend design that protects switches and connectors on the unique 14-gauge I-beam inner skeletal frame.

Though Radial's green DIs remain a tech rider standard and the most popular passive DI both on the road and in the studio, direct boxes with JT-DB-E transformers are available from several other manufacturers, including Whirlwind and Switchcraft.

Adding the cost of a Jensen transformer to the case, connectors and switches means that a DI can't be made for less than $100 and must sell for twice that. To serve a growing market of clubs and AV for affordable direct boxes, Radial developed the PRO line of transformer-based accessories, employing Eclipse transformers designed after Jensen principles but manufactured at lower cost. A current favorite is the PRO-AV2, a stereo passive DI with a variety of input options for less than a mono JDI.

Passive direct boxes tend to exhibit one or two degrees of negative phase shift in the highest octave, above 10 kHz, due to the behavior of their transformer, with better transformers exhibiting less VHF phase shift than inexpensive ones. Active DIs, on the other hand, tend to exhibit positive phase shift in their lowest octave, ranging up to 10 degrees, depending on quality and design.

GETTING ACTIVE

While the frequency response of passive DIs is nearly flat for active sources, pas-

The venerable Countryman Type 85 and newer Type 10.

sive sources with weak signals and high impedance can produce weak signals that require a lot of preamp gain and become colored when used with a passive direct box. Buffering a high-impedance passive input with an active DI is like giving that source its own local preamp and line driver before it gets to the mixer.

Piezoelectric transducers require a very high input impedance load to sound correct, and the transformers in passive DIs don't present a high enough impedance load for these sources. The rule of thumb is to match output impedance to input impedance that's up to 10 times higher. Fidelity is maximized when load impedance is significantly larger than the source.

The iconic and ageless black anodized aluminum Countryman Type 85 active direct boxes were designed to be "pin 3 hot" by Carl Countryman prior to AES declaring "pin 2 hot" an international standard. This matters if the Type 85's signal is being combined with a mic on the same instrument or its speaker cabinet, and it also matters if the Type 85 is being used on a stereo source and combined with a second DI of another make.

Carl pointed out to me many years ago that I was welcome to open my Type 85 and reverse its polarity with a soldering iron. However, he added that I should also clearly mark it, since its polarity would be the reverse of the majority of the thousands of other Type 85 DIs manufactured over the course of three decades.

Countryman's newer, silver Type 10 active direct box is designed to the current "pin 2 hot" standard employed by all other DIs and electronics today. Two green LED indicators confirm battery voltage and phantom power with the flip of a momentary switch. In addition to the -30 dB pad "speaker" position on the instrument/speaker switch, the Type 10 adds a second pad and simply labels the pad switch -15/-30 dB.

Most importantly, like the Type 85, its input impedance is 10 megohms without the pad, matching it to high-impedance sources like Piezo pickups; however, with the pads engaged, the input impedance is 10 kilohms to better mate with line- and speaker-level sources.

Radial recently released the PZ-DI, which has 3-way adjustable input impedance. Its ultra-high 10 megohm impedance compensates for the harsh tone that plagues Piezo pickups. A 1 megohm posi-

The Whirlwind Direct-JT doesn't play coy about what's inside.

tion provides the impedance commonly found on active DIs. It also has an adjustable high-pass filter and a 3 kHz low-pass filter for smoothing active instruments.

The PZ-DI's 220 kilohm setting warms up the tone of magnetic pickups to provide the tone of a tube amplifier. Magnetic pickups used in most electric guitars and basses are inductive and highly resistive due to the sheer amount of wire windings. An input impedance that's too low can cause HF roll-off and a dullness or lack of sustain.

HIGH LEVEL

Direct boxes have been used with speaker level sources for many years, though there is a strong aversion to any DIs by many electric players. The Countryman Type 85 has always had a switch with a "speaker" position that's a 30 dB pad, as does the Type 10, (and the Radial JDI), but they can't be used without a speaker cabinet (or other load) to soak up most of a guitar amp's power.

The isolation and consistency of speaker-level DIs allows the warmth of tube amps to be mixed in without adding the stage wash of loud gigs. Passive speaker level "Red Box" DIs have been employed for years by heavy metal bands whose engineers want that tube "crunch," often gaffe-taping them to Marshall guitar heads and combining them with miked cabinets to add dimensionality to 3- or 4-person bands.

The new Radial JDX-48 is an active speaker-level direct box that filters both the amp's output and the back EMF impulse from the speakers. Its emulation is a compromise between a closed-back 4x12 and an open-back 2x12 cabinet. It also provides bass extension and handles 300 watts for SVT applications.

The DI evolves for the digital age with the new Radial DI-NET (Dante-enabled) and BT-PRO (Bluetooth-enabled).

In late breaking news, Radial has also added two direct boxes to its line for the digital age. The new DI-NET is a Dante-enabled DI in a transmit (TX) version, with 1/4-inch, RCA and stereo 3.5 mm input jacks, as well as an RX with two male XLR and local mini-TRS headphone monitoring. Even more exciting is the new BT-PRO Bluetooth DI that allows reception up to 65 feet and also has local mini-TRS headphone monitoring. LSI

***Mark Frink** is a regular contributor to Live Sound International and a long-time touring engineer who's currently working with the Jacksonville Symphony Orchestra and available for a summer tour.*

DR. ADAM J. HILL

Merging diverse experiences in a fascinating, multifaceted career. *by Kevin Young*

Although he's only been in the audio industry for 12 years, Dr. Adam J. Hill has packed in a huge amount of work, splitting his time lecturing at the University of Derby in the UK and working "across the pond" with Chicago-area-based Gand Concert Sound (GCS) as an audio engineer.

At Derby, the 31-year-old teaches on the university's BSc (Hons) Sound Light and Live Event Technology program (considered by many to be the leading program of its ilk in the UK), and also runs the MSc Audio Engineering program, which he created along with colleague Dr. Bruce Wiggins. While his live sound work in the UK currently consists primarily of supervising the shows his students work on, that may change when he obtains dual citizenship in the next few years.

Each gig feeds the other, Hill explains. "Doing live sound is essential for my academic work. We have students who've gigged a lot, and if they get any hint you don't know what you're talking about, they'll tear you apart." Given the pace of technological change in the industry, he adds, "If I wasn't hands-on, my teaching would be almost irrelevant. I also love doing sound. GCS is like a second family."

IN THE FAMILY

Raised in Highland Park, IL in north suburban Chicago, Hill spent a good part of his childhood on stage with his father's band, Dr. Mark and The Sutures. "That's where I started out, but my grandpa taught me to play guitar at an early age. He was a professional jazz player who performed with guys like Les Paul and Wes Montgomery. Guitar was the first thing I noodled around on, but my dad made me learn piano first. Then I taught myself bass, drums and percussion."

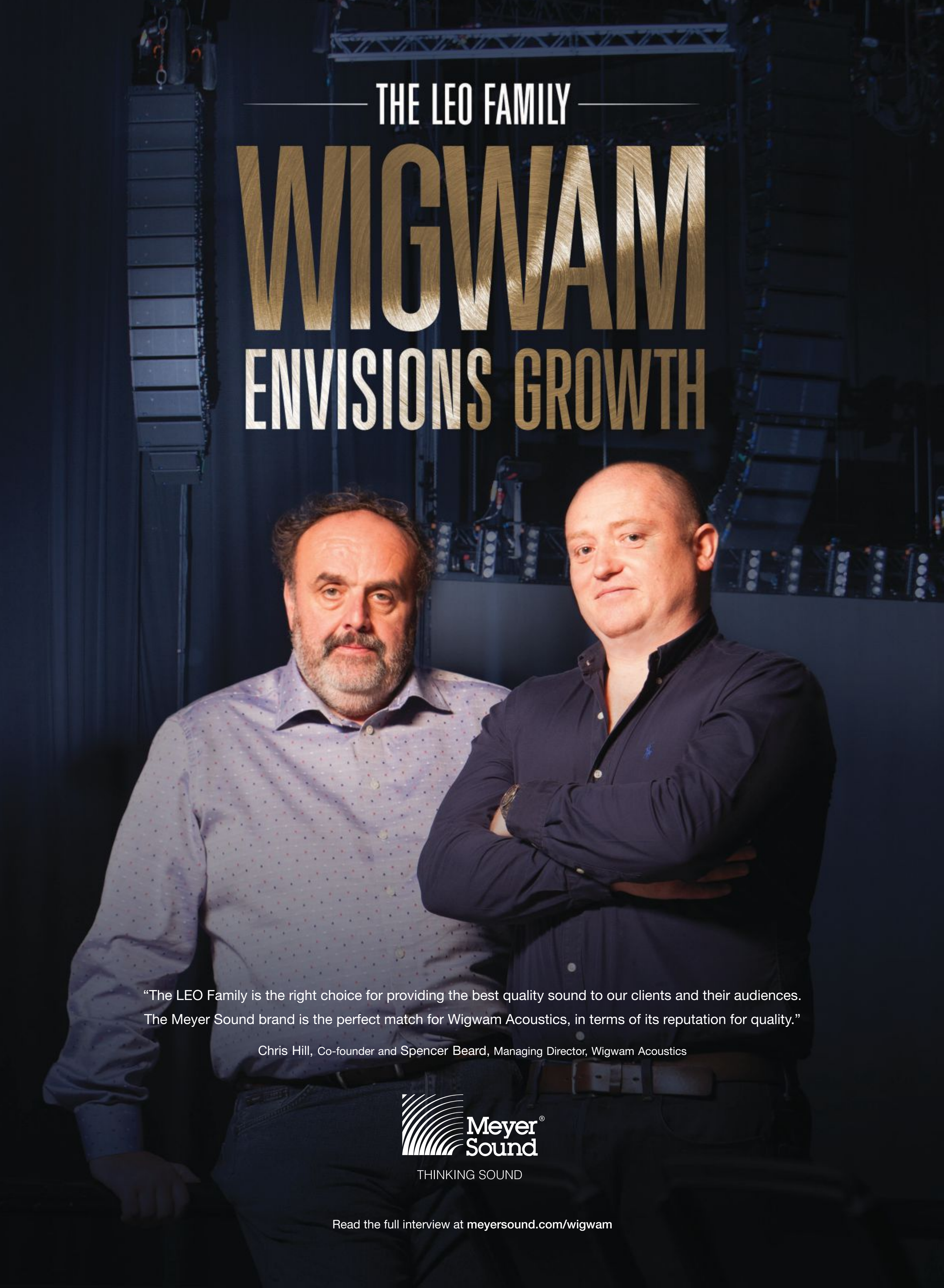
Later, he'd go on to play in various school ensembles and rock bands, but the experiences with his dad and bandmates stand out, including once opening for Lynyrd Skynyrd. Hill was also interested in what the band's audio engineer was doing, combined with an aptitude for math and science. "But as a teenager, I was going to be a rock star," he says, laughing. Several family members and friends who'd already gone the professional music route tried to dissuade that direction, with one bluntly advising him, "If you're good at something else, do that instead."

"The crazy ideas I dream up – even if they are indeed crazy – can be implemented in practice."

Despite those voices of experience, he admits that if it wasn't for his mother giving him a college application ("and partly filling it out herself," he adds), he might not have gone the higher education route. Fortunately, given his love of music and STEM aptitude, she suggested considering the study of audio engineering.

He hadn't. In fact it took some time to settle on a major at Miami University (Oxford, OH), his dad's alma mater and perhaps the last school on earth he actually wanted to attend. "I looked at it to appease my father, but it became clear that was where I needed to go."

Thus began the pursuit of a Bachelor of Science degree in Electrical Engineering, something he still didn't fully engage in until the influence of his academic advisor, Professor Jade Morton. "She helped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me test my limits," Hill states. "Without her I never would have made it to the UK. She was the one who said, 'take the opportunity, go abroad'."

PAYING THE DUES

Amidst the academic pursuits, Hill reached out to GCS founder/co-owner Gary Gand, seeking hands-on work in pro audio (a.k.a., a job) while at home during the summer. "It was like an internship. Gary threw me in the deep end with the sharks and I managed to survive," Hill says. "On one of my first days a whole bunch of gear had just come back from a circus gig and they sent me off to the loading dock to clean a 300-foot analog snake covered in, well, you can imagine. I also repainted lift gates, 'roach-bombed' speaker cabinets, mopped the floors; the typical stuff you're going to do if you're willing to pay your dues."

His knowledge of instruments and backline gained during the band days, however, soon led to working at gigs, where he also shadowed monitor and front of house engineers to watch them ply their trade. That first summer he got a shot at mixing monitors, and the next year he continued with monitors as well as front of house.

"I still prefer monitors," he notes. "A few years back there was a stretch of doing mostly FOH, but with monitors you can have more than 20 different mixes. It's more challenging, and I also enjoy working with the musicians."

The scope expanded over the years to working every live sound role, including system tech and design, with hundreds of diverse artists, including Alice Cooper, The Allman Brothers Band, Buddy Guy, Cheap Trick, Ben Folds, Debby Boone and Dierks Bentley. "That's what we do at GCS," he adds. "I'm not going out on tour for months. Every single day it's a different band or bands at one-offs and festivals."

Teaching fledgling audio professionals at the University of Derby.

A particularly memorable show came along in 2005, where he was working as the monitor system tech for a top funk/pop group. The stage was outfitted with upwards of 30 proprietary monitors that Hill spent most of his time swapping out because the band was convinced they were blown. Only one wedge actually proved defective, but he notes, "It's a gig that stands out for getting that rush of being in the moment and making it happen."

The relationship with GCS continues to this day, with Hill returning to Chicago following the conclusion of each academic year to work summers on the company's busy (and growing) schedule of shows, festivals and events.

MERGING TWO WORLDS

With the encouragement of Professor Morton, he pursued a broad range of study after being awarded his BSE degree from Miami, including a Master's (MSc) in Acoustics and Music Technology from the University of Edinburgh followed by a PhD in Electronic Systems Engineering from the University of Essex. During his time at the latter, Hill credits Professor Malcolm Hawksford – a highly respected and accomplished researcher and developer of analog and digital audio systems – with encouraging him to pursue adventurous ideas in research but to always focus on real world applications.

"That's what being involved in live sound provides," Hill explains. "The crazy ideas I dream up – even if they are indeed crazy – can be implemented in practice. So if I'm coming up with a new technique for, say, subwoofer alignment, the question becomes, 'Is this practical? Would I be able to use this on a gig?' If the answer is no, I do back to the drawing board and find a way to make it realistic.

"I can be engineering with GCS and an idea will pop into my head," he continues. "I'll be struggling with something and think, 'It would be great if I had this,' and the beauty is, I can go back to Derby and work on the solution, often in collaboration with Gary (Gand) and Adam Rosenthal at GCS."

Granted not all solutions call for research, he says in noting that, in his opinion, the most indispensable tool for live audio work is the Shure SM58 microphone. ("Which, in a pinch, I've also used as a hammer," he adds.)

Many of his ideas generate from discussions with undergraduates, postgraduates and colleagues, and the dialogue between he and his students, Hill maintains, is the most important part of his job as an educator. "Students might not have been working in this field for

Receiving the Lecturer of the Year Award from Derby vice chancellor, Professor John Coyne, in June 2015.

years and aren't dug into certain ways of thinking, so they'll come up with really interesting approaches to a problem." It's right in line with a quote from Frank Zappa that he counts as one of his favorites: "Without deviation from the norm, progress is not possible."

APPLYING CONCEPTS

It's also an ethic that has driven his academic work, including the development of modeling programs such as the Virtual Bass Toolbox and FDTD Simulation Toolbox – the latter created while working on his PhD and which has enabled him to do a significant amount of research.

"I specialize in low-frequency sound reproduction and reinforcement, from small rooms to large-scale designs, and needed to simulate low-frequency acoustics accurately in rooms with strange shapes and obstacles," he explains. "The programs that existed went down to about 63 Hz, but I was working at 20 to 200 Hz. The FDTD Simulation Toolbox addresses these needs, although it's in serious need of updating right now."

Additional areas of research include diffuse signal processing (DiSP), low-frequency sound source localization, calibration techniques for cinema subwoofer systems, and chameleon subwoofer arrays (CSA).

Developed as part of his PhD thesis, and covered extensively in an AES paper co-authored with Professor Hawksford, CSA was developed to minimize variations of low-frequency response caused by room-modes in closed spaces. "The idea is that, in its perfect implementation, each subwoofer would have multiple drive units that could be controlled individually, calibrated by measurement at certain points within the room," he expounds, adding that the technology also works with conventional loudspeakers.

He's also applied a combination of CSA and DiSP methods to a ground-stacked subwoofer array. "Only in simulations at this point," he notes, but with an eye to distributing low-frequency energy within the coverage area while dramatically reducing stray energy on stage. "The approach of DiSP is to de-coorelate signal going to each subwoofer by adding a slight amount of phase randomization, not enough to be perceptible but just enough to minimize constructive and destructive interference within the audience. It's an extension of a similar technique that has been used in the industry for quite a while."

KEEPING IT INTERESTING

To this point in his young career, Hill has authored and co-authored at least 25 papers for leading audio societies and organizations, including more than 10 for the AES. While many of them focus on low-frequency concepts and research, others carry titles such as "Live event performer tracking for digital console automation using industry-standard wireless microphone systems" and "The effect of background music in higher education learning environments."

And some of his work strays well beyond those norms. For example, there's a 2014 paper he co-authored entitled "Habitat quality affects sound production and likely distance of detection of coral reefs," prepared for the Marine Ecology Progress Series.

Coming at the conclusion of our conversation, Hill began chuckling before I even finished reading the title. "When I was in Essex, a friend of a friend was doing his PhD in Marine Biology, studying the health of coral reefs," he explains. "He was interested in whether there was an audio cue that could be used to judge a reef's health. I applied one of the algorithms used in the Virtual Bass Toolbox to analyze the recordings my colleague made.

"It's a random application of something I'd already done, but what amazes me is the number of different places this stuff can be applied. You've got to talk to people in different disciplines and collaborate. That's what keeps things interesting." LSI

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

Showcase

DEVELOPING THE GAME PLAN

Stage monitoring approaches for smaller multi-act festivals and events. *by Craig Leerman*

My company has worked countless smaller to mid-sized festivals and variety shows over the years, and through trial and error we've discovered several approaches and problem-solvers that make life a little easier when it comes to working with stage monitoring.

As always, the first step to success is advancing the gig. We never take the promoter's word that the riders we've received are current, so we call each artist's representative to make sure we've got the latest monitoring requirements. The same goes for local performers who may have neither a rider nor a representative. And in both cases if they have stage plots, even better. It never hurts to ask.

Armed with this information, we begin laying out the "monitoring plan of attack." First, how many mixes will be needed, and then how many for wedges and in-ear monitors (IEM)? Once that's determined, pad it by at least a few to account for last-minute changes as well as extra performers or guest artists invited to perform.

For most bands, it's pretty easy to figure out where the wedges go, short of special requests. Still, it's a good idea to specifically determine who needs a wedge and where they'll specifically be positioned, and then how that's going to change for the next act, and so on. Other gear to be added or struck should also be included in this ongoing choreography. Some of the festivals we handle also have acts such as dance troupes and choirs on the bill, and they often need area monitoring like side fills.

Basically it's a game plan, entered on our master show log, that we follow for the fastest, most efficient (and accurate) changeovers. The log should also include all input and output patching, microphone swaps, power drop requirements, etc.

WEDGE WORLD

We're fortunate to have a fairly deep inventory of stage monitors (wedges) in terms of sizes and types. This allows us to provide the best option for each act, joined by any additional drum monitors, side fills, and boxes for what we call the Front Line.

The term refers to an approach at smaller festivals with a limited number of monitor mixes where we place a row of wedges across the front of the stage, all receiving the same mix. This "front line" serves solo artists, acoustic duos and possibly trios, and singers with tracks, and can stay in place, saving us

Above: Stage wedges on the "front line." Opposite page: Give drummers a wedge with more low-end "oomph," and consider a sub as well.

precious time during changeovers. While fewer loudspeakers on stage is always the preference, it serves as a good compromise between dedicated wedges and/or side fills.

Another variation of this deployment is what we call the Front Row. It looks the same but every wedge is on a separate mix. The wedges can be pulled into position for performers then returned to a straight row, depending on what's needed. It provides more control over the wedges, reducing bleed onstage while allowing us to tailor the stage mix for individual performers. In either case, just remember to remove the front wedges if a dance act is on the bill – people want to see their feet moving on stage.

Switching focus to the back of the stage area, drummers always get a dedicated mix and wedge on our shows; they always seem to want vastly different things than the rest of the band. For acoustic acts with low-volume drummers (yes, they exist!), we deploy what we call a Drum Box, a wedge with a 15-inch woofer that can reproduce the kick and floor tom well. For louder rock or reggae drummers we will place a subwoofer to give the bottom end some added boost.

CONFIGURATION CONCERNS

Monitors are mixed on the house console at a good deal of the smaller festivals we serve. With analog consoles, particularly those with only a few pre-fader aux sends, this can limit the number of monitor mixes. While we can get creative and use matrix outputs and post-fader aux sends to get a few more mix channels, a much better alternative is to go digital.

Most compact digital desks are outfitted with multiple assignable outputs that can be configured for pre-fader monitor sends, making it easy to route enough mixes to the stage. Further, digital desks that offer control via a tablet allow the engineer to walk on stage to adjust mixes, both before and after the performers arrive. Something else that we're investigating are the newer digital console/monitoring platforms that provide individual monitor control to each person on stage via their smart phones. Talk about a time saver!

Of course, for larger-scale events, we provide a dedicated monitor board located stage-side, manned by its own engineer. When there will be a lot of playback needed for singers and/or dancers, we position the playback devices at the monitor position so the engineer can trigger the cues. If there's a change in the program, (and there usually is) the performers (or their representative) can communicate the changes directly with the playback operator.

WORKING WITH IEM

While wedges are still most common on stages for smaller. locally-based festivals and event, we are seeing an increase in the use of IEM, which have become quite affordable. One thing we've made standard operating procedure is making sure that mixes for performers on IEM are provided with some ambience from the audience (captured via a mic or two on stage pointed toward the crowd). This lessens the feeling of isolation that can particularly impact less-experienced performers as well as those new to IEM.

In most of these situations, we haven't worked with the artists before, and there's rarely any time for sound check. As a result, if it's a stereo IEM system (most often the case), we center each singer's vocal and instrument in their mix and pan everything else to the sides. With mono mixes, the emphasis is the same but of course it's spatially much more limited.

We're also a safety-first organization, so we try to do everything possible make sure that the performers aren't turning up their systems too loud. Further, a squeal of feedback or loud thud from a dropped mic can also cause hearing damage, so compression and limiting is always deployed. Depending on the event, we'll even insert "brick wall" limiting to control any unexpected spikes.

Quality antennas should be provided for use wireless IEM systems to help eliminate dropouts and other problems, particularly in this ongoing era of RF challenges. Directional "paddle-style" (a.k.a., log-periodic dipole array) antennas can provide up to 6 dB of gain, while Helical antennas can deliver up to 10 dB of gain.

A few more things to keep in mind:

- Label all wedges and cable ends with their respective mix numbers. It's more easily understood when a performer asks for something in "Mix 4" than just "in that box." Also label all IEM belt packs with mix numbers and performer names so they don't accidentally grab the wrong pack.
- Teach new performers basic hand signals so they can communicate their monitor needs to the audio crew during the performance instead of announcing their issues over the PA. Our method is to verify that the performer and monitor engineer make eye contact, then the performer points at what they want to hear, and then point up or down for volume. (You might be surprised by how many performers who've never used monitors before step on to our stages.)
- Adding a touch of reverb to the mix of an inexperienced performer can give them a confidence boost, lending a pleasant signature that's akin to a karaoke machine with effects. (Just be sure to mute the effects when they talk.)
- Make and carry some "Acoustic Aiming Devices" (ADD). I've referred to these in previous articles, but simply, they're small pieces of wood painted black that can be inserted under wedges to give them more tilt, putting the coverage pattern more firmly on the performers. (If they can't hear it clearly, they'll want more volume.)
- Place wedges in the rear or side null of the mics. With cardioid mics, we place the wedges right behind them, while with supercardioid and hypercardioid mics, we place them off to the side at about a 45-degree angle in the mics null zone.
- Spend time ringing out the wedges. Most of the feedback problems on these type of gigs comes from the monitors not getting enough attention before the show starts.
- After the ringing out process, check the monitors with both speech from a mic and with music playback. During testing, turn on the mains and check out how the house system affects the stage sound, especially the output from the main subwoofers. (We often find that side fill subs aren't really needed due to plenty of output – and spill on stage – from the main subs.)
- Embrace parametric equalizers. Not every problematic tone falls right on the center frequencies of a graphic EQ. Parametric EQ can deliver needed precision. It's included on many digital consoles, or outboard units can be inserted.
- To tame a mic when no parametric is available, use the input channel EQ on the console. If the mixing is being done of the FOH console, split the input signal into two channels, using one for the FOH feed and the second for channel EQ sent to the wedges. If a splitter is not available, use the direct output of the first channel to feed the second.

Use those analog channels wisely when mixing monitors from the house console.

- Roll off the low end on the wedges; don't put extra low-frequency energy on stage that adds to the mush. We usually roll off everything under 80 to 100 Hz except for drum and side fills. For wedges getting nothing but vocals, we typically high-pass up to 150 Hz (and even more with female singers).
- Provide a talkback mic even if it's a small stage. Communication is key, period.
- Noted before, but worth repeating: bring a backup gear, especially wedges, to the gig. Having an extra wedge comes in handy with announcers, guest artists and audience members brought onstage, as well as for talkback communication – and in case an IEM system decides to go south two minutes before the set is slated to start. (You know it happens!) LSI

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


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SHOCKING SITUATIONS

Failures in AC outlet testing exposed.

by Mike Sokol

I've been a professional audio engineer for 40-plus years and a musician for 10 years more than that, and during that time, I've witnessed hundreds of shock events on performance stages, recording studios, and even factory floors. A survey I ran on ProSoundWeb a few years ago revealed that 70 percent of the 3,000 musicians who responded had been shocked at least once on stage – some so severely that they were knocked unconscious.

I've also witnessed dozens of ground-fault current events where signal cables interconnecting sound gear plugged into different electrical outlets mysteriously arced, sometimes turning red hot and melting before my eyes. The cause of most of these (usually guitar-to-microphone) shocks appears to be from incorrectly wired electrical outlet grounds or damaged extension cords.

Yet while a broken-off ground pin on a power cord is the obvious culprit in most home or stage shock situations, many power outlets show they're wired correctly when checked with a 3-light outlet tester or even a voltmeter reading between H-N, H-G, and N-G, yet still present a shock hazard. Standard outlet testing methods fail to reveal one of the most dangerous miswiring situations possible, which I refer to as the "Reverse Polarity Bootleg Ground" (RPBG), as seen in **Figure 1**.

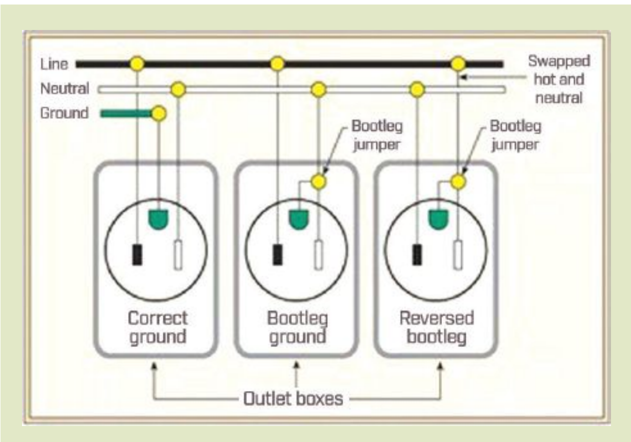

Figure 1: A demonstration diagram showing a correctly wired outlet, a bootleg ground wired outlet, and a reversed polarity bootleg ground (RPBG) outlet.

As the illustration shows, a bootleg ground (or false ground) occurs when an ungrounded electrical outlet in an older building or stage has been improperly upgraded to a modern NEMA 5-15 or 5-20 grounded outlet. Because sound stage, office building, and home wiring installed before 1965 didn't require a safety ground, there's no easy way to install a grounded NEMA 5-15 outlet. Per Section 250.130(C) of the 2011 NEC (National Electrical Code), in that situation, a GFCI (Ground Fault Circuit Interrupter) outlet should be installed with the ground wire unattached.

Under what condition can a 2-wire receptacle be replaced with a 3-wire receptacle when no ground is available in the box? Where no equipment bonding means exists in the outlet box, nongrounding-type receptacles can be replaced with [406.3(D)(3)]:

- Another nongrounding-type receptacle
- A GFCI grounding-type receptacle marked "No Equipment Ground"
- A grounding-type receptacle, if GFCI protected and marked "GFCI Protected" and "No Equipment Ground"

Note that GFCI protection functions properly on a 2-wire circuit without an equipment grounding (bonding) conductor because the equipment grounding (bonding) conductor serves no role in the operation of the GFCI-protection device.

Caution: Permission to replace nongrounding-type receptacles with GFCI-protected grounding-type receptacles doesn't apply to new receptacle outlets that extend from an existing ungrounded outlet box. Once a receptacle outlet (branch circuit extension) has been added, it must be of the grounding (bonding) type and must have its grounding terminal grounded (bonded) to an effective ground-fault current path in accordance with 250.130(C).

However, the Code states this outlet must be clearly marked on its front as being ungrounded. Because GFCI breakers don't need a ground wire to function properly, this type of outlet isn't a shock hazard. Even if the chassis of an instrument or appliance becomes electrically energized due to a high-pot failure, for example, anyone touching it and ground simultaneously would exceed the 4 to 6 milliamps (mA) GFCI trip threshold and be protected from electrocution.

Some electricians and "DIYers" take shortcuts when replacing receptacles on older stages and in residences by installing a grounded NEMA 5-15 outlet and strapping the ground screw to the neutral screw on back of the outlet, creating a classic bootleg ground situation. Although this practice is a Code violation, it

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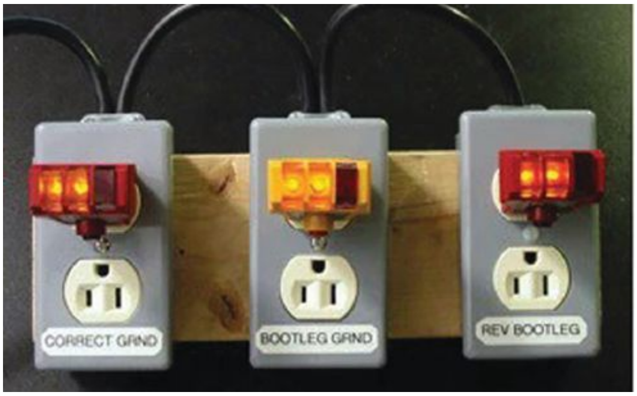


Figure 2: A 3-light "cube" tester is unable to detect a reverse polarity bootleg ground outlet.

occurs more often than you think. And because many electrical inspectors rely on a simple 3-light "cube" tester to verify correct wiring connections at outlets, RPBG outlets can go undetected.

WITH A 3-LIGHT TESTER

An RPBG outlet will test as being correctly wired using any 3-light tester (**Figure 2**), because there's no reference to actual earth potential. What makes an RPBG outlet so dangerous is that any sound gear (guitar amplifier), appliance (refrigerator), or even an RV or boat plugged into an RPBG outlet appears to operate normally. But the chassis of the "grounded" appliance is now directly connected to the "hot" wire in the outlet with a low-impedance current path, and anyone touching the body of the appliance will be electrically biased to a full 120 volts.

That situation itself is not dangerous due to the "pigeon on the power line effect." But if a person touches the strings of the guitar, door handle on the refrigerator, or metal door of the RV that's plugged into an RPBG outlet while also touching anything

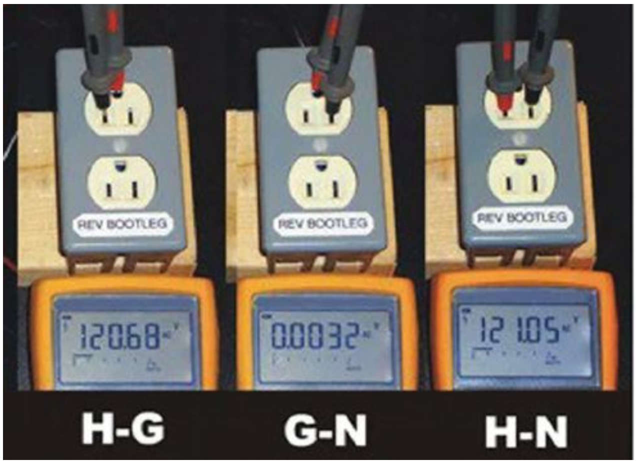


Figure 3: Even a voltmeter will fail to identify a reverse polarity bootleg ground outlet if you simply measure from hot-to-ground, ground-to-neutral, and hot-to-neutral.

that's correctly earth grounded, they will receive a potentially deadly shock (around 100 mA of current at 120 volts). Just 10 mA of current through your body will result in a painful shock, and 100 mA of current for a few seconds is probably lethal if not immediately treated by a defibrillator.

Note that these potential shock currents flow through the ground contact of the outlet, and avoid the neutral and hot contact current paths. So any GFCI outlet wired on a branch circuit extension downstream of an RPBG outlet will probably not sense unbalanced H-N currents and thus won't trip as designed. In fact, because the GFCI can't disconnect its own ground contact from the now electrified ground wire, even if it trips, there will still be the full branch current available for the ground fault path up to the circuit breaker trip current (typically more than 20 amps).

WITH A VOLTMETER

Don't be mislead that a voltmeter is any "smarter" than a 3-light cube tester. Simply measuring from H-G, G-N, and H-N using any analog or digital voltmeter shows that an RPBG outlet will appear normal (**Figure 3**). The only hint there's a bootleg ground is that the G-N reading will be very close to 0 volts, while in a loaded branch circuit with correctly isolated G-N outlets, you'll likely see at least a half to a few volts difference between ground and neutral due to voltage drops in the neutral bus.

WITH AN IMPEDANCE TESTER

Now that we know a 3-light tester and a voltmeter can't identify RPBG outlets, what other device can we try? A ground loop impedance tester, perhaps? Although it can properly identify a bootleg ground due to the G-N impedance being too low, it also cannot identify an RPBG.

Figure 4: A ground impedance tester can identify a receptacle with a bootleg ground, but it can't tell if the outlet has correct or reverse polarity.

In **Figure 4**, the ground impedance tester is indicating a false (F) bootleg ground, but the device can't tell if the outlet has correct polarity or reverse polarity. It only indicates the false ground with a flashing (F) because the ground-to-neutral impedance is too low.

But if an RPBG outlet is feeding an extension outlet on that same branch circuit, the extra resistance of the added wire length would fool the ground impedance tester, indicating 100 percent Code compliance. Even a visual inspection of a "properly wired" extension outlet fed from an RPGB outlet probably wouldn't turn up any obvious problems. Note that its neutral and ground connections are both at 120 volts with respect to earth potential.

A TESTING SOLUTION THAT WORKS

The gold standard method to identify this sort of miswired outlet is to use a voltmeter connected to a known good earth ground to test each hot, neutral, and ground conductor in all outlets. But in reality, this type of test isn't done except perhaps as part of a post-accident forensics investigation.

A simple test is to check the outlet with a mid-voltage non-contact capacitive voltage tester (90-volt to 1,000-volt range), commonly referred to as an non-contact voltage tester (NCVT) When this tester is used on a properly wired outlet, you'll hear a beep or see a light only when its tip is inserted in the "hot slot" of an outlet.

However, when this type of tester gets anywhere near the front or ground contact of an RPBG miswired outlet, it typically beeps/lights from inches away, because the internal ground strap in the outlet will be biased to 120 volts, which provides a large electrified surface area (**Figure 5**). The green light on the left and center outlets indicates no or low-voltage (under 40 volts), while the red light on the right outlet indicates a hot ground condition (typically more than 40 volts for most standard sensitivity NCVTs).

Although using a DMM to test between a known earth ground to each outlet contact is the gold standard, adding this NCVT check for RPBG outlets to your test procedure could save lives and dollars. LSI

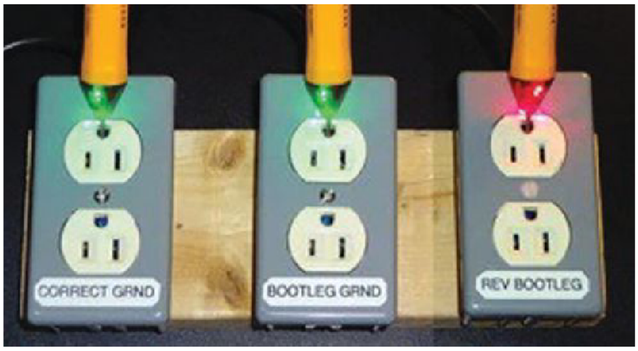


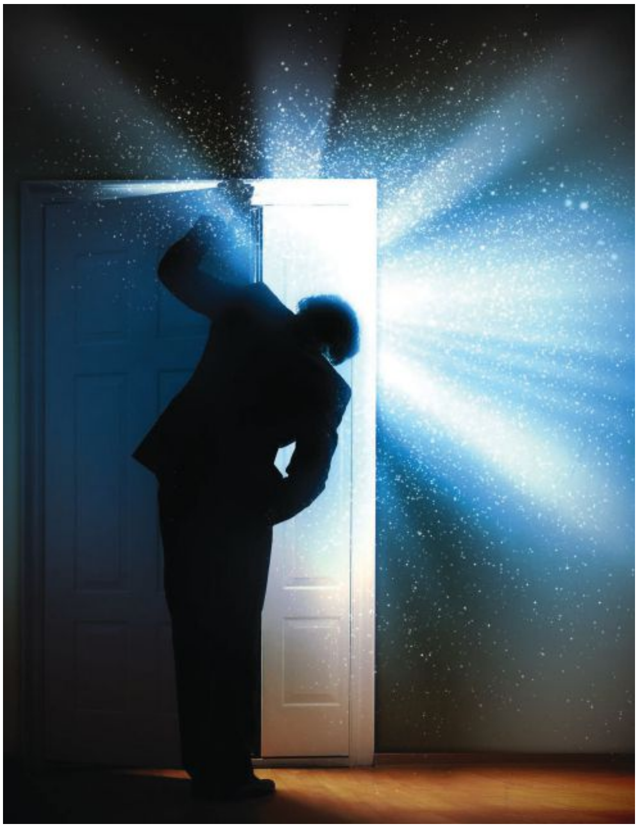
Figure 5: A simple non-contact voltage tester will beep or light up inches away from the face of an outlet that has been miswired in a reverse polarity bootleg ground arrangement.

***Mike Sokol** is lead trainer for Live Sound Co in Maryland, and lead writer of the Live Sound Advice blog (http://livesoundadvice.com). He's a veteran audio educator and is also an adjunct professor at Shenandoah Conservatory in Winchester, VA.*

THE WAY FORWARD

Getting (and keeping) the gig, earning repeat business, building a career and more.

by Nicholas Radina



Career-building and understanding the "biz of the business" is important for everyone working in pro audio, but perhaps even more so for the independent practitioner, a.k.a., Lone Audio Ranger. So let's step back a bit from the technical side and address some equally important techniques that can put you in control of the work you want and strategies to help you get more of it.

HIDING IN PLAIN SIGHT

When it comes to "getting the gig," being selected from a large pool of qualified people is a privilege that's earned. And how that opportunity is earned is the foundation of a successful career. I believe it boils down to six things: talent, taste, hard work, professionalism, personality and perspective.

One of my first high-level touring clients initially hired me as the monitor engineer. Weeks before the first leg started, the budget revealed only one spot: a merchandise seller. Although not the situation I was hoping for, and having zero experience hawking t-shirts, I took the job anyway – and took it seriously. I worked hard and sought to learn as much as possible.

A week into the run, a personnel change led to the offer of an additional position as tour manager. Faced once again with a role I had zero experience with, I took the gig, taking on this new opportunity with just as much determination as "t-shirt hawker." Over the next several months, my responsibilities grew and came to encompass the roles of tour manager, merch seller, stage tech, and yes… monitor engineer.

It turned into many years of wonderful work, learning, great music, and fun. Eventually I even made my way to the stage as a player, which was the cherry on top. The biggest lesson was simple yet profound: trust myself – my potential, my confidence and my work ethic. You can know what's around the corner if you let yourself take a peek.

#POWEROFPERCEPTION

Seemingly obvious, but it's a must to communicate that you exist and are available for work. Never has it been easier to leverage technology to do this. Resources such as Facebook, Twitter, Instagram, and the like offer powerful ways to tell your story, and to cast a wide net while doing so. Being able to communicate to potential clients who you are and what you do helps greatly.

Via an online presence, you can communicate your experience, qualifications, and your personality. It's important to always leverage social media in a professional manner. Create and maintain an informative website focused on the image you want to project, with qualifications to back it up. Don't forget to make it as easy as possible for anyone interested to contact you. All social media activity can flow back to this destination.

Further, it's always good form to pass along through these channels ideas and information you've come across that could be helpful to others. (Dave Rat, to point to one example, does this extremely well, and I strongly suspect it's worked out quite well for him.) Remember, the ones making the hiring decisions think first of who they know, and then next, who's made an impression. Make every effort to be in those memory banks, and for all of the right reasons.

NETWORKING (UGH)

Like many, I can be an introvert. The sight of the ubiquitous "networking event" usually has me thinking of walking in the other direction. Although helpful to some, particularly extroverts, these staged get-togethers can seem forced. For introverts, it's a true chore to talk with strangers, exchange the obligatory business cards, and trade rusty gig stories.

Yet networking comes in many guises. (In fact I'm networking with you right now.) We may not know each other personally (yet), but you can find me easily if you want, and I'm willing to provide help in any way possible regarding "things pro audio." The social media tools noted previously also offer great (and ever-expanding) networking benefits, and there are also like-minded online groups and industry forums to participate in.

Even simply meeting up with some local fellow "sound nerds" for a drink or a meal can open up new prospects. We work in a business that's made up of talented, creative, hard-working people who for the most part are very open to sharing knowledge and opportunities.

STILL LEARNING?

We can be totally into audio but that doesn't limit us to just mixing a bar band every week at the local pub, touring year-round, and everything in between. Tech skills are valuable in other environments, so it's a solid career move to step out of the audio box a bit and apply those skills to related fields. Learn the basics of operating today's video switchers and lighting consoles, or delve into teaching and writing.

Modern technology offers so many resources to learn new skills to make ourselves even more marketable. Take this challenge and reap the benefits.

LIKE WHAT YOU DO?

Although obvious to most, being passionate about work is paramount, because our relationship to that work is what guides attitude, work ethic, the ability to handle stress, and problem solving. We all have days, weeks or even months where we may question our career path, usually stemming from the time and physical commitment it takes and how those both relate to financial reward or even simple satisfaction.

When this happens, my approach is to step back a bit to identify why I'm struggling. Is it truly the overall job, or is it a specific gig? Maybe things have become mundane? Was someone difficult to work with? Answers to these questions provide peace of mind while helping us align ourselves with the types of people and organizations that best fit our particular interests and goals.

RESPECT & FOLLOW-UP

We should know the essentials of the gigs we're walking into. Don't assume that new clients understand our technical world. One of our goals is always to make the ones who hire us look good for hiring us.

And sometimes it's overlooked, but the relationship with whomever hired us doesn't end with the paycheck. A simple follow-up email to every client should be standard practice. In some circumstances, asking for a quick overview of their experiences with our work in return can help toward building a powerful portfolio of happy existing clients to show to prospective new clients.

WHAT ARE YOU WORTH?

This is a tough one because the amount charged for work can be relative to specific geographic areas, competition, and the overall specifics of a given market. But it's important to really analyze what our time is worth, and then set a fair rate to charge within those specific circumstances. Try to be consistent with what you charge clients, and never "nickel and dime."

In addition, don't fall into the trap of pricing yourself out of opportunities that, although initially low-paying, could turn into a long-term opportunity and revenue stream. Look at the potential of what something might become. (Recall what happened with that young fellow once hired by a tour to sell merch.)

THE WIN

After following these principles for a few years, I realized I was able to do more of the work I liked, and more often, because those who made the hiring decisions were specifically choosing me and what I could bring to the table. And that's a privilege I don't take for granted.

Our talent, experience and personality is the name of the game, far more important than the gear we own or have available. I've always tried to put into perspective how fortunate I am to spend my time (and get paid for) doing something I love while building and maintaining a solid reputation. I wish the same for all Lone Audio Rangers. LSI

Nicholas Radina *is an audio engineer and musician based in Cincinnati. In addition to keeping up with a busy freelance schedule, he smacks cowbells with local Salsa bands and tours as the monitor engineer with the band O.A.R. He invites your input via his website at NicholasRadina.com.*

TRIPLE-THREAT **PRODUCTION**

Simultaneous sound reinforcement (and more) at three venues for Passion 2016. ***by Steve Harvey***

CREDIT: JAY RIGBY

L-Acoustics K1 and K2 arrays delivering sound in the round at Philips Arena in Atlanta at Passion 2016.

FOR NEARLY TWO decades, the 268 Generation, a Christian organization that hosts gatherings worldwide of college students between the ages of 18 and 25, has traditionally held its annual U.S. national Passion Conferences at multiple venues throughout January and February. For the recent Passion 2016, however, organizers vowed to bring an identical experience to all attendees, ambitiously staging simultaneous three-day events at three locations, with real-time audio, video, and control data streaming and synchronized between the locations, and with a webcast to the world.

This year's convocation attracted tens of thousands of young adults to Philips Arena in downtown Atlanta, Infinite Energy Arena in Gwinnett County (northeast of Atlanta), and Toyota Center in Houston. Pastors at each of those venues – respectively, Brad Jones, Clay Scroggins, and Ben Stuart – led the daily session schedules, interacting with each other via IMAG screens at each location. Pastor Louie Giglio, who founded the conference in 1997, together with the Passion band plus artists and musicians including Hillsong United and Rend Collective, traveled between the three cities over a period of less than 48 hours to appear before audiences at each arena.

The locations had identical control and broadcast packages, reports Tom Worley of Rat Sound Systems (Camarillo, CA), which supplied audio production at all three venues and has provided sound for two previous Passion Conferences. Each venue was outfitted with three DiGiCo SD7 consoles: one at front of house and one at monitors, plus another for broadcast located backstage. Each system also incorporated six Waves DiGiGrid servers.

In addition, there were two Midas PRO9 consoles at each location (one for FOH and the other for monitors), provided at the request of Hillsong United, a major band from the Australian Christian


CREDIT: JAY RIGBY

The view from behind the DiGiCo SD7 at front of house at Philips Arena.

community. Artists performing throughout each day alternated between the two console platforms.

MAKING IT THE SAME

Since the bands played all three arenas, having identical systems at each location enabled show files to be easily updated and shared, Worley notes, adding that all crew members had access to the latest information on Dropbox or Google Drive. The microphone complement was also identical for each band at each venue. "If you were using a [Shure] KSM32 on overhead in one venue it was the same in the other venues," he says. "This gave us peace of mind knowing that the show files would transpose between venues and allowed us to have the consistency required to pull off an event of this nature."

Wireless systems were similarly duplicated from one venue to the next, including 16 channels of Shure UHF-R mics, 12 channels of Sennheiser G3 in-ear monitoring systems (working with the Midas consoles), and 12 channels of Shure PSM1000 personal monitoring systems (working with the DiGiCo consoles). RF technician Tom Jones coordinated wireless frequencies at each site.

Rat Sound PA tech Andrew Gilchrist designed the house reinforcement systems; the two Georgia locations were provided with in-the-round coverage while Houston was configured as an end stage. The systems were virtually identical, scaled relative to each venue's size.

Coverage at Philips Arena was delivered by eight line array hangs comprised of 41 L-Acoustics K1 and 60 K2 cabinets. Four hangs were positioned in the four corners of the room, with the mid-section handled by upper and lower hangs to work around a large-format video screen. Infinite Energy Center was reinforced with six hangs of 18 K1 and 59 K2 boxes, and at the Toyota Center in Houston, there were four hangs of 28 K1 and 44 K2. Each rig also included two-dozen SB28 subwoofers as well as ARCS and Kara loudspeakers for various fill needs.

"Philips Arena was the hub of the whole system; that's where all the rehearsals were," says Worley. "We had one day to set up then went straight into rehearsals. It's a big space to fill in-the-round, and every seat was full, right down to the front of the stage. A lot of the hangs were eight K1 with eight K2 underneath, or a similar combination – it was great to have the width of the K2 at the bottom. The curvature on the PA was pretty phenomenal."

In Houston, 18 L-Acoustics LA-RAKs on the floor (stage left and right) provided amplification and loudspeaker control, while at the Georgia venues, these units (26 LA-RAKs in Atlanta and 22 in Gwinnett) were positioned on the bumpers of the arrays. "We took drive lines from front of house up to the catwalk using Riedel RockNet (digital audio networking). That was distributed out over AES, with an analog backup, to every hang. Power distribution was up there as well," Worley notes.

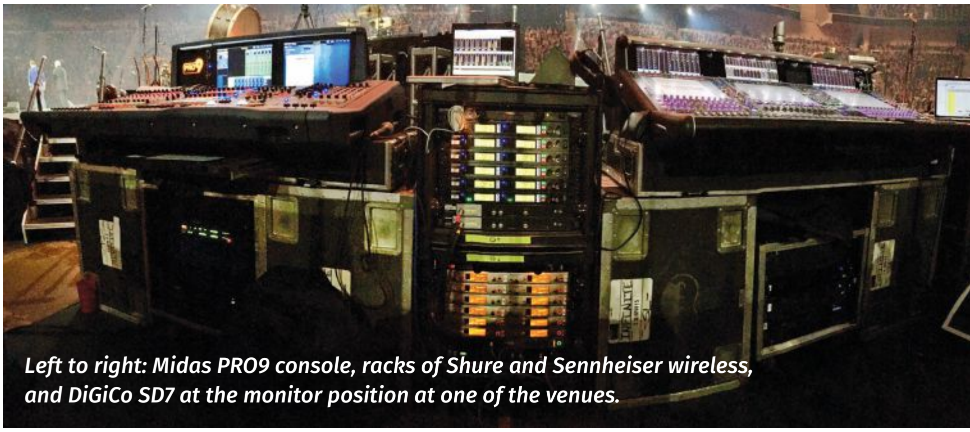
HERE TO THERE

The three SD7 desks at each venue were networked over the Optocore fiber-based transport platform in a configuration that linked each console through four DiGiCo SD-Racks to form a star-shaped loop. "We went from the SD-Racks, which lived in monitor world, out to front-of-house, then back to another SD-Rack," Worley elaborates. "Then from the other side of the SD-Rack to broadcast and back into the other side of an SD-Rack. Monitors closed and opened the loop."

The two PRO9 consoles at each venue interconnected with three DL431 I/O-splitter racks over AES50, and in turn were connected over an analog link into two dedicated DiGiCo SD-Racks. "Both Midas consoles had head-amp control," says Matt Manix of Method Production Group (Nashville), who was responsible for designing and coordinating the audio networks within and between the locations. "We didn't do any gain tracking on the DiGiCo side. Two of the racks were fully controlled by monitors and everybody else had trim control. The SD7 broadcast engineers had head amp control over the two racks that took all the Midas inputs."

Manix says of the SD7 package, "Having shared I/O and being able to have clean audio at every position without utilizing analog splitters, we were able to get the channel count up really high and retain the quality. It has to have one of the largest output bus counts out there and is arguably the most flexible system, as far as routing anything anywhere. And with the SD7's A and B engine we didn't have to rent extra desks or engines for redundancy."

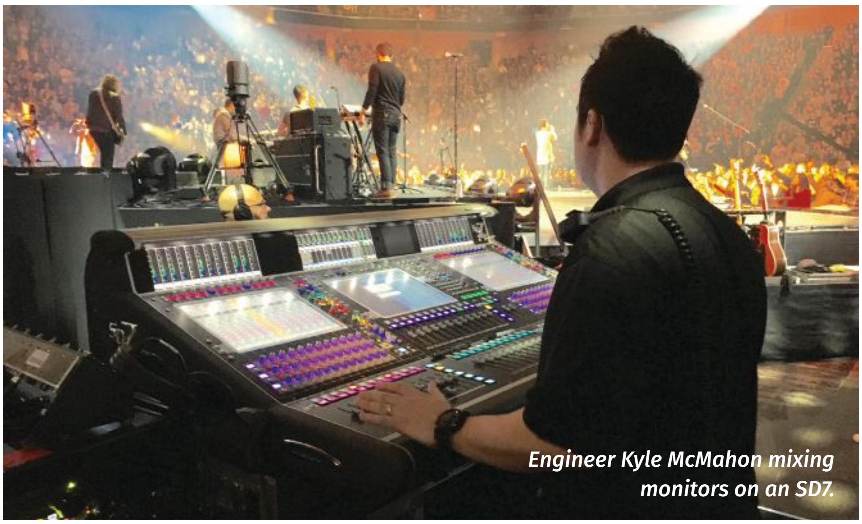
Philips Arena in Atlanta was the hub for the fiber link transmission, with satellite backup, between venues. "It was a hub and spoke, with Philips as the hub for all the HD-SDI transports," Manix notes.



Left to right: Midas PRO9 console, racks of Shure and Sennheiser wireless, and DiGiCo SD7 at the monitor position at one of the venues.

"All of the video going between all three venues made its way through Philips, as did the audio. Philips could 'talk' to Toyota Center and Gwinnet independently, but there wasn't fiber between the two. We never had to use the satellite backup because the fiber network was rock-solid."

An HD-SDI stream with 16 embedded audio channels traveled over rented OC-192 lines between cities. Transmission speeds over the optical carrier are typically comparable to 10 Gigabit Ethernet. The Optocore network extended out to each arena's loading dock, where a DD4 fiber-to-MADI converter fed video-link world. Audio signal passed via a DiGiCo SD-Mini Rack located in the outside broadcast production truck supplied to each venue by TNDV of Nashville, which provided a complete camera package and handled video switching. "They could pick up what they wanted from the broadcast or front of house engineers, and shoot that off via fiber to the other venues," says Manix.



Engineer Kyle McMahon mixing monitors on an SD7.

Using a Riedel MediorNet fiber-based real-time media transport network, link transmission operators were able to select which of the broadcast mixes from the three venues to webstream. "The link truck had access to all embedded audio from each venue at all times. Additionally, each DiGiCo desk at each venue had access to all embedded audio from all venues," Manix explains. "There was a master clock from the OB truck, sent tri-level sync to MediorNet, which provided word clock to our audio network. Then it was re-clocked at every venue. It was perfect."

GETTING IN SYNC

The pastors emceeing at each venue were outfitted with DPA headset microphones with integrated earpieces, through which they received an IFB (interruptible foldback) feed. "They wore an IFB pack with essentially a mix from monitors of producer talk and host ISOs from the other venues. There were three producers and three assistant producers calling the show at the venues," Manix says.

When the pastors were speaking, he continues, "We received the live ISO of the headsets from the other venues via the MediorNet into the DD4, and sent out our local ISO mics in the reverse fashion. The ISO of any person speaking was sent out discretely so it could be processed in the receiving venue separately."

Further, he says, "Each broadcast desk sent a mix-minus to the other two venues, which were picked up by front-of-house, broadcast, and monitors. We also sent the front-of-house program feed to each venue as a backup to the broadcast mix and mix-minus. We had complete redundancy everywhere."

The TNDV truck at Philips Arena was the sole source for any pre-produced video packages, but all three trucks captured the feeds from the other two venues on EVS and MIRA instant replay machines for delayed playback. "There were some moments where a person at one venue would start speaking, and the other two venues would capture that then delay the playback so it could be timed with the end of their sessions," Manix says.

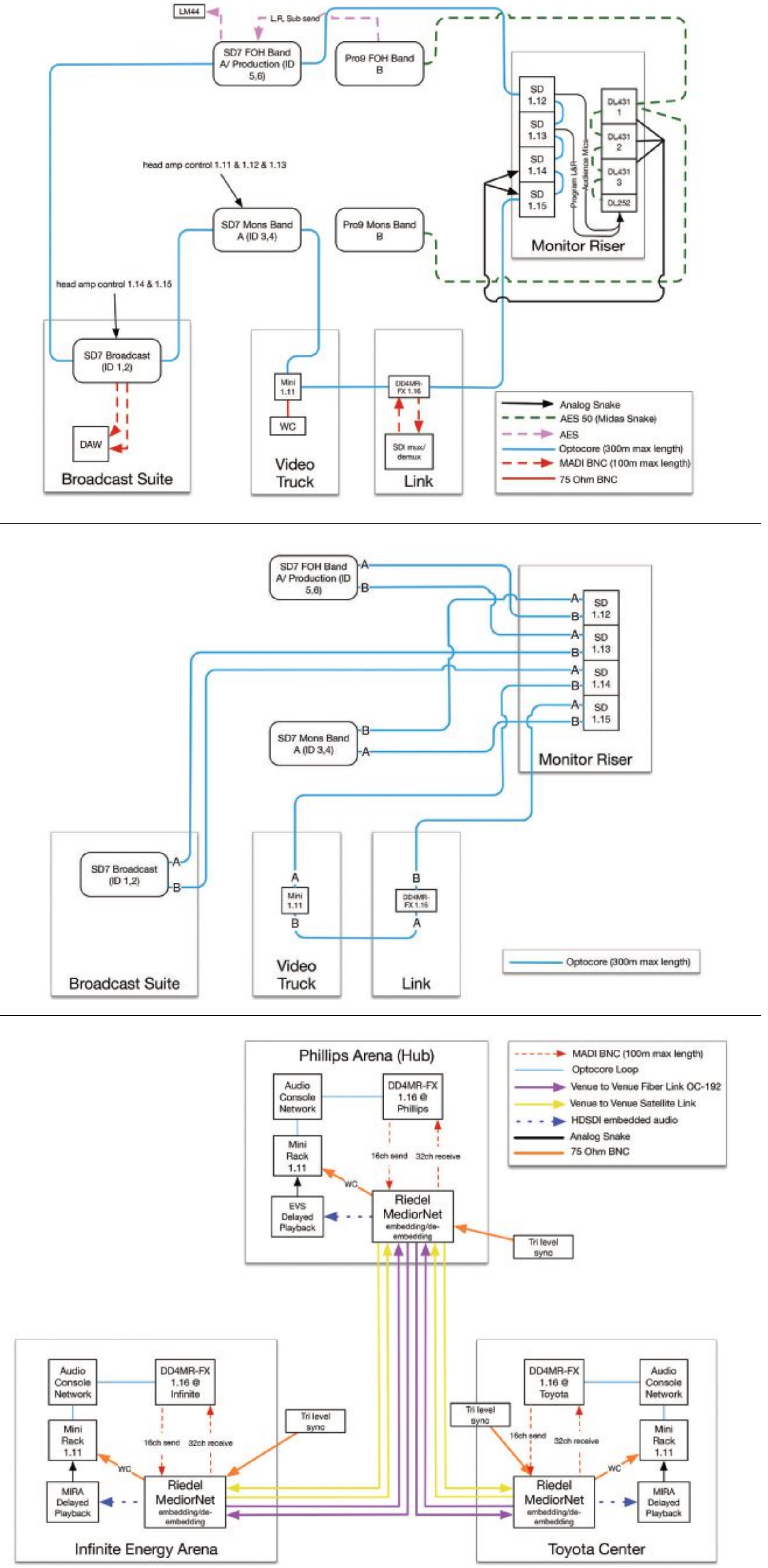
Production intercom between venues was also transported over the fiber links: "The Riedel guys set us up so that I was able to talk instantaneously with the other two venues and coordinate line checks over fiber from Philips, which is where I was located. That was super cool."

In addition to audio, video, and comm traffic, says Manix, "There was also some timecode sent across for some of the lighting and video sync elements. If it was generated in one venue then it would be synched at the other venues."

The roundtrip delay between Phillips Arena and Toyota Center was about 50 milliseconds (ms) and around 20 ms between the two Georgia venues. "We had one pretty cool sync moment where all three venues had bands playing at the same time," Manix reports. The performance featured rappers Lecrae and Trip Lee at the Philips Arena, KB at Infinite Energy Center, and Tedashii at the Toyota Center.

"Click was generated at one venue and received at the other two venues," he recalls. "We received the band mix-minus back from them. We ended up delaying the click and tracks in the source venue, where it was generated, so that when we received their live audio back it was in time. We had to do something similar for the Georgia venues. It was a bit tricky doing the math, but everything worked flawlessly. It was pretty great." LSI

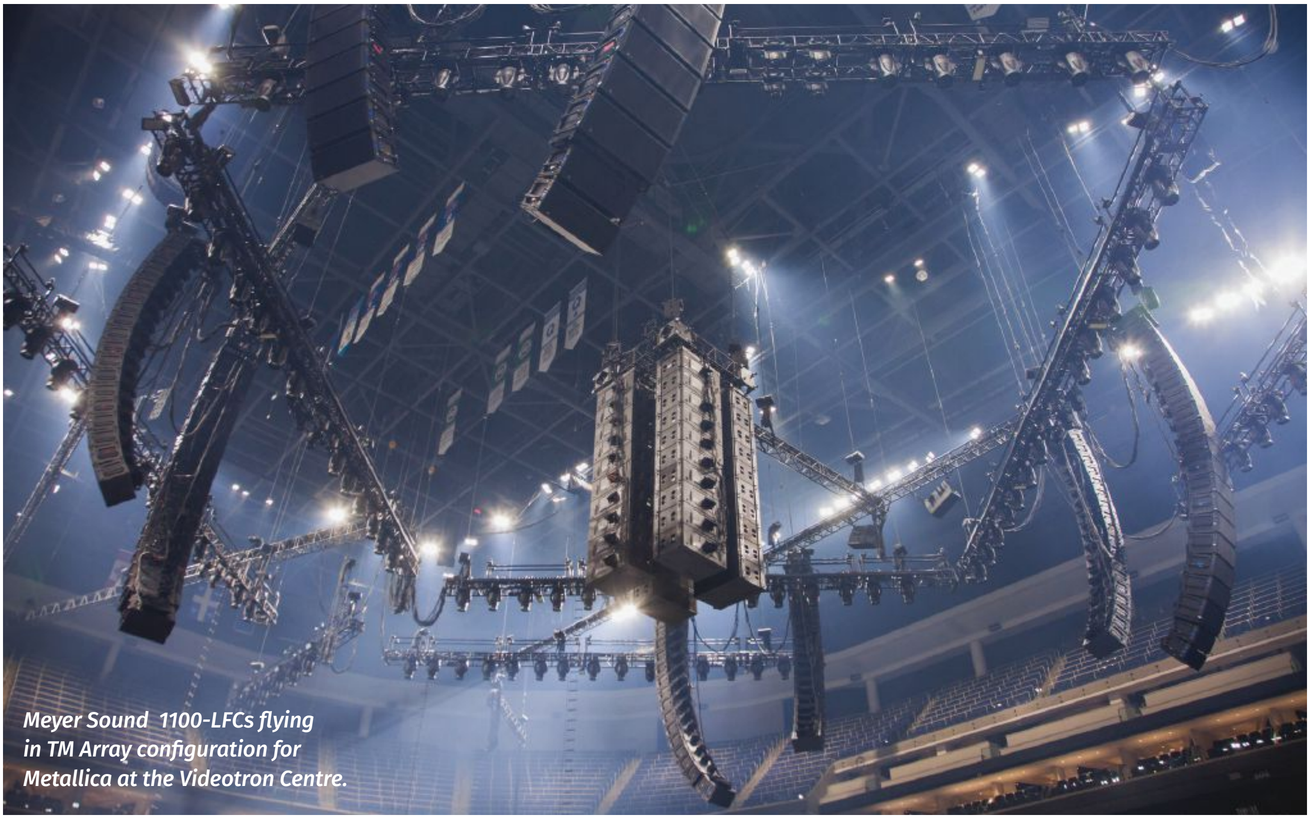
*Based in Los Angeles, **Steve Harvey** is a long-time pro audio journalist and photographer.*



The top diagram shows the system's DiGiCo-Midas network, followed by the DiGiCo-Optocore signal transport loop, and then transmission routing between the three venues.

CONTINUOUS IMPROVEMENT

Interesting new upgrades and directions in a range of projects. ***by LSI Staff***



Meyer Sound 1100-LFCs flying in TM Array configuration for Metallica at the Videotron Centre.

BIG LF IN THE ROUND FOR METALLICA IN QUEBEC

For the first time, Metallica used an in-the-round Meyer Sound LEO loudspeaker system with 1100-LFC low-frequency control elements in a TM Array configuration when the band recently inaugurated the 20,000-capacity Videotron Centre in Quebec City, Canada.

The TM Array configuration comprised 40 1100-LFC elements configured in four arrays suspended directly over Lars Ulrich's drum kit to spread a uniform, donut-shaped pattern of low-end frequencies throughout the seating area. "Never before have I heard that amount of low end – with such tightness – when we played in the round," reports "Big Mick" Hughes, Metallica's long-time front of house engineer. "The TM Array works fantastically well. The 1100-LFCs just have an ungodly amount of power."

The main arrays were made up of 72 LEO-M line array loudspeakers for the long ends of the arena and 72 LYON-W wide-coverage line array loudspeakers for the shorter side throws, while down fills for the end hangs were four-each LYON-M and LYON-W loudspeakers. Eight MICA and 16 M'elodie line array loudspeakers provided front fill, four JM-1P arrayable loudspeakers were suspended for side down fill, and 48 700-HP subwoofers covered the lower seating area. A Galileo Callisto loudspeaker management system supplied system drive and alignment.

"Working in the round, you have to get a grip on what you're doing – fortunately, LEO is very controlled at all frequencies," Hughes adds. "You're in the near field of one part of the PA, but you can have problems with room ambience generated by all the other array elements going off in other directions. LEO makes it completely manageable."

A MIXING SYSTEM UPGRADE FOR KING CHARLES IN LONDON

An Allen & Heath dLive mixing system was deployed recently at London's O2 Arena for King Charles supporting Mumford

Engineer Ken Lewis with the Allen & Heath dLive S3000 control surface for mixing King Charles in London.

& Sons on their UK tour, with engineer Jon Lewis employing a dLive S3000 console and a DM64 MixRack at front of house. The S3000 is equipped with 20 faders and a 12-inch touch screen that allows for up to 120 fader strips.

"From the build quality to the GUI to the flexibility of the console, everything's taken a huge step up," Lewis says of the dLive platform. "It's also extremely intuitive – we only had a day's rehearsal with the console before the O2 show with King Charles but that's all I needed to really get to grips with the dLive.

"One of my favorite new features is the added compressors on the channel strip, particularly the 'opto' comp, which allows you to dial in the ratio of wet/dry signal you need, effectively giving you parallel compression," he continues. "It sounded absolutely fantastic on the drum group at the O2, for example." An Allen & Heath GLD Chrome console was deployed for monitor duties.

SUPPORTING THE ARTS AT THE HEMMENS IN THE HEARTLAND

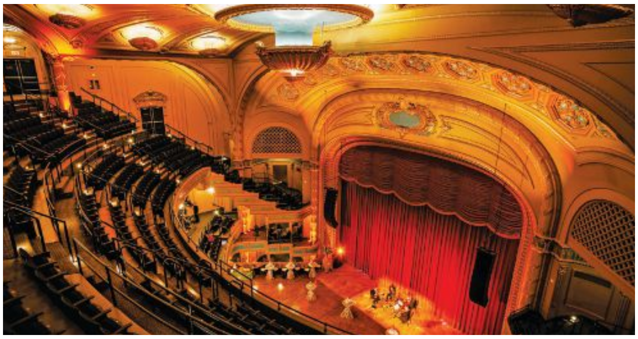
A new house system at The Hemmens Theatre is headed by NEXO GEO line arrays and a Yamaha CL5 digital console. The system was specified and installed by Gand Sound Installations (Elk Grove Village, IL) to serve the 1,200-seat performing arts facility that's a cornerstone of a growing arts community in Elgin, IL.

NEXO GEO arrays implemented at The Hemmens Theatre in Elgin, IL.

Specifically, the main left and right arrays each include eight GEO S1210 modules joined by a GEO S1230, while a center array incorporates a GEO S1210 and three GEO S1230 modules. In addition, dual NEXO RS18 Ray Sub install subwoofers are flown above the venue's acoustic clouds, with additional RS18 Ray Sub tour subwoofers provided for situations where extra low-end is required. All of these loudspeakers are driven by NX4x4 amplifiers.

The CL5 console at front of house is joined by two Rio3224-D I/O boxes and a Rio1608-D I/O box, all Dante networking compatible. "We chose the GEO S12 for the fidelity, clarity, flexibility, pattern control, ease of installation, and power," says owner Gary Gand. "The CL5 was selected due to its user friendliness and the ease of communication with Dante, accessible throughout the system from the Shure ULX-D wireless systems and all the way through to the NEXO amplifier outputs."

The entire system also needed to be easy to move and store for concert dates featuring the Elgin Symphony Orchestra when it's not needed.



CREDIT: JAMEY SHAW

SOUND FOR REVITALIZED RKO ORPHEUM IN THE BIG EASY

The revitalized RKO Orpheum in New Orleans outfitted with Martin Audio MLA Compact arrays.

Originally built in 1918 and shuttered since Hurricane Katrina, the 1,500-seat RKO Orpheum in New Orleans has new life following an 18-month restoration, including a new sound reinforcement system headed by a Martin Audio MLA loudspeaker system. The theater re-opened late last year with a performance of Mahler's "Resurrection" symphony by the Louisiana Philharmonic Orchestra, a homecoming for the ensemble.

Venue staff worked closely with installer Don Drucker of Pyramid Audio Productions (Jefferson, LA) on the new house system. "Don helped us make decisions with the understanding that in an acoustically sound structure, installing an amplified audio system needed to be done with care and an understanding of all the types of events being held at the Orpheum," explains general manager Kristin Shannon.

"We looked at several manufacturers who submitted designs for the site, rated those designs, made necessary adjustments and then presented the better proposals to the client," Drucker says. "Of all the submissions, Martin Audio required the fewest adjustments and that's how we came to the decision.

"We also knew the system had to be of the highest sound quality, very controllable, aesthetically friendly and compatible to the space while the response and articulation had to cover every seat in the house. We didn't want to upset the hall's acoustics and felt Martin Audio MLA would have the best fit into the acoustic atmosphere of the theater."

The new system includes arrays of 12 MLA Compact modules flown per side, with four DSX subwoofers ground-stacked near the stage, used for certain rock bands. In addition, 30 DD6 dual differential dispersion loudspeakers are deployed under the room's the two sets of balconies, and they're also used as front fills at the front edge of the stage, removed for dance events.

SONIC REFINEMENT AT OPPIKOPPI IN SOUTH AFRICA

The recent Oppikoppi music festival, held annually in the Limpopo Province of South Africa, welcomed more than 160 artists who performed over three days on seven stages. Blue Array Productions (Pretoria) supplied the systems for all but one of the stages, deploying a range of d&b audiotechnik gear, including J-Series (arrays of J8 and J12) for the Main Stage, joined by J SUBs that were both ground-stacked and flown behind the main arrays.

Most significantly, it was on the Main Stage that Blue Array also chose to implement d&b's recently introduced ArrayProcessing. This function within d&b ArrayCalc simulation software helps with optimization by applying a combination of FIR and IIR filters to each individual loudspeaker within an array to achieve a common tonality and consistent sonic result over the audience area, defined by the vertical coverage.

Blue Array managing director Kobus van Rensburg states, "Not having used ArrayProcessing before, I was intrigued to experience both the process and the result. We designed the system in the d&b ArrayCalc software to give us the best front-to-back coverage, as we would normally do. Then we applied the ArrayProcessing algorithm to the main array to give us a 2 dB drop in SPL from front to back, which was over 80 meters away. This whole process only took about five minutes, from design to implementing and loading the parameters into the amplifiers.

"It's no exaggeration to say the result was absolutely astounding," he continues. "The front-to-back coverage was smooth and consistent without any tonality changes at all, just that slight defined 2 dB drop in SPL as you moved towards the back. I've not experienced such even tonality from any PA system ever before. We also found that there was no distortion or audible artifacts present at any volume, high or low."

An improved listening experience thanks to d&b ArrayProcessing at Oppikoppi in South Africa.

CREDIT: CHARLEMAGNE OLIVIER

MEETING EXPANDED REQUIREMENTS AT SYRACUSE U

To improve both the quality of music and speech at Hendricks Chapel, which for 85 years has hosted a wide range of events on the campus of Syracuse University (Syracuse, NY), DCI Sound (Marcellus, NY) recently implemented Renkus-Heinz Iconyx Gen5 loudspeakers.

Renkus-Heinz Iconyx Gen5 IC24-RN digitally steerable arrays providing coverage while blending in at Hendricks Chapel at Syracuse.

Specifically, a left-right pair of Iconyx Gen5 IC24-RNs, which are digitally steerable arrays, are mounted on each side of the proscenium to provide primary coverage in the highly reverberant space that has domed ceilings, curved walls, and deep balconies. The lower end of the spectrum is reinforced with Renkus-Heinz CFX12S 12-inch subwoofers.

"The space was underutilized until about 15 years ago," notes David May, owner of DCI Sound. "At that time, we installed a sound system that was primarily Renkus-Heinz loudspeakers and first-generation steerable arrays. Finally, people could hear, and usage of the space increased. But that system was designed for speech intelligibility, and Hendricks Chapel now hosts many musical events. So the university asked us to design and install a new system that would address the chapel's expanded requirements."

DCI has extensive experience with Iconyx, May explains. "In addition to the advantages of steered beams, Iconyx is much more musical and has higher output capability than the systems we used before," he says. "Iconyx Gen5 is even better because you get a more flexible selection of configurations, with even greater precision. That means we can customize Gen5 systems even better for each venue, and can deliver a high-quality system for less money."

The IC24-RN employs 24 four-inch coaxial transducers, each with three tweeters, in a slim, low profile design. "Because we can choose the location of multiple beams, we were able to mount the arrays up high, getting the beams above the lectern mics," May notes. "That improved gain before feedback." He also utilized new RHAON II software to program the loudspeakers. "The original RHAON was good, but RHAON II renders much faster. It's a big improvement," he concludes. LSI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WAVES AUDIO SUB ALIGN

Checking out a new plugin for PA systems.

by Phil Hagood

The new Waves Audio Sub Align plugin allows live engineers to finally have control over PA systems that have no accessible DSP. In short, it adds a crossover with gain and delay adjustment to PA drive lines. Many times in small venues, there's only a left and right feed to the PA but the system doesn't have a proper tuning in the processor.

I recently evaluated Sub Align, and as with most Waves products, I was able to first check out an instructional video that provided a complete demonstration of the operation of the plugin as well as good explanation of what it would do. As usual, the download and authorization process was easy. The new Waves Central license software allows quick and easy viewing of your purchased products and where they're authorized. When purchasing or demoing the product, it immediately shows up in Waves Central and allows you to authorize the plugin on a USB drive or computer. When using plug-ins in a live scenario, having an easily transferred license can be crucial when dealing with venue owned consoles.

Upon completing my download, I found the graphic user interface (GUI) of Sub Align to be sleek and easy to operate, both for novice engineers and seasoned pros. It provides a crossover frequency for both left and right, plus individual gain, delay, and polarity control for left and right. This allows you to adjust and correct subwoofers and mains that aren't set up symmetrically. In addition, the ability to move into negative delay provides an extremely powerful tool for engineers that hasn't been previously available in a traditional DSP.

The graphic user interface of the Waves Sub Align plugin.

SETTING PARAMETERS

To evaluate this plugin, an SSL L300 Live console and a Waves MGO + Server One was set up to drive two different PA systems. The first system was comprised of stereo powered Electro-Voice two-way, 15-inch loudspeakers joined by stereo powered JBL subw oofers, similar to what would be found in a small club or church. The second system utilized NEXO concert-level line arrays joined by both flown and ground-based GEO subs as well as several fill loudspeakers. For both systems, there was a laptop running Rational Acoustics Smaart v7 with a iSEMcon tuning microphone to provide a transfer function reference to test the results of the plugin.

The first (smaller) system was intentionally set up with asymmetric subs to test the functionality of Sub Align. The crossover point was established at 80 Hz, then the delay times were moved back and forth until the kick drum "felt" good. Full-range test tracks were then used to reference transient base response as well as tonal bass response, and then Sub Align was again adjusted until the music "felt" right.

As a further test, the subs were then timed and tuned using Smaart v7. The results were several milliseconds different at the front of house position using the Smaart tuning versus the plugin tuning. That did not effect the 40 to 50 Hz punch but did change the clarity and smoothness of the 60 to 80 Hz range.

WITH THE LARGER SYSTEM

We then moved along to the second (larger-format) system, and the same process was followed. With the combination of subs in two different locations, they never "felt" right. Either the flown subs were timed in, or the ground subs were timed in, and the tonality of them never felt smooth.

Once the results were cross-checked with Smaart, we could visually see the issue. When using ground and flown subs, the plug-in does not give the granularity required to properly calibrate the system. This resulted in uneven frequency response and inconsistency across the venue in terms of low-end response.

The other side effect is in the front and delay fills of a larger system; if the crossover point of the fills and main PA are not the same, there can be unwanted low frequency timing issues in the fill boxes. For example, if you're using a full-range delay fill, you're now changing the timing within the low-frequency component of that single loudspeaker.

The bottom line is pretty clear. Sub Align is a great tool that can be useful in smaller systems and venues, but it's not one size fits all. If a PA has full-range signal being fed to a simple stereo system with subs and mains — and without fills — the plugin can be used to great benefit. Adding fills and multiple subs in multiple locations compromises its effectiveness. That said, virtually all situations with larger systems will have an accessible DSP and/or system engineer to remedy any issues in sub timing and levels.

Waves confirmed this assessment, expressly telling us that in any conventional situation, they recommend using a system's DSP for important adjustments such as time-aligning subs, but that the primary purpose of Sub Align is to serve as a "show-saver" in situations without access to DSP.

The company continues to develop innovative and original plugins that allow engineers to truly be able to focus on the music and creativity in mixing. Sub Align is an excellent tool for engineers using smaller systems. It definitively provides a way to quickly and effectively recalibrate these types of systems for a better mix and a better show.

The Waves Sub Align plugin is available for $129 at www.waves.com. LSI

***Phil Hagood** is director of operations for Morris Integration, based in Nashville.*

Sub Align Overview

SUPPORTED PLATFORMS
Most Audio Hosts
MultiRack Native
MultiRack SoundGrid
StudioRack Native
StudioRack SoundGrid
eMotion ST Mixer
eMotion LV1 Mixer

OPERATING SYSTEMS
Mac 10.9.5 - 10.11.3
Mac 10.8.5 for ProTools 10 only
Windows 7 with SP1 64 bit
Windows 8.1 64-bit
Windows 10 64-bit

INCLUDED IN BUNDLES
Mercury
Pro Show
SD7 Pro Show

PURPOSE BUILT

A look a vocal microphone design principles and a look at recent models.

by Craig Leerman

The textbook definition of a microphone is a transducer that converts acoustical energy (sound waves) into electrical energy. Basically, a sound wave hits a diaphragm or membrane and causes it to vibrate. These vibrations are turned into electricity and flow out to the console.

Mics are usually categorized by the conversion process that's used to turn acoustical energy into electricity. The most common type used on live stages are dynamic designs that work on the electromagnet principle where a coil of wire is attached to the diaphragm and moves by a magnet, creating electricity as the sound waves push against the diaphragm. Dynamics can

be very rugged and resilient to rough handling, but because there's added mass attached to the diaphragm, they may not respond as quickly to changes in sound pressure as other types.

Condensers have gained in popularity in the live world, with many newer models robust enough to withstand abuse. They have two plates with a voltage between them. One plate is made of very light and flexible material, and acts as the diaphragm. The diaphragm vibrates when struck by sound waves, changing the distance between the two plates, which in turn changes the capacitance, producing an electrical signal.

Condenser diaphragms are not connected to a coil mass. They respond to transients and higher frequencies very well, enhancing sensitivity and the ability to capture high-frequency detail with more accuracy. A variation of the condenser is the electret, which uses a fixed charge on one of the plates. Note that condensers require a power source (phantom power or batteries).

The vast majority of vocal mics are unidirectional, with pickup patterns including cardioid, supercardioid and hypercardioid. In general, they reject sounds from directions other than the front (at various degrees), and this is highly desirable given all of the noise on a typical stage as well as from the house.

Cardioids are more forgiving to a singer's technique and the angle at which the mic is addressed. Supercardioids are more focused to favor sounds from the front and have greater attenuation at their sides – about 10 dB – exhibiting minimum sensitivity 140 to 150 degrees off-axis, with a minor secondary response lobe at their rear. Hypercardioids are even slightly more focused and distinguished by minimum response 110 to 120 degrees off-axis at the expense of a slightly larger rear lobe.

Another way mics are categorized is by their intended use on vocals or instruments. Many sound good in either application, but higher-end vocal models often have added features to benefit vocals. One phenomenon that comes into play is proximity effect – as a singer gets right up on the mic, the lower frequencies become more pronounced. Many premium vocal mics address this issue with acoustic tuning, low-cut switches, or electronic processing so the singer's voice stays more consistent.

Vocal mics should be designed to deal with "pops and plosives," the breath noises that happen when a singer uses a hard consonant like a P or B. Mics designed for live performance have windscreens, but the better vocal models tend to use multi-stage or layered screens to reduce the problem. Handling noise is another issue. As it's moved around, noise from the mic body can get transmitted to the capsule. Capsule isolation to curtail this issue is a hallmark of well-designed units.

Mics specifically designed for vocals should have a frequency response tailored for the human voice. Some models will offer a reduced frequency range while others may add a boost or cut. When it comes to vocal mics, it's not necessarily always about having a flat-graph response, it's about making the voice sound great in the mix. Enjoy this tour of a range of dynamic and condenser mics for vocals. LSI

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

RWG Spotlight Listing

Audio Technica Artist Elite AE5400

www.audio-technica.com

The Artist Elite Series is engineered for superior performance, with the AE5400 leading the way in delivering unmatched vocal quality. Audio-Technica partnered with leading engineers and artists in developing the AE5400, applying those insights with an extensive knowledge base in the creation of a true high-performance instrument.

A large-diaphragm element and a true condenser design are combined with superior anti-shock engineering to ensure low handling noise and overall quiet operation that keeps the focus solely on the vocal. This is furthered by excellent protection against plosives and sibilance, and without compromising high-frequency clarity that plagues many models.

The superb sonic design, which is also very rugged to assure dependability on the road, matches the advances found in modern live sound systems in delivering an unparalleled natural, open vocal signature.

TECHNOLOGY FOCUS: *An integral 80 Hz hi-pass filter provides easy switching from a flat frequency response to a low end roll-off. The roll-off position reduces the mic's sensitivity to popping in close vocal use. It also reduces the pickup of low-frequency ambient noise, room reverberation and mechanically-coupled vibrations.*

OF NOTE: *A carefully developed uniform cardioid pick-up pattern virtually eliminates the possibility of feedback. Supplied with AT8470 Quiet-Flex stand clamp.*

KEY SPECIFICATIONS:
Type: Condenser
Polar Pattern: Cardioid
Frequency Response: 20 Hz to 20 kHz
Sensitivity: 10.0 mV/Pa
Impedance: 150 ohms
Dimensions: 7 x 2 inches
Weight: 11.6 ounces

Shure KSM8 Dualdyne
www.shure.com

Type: Dynamic
Polar Pattern: Cardioid
Frequency Response: 40 Hz to 16 kHz
Sensitivity: -51.5 dBV/Pa
Impedance: 300 ohms
Dimensions: 7.4 x 1.9 inches
Weight: 11.6 ounces
Of Note: Patented cartridge design featuring two ultra-thin diaphragms (one active and one passive) and inverted airflow system, advanced Pneumatic Shock Mount rejects handling noise without loss of low-frequency response, capsule available for wireless transmitters

Sennheiser E 965
www.sennheiserusa.com

Type: Condenser
Polar Pattern: Cardioid or Supercardioid
Frequency Response: 40 Hz to 20 kHz
Sensitivity: 7 mV/Pa
Impedance: 1000 ohms
Dimensions: 7.8 x 1.9 inches
Weight: 14 ounces
Of Note: Switchable pick-up pattern, low-cut and pre-attenuation options, large diaphragm

Earthworks SR40V
www.earthworksaudio.com

Type: Condenser
Polar Pattern: Hypercardioid
Frequency Response: 30 Hz to 40 kHz
Sensitivity: 10 mV/Pa
Impedance: 600 ohms
Dimensions: 7.25 x 1.95 in
Weight: 13.3 oz
Of Note: High SPL handling (139 dB), low self-noise, very fast transient response, capsule available for wireless transmitters

CAD Audio D90
www.cadaudio.com

Type: Dynamic
Polar Pattern: Supercardioid
Frequency Response: 50 Hz to 16 kHz
Sensitivity: 2.8 mV/Pa
Impedance: 500 ohms
Dimensions: 7.5 x 2.1 inches
Weight: 18 ounces
Of Note: Proprietary Trueflex diaphragm and PowerGap high-gauss neodymium magnets help produce articulate high output

RWG Spotlight Listing

Audix OM7 | *www.audixusa.com*

The Audix OM7 is a dynamic vocal microphone used by professional sound companies, artists, front of house and mixing engineers, as well as high profile fixed installations. It's known for its ability to provide unprecedented gain-before-feedback on concert level stages, without sacrificing sound quality.

In addition, the OM7 (pictured here with Eddie Vedder) is very resistant to feedback on extremely loud stages and for performers who tend to "cup" the microphone with both hands. In order to achieve these exceptional performance benefits, the mic is designed with an unconventionally low output level that acts as a natural "pad" at the capsule in order to maintain high fidelity at the source.

Well balanced, comfortable to hold and durable, the OM7 is manufactured with very high standards and tight tolerances. It includes a precision die cast zinc alloy body, durable black E-coat finish, dent resistant steel mesh grille, and gold-plated XLR connector.

TECHNOLOGY FOCUS: *A super-tight hypercardioid polar pattern effectively isolates the vocals from the rest of the instruments on stage. With a wide frequency range of 48 Hz to 19 kHz, the OM7 employs a proprietary VLM (Very Low Mass) diaphragm for a very clean and punchy sound with exceptional transient response.*

OF NOTE: *Supplied with heavy-duty MC1 nylon molded clip with brass insert and carrying pouch. Range of accessories includes impedance matching transformer for seamless connection to a high impedance input.*

KEY SPECIFICATIONS:
Type: Dynamic
Polar Pattern: Hypercardioid
Frequency Response: 48 Hz to 19 kHz
Sensitivity: 0.8 mV/Pa
Impedance: 50 ohms
Dimensions: 6.9 x 2.1 inches
Weight: 10.8 ounces

Telefunken M80
www.telefunken-elektroakustik.com

Type: Dynamic
Polar Pattern: Supercardioid
Frequency Response: 30 Hz to 18 kHz
Sensitivity: 1.4 mV/Pa
Impedance: 300 ohms
Dimensions: 7.2 x 1.9 inches
Weight: 13.1 ounces
Of Note: Low mass, thin yet rugged capsule membrane, available in a wide range of finishes and custom colors

DPA Microphones d:facto
www.dpamicrophones.com

Type: Condenser
Polar Pattern: Supercardioid
Frequency Response: 20 Hz to 20 kHz
Sensitivity: 5 mV/Pa
Impedance: 100 ohms
Dimensions: N/A
Weight: N/A
Of Note: High SPL handling (160 dB), three-way pop-protection screen system, adapters to convert mic head to most wireless systems

AKG C535 EB
www.akg.com

Type: Condenser
Polar Pattern: Cardioid
Frequency Response: 20 Hz to 20 kHz
Sensitivity: 7 mV/Pa
Impedance: 200 ohms
Dimensions: 7.2 x 1.8 inches
Weight: 10.6 ounces
Of Note: Gold-plated capsule, switchable pre-attenuation pad, bass-cut filter

beyerdynamic TG V71G
www.beyerdynamic.com

Type: Dynamic
Polar Pattern: Hypercardioid
Frequency Response: 35 Hz to 18 kHz
Sensitivity: 3.2 mV/Pa
Impedance: 420 ohms
Dimensions: 7.3 x 2.1 inches
Weight: 12.2 ounches
Of Note: Progressively damped capsule suspension for reduced handling noise, compensated proximity effect

Electro-Voice ND76
www.electrovoice.com

At the heart of the dynamic models in the new Electro-Voice ND Series is a new large-diaphragm capsule design that takes the technology of the original N/Dym to new levels of sonic performance. Vocalists can select a specific ND Series model to provide optimal results according to venue size and/or stage volume.

The ND76 dynamic cardioid produces very clear and balanced vocal tone, and is particularly effective in small-to-medium sized venues. Specialized capsule tuning ensures that vocals are strong, clear and forward in the room and mix. In addition, the shock-mounting of the capsule virtually eliminates handling noise – no "bumps and thumps" in the PA.

TECHNOLOGY FOCUS: *Considerable attention also went into the humbucking coil to guard against line him. The ND76, and all ND Series models, can be used with confidence near speaker cabinets and EMF-generating equipment racks.*

OF NOTE: *The ND76 is also available with an on/off switch (ND76S). The durable Memraflex tight-mesh grilles are designed to withstand exceptionally rough treatment.*

ADDITIONAL ND MODELS:

- **ND86 dynamic supercardioid** for at large concert and festival-sized venues.
- **ND96 dynamic supercardioid** with tailored frequency curve for vocals on loud stages of any size.
- **ND44 dynamic cardioid** for drums and range of instruments; includes clip-on mount.
- **ND46 dynamic supercardioid** mic for drums and range of instruments; includes locking pivot mechanism.
- **ND68 dynamic supercardioid** specially voiced for kick drum.
- **ND66 small-diaphragm condenser cardioid** for drum overheads, hi0hat, close miking, piano and more.

Neumann KMS 105
www.neumannusa.com

Type: Condenser
Polar Pattern: Supercardioid
Frequency Response: 20 Hz to 20 kHz
Sensitivity: 4.5 mV/Pa
Impedance: 1000 ohms
Dimensions: 7.1 x 1.2 inches
Weight: 10.6 ounces
Of Note: Electronic compensation used to control proximity effect, low self-noise, high-pass filter

Heil Sound PR 35
www.heilsound.com

Type: Dynamic
Polar Pattern: Cardioid
Frequency Response: 40 Hz to 18 kHz
Sensitivity: 2.26 mV/Pa
Impedance: 370 ohms
Dimensions: 7.7 x 2 inches
Weight: 9 ounces
Of Note: Large 1.5-inch shock-mounted hum-bucking voice coil helps ward off unwanted handling noise and electronic interference

Blue Microphones encore 200
www.bluemic.com

Type: Dynamic
Polar Pattern: Cardioid
Frequency Response: 50 Hz to 16 kHz
Sensitivity: 2.25 mV/Pa
Impedance: 25 ohms
Dimensions: 7.3 x 2 inches
Weight: 14.1 ounces
Of Note: Active dynamic phantom power circuit coupled with electronically transformed output provides higher gain and the ability to drive long lines

Audio-Technica AE6100
www.audio-technica.com

Type: Dynamic
Polar Pattern: Hypercardioid
Frequency Response: 60 Hz to 15 kHz
Sensitivity: 1.7 mV/Pa
Impedance: 250 ohms
Dimensions: 7 x 1.9 inches
Weight: 10.9 ounces
Of Note: Superior anti-shock engineering for low handling noise, layered grille and layers of foam for protection against plosives and pops

Countryman H6
www.countryman.com

Style: Headset vocal
Polar Pattern: Omni, cardioid or hypercardioid
Transducer Type: Condenser
Frequency Response: 20 Hz to 20 kHz (omni) / 30 Hz to 15 kHz (directional)
Sensitivity: -43 dB/-54 dB/-63 dB (omni W5, W6, W7)
Maximum SPL: 140 dB (omni W7)
Diameter: 0.1 inch
Weight: .07 ounce
Of Note: Adjustable boom length, supplied with detachable cable, case, cable clips, windscreen and protective caps

Point Source Audio
SERIES8 CO-8WD
www.point-sourceaudio.com

Style: Headset vocal
Polar Pattern: Omnidirectional
Transducer Type: Back Electret Condenser
Frequency Response: 20 Hz to 20 kHz
Sensitivity: -43 dB
Maximum SPL: 148 dB
Weight: 0.6 ounces
Of Note: 360-degree bendable boom, left or right ear wearing, IP57 waterproof rating, interchangeable X-connectors for swapping wireless terminations, available in beige or black

DPA Microphones d:fine
www.dpamicrophones.com

Style: Headworn vocal
Polar Pattern: Both omnidirectional & cardioid available
Transducer Type: Condenser
Frequency Response: 20 Hz to 20 kHz
Sensitivity: -44 dB
Maximum SPL: 144 dB
Diameter: 0.21 inch (omni)
Of Note: Can be worn on either the right or left ear as well as changed from single-ear to dual-ear and vice versa, two boom sizes available, series now offers 4066 and 4088 capsules as well as broadcast models

Audio-Technica
BP894 MicroSet
www.audio-technica.com

Style: Headworn vocal
Polar Pattern: Cardioid
Transducer Type: Condenser
Frequency Response: 20 Hz to 20 kHz
Sensitivity: -49 dB
Maximum SPL: 135 dB
Diameter: 0.11 inch
Weight: .07 ounce
Of Note: Rotating capsule housing with talk-side indicator for use on either ear and optimal polar pattern placement, converts to single or dual-ear mount


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IN MEMORIAM: MRS. ROSE L. SHURE

Mrs. Rose L. Shure, beloved chairman of Shure Incorporated, recently passed away peacefully at her home. From a company statement:

"For more than 60 years, Mrs. Shure has served as an inspiration to all Shure associates, past and present. She was a role model for Shure's Core Values and basic principles, created by her husband and company founder, Sidney N. Shure, that have guided the company.

"The welfare of Shure and its associates was her highest priority. Business integrity, respect, and fair treatment for all were her motivators. She provided a work environment that inspired creativity, fostered pride in making products of the highest quality, and encouraged all Shure associates to reach their personal potential."

In lieu of flowers, donations may be made to the Sidney N. Shure Kehilla Fund, the Jewish Community Center of Chicago, 30 S. Wells Street, Chicago, IL, 60606, or a charity of your personal choice.

Gene Joly has been named vice president of the **QSC Professional Division**. He brings nearly 40 years of sales and executive management experience in pro audio and MI to his new role. Most recently, he was president of Musician's Friend, and has also served on the board of directors of NAMM and the board of trustees of the Berklee College of Music.

"I cannot express how thrilled I am to now be a part of the QSC family. I've always been impressed with every aspect of this company, from the character of the founders, to the leadership of the management, the dedication of the sales and service teams, the outstanding quality and reliability of the products, and their reputation for service and solid relationships with their customers," says Joly.

Allen & Heath has expanded the product management team based at its UK headquarters. **Scott Mason** (pictured above) has been named pro digital sector specialist, where he is supervising and promoting the company's digital product line, including GLD and dLive mixing systems. He has more than 30 years of experience in sound engineering, system design, technical training and pro audio sales, and has worked with companies such as Yamaha, NEXO and Martin Audio.

In addition, **Ian Thomas** has been appointed install sector specialist. He offers a solid background in AV installation system engineering and project management, and for the past eight years has been working for UK-based international system integrator LSI Projects. Thomas is supporting fixed install market integrators and consultants on system design, proposals, testing, and commissioning.

Greg Kirkland has joined the technical sales team at **VUE Audiotechnik**, bringing more than 30 years of experience in sound system design and engineering to the role. He's mixed tours for Liza Minnelli, Andy Williams, Burt Bacharach, Julie Andrews, Rosemary Clooney and Creedence Clearwater Revisited, among others, and as a

EAW'S KENTON FORSYTHE RETIRING

Eastern Acoustic Works has announced the retirement of Kenton Forsythe, a gifted loudspeaker design engineer and an original founder of the company. He first entered the pro audio industry in 1975 with the introduction of the Forsythe Audio SR215 dual 15-inch bass horn, which was the first design of its kind and could fit through a 30-inch door.

His SR109 loudspeaker and BH212 subwoofer were the first products introduced under the EAW name in 1978. By the late 1980s, Forsythe's KF850 had become the number one touring loudspeaker in the world, and he followed this up in the 1990s with the development of multiple loudspeaker solutions for the stadium market – many of which are still in play today.

"One element of Kenton's brilliance is his ability to focus on the application for a product," states Jack Wrightson, partner at leading system design company WJHW. "Everyone wants to design a great loudspeaker. Kenton wants to design a great solution. That's made all the difference in our projects where EAW was able to provide terrific solutions to a problem, not just a great sounding loudspeaker."


CREDIT: ALEX ALEVROYIANNIS

Nitelites has become the first UK company to acquire **RCF's** flagship TTL55-A system, both to provide coverage at large festival sites in addition to scalable options for smaller events. The consignment includes 56 TTL55-A three-way enclosures, 24 TTS56-A dual 21-inch subwoofers, and 24 TTS36-A dual 18-inch subs. In addition, the company added two dozen smaller TTL33-A cabinets to be used for down fill boxes while 16 TTL36-AS will provide further LF extension. Pictured: Andy Magee (left) and Jamie Moore (right) of Nitelites with RCF's Mick Butler and some of the new inventory.

founding partner of Thomas Gregor Associates, designed and programmed audio systems and oversaw production of the company's larger and more complex projects.

"We're extremely excited to have someone with Greg's extensive background to help VUE fully support the larger projects we're working on," says **Ken Berger**, CEO, VUE Audiotechnik. "With his substantial skills in design, production, and management, he will be a key part of growing our relationships with leading acoustical consulting firms throughout the world."

Neutrik USA has announced an exclusive arrangement with **BTX** to sell the XIRIUM PRO product line, including the new wireless digital audio system that's expected to be available in quarter one of this year.

"For several years, BTX has worked on market development and design criteria for XIRIUM PRO in conjunction with Neutrik in the U.S. and Liechtenstein," states **Tom Chudyk**, national sales manager for Neutrik USA. "BTX's CTS-certified field staff will provide excellent support for the product launch, working directly with audio engineers and other Neutrik partners to fully explain and demonstrate the product."

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TUNING TIME

Identifying your particular type when it comes to checking a PA.

by Craig Leerman

There are many different ways of checking PA systems. Some of us play music and walk around the venue, while others use sweeps or pink noise and deploy on a measurement program. Many do a combination of both. Regardless, at some point in the process, we usually grab a microphone and listen to the PA with our own voice. Witnessing hundreds (thousands?) of these voice checks over the years has led me to compile the following "types." Which one do you resemble?

The Assistant Principal: "Tap-Tap-Tap, Is this thing on?" Also called the Rotary Club Presenter, and inevitably followed by, "Can you hear me in the back?"

The Singer: "Feelings, nothing more than feelings." You're a frustrated karaoke star and can't help but break into song when a mic is in your hand.

The Rapper: "I like big bass and I can't deny…" You like to bust out some rhymes whenever you get a chance. Bonus points if you cup the mic.

The Dad: "Microphone test channel one, this is a microphone test." This is how my dad started every recording on his cassette tape recorder. It only works with gear built in the 1970s.

The Drive-Thru: "Do you want fries with that?" You're hinting at a previous career in an illustrious field. The only question: why have you fallen so far?

The Not-So-Sophisticated: "Hey, watch this! Ow, oh, arghh…" Didn't Jeff Foxworthy base an entire comedy routine on your life?

The Willie Mays: "Hey, Hey, Hey." Unlike the real Say Hey Kid, you have not hit 660 home runs, you do not possess a career batting average of .302, and there is no plaque dedicated to you in the Major League Baseball Hall Of Fame. So please stop saying "Hey."

The Tom Hanks: "Sibilance, Sibilance, Sibilance." We all love Tom Hanks in *Wayne's World* as Aerosmith's roadie, but the only time you're allowed to say "sibilance" more than twice is when you're tying the scarves onto the lead singer's mic stand.

The Joker: "A horse walks into a bar. 'Why the long face?' asks the bartender." You crack yourself up telling jokes on the mic. The key word here being yourself.

The Whistler: "Tweet, Tweet." When I was a young soundman, The Old Soundman told me to never whistle in a mic. Now that I'm an old soundman myself, I'm telling you.

The Checker: "Check, Check, Checking." Nothing but checks. Hey, I'll pick up the dinner check if you stop saying "check." I swear, the check's already in the mail. Go check.

The Tester: "Test, Test, Test." Sometimes you might mix it up and really put it out there by swapping in the word "testing." How about swapping in the word "annoying" instead?

The Fence Sitter: "Test, Check – Test, Check." You're perpetually undecided about which word to use, so you go with both. At least it's better to be sitting on the fence than thrown under the (tour) bus.

The Counter: "One, Two – One, Two." Come on, everyone, let's say it together: Never count to three, because on three you lift.

The Professional: "Test, Check – One, Two." Combining the best of both worlds. You're thorough but making it look easy, following the mantra of never letting them see you sweat.

The Overachiever: "Tap-Tap, Test, Check, One, Two, Hey, Hey, Huh, Yo, P, B." You're a true audio person of the world, multilingual in the art of PA checks and not afraid to use it. Just be careful: mixing physical action, words, numbers and random consonants is not for the faint of heart, but you know it's the only way to true PA excellence. Bonus points for using an SM58 exclusively! Double bonus points if you play Steely Dan tracks through the system as well! LSI

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