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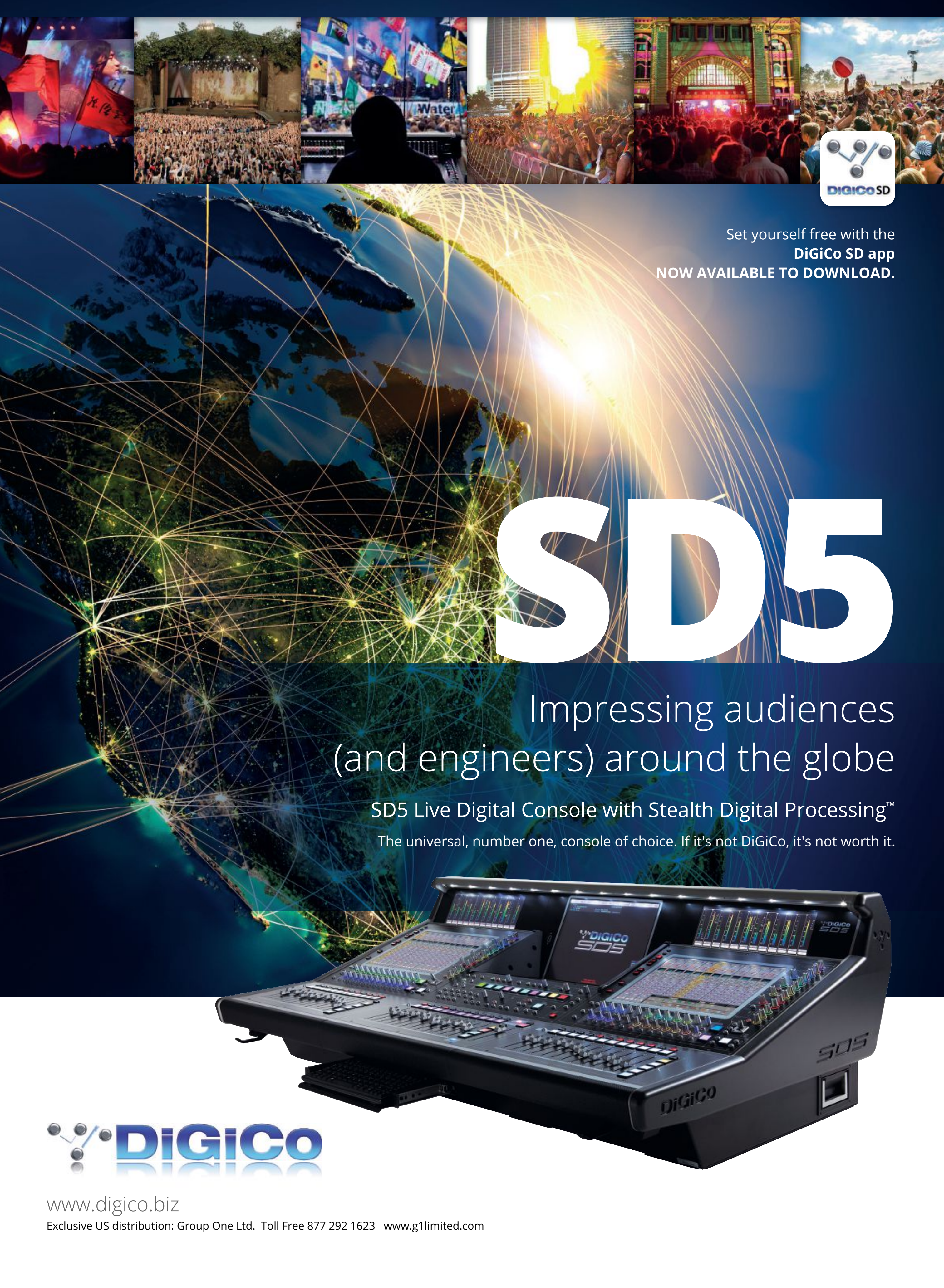
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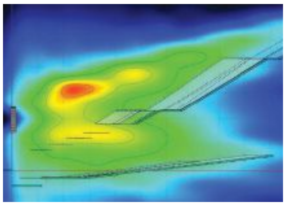

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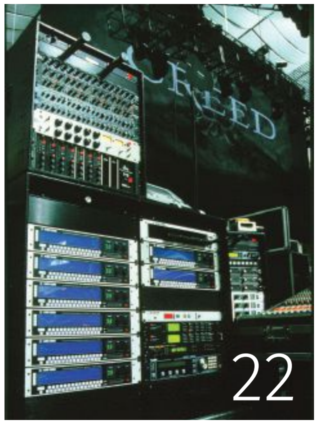

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Pat Guertin, Marie-Mai (FOH)

Rich Steeb, Blue Rodeo (FOH)

Duke Foster, Blue Rodeo (monitors)

Myles Hale, Black Sabbath (monitors)

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From the Editor's Desk

THE COVER STORY for this issue ties into an attraction at the location of this year's InfoComm show in June, and in addition to that connection, it also exemplifies the technological changes happening at rapid pace in pro audio. I recall having the opportunity to take a "behind the scenes" tour of the audio system for the then-new Fremont Street Experience in Las Vegas shortly

after it opened in the mid-1990s, and it was an impressive endeavor.

Fast forward to the present, and it's even more so. Racks and racks of analog gear have been replaced with a minute selection of digital, networked devices that were merely the subject of system designer dreams 20 years ago. Ditto the loudspeakers that are self-powered and a whole lot more. It's an interesting time to be working in this market, particularly against the backdrop of what was "state of the art" just a relatively short time ago.

Elsewhere in the issue, Mark Frink has provided an excellent primer on several key factors of the technical rider, while Ike Zimbel talks about ways to effectively work with RF coordinators, an increasingly important aspect of work in live sound reinforcement.

In addition, Mike Sokol contributes another significant discussion on electrical safety, something that should be a priority in the minds of everyone working with systems. We're also pleased to present part 2 of Bob McCarthy's look at the evolution of system optimization, this time focusing on equalization, and in a way, an interesting companion to our cover story.

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

Keith Clark

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

25th ANNIVERSARY

ON THE COVER: An RCF line array flying at one of the performance stages at Fremont Street Experience in Las Vegas. *(Photo by Paul Natkin)*

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

PUBLISHER Kevin McPherson, kmcpherson@ehpub.com
EDITOR-IN-CHIEF Keith Clark, kclark@livesoundint.com
SENIOR EDITOR M. Erik Matlock, ematlock@livesoundint.com
SENIOR CONTRIBUTING EDITOR Craig Leerman
cleerman@livesoundint.com
CHURCH SOUND EDITOR Mike Sessler
msessler@livesoundint.com
TECHNICAL CONSULTANT Pat Brown, pbrown@synaudcon.com
ART DIRECTOR Katie Stockham, kstockham@ehpub.com
CONTRIBUTORS: Andy Coules | Kevin Young | Mark Frink
Jonah Altrove | Mike Sokol | Bob McCarthy | Greg DeTogne
Bruce Main | Ike Zimbel

ProSoundWeb.com
EDITOR-IN-CHIEF Keith Clark, kclark@prosoundweb.com
SENIOR EDITOR M. Erik Matlock, ematlock@livesoundint.com
PRODUCT SPECIALIST Craig Leerman,
cleerman@prosoundweb.com
WEBMASTER Guy Caiola, gcaiola@ehpub.com

ASSOCIATE PUBLISHER Jeffrey Turner
jturner@livesoundint.com | 415.455.8301 | Fax: 801.640.1731
ASSOCIATE PUBLISHER ONLINE Mark Shemet
mshemet@prosoundweb.com | 603.532.4608 | Fax: 603.532.5855
AD PRODUCTION DIRECTOR Manuela Rosengard
mrosengard@ehpub.com | 508.663.1500 x226
AD PRODUCTION MANAGER Jason Litchfield
jlitchfield@ehpub.com | 508.663.1500 x252
CLIENT SERVICES MANAGER Jeffrey Miller
jmiller@ehpub.com | 508.663.1500 x253

Circulation and Customer Service inquiries should be made to:
Live Sound Customer Service
EH PUBLISHING
Phone: 800-375-8015, ext 294
(Outside the U.S.: 508.663.1500 x294)
Fax: 508.663.1599
customerservice@livesoundint.com
111 Speen Street, Suite 200
Framingham, MA 01701

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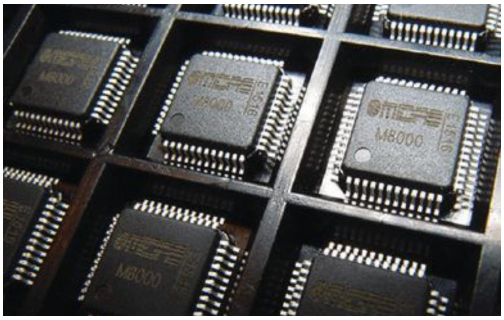

Midas M8000

A 192 kHz, 8-channel A/D converter chip for Midas mixing consoles. Based on multi-bit delta-sigma architecture and a high-speed TDM interface, the converter is rated to deliver close to 120 dB of dynamic range with sampling rates up to 216 kHz. It is designed to capture the nuance of the console microphone preamps and convert that performance into the digital domain. Midas will begin converting its consoles to the new chip by the end of 2016, and may also offer an upgrade kit to existing customers. ***www.music-group.com/brand/midas/home***

JBL Professional Control SB2210

A dual 10-inch subwoofer that combines high output and compact size for applications in the install market and to serve as a complement to recently updated Control Contractor surface-mount loudspeakers. Power capacity has been increased to 500 watts, and it is also stated to have a flatter frequency response with an improved bass extension to 38 Hz. The newly designed 10-inch drivers have woven fiberglass cones to enhance performance and a highly damped butyl rubber surround and linear suspension spider to lower distortion. Installation options include floor placement, wall or ceiling attachment via an optional U-bracket, or suspension via 13 included M6 insert points. The cabinet is available in black or white and is paintable.***www.jblpro.com***

Radial Engineering mPress & Exo-Pod

A press distribution system configured in two parts – a master host (mPress) and external slaves (Exo-Pods). mPress is housed in a standard 1RU enclosure and offers two microphone inputs, both with a variable level control high-pass filter. For podium mics, 48-volt phantom power is available and activated using a recessed front panel switch. Selecting between mic channels is also done via front panel switches. Two knob limiters provide threshold and release. In addition, mPress is equipped with 1/4-inch, RCA and 3.5 mm stereo inputs along with a separate level control. There are eight balanced outputs, two on the front panel and six on the rear. Each can be configured using a recessed switch for mic level or high-output drive to feed an Exo-Pod. This allows the mPress to be used as a 2 x 8 distro for smaller events or expanded by adding Exo-Pods on the outputs. The Exo-Pod is a passive floor box with an XLR input, a thru-put for expansion, a local level control, plus 10 XLR mic outputs and four 3.5 mm mini TRS outs. With eight Exo-Pods, signal can be distributed to as many as 112 users. ***www.radialeng.com***

Verity Audio AMBIENCE

A column loudspeaker series with three full-range models and a subwoofer. The COL4.3, COL8.3 and COL16.3 columns are composed of four, eight and sixteen 3-inch neodymium drivers, respectively. They offer output with a narrow vertical angle, helping to reduce ceiling reflections. The COL8.3 and COL16.3 also have vertical aperture angles of the sound beam that can be selected by means of a slide switch for more precise focus on the audience (10 to 20 degrees for the COL8.3 and 15 to 25 degrees for the COL16.3). The COL4.3 vertical aperture angle is fixed at 20 degrees. The SUB210S subwoofers is loaded with dual 10-inch cone drivers. The symmetrical positioning of the drivers and reflex ports on the bottom side of the cabinet helps cancel vibration and distortion. All of the enclosures are made of MDF wood. ***www.verityaudio.fr***

Fulcrum Acoustic FLS115

A subcardioid subwoofer incorporating proprietary Passive Cardioid Technology designed to reduce excessive rear low-frequency radiation without the need for additional drivers, amplifiers, or signal processing. As with the companion FL283 subcardioid line array, the FLS115 enclosure and rigging can accommodate up to 20 degrees of splay between adjacent cabinets, providing for more sharply curved arrays. Specially designed acoustical circuitry, combined with Fulcrum's TQ processing, enhance pattern control and transient response, even at high SPLs. ***http://fulcrum-acoustic.com***

Soundcraft Ui Series Firmware

An upgrade for Ui Series digital mixers that adds several new features and extends browser and device compatibility. Most notably, the software now supports Microsoft's latest Edge browser. In addition, the Ui12 now includes stereo recording capability previously found only on the Ui16. Additional functions include a more secure password-protected Access Limitation System, a new sync ID that allows users to maintain channel sync across multiple browser windows (on different displays and devices), MoreMe portrait on/off mode, enable/disable SSID broadcast, channel mute that now mutes auxes in PFL mode, new global pre-fade aux send point selection, and AFS2 preset manager. The free software update is available for download from the company website. ***www.soundcraft.com***

WorxAudio Technologies PDA-1000R

A power amplifier with PreSonus Active Integration (AI) technology to create networkable Dante-enabled loudspeaker systems with DSP that can be controlled by a computer running WorxControl (a loudspeaker management and remote control/monitoring application) over a standard LAN. The amp can be rack-mounted or housed in loudspeakers. The 2-channel, class D power unit is rated to deliver 500 watts per channel and incorporates two onboard presets, high-pass filter, temperature, signal, -3 dB, limit and clip indications, as well as XLR input, XLR pass-thru, and Dante networking. WorxControl provides a suite of editing controls to compensate for room anomalies, create delay systems, eliminate feedback, and more. Key aspects of WorxControl include an 800 ms alignment delay adjustable in 0.1 ms increments. There's also a limiter with fully variable threshold to control dynamics without a mixer or an output processor. ***www.worxaudio.com***

FBT PROMAXX

A range of portable loudspeakers available in active self-powered or passive versions, with both types incorporating custom transducers in bass reflex polypropylene enclosures. Active models include the PROMAXX 110A (10-inch), 112A (12-inch) and 114A (14-inch). The high-frequency section has a 1-inch-exit B&C Speakers compression driver (1.4-inch-exit in the 114A) feeding a 90- by 60-degree (h x v) constant directivity horn. All models also have stage monitoring cabinet angles. The amplifier module, with switch-mode power supply, is rated to deliver 700 watts (via class D topology) to the LF and 200 watts (via class H/AB topology) to the HF. A GUI on the rear of the enclosure provides access to all DSP. The menu-driven approach is navigated using a single rotary control with push-to-select functionality. Cabinets include a pole-mount socket, M10 suspension points, and a wall bracket plate. ***www.fbtusa.com***

Peavey RBN Series

Includes the RBN 112 full-range loudspeaker and RBN 215 subwoofer, both self-powered. The RBN 112 is driven by a power amplifier rated to deliver 1,500 watts. It includes a proprietary 12-inch dual-voice-coil Scorpion cone driver with field-replaceable basket as well as a proprietary 120 mm ribbon high-frequency driver on a low-coloration waveguide. Also included are XLR/1/4-inch combo inputs, a 3.5 mm input, and a mic/line level selection. Each input offers digital infrasonic high-pass filters, a 9-band graphic EQ, and delay, while each output has a compressor/limiter and fourth-order high-pass/low-pass filters for crossover function and external subwoofer incorporation. The RBN 215 is rated to provide full output down to 36 Hz, with dual 15-inch woofers driven by a 2,000-watt (peak) amplifier. It also has an assignable crossover. Both models offer a pole mount socket, and there are several other mounting options for the RBN 112. ***www.peavey.com***

Clair Brothers 8CX & 5CX

Compact coaxial loudspeakers with 8-inch (8CX) and 5-inch (5CX) neodymium transducers. The 8CX offers a pair of options in horizontal dispersion angles: a 100-degree conic horn for near field coverage applications and a 70-degree conic constant directivity horn for applications requiring tighter coverage and increased projection. The 5CX has a 70-degree conic constant directivity horn. Both models offer an integrated pole mount and rigging points, along with a variety of other mounting options. ***www.clairbrothers.com***

LOST IN TRANSLATION?

Pro audio has a language all its own.

by Jonah Altrove

I first started reading *Live Sound International* during high school (literally, *during* school), and I distinctly remember feeling frustrated by the professionals' jargon. To that point, the sum total of my experience was "guess/test/revise" with whatever gear my school had lying around. I understood some concepts, but not the terms for them. I knew how to ring out a monitor, but I had no idea that process was called "ringing out." I just knew it was a thing you were supposed to do with monitors.

Flipping through my first issue of *LSI*, which I found in a crate in the school's tech booth, had me feeling as though there was a whole wealth of knowledge just beyond my grasp. I could understand the concepts if I could only understand the language.

As professionals, we often don't realize how terminology-laden our dialogue is until a lay person points it out. ("What is polarity/XLR/HPF/PFL?"). When we hear "58" our brains sort of "autocomplete" and pull up a mental database entry for "Shure SM58, a cardioid dynamic handheld vocal microphone," which is a big concept to communicate just by saying "58." If we could visually externalize this mental "unpacking" process, I imagine it would look a bit like the Heads-Up Display inside the Terminator or Iron Man's helmet. Usually, though, this happens so fast we hardly notice it.

With such "information density," a small error can become a big error, so precision is required. A simple typo can turn a spec for a D5 digital console into a D6 kick drum mic, or an S16 digital snake into an S10 loudspeaker or a Qu-16 mixer. Clearly, a high degree of accuracy is required in our technical language. (This extends to numbers, too – a misplaced zero or decimal point on an invoice or rigging weight estimate can lead to a big problem.)

GREETINGS & SALUTATIONS

Engineers who can rattle off specs and model numbers are said to be "speaking our language." What I'm more interested in is the literal sense of it: what if another sound engineer is not speaking our actual language?

I experienced this first hand while working as a house engineer for a college. As part of a cultural programming series, the college was bringing in a traditional Japanese Taiko drumming and dance production. The troupe consisted of about 40 Japanese college students who had flown to the U.S., accompanied by an instructor who was also their director, production manager, and the only member of the group who spoke English. He introduced me to Suki, the troupe's sound engineer, who greeted me with a smile and formal bow, which I attempted to return. I must have looked ridiculous because she laughed and held out her hand for shaking. Introductions made, the director departed to work with our lighting crew.

There's a wonderful harmony here because dance gigs are pretty unique in terms of their audio requirements,

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so they can seem a bit foreign to engineers who haven't worked in that capacity before. For example, it's primarily a visual medium so lighting is paramount, and hours are spent tweaking, focusing, and programming. Audio is largely a support role: usually playback only, so my job was to ensure that everyone could hear what they were supposed to hear, at the proper volume.

My mantra for dance gigs is "stay out of the way." This applies not only to my work process (I stagger my meal breaks with the rest of the production, so I have a couple of "empty-room" periods in which to do what I need to do), but also to my setup itself: absolutely clean stages, no visible loudspeakers or cables, no gear blocking the wings where the dancers enter and exit, and I'm usually not even visible to the audience myself. I work from the lighting booth, since I have no live mics and need to call cues for the rest of the crew based on the playback timer. At the booth I can also communicate freely with the lighting team without distracting the audience by hollering into a headset.

We both shared a passion for music and sound, a passion driven by what is heard, not what is spoken.

Suki had a 2U rack of solid state media players (bonus point to Suki – I don't trust CDs, either.) So I helped her patch into the console and built a custom fader layer so that she could run the show without banking to the output layer. (For dance, I like to have easy access to the monitor bus master fader, so I can pull a fade-out without having to run the monitors post-fader.) Suki borrowed my board tape and labeled her faders in Japanese (cool!).

WORKING TOGETHER

Down on the stage, we looked at placement options for the side fills. This is a tricky aspect of dance audio – you can't obstruct the wings where the dancers pass, you can't block the lighting booms or the sides of the stage, and it's also unlikely they can be flown because the "pipe end" or "high side" position is prime lighting real estate.

We decided on a placement behind the first set of legs, and Suki asked through gestures if it would be possible to add another pair of fills further upstage. I nodded, pointed to my watch and held up two fingers – "Give me two minutes."

I went to a storage room to get extra wedges and cabling, and when I turned around, Suki was there with two dancers to carry the gear.

Normally I'd feel like a slacker if the talent is on stage carting wedges around, but these guys and gals were adamant about helping as much as possible. They placed the wedges, bowed, and hurried off. I cabled up the wedges while Suki applied a liberal application of white gaffers tape for high visibility in the dark wings.

Side fills sorted, I hovered in the background while Suki tested playback levels and recorded them on a notepad to the tenth of a dB. I'd periodically give an "everything good?" gesture, and she'd respond with a thumbs-up. She needed her director to translate only once: a question about gain structure and noise floor in the main amp rack. Good luck communicating that with hand gestures! After the show, the whole group signed a T-shirt for me.

COMMON EXPERIENCE

The reason I share this is the same reason I remember it so well: Suki and I didn't understand a word of each other's language, but we had absolutely no trouble communicating. Signal flow is signal flow, decibels are decibels, and amplifiers are amplifiers – no matter your word for them. As Shakespeare might have said if he were a sound engineer, "A system matrix processor by any other name…"

This is quite silly, but what's cool is that Suki and I actually did have a language in common, the language of pro audio. We both shared a passion for music and sound, a passion driven by what is heard, not what is spoken. This is astounding to me when I stop and think about it – live audio gave us so much common ground that we literally did not have to speak to each other to put on a show together.

And that, I think, speaks for itself. LSI

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

SITUATIONAL AWARENESS, TAKE 2

An additional set of common live mix mistakes and solutions.

by Andy Coules

In a continuation of my previous article (*LSI April 2016*), I'm taking a look at five more of the most common live mix mistakes I've encountered when attending gigs as an audience member and offering some potential solutions to these everyday issues.

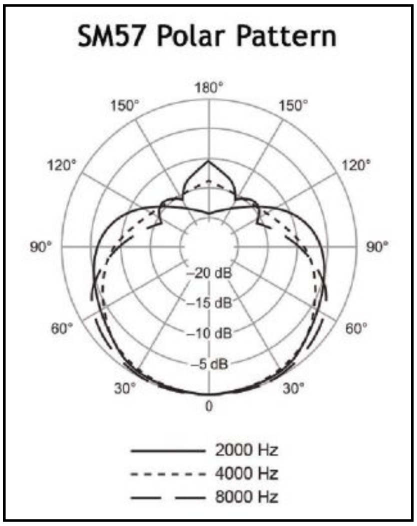
Excuse me while I kiss this guy. It still surprises me when I go to gigs where either the vocals can't be heard or they're so buried in the mix that it's impossible to make out what they're saying. The vast majority of music performed by bands at gigs is song-based, and arguably the most important element of song-based music is the lyrics. So if they can't clearly be heard above the band then the full artistic vision of the artists is not being adequately conveyed – and that's often the fault of mix engineers.

We all know that there's a limited amount of gain that can be applied to any given microphone before feedback is inevitable and there's no magic solution to this issue, but there are a number of things that can improve the chances of getting the vocal heard.

The battle against feedback starts at the very beginning. The positioning of the loudspeakers and the initial reflections can have a huge bearing on this issue, so try to keep microphones behind the main loudspeakers and exploit the polar patterns of the mics to best position any monitors.

The tuning of the PA can also help you out. I always ring out the system using the vocal mic placed in the most likely position it will be in during the show, but if I anticipate a particularly troublesome room, then I EQ more extremely than a properly treated room (that has decent separation between the stage, the room and the PA).

If you're doing monitors from front of house, try cloning the main vocal (using either a Y lead or via soft patching) and treat the copy sent to the monitor with less EQ and compression than the copy that's going to the PA. Another classic trick is to invert the polarity of the vocalist's monitor send, as this will invert any standing waves and might just shift a troublesome frequency out of the mic's path.



or two. So you have a bright idea: "What if I take the stand on the guitar amp mic and wrap the cable round the handle so the mic dangles in front of the amp? Brilliant!"

To find out why this is a bad idea, you only need to look at the polar pattern of most mics commonly used on guitar amps. Any signal entering the side of a Shure SM57, for instance, is going to be 6 dB quieter than if it were the same distance from the end of the mic – and that's at 2 kHz. As the frequency increases, the sensitivity drops off to -10 dB (at 4 to 8 kHz). Therefore you're getting a weaker signal which means inevitably having to

The whole process took about 20 minutes and never got resolved, so the band played on without their trusty drum samples.

You're miking up the floor! This is a common issue and most of us (me included) have been guilty of perpetrating it at some time. The situation is as follows: you have limited resources and the band just turned up with a full horn section that no one warned you about and now you're short a mic stand

increase the gain and thus increasing the risk of feedback.

In addition, you're getting a less consistent frequency response because the end of the mic (i.e., where the signal should be) is pointing down at the floor, so in effect you're miking up the floor. I once saw an engineer do this while the guitar amp

was sitting on top of the bass amp and he couldn't figure out why the bass was still so loud, even when the bass channel was off!

The engineer's best friend can help us out when we're short of mic stands; yes, I'm talking about gaffer tape. Drummers often have too many cymbal stands so they can easily be pressed into service as impromptu mic stands, and I've even taped mics to chairs in the past – it might not look pretty but it sounds much better than miking up the floor.

All I can hear is guitar. This ties in with my initial point, as one of the most common causes of not being able to hear the vocals is that the guitars are too loud. Guitarists are a key part of most genres of music, and it takes a certain amount of skill and practice to get to the point where they can get up on that stage and show the world what you can do, but unfortunately, it also takes a fair amount of ego. Guitarists like to be loud not just so they can hear all of the intricacies of their playing but also because they *like to be loud*.

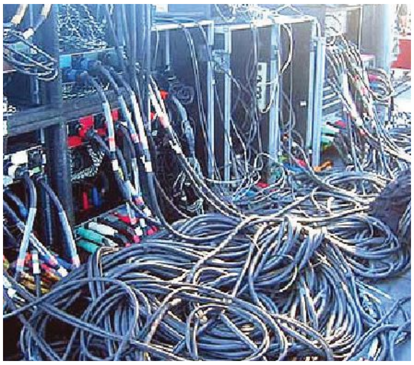
I once worked with a guitarist who had a very expensive amplifier that he cranked up so loud that you couldn't hear anything else – not the drums, the bass or any of the other instruments; the mix was just guitar. So I asked him nicely to turn it down and he proceeded to explain why this particular amp only sounded good when the level was above a certain point. I tried to explain to him that at those levels all the audience will hear is the guitar and I swear he smiled. (That was clearly not the correct logic to use on a guitarist.)

This particular problem gets worse the smaller the venue. Not only can the guitar dominate the mix if the amp is too loud, but the directional nature of guitar amplifiers means the mix can be drastically different depending on whether you're standing in front of it or off to the side.

I've even taped mics to chairs in the past – it might not look pretty but it sounds much better than miking up the floor.

The best way to avoid the problem is to work with the guitarist during sound check to get the appropriate levels. One of the most common causes is guitarists not being able to hear themselves, so I always raise up the amp and make sure it's pointing at the guitarist because no one has ears in the backs of their knees. If that doesn't help, I make sure they get plenty of level in their monitor – you can actually get a guitarist to turn down their amp by increasing the level in the monitors (although it doesn't always work). Many musicians aren't even sure of the appropriate level for their amps on stage, so I always advise them in small venues to set their amps at a level so they can hear themselves clearly over the drums, but no more, and then leave the rest to me.

If, however, you're faced with the kind of guitarist who refuses to turn down the amp because it compromises the purity of it's tone, then it's time to introduce him to the concept of a power soak.



Terrible troubleshooting. I was at a gig where the band was about to go on and the customary line checks were being performed. It quickly became apparent even to the unknowledgeable onlooker that the signal from the SPD drum sampler was not reaching the desk.

The engineer proceeded to kick the DI box, pull the XLR cable out and put it back in again, prod the SPD, fiddle with some controls, and generally stand and stare with a puzzled look on his face, all this while repeatedly returning to the desk to see if the signal had magically re-appeared. The whole process took about 20 minutes and never got resolved, so the band played on without their trusty drum samples.

Basic troubleshooting skills are a vital part of sound engineering and can mean the difference between a smooth-running night that everyone involved enjoys and the kind of hellish experiences that get discussed in hushed tones wherever engineers gather.

There are two distinctly different approaches to troubleshooting, the choice of which depends on whether it's pre-show or in-show mode. If in pre-show mode, such as sound check, you can rather leisurely troubleshoot any issue by logically following the signal chain and swapping out consecutive components until you're able to isolate and replace the faulty item.

However, if you're in show mode there's no time to methodically inspect the signal chain, so the best option is to duplicate it. Plug a new XLR into a different channel on the multi-core which goes to a different channel on the console, and plug that into a different mic or DI, swapping out the whole lot in one fell swoop. Nine times out of 10 it will work perfectly well and you can just carry on; if it's an analog desk you'll need to copy across the channel processing, but if it's digital you should be able to soft patch the input so the new signal comes up where the old signal should have been.

Too much track. This isn't technically a mistake but something of a personal bugbear that I think we should all be a little wary of. More and more bands are using backing tracks live, partly due to the fact that it's so much easier to take a laptop on stage, and also because it cuts down on musician costs and enables bands to produce a bigger sound on a small budget.

However, we should never forget that the audience is there because they've (usually) paid to witness and participate in a live event. Few people turn up expecting to hear the music rendered note perfectly – if that was the case then most would stay at home and listen to the album on their hi-fi (and avoid all of the overpriced beer).

So when bands deploy too much track featuring ghost choirs and invisible horn sections, I find it detracts somewhat from the live experience. It also breaks a kind of unspoken agreement whereby we don't mind if they use track elements to enhance the show as long as it doesn't overshadow the live dynamic.

It's a fine line and I'm aware that not everyone will agree with me that too much track is a bad thing, but I do have a little advice for how track elements should be mixed. The key is to get the live elements sorted first, get the band rocking along, and then add the track elements subtly to support and enhance the band. If you do it right, most people in the audience won't even realize that backing tracks are being used – they'll just think that the musicians on stage are incredibly talented to create such a big sound.

I hope that my observations have helped not only identify some of the most common mix mistakes, but also provide some useful ways to potentially deal with them. Remember that to err is human but to not learn from your mistakes, or those of others, is downright foolish. LSI

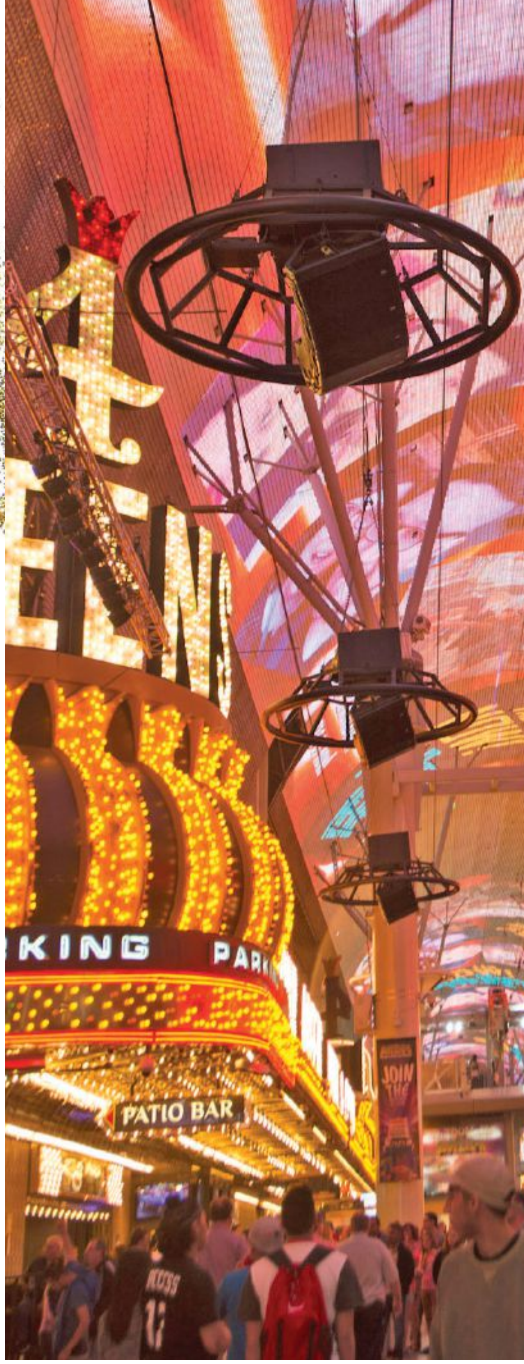
Andy Coules *(andycoules.co.uk) is a sound engineer and audio educator who has toured the world with a diverse array of acts in a wide range of genres.*

Cover Story

VIVA LAS VEGAS

Latest upgrade brings new audio to Fremont Street Experience.

by Gregory A. DeTogne, photos by Paul Natkin



FREMONT STREET is where it all really started in Las Vegas. As the site of many firsts for both the city and Nevada, consider that within an area of five fabled downtown blocks, the city's first hotel was built (the Hotel Nevada in 1906), Nevada's first gaming license was issued (to the Northern Club), and the town's first high-rise took shape (the Fremont Hotel in 1956).

By 1992, however, Fremont Street was no longer leading the way, a fact underscored by Las Vegas' casino market, 80 percent of which had moved onto The Strip. Downtown was becoming downtrodden, and as the centerpiece among ideas brought forth by a group of the street's hotel and casino operators, the Fremont Street Experience was born to help revitalize the district.

Completed in 1995, the pedestrian mall and attraction built upon a show concept revolving around a barrel vault canopy rising some 90 feet high above street icons such as the Vegas Vic neon cowboy sign and an area known as Glitter Gulch. Extending four blocks from the foot of The Plaza Hotel on Main Street to Fourth Street on its eastern edge, the canopy – or Viva Vision Light Show as it's known officially – offers visitors an eyeful of imagery extending well beyond even this town's exaggerated sense of mind-bending dazzle and flash. With each light show running about six minutes, typically from dusk to midnight depending upon the season, the canopy now draws upon 12.5 million energy-efficient LED lamps for illumination.

STREET LINED WITH SOUND

Equally impressive in its own right, audio accompanying the show is concert quality, and just like everything else in Vegas, has constantly changed and reinvented itself with an eye on making steady improvements over the years. With permanent stages now numbering three in total, added beginning in the early 2000s to eliminate the need to bring in temporary staging for special events, the mall sound system was first upgraded in 2001 and is now undergoing another complete renovation.

When it's finished in September, a total of 50 new clusters will line the street from above, providing sound with the aid of HDL 20-A full-range cabinets from RCF USA, along with the company's SUB 8005-AS subwoofers. The stages will get a sonic redux as well, once again using an assortment of line array, subwoofer, and monitoring technologies supplied by RCF.

An unobtrusive oasis in this sea of sensory overload, the control room for the Fremont Street Experience lighting and audio systems looks out toward the SlotZilla Zip Line tower a mere 50 feet away. Staring out at the Zoomline, which transports riders Superman-style in a prone, flying position starting at 114 feet up all the way down the length of the Viva Vision canopy, Fremont Street Experience production manager Joe Pizzo took the ride himself before the attraction was open to the public.

"I went down the Zoomline while one of the canopy shows was actually going on," Pizzo recalls. "I was like four feet from the lights. There's a nice view of the audio

Looking down (and up) at Fremont Street, where sound to accompany the overhead video program is provided by groups of RCF HDL 20-A arrays.

Above, Fremont Street Experience production manager Joe Pizzo on site, and below, a closer look at one of the 50 distributed clusters, innovatively utilizing locations and hardware from the previous system.

clusters from up there, I'll say that. You gain a unique perspective of the area of coverage you're dealing with. Honestly, it's a weird environment for the amount of audio we have. So many nooks, crannies, reflective surfaces…"

Joining Pizzo on staff at the Fremont Street Experience to spearhead the project are entertainment operations director Len Turner, manager of entertainment operations Steve Winchester and maintenance manager Ron Ischer, who is currently doing double-duty installing the clusters and taking care of the video canopy. As of this writing, the audio upgrade is proceeding apace with four clusters being swapped out every two weeks, a process that began in March.

"We'll be done by September 1," Pizzo says. "We were originally going to do this in 12 to 18 months, but once the powers that be heard the first four new clusters, they told us to go at it full-throttle and be done in six. We're cruising now with the install. It's going to be a good summer."

PAVING THE WAY

The need for the upgrade was initiated by age-related issues bearing down on the existing system, which was beginning to deteriorate notably under the outdoor stress of the harsh, unforgiving desert environment. Larry Hall of H.A.S. Productions (Las Vegas) served as the impetus for the project when he delivered 24 RCF boxes to the Main and First Street stages for installation.

"Larry has been a friend and a vendor of ours for years," Pizzo notes. "Everyone loved the sound of the RCF boxes, and since we were in the process of sending a bid out for the upgrade, he helped us to find out if they were interested in participating in a demo."

With some able direction coming from RCF USA national sales manager Tarik Solangi, the company put a demo together in front of the Main Street stage that rocked the mall and paved the way for the installation of the new RCF-based clusters. As they are being installed right now, each cluster goes airborne using modified versions of the existing framework and rigging components built for its predecessors. A half-dozen RCF HDL 20-A enclosures are incorporated into each along with a single SUB 8005-AS subwoofer.

New sonic life is being injected into the three stages as well. When all is said and done, HDL 20-As will be on the Main and First Street stages, and TTL55 components will be on the Third Street stage. Subs on the stages include RCF TTL56s, AS-8006s and TTL36s; wedges are TT25-CXA models. DriveRack Venu360 boxes from dbx help orchestrate events on each riser, while mixing is a task given over to Soundcraft Si Performer 2 consoles obtaining signals with the aid of Soundcraft 32 x 8 MADI format compact digital stage boxes.

"We first put HDL 20-As into our First Street stage three years ago," Pizzo relates. "We've thrown everything we have at them since then seven nights per week, 365 days per year, out there essentially hanging in the open unprotected, and they just keep going. Combine this with the fact that everyone loves the sound, and you'll understand why we chose RCF for the rest of the mall as well."

Being self-powered, the presence of the HDL 20-As in the 50 mall clusters have definitively brought to a close the old days on the street of crowded amp rooms and long loudspeaker cable runs. Within the previous system, 12-gauge wiring fed the clusters from conduit extending down to each cluster from the canopy. In an artful act of repurposing, Pizzo and the installation crew found it to be an easy enough task to pull back on that 12-gauge wiring and simply drop in the new 30-amp lines the design required.

"We have super-clean power up there right now," Pizzo notes. "And we wired it right in. We were able to wire a backup circuit in too, just in case. But we don't push the mall speakers that hard, at least not like at the stages, where we post 106 dB-SPL levels C-weighted for cover bands on any given night, and 105 A-weighted for headliners. That's loud."

SEPARATE AND TOGETHER

There are nights when headlining acts will play back-to-back on each of the stages. In a general sense, audio is usually compartmentalized to remain in the areas serving each of the stages, but there are other times when a show on the Main Stage, for example, will be broadcast visually down the entire canopy and the audio will follow suit a good portion of the way down the mall too.

Another view of some of the loudspeaker sets inside Fremont Street Experience, and inset, Larry Hall of H.A.S. Productions, who was an impetus for the project.

To meet the challenges of controlling and distributing the audio so that it arrives where and when it's supposed to and stays out of the areas it's not intended to serve, a relatively simple blueprint was developed using a front end system based around a TiMax SoundHub. Working in conjunction with a fiber-fed network of BSS London BLU-806 processors numbering five in total – one in the control room and four running down the length of the mall two per side – the TiMax SoundHub manages all playback and distribution, in effect telling the BSS boxes which cluster will get what levels at any given moment, and additionally providing the clusters with functions like delay when program material emanates from the stage. Audio is sent to and from the stage using BSS London BLU-326 processing.

"The way we set it up, it's almost like a theatre application," Pizzo says, giving the blueprint better illustration. "We wanted to make the system fully functional, yet keep it elegantly simple. On this level we've succeeded quite well. If you look around the control room, there's really not a lot of space dedicated to audio. It's 99 percent video. We come out of the TiMax SoundHub with Dante, that gets picked up by the first BSS box, and then it's literally off to the races with whatever level of control and distribution we might need.

"There is very little audio processing going on," he continues. "The four BSS processors out in the mall are fan-cooled, mounted about 30 feet up on columns. Dante travels via fiber to each, we convert back to Ethernet at each of those locations, and then have analog outs that we run to the clusters in either direction."

The system was tuned using little more than a spectrum analyzer and good sets of human ears, with the resulting performance from the clusters resembling that of a constant curve type of array. "When we're done it will be like a half-circle pattern that we've mounted in here," to use Pizzo's straight-to-the-point analogy. "Every one with a stake in this project listened to one another, and we really worked as a team to put it together. What we've created will take us well into the coming decades. Vegas, as a concept, is indeed constantly changing. Now we have the tools to deal with whatever comes our way both today and tomorrow." LSI

__Gregory A. DeTogne__ is a writer and editor who has served the pro audio industry for the past 35 years.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SIGNAL PROCESSING

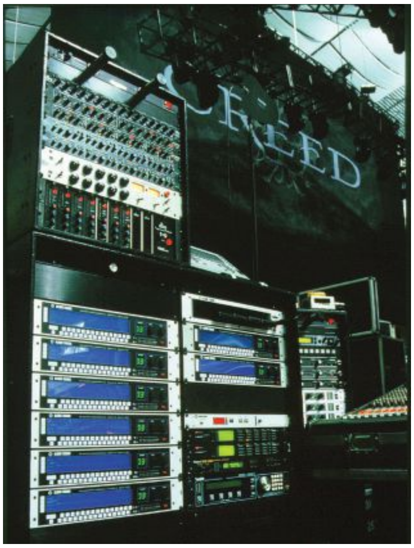
The evolution of large-scale sound system optimization, part 2. *by Bob McCarthy*

"In the beginning there was graphic EQ." The first standard tool for system equalization was the graphic equalizer. Early versions were 10 bands at octave intervals, but the 1/3rd-octave version took over the market completely by the late 1970s. The 31 bands were standardized to a series of 1/3rd-octave intervals beginning with 31 Hz. There was no standardization of the shape of the filters, however. One model might use 1/3 octave filters, another would use octave filters.

One of the primary attractions of the graphic equalizer was that its front panel settings seemed to represent the response it was creating (hence the name "graphic"). This was mostly true if the settings were all flat, but once the sliders were moved the resemblance faded because the parallel filters also affect the range of their neighbors. The parallel filter interaction dramatically affected how closely the "graphic" shape of the front panel actually resembled the curve that was being created.

The reality is that the picture on the graphic EQ front panel was never accurate but some (especially the ones with wide filters) were wildly inaccurate. This confused people because they attempted to use these tools for what I call "ear to eye training." Engineers learned to distrust the "graphic" settings. They knew it wasn't doing what it showed on the front panel – but they didn't know what it "was" doing. The inaccuracy of the graphic EQ created a lot of false conclusions.

Graphic equalizers with narrow filters create a ripple in the response, which increases as the cuts deepen. The center frequencies cut deeper than the mid-point between. Wider filters reduce the ripple, but increase the overlap, which decreases the accuracy of the front panel. Graphics with narrow filters more closely correlate to the panel but have higher ripple. I know of only one graphic EQ that old engineers still have romantic feelings for: the Klark Teknik DN360 (wide filters, low ripple and low accuracy). Bottom line: front panel accuracy doesn't matter much when you are tuning by ear, but ripple does.



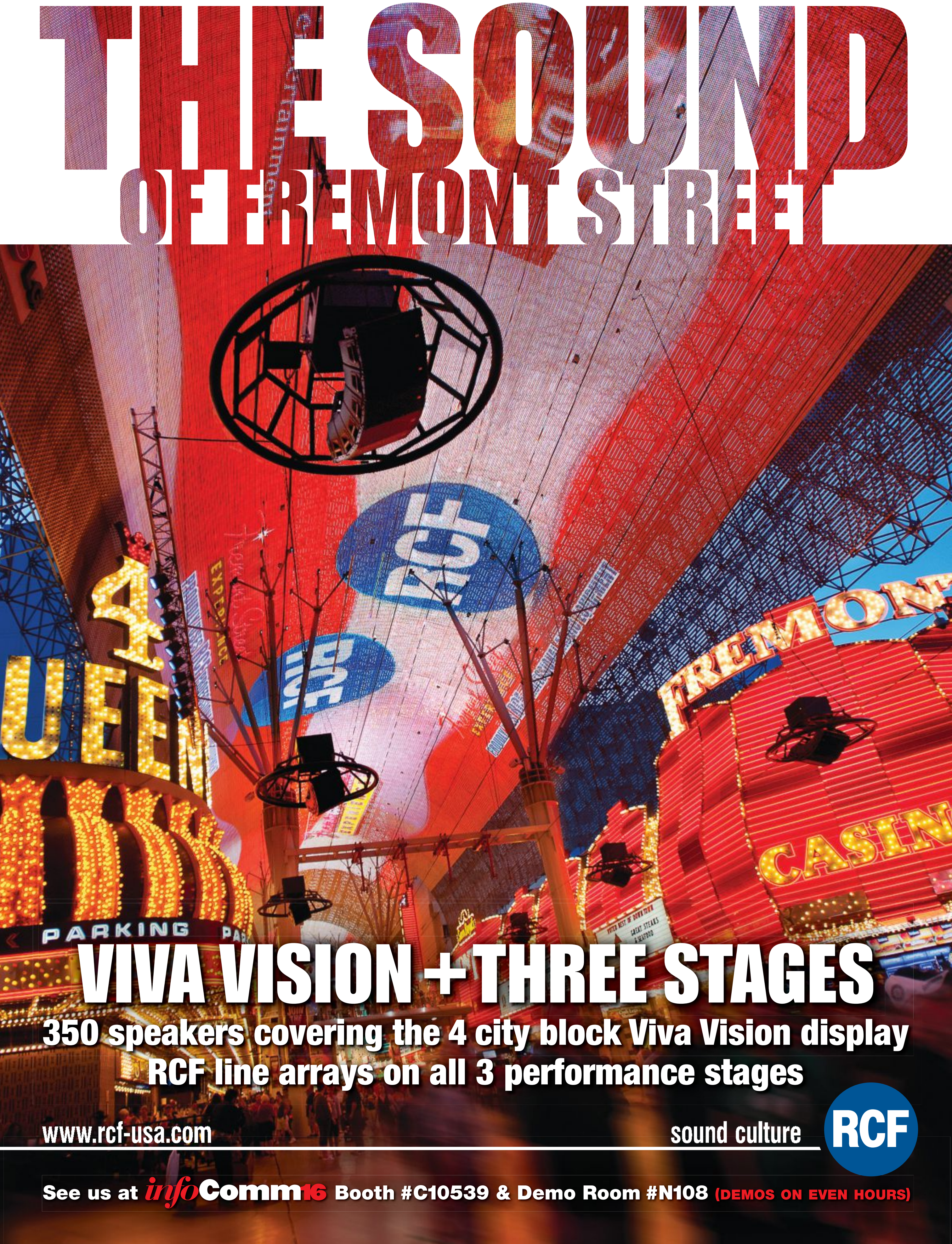
The monitor position for Creed on tour in the late 1990s, chock full of Klark Teknik DN3600 programmable graphic equalizers and a whole bunch of other analog processing gear. Scorpio Sound (West Bridgewater, MA) was the touring company.

THE GRAPHIC DETAILS

A substantial culture arose of what I would call the "graphic EQ code of conduct," a set of visual rules that governed the fader placement. The foremost of these was the "move the neighbors" rule, which mandated that a deep cut at 500 Hz meant you had to move the 315, 400, 630 and 800 Hz faders down as well to make it look like a gradual curve. Never mind that this causes 500 Hz to cut much deeper and wider than you intended.

Another of the folk legends was the belief that cutting more than 6 dB would create a "phase problem" of some mysterious unquantifiable variety. This was taken seriously: Everyone knows you don't push that fader past 6 dB! I can't say this phasor vortex never happened to somebody, but I can say I never saw

The ubiquitous (at the time) Klark Teknik DN360 graphic EQ, joined by other popular models from BSS (FCS-960) and dbx (231).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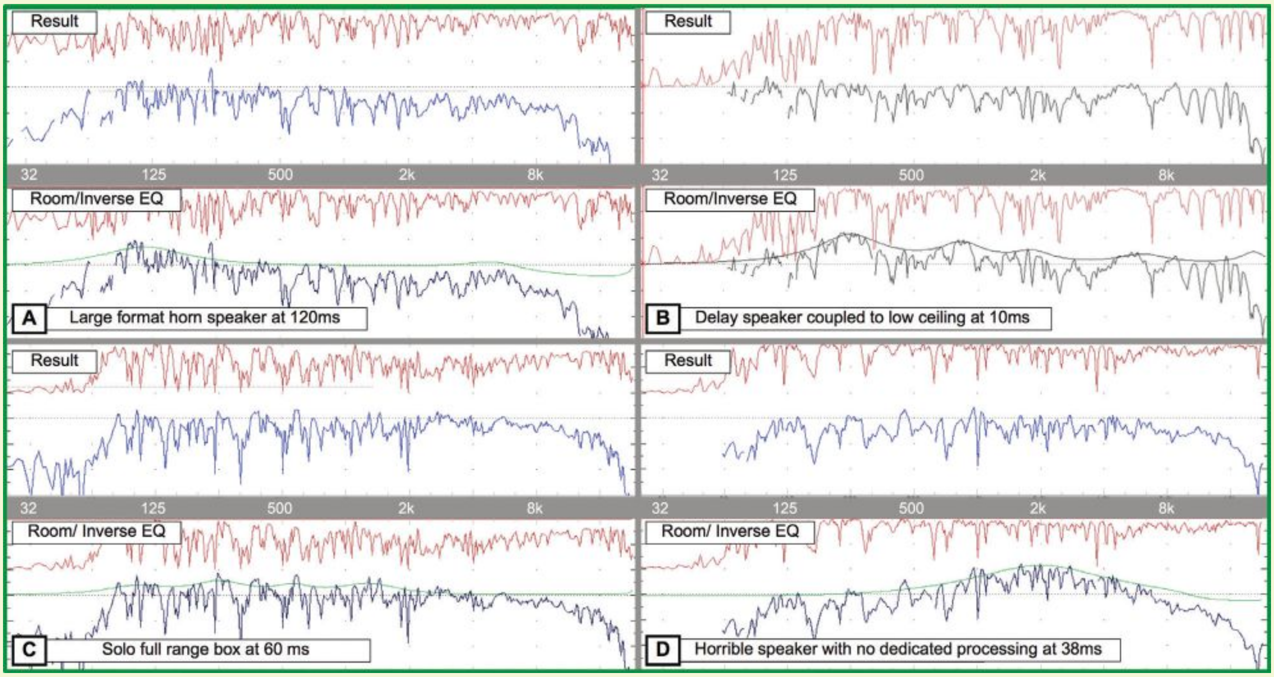

SIM measurements that highlight some of the problems with graphic EQ.

such a thing occur on my analyzer (which reads phase). The phase problems that we did see were primarily the side effects of amplitude problems associated with ripple and having the wrong center frequency and bandwidth to do the job.

This gets to the heart of the graphic EQ's principal shortcoming: fixed center frequencies and bandwidth choices. Simply put, the graphic could never succeed as an optimization tool because the problems it is trying to solve do not have a single fixed bandwidth and are not obliged to fall on the ISO approved center frequencies. Our challenge is more complex than that.

At one of the first concerts that John Meyer and I measured with SIM, we came up against the graphic EQ rule mentality. We measured a 10 dB peak at 100 Hz and knocked it out with a single cut on the graphic EQ. We high-fived each other at the perfection of the lucky match of center frequency and bandwidth as we watched the amplitude and phase flatten out.

A short time later the system engineer saw the single deep cut on the graphic and freaked out. He then reworked the settings and made them look nice and gentle on the graphic, which looked good there but no longer solved the problem. He explained to us that he needed to do this because we were messing up the phase response. It never occurred to him that we could actually see the phase response right there on the analyzer. On that evening John and I knew that we could never beat the graphic EQ police and needed to make a better tool.

PARAMETRIC EQ

The inadequacies of the graphic equalizer became totally apparent once we began to see high-resolution frequency response data. Our analyzer could now show us a problem centered at 450 Hz, but we were stuck with the graphic EQ's fixed filters at 400 Hz and 500 Hz in that area. The inability of the graphic EQ to create a complementary response to what we measured was impossible to ignore.

The parametric equalizer was immediately seen as the superior tool, since it had independently adjustable center frequency, bandwidth and level. Anything that we could see on the analyzer that was worth equalizing could be precisely complemented with the parametric filters.

The high-resolution transfer function analyzer put an end to the usage of the graphic EQ (although it took a long time to die). Now we could see the phase response (one mystery solved) and we could also see the actual amplitude response of the combined filters (second mystery). The analyzer proved that the graphic could never satisfy our needs. The graphic EQ is now used only for gross tonal shaping by ear, i.e., an artistic tool or for combat EQ (stage monitors), not a system optimization tool.

There were several reasons why parametric equalizers had attained such minimal acceptance before that time. The first was that people had trouble visualizing in their mind what the filters were doing. Filters could be set anywhere – including right on top of each other. You had to look at all the settings and then conjure in your mind what it all meant. This made many engineers understandably insecure. Most modern parametrics accurately display their response on their front panel display or software program, even incorporating filter interaction. The parametric response is no longer a mystery.

The second issue was that most commercially available parametric EQs used a filter topology that was poorly suited to system optimization. The filters were asymmetric, having a different type of response for peaks and dips. The dip side of these devices used a notch filter topology, which does not properly complement the peaks it's trying to treat in the sound system. The notch-type parametrics with wide peaks and narrow dips actually mimic the problematic comb filter effects rather than compensate them.

The high-resolution measurements taken with SIM showed us the advantages of complementary phase equalization: a parametric EQ with symmetric second-order filters with minimum phase shift, which became the guiding principles behind the 1984 development of Meyer Sound's CP-10 equalizer. We could now put the "equal" in equalization by producing an equal and opposite amplitude and phase response to the peaks and dips found in the field.

As previously stated, there was a lot of resistance to parametric EQ in those days because of the lack of a graphic user interface. The SIM analyzer gave us something better than a graphic interface. We could view the actual measured amplitude and phase of the EQ without repatching or taking it off line. Transfer function measurement allows us to probe across any two points in the signal path of our sound system. We can monitor the EQ output versus the EQ input and see precisely what response the device is creating. From the outset, the SIM system was always set up to be able to view the EQ electrical response as well as the response of the speaker system in the room.

THE DIGITAL AGE

The digital era dawned with the introduction of digital delay lines. These replaced the previous generation of analog delays (yes there were such things but their dynamic and temporal range was very poor). The first-generation digital delay was a noise floor choke point, so it was used only sparingly, when absolutely needed. The digital delay within the modern DSP is different from its first-generation version only for its higher dynamic range and resolution (and better A/D conversions).

The systems of that era had digital delays, analog equalizers, and analog level distribution, all in separate devices, each of which had only a few inputs and outputs. Once digital equalizers started

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to hit the market we quickly reached a tipping point in favor of merging all of these functions under one roof. Go to a rental house tomorrow and ask for a component digital delay or analog EQ. There will be hundreds to choose from once you blow the dust off.

There is great advantage to minimizing the number of A/D conversions, the wiring, patch bays, ground references, power supplies, etc. All of these functions are now done with multichannel input and output devices.

We have evolved to two families of DSP: open topology and fixed topology. The open topology systems (e.g., BSS Soundweb, QSC Q-Sys, Peavey MediaMatrix) are inputs, outputs and a mountain of malleable memory. They are an open interior waiting for us to arrange the furniture. Users can pull "devices" off the virtual shelf and "wire" them up to customize them as needed.

Fixed topology devices (e.g., Lake Controller, Meyer Sound Galileo) have pre-ordered the parameters and signal routing, incorporating all the features relevant to system optimization (and more). The filters in the modern DSP mime the filters of our analog world but can also go beyond them to make exotics. Very few of our optimization needs can't be solved by the analog filter shapes (parametrics, band filters and all-pass filters), so these are the still the workhorses.

Meyer Sound CP-10 parametric EQs in the racks at Carnegie Hall circa 1988.

The digital exotics such as FIR filters and free-pass filters require adult supervision. But there's good news: a digital version of the graphic EQ can still be found as an option in most of these devices. Works great with vinyl. LSI

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

CONSTANT TIDE

Software keys the growth of digital console platforms.

by Craig Leerman

A console that has a computer for a brain, rather than being limited to hard-wired analog paths, offers many advantages. Easily one of the most significant is that new features and capabilities can be easily (and at low to no cost) added via software updates. This provides much better return on investment to those who purchase the gear while also making life easier on the mix and tech side of the equation.

The software-based approach also means that firmware upgrades can be implemented to fix glitches that only reveled themselves after the product was released to the public. This is great because it means that, in most cases, the console doesn't need to be shipped back to the manufacturer (or to a service center) to get problems sorted out; rather, users just download the latest upgrade or patch, implement it, and then get right back to work.

Yet another advantage is additional flexibility. Users are provided with alternative methods of control for their mixes and related factors like automation. They can use a console's physical controls, a combination of those controls and touch screen interfaces, or increasingly, the touch screen almost exclusively. Of course, this extends even further to external tablets and smartphones serving key roles in the mix process, even "delegating" some mixing to individual users (i.e., musicians doing their own monitor mixes).

The listing of capabilities could go on and on, but you get the idea. So let's take a look at recent software updates from the digital console world in terms of the new capabilities and features they provide.

One of the niftiest aspects of the new **Allen & Heath** dLive platform is the DEEP processing architecture that embeds plugins directly within the input and mix channels. An array of plugins, including graphic and EQs as well as compressors (and multiband compressors) can be inserted on the fly, without taking up FX slots or requiring external plugins. In addition, dLive offers the RackExtra FX portfolio of "boutique" plugins that are conveniently already onboard, eliminating any latency concerns. This integrated library provides a host of reverb, delay and modulator algorithms, with 16 FX slots available, each with a dedicated stereo return.

Three of the many selections in the Allen & Heath dLive RackExtra FX portfolio, top to bottom: stereo tap delay, dynamic EQ, and gated verb.

At the other end of the spectrum, the newest member of the A&H Qu Series of digital mixers, the Qu-DB, is a surface-less freestanding or rack-mount unit that takes the form of a stage box, with all control parameters are provided via the Qu-Pad iOS tablet app.

At the recent Prolight + Sound show in Frankfurt, **Avid** announced new VENUE 5.1 software that delivers enhanced capabilities for VENUE | S6L mix systems. Version 5.1 now supports two AVB-192 Ethernet AVB network cards, which enables users to expand their S6L systems to support up to 192 remote mic-preamps and 96 outputs using three fully loaded Stage 64 I/O racks.

In addition, the new Spill Mode allows users to quickly spill any aux, group, or VCA members onto the surface faders for immediate access to these channels. The new upgrade also offers enhanced visual feedback of parameters and states on the high-resolution OLED displays for faster navigation and mixing, as well as improved VENUE show file compatibility with other VENUE systems.

It should also be noted that the VENUE | S6L is a highly

The Avid VENUE | S6L System interface sports a darker new look but functions just the same as its predecessors, albeit with more capabilities.

integrated platform, supporting Avid and third-party 64-bit AAX plugins. It also fits within the Avid MediaCentral Platform, an end-to-end media management and distribution platform, with one highlight being seamless integration with Pro Tools recording and playback capabilities.

Yamaha has been quite active, including new firmware update V2.5 for the TF Series. It includes four matrix outputs with delay parameters that are well-suited for setting up delay compensation for live loudspeaker systems in large venues or installations. Users can set delay time in meters, feet or milliseconds. TF V2.5 also includes a simple output delay that can be used as an insert FX on the aux 9/10 or aux 19/20 buses to manage the timing of signals sent to loudspeakers at different locations.

V4 for the Yamaha CL Series adds a new MBC4 multiband compressor Premium Rack device.

The CL Series also gets an update, V4, that adds numerous upgrades, including new "Precise," "Aggressive," and "Smooth" channel EQ algorithms and the new MBC4 multiband compressor Premium Rack device. In addition, V4 allows CL consoles to support control and monitoring of Shure ULXD4D and ULXD4Q digital wireless receivers. Meanwhile, the MonitorMix iPhone app, which accommodates use of up to 10 devices simultaneously, now supports GEQ gain control from the "Touch And Turn" knobs on the console surface, as well as head amp control of the Tio1608-D I/O rack and more.

Steinberg Nuendo Live DAW software, previously included with the CL Series, is also now being bundled with QL and TF Series consoles, providing multitrack recording and virtual sound check capabilities. And, several new features have been added to StageMix V5.1 for CL, M7CL and LS9 consoles, including 121-band RTA support that expands on the 61-band capability of the current RTA by using the built-in iPad microphone. New aspects of StageMix V5.1, for CL/QL Series only, include an improved USB recorder that enables the timeline dot to be dragged to change the playback point, or the desired playback point (time) can be entered numerically.

DiGiCo recently announced Stealth Core2, an upgrade to its proprietary Stealth Digital Processing, that's available as an option to all existing SD consoles. It provides additional processing from the audio core as well as features that include a new look application code to provide a new screen graphical interface.

In addition, full dynamic EQ will now be available on every channel and bus, and it also provides an increase in the number of channel strips, multi-band dynamic options, DiGiTuBes, and FX units. For example, the SD9 channel count goes from 48 to 96 channels of processing at 96 kHz and the SD7's total number of processing strips increase to 600 (also at 96 kHz).

The subsequent Stealth Core2 upgrade announced for the compact SD11 and SD11i models takes the SD11i's channel strips from 40 to 80 and doubles the output buses to 24, plus master and 8 x 8 matrix. It also increases the six DigiTuBes to 114, the six dynamic EQs to 114, and the multiband dynamic options from six to 114. And, the number of digital FX rises from six to eight units.

Earlier this year **PreSonus** released a series of related updates for the StudioLive RM16AI and RM32AI rack-mount digital mixers; StudioLive AI-series consoles; the StudioLive CS18AI control surface; UC Surface touch-control software for Mac, Windows, and iOS; Capture 2 recording software; and Studio One DAW software.

With the updates, two StudioLive RM-series mixers can be cascaded via AVB Ethernet to create larger mixing systems. Both mixers in the expanded systems can be controlled from a StudioLive CS18AI, from UC Surface, and from Studio One Remote. The RM-series update also adds a new Stage Box mode that allows the user to use the mixers as simple I/O devices in conjunction with StudioLive AI-series consoles, in addition to the previously available remote I/O and monitor-mixer mode.

PreSonus simultaneously released Studio One 3.2 that adds the ability to remote-control StudioLive RM-series XMAX preamps from the DAW. Users can also control the hybrid Fat Channel processing for RM-series and AI Series consoles from within Studio One. In addition, the updates also add remote control of Capture live recording software from a CS18AI and from UC Surface.

While technically not a software update, the new **Waves Audio** eMotion LV1 is a software-based console designed for front of house, monitor and broadcast duties. Based on SoundGrid technology, 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, allowing engineers to move between live and studio environments.

The channel EQ, filters and dynamics processing in the mixer

The new Waves Audio eMotion LV1 software-based console.

The Mackie AXIS system's DC16 control surface that works with Master Fader v4 and can dock up to three iPads.

is handled by Waves eMo Q4, F2 and D5 plugins. 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 touch screens to a single laptop or tablet.

Recently released **Mackie** Master Fader v4.0, the control app for the DL32R, DL1608 and DL806 digital mixers, marks the integration of iPad (including the new larger iPad Pro), iPhone and iPod touch support within a single app, doing away with the need for the separate My Fader app for personal monitor mixing.

Master Fader also offers additional new features, including an RTA on each output, built-in oscillator (with a choice of pink noise, white noise or sine wave with selectable frequency), the ability to copy and paste channels and mixes, and a quick assign function. There's also an updated take on recording and playback.

Meanwhile, Mackie has also introduced the DCi6 control surface for the DL32R. Dubbed the AXIS system, the control surface allows adjustment of every parameter on the mixer, providing full-color channel displays and the ability to dock up to three iPads in the SmartBridge. This provides users with easy customization over each iPad view to meet specific workflow needs.

The latest software revision for the **Roland Pro AV** M-5000 O.H.R.C.A. platform is version 1.101, providing the latest M-48 personal mixing management and new GP I/O functions from the built-in 25-pin port on the rear of the console. The M-48 provides 40 channels that are mixed as 16 stereo groups to each musician on stage via a Cat-5/6e cable and allows each musician to control their own mix.

Users can also set up two monitor mix zones using REAC A and REAC B ports, which allows a separate set of 40 input sources that can be assigned in 16 stereo groups, providing multiple monitoring mix options without taking up any console resources. The "Engineer's Monitor" function allows an engineer at front of house to mirror a monitor mix to a local M-48 at the console from any M-48 on the network, listen to it, make adjustments, and send it back to the musician on stage.

Soundcraft just announced the availability of a firmware update for its Ui Series digital mixers that adds several new features and extends browser and device compatibility. The software now supports Microsoft's latest Edge browser for both models, while the Ui12 now includes stereo recording features previously found only on the Ui16.

The update also provides a more secure password-protected Access Limitation System that now lets users assign secure access privileges to other users. A new sync ID feature has also been added that maintains channel sync across multiple browser windows, on different displays or even devices. Further enhancements include a MoreMe portrait on/off mode, the channel mutes now mutes auxes in PFL mode, and new global pre-fade aux send point selection.

Cadac has unveiled V3.01 for its CDC consoles, an update that includes enhanced channel copy and paste functions, a router GUI and network gain compensation, input and output patching enhancements, and direct outputs with additional options. Further, EQ, effects and dynamics settings can be stored in a user library and utilized to make custom fader layers. These can consist of any mixture of inputs, buses, VCAs and monitor faders.

In addition, the new CDC Offline Editor mirrors the console's operating system using an identical GUI, enabling full offline functionality. Show files can be created or pre-existing files can be edited and then uploaded to or downloaded from the consoles via a USB key. CDC Offline Editor is anticipated to launch on the Apple App Store in the third quarter of the year.

The Cadac CDC six, now with V3.01 software.

Last year, Solid State Logic upgraded the feature sets of the SSL.Live console range with V3 software, which introduced over 40 software and hardware features. The L500 console was renamed the L500 Plus, increasing from 192 mix paths to 256 with a doubling of effects processing power (depending on the effects selected), while the L300 also offers an increase from 128 to 192 mix paths.

Additional highlights of V3 includes VCA control of matrix channels, new implementations for the user assignable buttons, and a new stereo version of the Fixed Point Per Octave Analyzer called the FPPO-lyser. It includes a difference mode, allowing the difference between two signals to be monitored and two sets of crosshairs to track the maximum level/frequency and the level of a specific frequency. LSI

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


Desired directivity
and better impulse response

SIZE MATTERS

Observations on loudspeaker directivity.

by Bruce Main

EDITOR'S NOTE: This fine article was featured in the April 2010 issue. We reprint it here in celebration of our 25th anniversary.

Trap boxes and line arrays get all the attention. And that's no surprise – they're big and loud, and dare I say it, glamorous. But the truck rarely rolls without a complement of two-way loudspeakers sporting a 12-inch or 15-inch woofer and a horn. Whether its monitor wedges, drum fill, front fill or just "speakers on sticks," small 2-way boxes do many of the everyday jobs that make up a typical sound reinforcement day.

We take the performance of these boxes for granted, but they can be used to better effect if we really understand their directivity characteristics and what makes them perform the way they do. They're often described as a "90 by 60 box" or some other dubious reference.

But 90 degrees by 60 degrees at what frequency? Certainly not from DC to light.

There are four principle ingredients that govern the dispersion pattern of these loudspeakers, including the cone driver, horn, crossover and cabinet. Let's look at these one at a time and assess their contributions.

Before we go through our list, though, let's review some basics. The amount of directivity any device can exert on a sound wave is directly related to the proportional sizes of the device and the sound wave. To understand this relationship it is important to have a good grasp of how big or small a sine wave is at a given frequency. Sound at sea level at 72 degrees Fahrenheit travels at approximately 1,130 feet per second. We express frequency or cycles (sine waves) per second as Hertz (Hz).

So if the frequency of a wave is 1 Hz, the wave is 1,130 feet long. Logically, a 10 Hz wave is 113 feet long, a 100 Hz wave is 11.3 feet long, and a 1,000 Hz wave is 1.13 feet long, etc. While it's not overly difficult to do the math to determine the wavelength of any given frequency, there is an old "cheat" called the rule of 5-2-1:

20 Hz = 50 feet
50 Hz = 20 feet
100 Hz = 10 feet
200 Hz = 5 feet
500 Hz = 2 feet
1,000 Hz = 1 foot
2,000 Hz = .5 foot
5,000 Hz = .2 foot
10,000 Hz = .1 foot

While not perfectly accurate, it fills the bill for "quick and dirty" calculations. Physics dictates that a source be physically large in comparison to a wavelength to exert directional control over it. So let's look at the low-frequency directivity of a 12-inch driver in a 2-way loudspeaker with a 90-degree by 60-degree horn.

MATTER OF CONTROL

Remember that the low-frequency driver's only means of controlling the dispersion of the sound wave in a front-loaded loudspeaker are its cone diameter, and to a lesser extent, some boundary effects (we'll discuss that later). At 100 Hz, the driver is physically small in comparison to the 10-foot wavelength and provides almost no directivity (**Figure 1**).

If we increase the frequency gradually, the 12-inch driver does not suddenly exert pattern control over the sound wave when

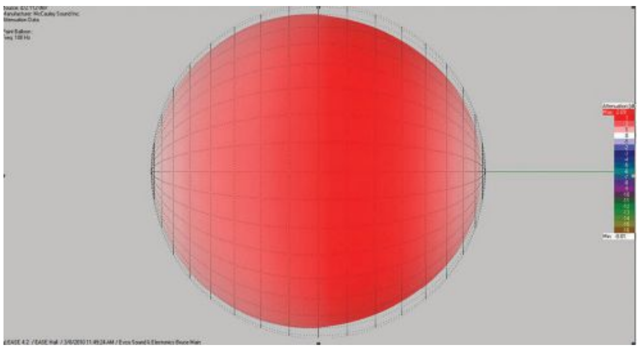

Figure 1: Horizontal directivity balloon of a 12-inch 2-way loudspeaker at 100 Hz (box facing left).

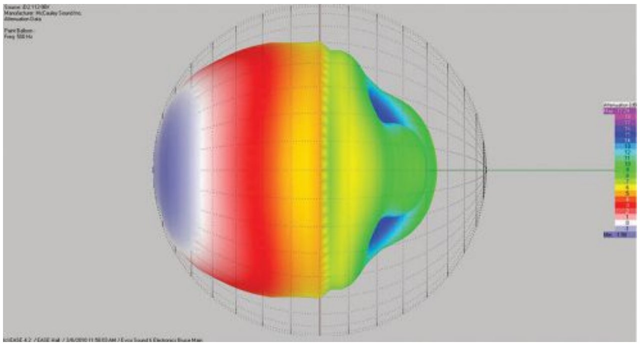

Figure 2: Horizontal directivity balloon of a 12-inch 2-way loud-speaker at 500 Hz (box facing left).

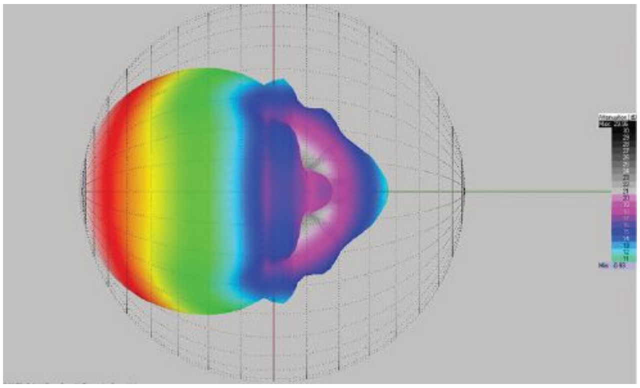

Figure 3: Horizontal directivity balloon of a 12-inch, 2-way loud-speaker at 800 Hz (box facing left).

it reaches 1,000 Hz (1 foot), and is the same size as the driver itself. Rather, it has more and more effect as the frequency gets higher and the wavelengths get shorter (**Figures 2 & 3**).

In this frequency range, the cone driver is actually providing approximately 90-degree horizontal dispersion. But also realize that since this pattern is conical (the driver is round), it is not producing the specified 60-degree vertical pattern. As the frequency increases the driver exerts more and more control until it begins to "beam" at higher frequencies. But by the time it narrows that much, it's above the crossover frequency.

This particular loudspeaker crosses over about a half-octave above the balloon in **Figure 3**. This has an overriding effect on the polar behavior of the box, especially in the vertical domain, so we will discuss the range from 1,000 Hz to 1,500 Hz when we discuss the crossover.

Now, on to the horn.

DOMINATE THE WAVELENGTH

There are multiple elements in a horn's design that contribute to its ability to achieve pattern control at a given frequency. Some of them are throat geometry, length and flare rate.

But the most obvious factor is the size of the horn mouth. The same rules apply here as to the cone driver. Size matters. The horn mouth must be large enough to dominate the wavelength in question in order to provide complete directivity at that frequency.

So if a horn mouth is 6 inches wide by 3 inches tall it will be somewhat omnidirectional at 1,000 Hz. It will not dominate the sound wave until the frequency reaches about 2,000 Hz in the horizontal plane and 3,000 Hz in the vertical plane. It may provide a 90-degree by 60-degree pattern above 3,000 Hz, but almost certainly not at lower frequencies.

Cone drivers and horns by themselves are fairly predictable devices. But combining the two in close physical proximity can be quite challenging. The first problem is physical offset. In a typical 2-way box, the devices are located one above the other, and may also be at different depths. Even if we use delay to correct the time alignment between the drivers on axis, any other vertical angle will skew the time arrivals from the horn and the cone driver.

Because the bandpasses and vertical dispersion patterns of the drivers necessarily overlap in the crossover region it is probable that at any vertical angle that is off axis we will be hearing contributions from both drivers out of phase. This means there will be lobes and nulls (**Figures 4 & 5**). This particular box was crossed over at 1,350 Hz with a symmetrical Linkwitz-Riley 24 dB slope.

These lobes will vary in direction and intensity based on driver offset and pattern control, crossover slope, and overlap and alignment delay settings, but they will always occur in multiple driver boxes with physically separated sources. If a cabinet is laid on its side we get the same phenomena in the horizontal plane. Floor wedges, anyone?

This is one reason there has been resurgence in coaxial boxes. Because there is no vertical offset between the sources, we only have to correct for the variation in depth between the acoustic origin of the cone and the horn driver, and that distance stays

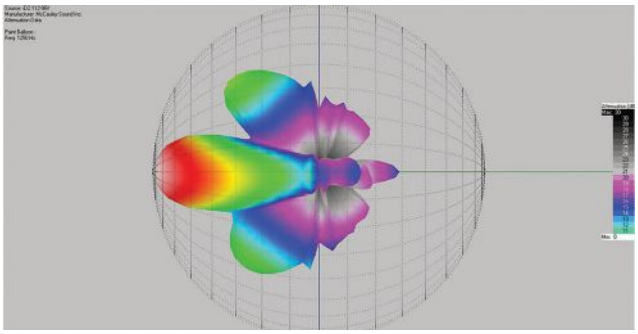

Figure 4: Vertical directivity balloon of a 12-inch, 2-way loud-speaker at 1,250,Hz, crossover at 1,350 Hz (box facing left).

more constant with off-axis listening positions.

The trade-off is that many coaxial designs use the driver cone as the horn flare to guide the high frequencies, and while this may be fine for monitors or other near-field applications, more precise pattern control is often required for sound reinforcement duties.

BAFFLES, BOUNDARIES

The final piece of the directivity puzzle is the cabinet itself and the boundary effect created by setting it on something. Fractional space loading is created when we decrease the space that a device is radiating into.

As we saw in Figure 1, low frequencies are omnidirectional, so when we set a loudspeaker on the floor, we effectively halve its radiating space at low frequencies. This produces an additional 3 dB of output (double the power) in the hemisphere that it is still exciting.

If the baffle on the cabinet is physically large enough versus a given frequency, it can act as a boundary to create half space loading. This is what is sometimes called “baffle step.” In modern cabinets, the baffle is rarely much larger than the driver that is mounted in it, because generally, priority is given to things like weight, truck pack, handle location, flying hardware,

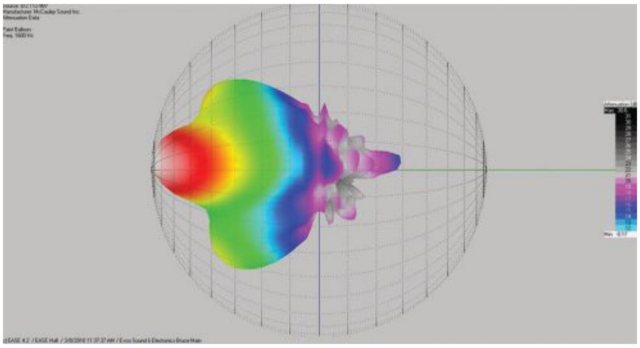

Figure 5: Vertical directivity balloon of a 12-inch, 2-way loudspeaker at 1,600 Hz, crossover at 1,350 Hz (box facing left).

arrayability and profile.

Technology has gone a long way towards providing a ton of output and fidelity from small packages. But physics hasn’t changed. When it comes to pattern control, size still matters! LSI

Bruce Main *has been a systems engineer and front of house mixer for more than 35 years, and has also built, owned and operated recording studios, and designed and installed sound systems.*

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GREAT EXPERIENCES

Inside the world of loudspeaker designer Ralph Heinz. *by Kevin Young*

THE CORPORATE BIO on Ralph Heinz states, "as the son of Renkus-Heinz founder and chairman, Harro... he was born to design loudspeakers." While his family has a history of being entrepreneurs, Heinz, now senior vice president, is quick to point out that there wasn't an automatic route to the role he'd ultimately play when joining the company in 1989.

"But I do treat loudspeaker design and my role in product development and management with an entrepreneurial spirit," he adds. "For me, it's important to come up with products – after having polled the marketplace and interpreted users' needs – that will truly solve their problems in a unique and novel way."

VARIED INTERESTS

Growing up in Queens, NY, Heinz was aware that his father was involved with audio, and when the family moved to Chicago in the early 1970s as Harro took the role of president of Rauland-Borg, Heinz began working for his father part-time during high school, assembling loudspeaker cabinets. "I was already a closet audiophile," he says, "but I never thought of that being anything but a summer job."

When Harro moved the family to southern California in 1979 to start Renkus-Heinz, the son developed a greater interest in his father's work. Targeting a career in mechanical engineering, after his first year of college, Heinz was again enlisted to help out at the fledgling company's new shop.

Things still didn't come together that time, either, with Heinz eventually departing to work with Sieger Engineering, a job shop south of San Francisco. Later he joined the NASA Ames Research Center in Mountain View, CA as a machinist/CNC programmer/tool and dye maker, manufacturing components that would later end up being a part of space shuttle experiments. Married by now, he was also continuing his studies of engineering at San Jose State University.

"But my father wanted me, my wife at the time, and our son to live closer, so he said, 'Come work for me and I'll make sure you're paid enough to continue school,'" he explains. "How can you turn down a deal like that? It worked well for a couple of years, but after I designed my first speaker and started going to trade shows, I no longer had time for the lab classes, which required my attendance."

Thus in 1989, he rejoined Renkus-Heinz, this time as a mechanical engineer, handling the tooling for compression drivers. Another focus was implementing ways to reduce cycle times and waste while improving yield and performance through various manufacturing techniques.

TAKING THE CHALLENGE

Partnership with users, clients and customers, and their education in the use of Renkus-Heinz products is core to the company's philosophy, Heinz says in citing the impact of a critique of one of his first designs had on his future work. Shortly after re-joining the company, while attending Musikmesse in Frankfurt for the first time, he spoke with the R-H distributor from Japan who was looking for a loudspeaker that would compete with another manufacturer's model. Upon his return, Heinz took up the issue with the chief design engineer.

"He replied, 'Good luck with that. I just took a job at JBL.'" Heinz, says, laughing, something he does easily and often when discussing his path in becoming a driving force in the company's development. "Because I was the son of the owner and founder," he continues, "the last thing anyone in the company wanted was for me to step into the design role, but I thought I was probably the most prepared to take on the challenge."

So he turned to noted loudspeaker designer D.B. (Don) Keele to discuss the distributor's request. Keele had consulted for the company on a range of large-format horn designs, and Heinz had taken that opportunity to increase the use of CNC equipment for prototyping those horns. The result of their discussion led to the creation of the C2 loudspeaker, which

included a 10-inch mid range horn and a co-axially mounted 2 by 1-inch manifold for high frequencies that Heinz felt would improve performance of the highs.

"It also had sort of a quasi-horn-loaded, bandpass arrangement for the dual 15-inch cones," he adds. "It was a 3-way system, fully horn-loaded. We showed it around the company and they thought it looked interesting and sounded good, so I said, 'Well, let's give it a try.'"

After seeing and hearing the C2 – housed in a 3-feet-deep, 200-pound enclosure and sounding like a studio monitor during its demo at AES – system designer Craig Janssen (of Idibri) later told Heinz: 'It's interesting. It sounds really good. But here's why I can't specify it.'

"He explained to me that the horn-in-horn coax, while offering improvements over non-coincident designs, compromised the directivity of the high-frequency horn because it was very small, and would therefore reduce intelligibility in the big venues he was designing systems for," Heinz says.

Getting into design, he continues, was therefore a bit of a trial by fire, but the critique, both in content and spirit, impacted his designs and design process going forward. In addition, his studies, albeit still incomplete at that point, informed his work enough that he was named the company's principal designer in 1992. "Fortunately my background in mechanics has served me quite well, as it turns out that that it applies to designing loudspeaker enclosures, waveguides, rigging systems, and other key aspects," he adds.

In the R-H assembly plant with one of his many loudspeaker creations.

Heinz "noodling" with a guitar during a break from designing loudspeakers.

TRENDING TO INNOVATION

Numerous design patents registered by Renkus-Heinz are attributed Heinz, including a method of arraying loudspeakers with improved phase coherency (1998) as well as a multiple driver horn (1996) that led to the company's impactful Complex Conic and CoEntrant technologies, respectively. He fondly remembers getting feedback from industry leaders Don and Carolyn Davis (co-founders of SynAudCon) on his early Complex Conic prototypes, a design he describes as being inspired by his opportunity to "look over the shoulders of giants."

"Don and Caroline were visiting, and Harro asked me to show them the new horn design I'd been developing. Having read and absorbed all of the latest AES papers on horn design, I always imagined that if Don Keele, Cliff Hendricks and Earl Geddes sat together at a table to design a horn, Complex Conic is what they would have come up with too. When Don and Carolyn heard it, they said, 'I think this is the future of sound, right here.' That was very rewarding to hear, especially from people who are so accomplished."

In getting to CoEntrant, he also references Janssen's earlier critique of the C2: "I didn't want to give up on the horn-in-horn coaxial design, so I thought about it for probably the next year and a half or so, and it occurred to me that if I could get the mids and highs to go through the same horn, I'd resolve the issues he mentioned. It would be something that was still coaxial, but with better coincidence in time and space for the mid- and high-frequency sources."

While Renkus-Heinz wasn't the first to market with coaxial loudspeakers, he does note, "It was already in our DNA, so it felt natural to design our next generation of products around that format."

In time, Heinz also began focusing on steerable sound and its potential to solve the serious issues presented by challenging acoustical environments. It eventually led to the Iconyx Series, a highly successful set of digitally steerable line arrays in a column loudspeaker format that's become a major product category in a wide range of pro audio applications, including performance venues, churches, and transportation facilities.

He readily credits the work of others – both at the company and outside of it – who have impacted a design, and this is no exception. "Iconyx was a full team project," he states, "and it's one that, because of the credibility we earned by producing a stable of good-sounding loudspeakers, put us in the sights of a consultant named Johann van der Werff."

About five years before the kick-off of the Iconyx development process, van der Werff tracked down Heinz following an AES presentation he'd done in Holland,

pointing to his own AES paper on what he called "A Loudspeaker Array with Exceptional Properties." "Johann also showed me simulations that displayed side lobe free beams and absolute constant directivity over a very wide passband, and asked, 'Does this look interesting to you?' It certain did. However, we didn't have the electronics and software resources, along with other capabilities we now have in house, but it was that introduction that led to Iconyx."

The ability to recognize and act on concepts and ideas that are both promising and consistent with the company's core design values has been integral to driving innovation, Heinz insists. "If I discover a good idea, I'll certainly try to adapt it and put a twist on it to make it our own," he says. "I believe it's important, as a designer, to come up with product differentiators, whether in performance or user experience, or preferably both."

MOVING FORWARD

What drives Heinz personally is simple, direct and right in line with the organization's founding principles: "Ensuring that everybody in an audience listening to a Renkus-Heinz system has a great experience" – an ethic driven as much by his passion for design as it is by his love of photography, live music, cars, motorcycles, and American-made guitars.

"I like nothing more than seeing a show in an intimate club," he says, referencing several venues in the region near his home in Silverado, CA. "I noodle on guitar. I'd hoped having instruments around would inspire my kids to play, but I didn't lead by example."

Essentially, all of his interests are rooted in the satisfaction he experiences in learning how to harness the power of technology. "Basically, I want people to feel as good listening to our loudspeakers as I feel about holding and strumming a Les Paul, zooming in on a nice Canon lens, or listening to a Porsche's engine warm up in the morning."

As for working with family in general for nearly three decades of the company's 37 years in business? "Well, it's difficult to turn off the business part and just be with family, but working with my dad has been very rewarding," he says. "He's allowed me the freedom to push things in one direction or another as long as I can make a convincing case."

One of Heinz's two sons also currently works at Renkus-Heinz as an application engineer, he concludes, laughing once again. "The other is studying to become a teacher, but if he doesn't get a job in the public sector, well, maybe we'll put him to work as an Iconyx trainer." LSI

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

LETTING IT RIDE

Prepping input lists, stage plots and other key aspects of the tech rider.

by Mark Frink

CREDIT: FRANK DONNELLY

Dr. John on a tour stop last year at Bull Run in Shirley, MA.

The input list and stage plot is the audio core of any technical rider and the road map for organizing stage equipment and console inputs. Accurate advance information allows risers and backline to be placed, microphones and wedges cabled, and even a line check when the touring crew's travel is delayed.

Working for clubs, festivals or sound companies, we're often frustrated by inaccurate paperwork reflecting a version of a band that's months or years old. The reason for out-of-date paperwork is clear when the process for booking shows is understood.

The basic information, the stage plot and input list, is one or two pages of a larger "Technical Rider" document that spells out everything a performer's staff asks to have provided to put on their show – sound, lights, backline, stage, electricity and risers. "Rider" means it's an attachment to the performance contract, a part of a larger "Contract Rider" that adds sections covering catering, dressing rooms, security, bus and truck parking or flights, hotel rooms, and local transportation.

Outdated tech riders exist because agents often book shows long before the band has rehearsed or the crew is hired. In order to execute a contract, the agency must include a rider that describes expenses beyond the artist's fee. Lacking that, the agency sends a previous rider. An up-to-date rider is eventually forwarded to the agency, but often not passed along. Meanwhile changes to inputs and plot affect the budget minimally.

My own solution is the "One-Page Tech Rider Update," a combination input list and stage plot that fits on a single page. E-mailing it a month or a week (or both) before the gig shows you care. Finally, to grease the wheels of progress, a few printed copies handed out at load-in immediately replace previous info with a current version without having to find a printer: one copy for the stage box, another for the mic workbox and a third for the backline vendor.

MAKING A LIST

The largest bands bring everything they need, from mics and direct boxes (DIs) to consoles and monitors. But for bands not carrying much, locals rely on accurate stage info. File-based tours require not only that the input list is in the correct order, but also that each input has the exact channel number, so the console's channels line up with physical inputs. Additionally, if travel goes awry, it provides enough info to prepare the stage for a late arrival that may even preclude a sound check. It happens.

The best practice is to incorporate both into a one-page document if possible, so that both inputs and plot can be seen at once and the two never get separated. Obviously larger acts may use an entire page for 48 or more inputs, but at that point, it's likely they're either the headliner or carrying all their equipment. To shorten input lists, stereo or multiple inputs using the same mic or DI saves one or more lines.

Input channels are arranged by instrument type and proceed in a standard order, with drums first, followed by bass, keyboards and guitars, any horns or acoustic instruments and vocals last. There's even a convention for the order of drum inputs.

Sub-snakes usually have 12 channels, so the first dozen often follow a standard festival drum patch, with two kick drum mics (inside followed by outside), snare mics (top before bottom), then hi-hats, followed by up to four toms, and ending with maybe

Version 3/23/2016 Prepared by Mark Frink (904) 307-XXXX monitors@MarkFrink.com

Dr. John & The Nite Trippers: Summer 2015, 'Fly-in' shows

Snake/PP	Sub-snake	Input	Microphone/DI	Notes	Stand
1. PP	D 1	Kick Drum Inside	Beta 91		-
2.	**D 2**	**Kick Drum Outside**	**Beta 52**		**Short Boom**
3&4.	D 3&4	Snare Over & Under	(2) SM 57		(2) Short Booms
5.	D 5	Side Snare	SM 57		Short Boom
6. PP	**D 6**	**Hi Hat**	**KSM 137**		**Short Boom**
7-10. PP	D 7-10	(4) Toms	(4) Beta 98 #4		-
11. PP	D 11	Ride	KSM 137		Short Boom
12&13. PP	D 12 / US 1	(2) Overheads	(2) KSM 137 #4		(2) Tall Booms
14.	**US 2**	**Bass DI**	**Aguilar XLR**		-
15.	US 3	Bass Mic	Beta 52 #2		Straight Stand #1
16&17.	US 4&5	'OPEN'			
18.	**US 6**	**USR Elec. Guitar**	**SM 57 #4**		**Short Boom**
19.	**DS 1**	**DSR Doc Elec. Guitar**	**Beta 57**		**Short Boom #9**
20.	**DS 2**	**DSC Doc Nord keybaord**	**Radial JDI**		-
21. PP	**DS 3**	**DSC Piano Pick-up**	**Countryman with BB CS-4000**		-
22&23. PP	DS 4&5	DSC Piano Low & High	*(2) Beta 181/C*		-
24.	**'HomeRun'**	**Trombone RF**	**XLR**		-
25. PP	**DS 7**	**Trombone FX**	**Radial JDI #2**		-
26.	**DS 8**	**DSL Trombone Vocal**	**Beta 58**	**'Sarah'**	**Tall Boom**
27.	**DS 9**	**DSR Doc Guitar Vocal**	**Beta 57**	**'Mac'**	**Tall Boom**
28.	**DS 10**	**DSC Doc Nord Vocal**	**Beta 57**	**'Mac'**	**Tall Boom**
29.	**DS 11**	**DSC Doc Piano Vocal**	**Beta 57 #4**	**'Mac'**	**Tall Boom**
30.	**US 7**	**USR Guitar Vocal**	**Beta 58**	**'Jamie'**	**Tall Boom**
31.	**US 8**	**USL Bass Vocal**	**Beta 58**	**'Roland'**	**Tall Boom**
32.	**US 9**	**USL Drum Vocal**	**Beta 58 #4**	**'Herlin'**	**Tall Boom #9**

Dr. John input list created with Microsoft Word Table.

a ride cymbal mic and finally two overheads. One or two more sub-snakes cover the remaining upstage backline inputs on each side of the drum riser, with another downstage for front line vocals and downstage instrument mics or DIs.

The audio tech that patches the stage box and labels the sub-snakes might not care about the musicians' names, but needs to know where each input is located on stage, so other than drum inputs, many input list channel names should end with three-letter abbreviations to be clear: DSL, DSC or DSR (downstage-left, downstage-center or downstage-right) and USL, USC or USR (upstage-left, -center or -right), with stage directions meant as facing the audience.

Input lists begin as a column of consecutive channel numbers on the left, followed by a column of input names with specific mic models. Additional columns show phantom power, sub-snake assignments, alternate mics and mic stand choices, each separated from the previous by a tab. Abbreviations for these properties are the headers for each column, in a logical order.

CONTINUING THE PATH

Experience dictates that channel numbers, abbreviated "CH," are followed by phantom power assignments, or "PP" for short, then "SUB" for sub snake assignments, making it easy for others to quickly review assignments without having to read too far across.

Next comes input channel name or "INPUT" for short, then "MIC/DI" for microphone and/or DI choice, followed by alternate or "ALT" and "NOTES," and finally, the type of stand, if any. Typing these column header abbreviations, separated by tabs, is the way to start an input list: CH (tab) PP (tab) SUB (tab) INPUT (tab) MIC/DI (tab) ALT (tab) NOTES (tab) STAND.

It's easier to type several inputs before converting it to a table, so underneath add 1. (tab) PP (tab) D-1 (tab) Kick/Inside (tab) BETA91 (tab) e901 (tab) On foam (tab) None" and underneath that add "2. (tab) (tab) (tab) (tab) (tab) (tab) (tab)," which spaces out the entry for the next channel.

Microsoft Word is the most common word processing software. To convert an MS Word list to a table, highlight the entire list and select "Convert Text to Table" from the Table menu. Bazinga! Grid lines suddenly delineate the list and greatly improve readability. As a monitor engineer, I also call out channels used in the monitors by making them bold.

Tabs can be reset to adjust column width by selecting the entire table and moving their markers on the overhead ruler. Columns can even be added or deleted under the Table menu. It may take some getting used to, so practice with your first few drum inputs. When you start getting the hang of it, remember to "Save" often, and there's always "Undo." After your first input list, editing a previous one is easier than starting from scratch.

When laying out an input list, stick to the standard conventions of starting with drums and ending with vocals. The "ALT" column for alternate mics or DIs can indicate where substitutions are acceptable (or not). It's helpful to total the numbers of each mic, DI or type of stand, so they're easily counted by locals and this can be indicated right after the last of each type: Beta58 #3, JDI #4, or tall boom #8.

Google the words "stage plot input list" and many examples can be found. I've used Word to create them for years, but so many bands use StagePlot Pro's $40 application (www.stageplot.com) that it's unusual to see anything else at festivals, unless it's the Sharpie/cocktail napkin variety.

SPARE CHANNELS

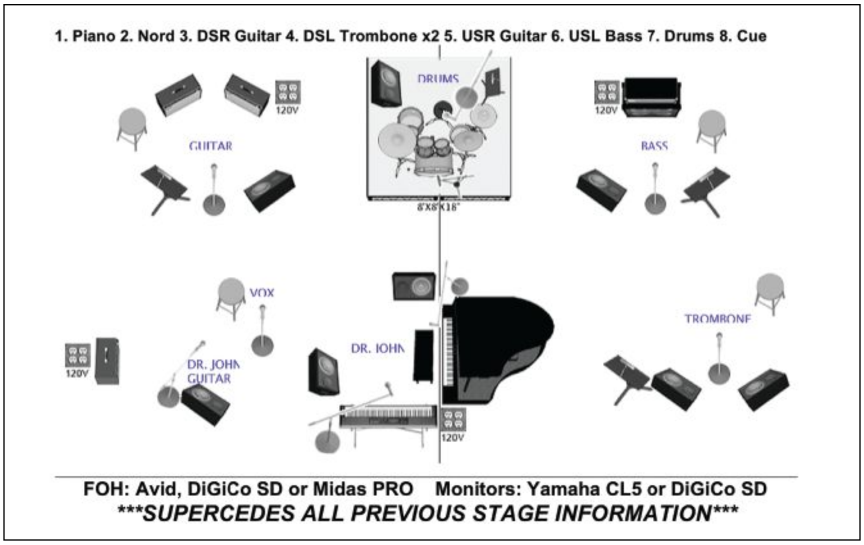
Digital consoles expect to find each input in a specific place, so it's important to keep each input in the same channel from one show to the next over the course of a tour. In my analog days, I'd label two channels between the instruments and vocals "Next-To" and "Nothing," so when an instrument or vocal suddenly got added, my partner at the snake's other end would already know where it was going.

It's better to have a few rarely used

channels than to need extra inputs with nowhere to put them. Adding horns or strings for special shows can be expected if they were used on the album for even a few songs. The same applies for supplemental acoustic instruments, even if it's just for a ballad or a breakdown section to help longer sets.

A more likely use for extra inputs is for a guest musician to sit in, even if the news isn't forthcoming until the last minute. Support or festival bands often make friends who occasionally sit in for a song. A block of four guest channels might include a spare vocal and active DI downstage, plus a guest guitar amp mic or a passive DI for a spare wireless or keyboard. Naturally each should be labeled and pinned to a sub-snake daily.

Engineers that started on Yamaha LS9 or Behringer X32 consoles are used to 32 channels, which is becoming a new standard for live sound with the advent of 96 kHz MADI. In a 32-channel footprint,



Dr. John Stage Plot created with StagePlot Pro.

putting drums in the first dozen inputs and leaving four channels open starts instruments in channel 17, followed by vocals. A dozen drum mics and a dozen more backline or vocal inputs leaves a half dozen spare channels and two more for console talkback.

A PICTURE = 1K WORDS

Neatness counts, and good-looking documentation gains respect. The stage plot is a "plan view" of a stage setup using symbols to indicate relative locations for risers, backline, monitors, electrical connections and audio inputs.

Stage plots are most easily understood as viewed from the audience, with a line at the bottom representing the downstage edge. It might not be exactly to scale, but labeling each riser's dimensions (8 x 8 x 1) and their relative positions provide much of the needed stage dimensioning.

One method to creating a stage plot is using software that's already on most computers, allowing documents to be updated. MS Word actually has features that can create a stage plot: Insert Shape, Picture and Text Box, but it takes time and patience to build from scratch.

In Word, choosing Shapes from its Insert menu and a window appears with various basic shapes that can be dragged and dropped into the document that can be resized and formatted. Alternately, photos or clip art can be inserted, so a good image of the exact drum kit can be easier to insert than to draw from scratch. Similarly, amp, keyboard and wedge images can also be used if they're clear. Stick to black and white, as it will likely be printed and faxed long after you've made it a PDF.

A small circle is a common symbol for a vocal mic. Although mics for drums are obvious, each of the other instrument mics and DIs (circles and squares) should show their stage positions and each should be identified with a channel number using a Text Box with its margins reduced to zero. And *please* put a little box labeled AC everywhere stage power is needed, including stage techs and monitor beach.

Double-ended arrows can be used to show important dimensions, such as distance from the downstage edge or drum riser. Once the initial document is built, making incremental changes is easy. Word has some very non-word processing tools waiting to be discovered.

Though it's possible to make input plots using Word, StagePlot Pro, available for both Windows and Macs for $40, has become an industry standard. (A free 30-day demo is available the website noted earlier.) While its individual graphic elements or "icons" are fairly standard, its input list section lacks some word processor features.

Word has good table formatting, with custom column widths and horizontal lines, but any stage plot image, including StagePlot Pro, can then be pasted into the bottom of a portrait-oriented input list. The same can be accomplished with MS Excel, for those more comfortable with it. Either of these widely available programs make it easy to correct and update the input list as it evolves, however best practice is to only make changes and republish this document once a year or season.

START IT OUT RIGHT

The input list and plot should have the name of the artist as a headline in large, bold letters, and include the current season and year to help future locals understand when it may be stale. Finally, I'd recommend adding the words "TV Promo," "Fly-in" or "Festival" to the headline to further indicate the type of show for which it's intended.

Use "Header and Footer" in Word to insert a name, e-mail address and cell phone number across the top of the page to serve as primary audio contact. Across the bottom (again in large, bold letters) should appear the version's date and the words: "Supersedes all previous information" or "Disregard previous plot and input list." I also like to offer a reminder of the preferred FOH and console choices here. LSI

***Mark Frink** is a long-time audio professional who's toured as monitor engineer for several top artists over the decades.*

RF FAQ

Successfully collaborating with a wireless frequency coordinator.

by Ike Zimbel

As the use of wireless systems continues to grow and the available wireless spectrum continues to shrink, the need for someone to coordinate all of that wireless on any given event becomes greater. It's why you're increasingly likely to encounter a frequency coordinator on a show or tour. With this in mind, and as a frequency coordinator, I've come up with a list of frequently asked questions about what I do.

What is a frequency coordinator?

A person (or team of people on a large event) who is responsible for making sure that all wireless systems in use are compatible and have been assigned unique, coordinated frequencies to prevent interference. This is done with a variety of tools, including spectrum analyzers and wireless frequency coordination programs such as IAS from Professional Wireless Systems.

Frequency coordinators are also tasked with the job of monitoring and tracking down any "rogue" wireless users and either getting them coordinated or shut down. For a backgrounder on frequency coordination and why we need it, check out my previous article, "Avoiding Intermod" (*LSI August 2014*).

I've got a festival date coming up, and I'm told there's a frequency coordinator on it. What is needed from me?

A list of all RF gear that your act is using. This must include make, model, frequency band and quantity of each system in use, as well as where it gets used. So, the simple part is the main wireless mic and IEM systems, say 4 x Shure UHF-R, J-5 band (stage mics) and 8 x Shure PSM-900, G-7 band (IEMs). After that, list all backline systems, with each listing including the musician's name and/or application (i.e., guitar, bass, etc.), make/model of their systems, their position on stage, the frequency band each system occupies, and any other special notes.

Note that it usually isn't necessary to

list actual frequency ranges as long as the Band/Block info is correct (so "J-5" is sufficient as opposed to "J-5, 578-638 MHz"). Also, it's important to note if you have any custom frequency banks; for example, one act I've worked with a number of times has some Sennheiser G3 systems in its backline that only tune from 534 to 558 MHz (I've never found out why).

In addition, note any mixed systems that tune differently from the norm, say Shure transmitters into Lectrosonics receivers (this is possible) or Sennheiser 2000 Series transmitters, which tune in 25 kHz steps into Sennheiser 3732-II receivers, which tune in 5 kHz steps. In each case, it's the limitations that govern what the coordinator needs to know.

OK, so I've given the coordinator all of the above. What happens now?

The coordinator will provide a list of compatible frequencies to use with your systems. On really busy, RF-intensive events, there may also be a timeframe specified as to when transmitters can be turned on. For example, the band is on at 5 pm, so you may be asked to not to turn on your transmitters until 4 pm, and also to have them off by 6:15 pm. Important! Also make sure the backline people turn off their transmitters at the end of your show. A guitar vault with 4 to 6 transmitters, still on, in close proximity to each other is a great little intermod generator, which is not something you want parked backstage while the next act is going on.

What if there are problems with some of my frequencies?

Let the coordinator know ASAP and he'll find alternates. Sometimes your list will be provided with alternate frequencies already. If so, try to keep track of which ones are bad and which ones you've moved to, as the coordinator will need to know these things if further work is required.

What you should *not* do is go "off-roading" to try to find your own frequencies. Here's a little anecdote to explain why: One time while working on an awards show, I gave the monitor engineer from an act his list of frequencies. He looked it over and said, grudgingly, "OK, I'll put these in, to start, but I *will* scan around because I've been screwed before." (And he did have a very nice set-up with mics and IEMs from the same manufacturer, linked with a good monitoring/coordination program.)

I let that ride and made a mental note to keep a close eye on his frequencies from my own monitoring set-up (I did, he didn't change anything), but here's why

It's really important to be on top of this when powering up, so you don't start what I call a "frequency stampede."

that would be a really bad idea. Sure, he has a state-of-the-art scanning set-up, but what *I* have is a list of 240 frequencies that are, *or will be*, in use.

In particular, this includes at least 24 channels of intercom belt packs, which are almost without exception set in the "push-to-talk & transmit" mode -- meaning that they are only transmitting when someone is talking. Which means that the likelihood of his scan picking them up is almost nil, which means that his scan would *appear* to show some wide open real estate to move his systems to when in fact, he might park his front man's mic channel on or near the floor director's transmit frequency... in which case her intercom chat could come out the lead singer's mic channel. This is also why it's never a good idea to go looking for new frequencies after you've been rehearsed and sound checked.

What can I do to be a good RF citizen on an event with a coordinator?

The biggest thing is not transmitting until your allotted time slot. This is relatively easy now that most systems have a "non-transmit" mode. Using that mode while setting up and tuning your gear is very helpful RF wise, and has the advantage of letting you get more of your set-up done in advance (like checking the patch and input levels to your IEM transmitters).

It's really important to be on top of this when powering up, so you don't start what I call a "frequency stampede." A frequency stampede occurs when "Tech A" turns on a piece of RF gear, causing a hit on "Tech B's" RF gear, which in turn causes Tech B to search for a new frequency. Meanwhile, as they're tuning, they step on "Tech C" or back on Tech A, who then wants new frequencies. Rinse and repeat.

I'm a backline tech for a major act and I hear we are going to have a frequency coordinator on our next tour. How will this affect me?

It's important that absolutely everyone using RF is on board. In practice, the frequency coordinator will provide a list of frequencies to tune your systems to at each new venue, and will also be available to resolve any issues that arise. This should make your life a lot easier and let you concentrate on tuning instruments instead of tuning wireless systems.

The same thing applies to the monitor position and other backline folks as well. It's been my experience that most people are really happy to have the wireless aspect of the gig off their plate. It's a funny thing, but because I only do shows with an RF coordinator on them (either me or someone I'm working with), I've never witnessed the struggles that seem to be a daily occurrence for a lot of techs. Which leads me to close with a conversation I've had with more than a few monitor engineers I've worked with.

Me: "How's it going?"
ME: "Are you (expletive) kidding me? I'm having the best day of my life!" LSI

***Ike Zimbel** has worked in pro audio for 35-plus years and runs Zimbel Audio Productions (zimbelaudio.com) in Toronto.*

PLAYING SAFE

The what and how of GFCIs, and why they really matter.

by Mike Sokol

Fig. 1

No, it's not the name of an insurance company or a European sports car – GFCI is an abbreviation/acronym for Ground Fault Circuit Interrupter. They've been required in many localities for electrical outlets located near sinks or the outside of residences since the early 1970s.

The two types of GFCIs you'll encounter are either built into the power outlet itself (**Figure 1, left**) or inside the circuit breaker at the power panel (**Figure 1, right**). Both do exactly the same thing: they watch for electricity that's going someplace it shouldn't in an electrical *Circuit* by way of a *Fault to Ground* and then *Interrupt* the flow by tripping the circuit breaker. Rearrange the letters and we get GFCI. That's how the name is derived.

WHY DO WE NEED A GFCI?

Human heart muscle is very sensitive to electrical shock. While it takes around 8/10ths of an amp (800 milliamperes, or 800 mA) of current to power a 100-watt light bulb, it takes less than 1 percent of that same current (10 to 20 mA) to send a heart into fibrillation, which can cause death by electrocution. That's why the NEC (National Electrical Code) now requires a special type of circuit breaker for damp locations that can tell the difference between the normal currents feeding an electrical appliance and the currents accidentally flowing through you to ground.

And while a GFCI sometimes trips unexpectedly, it's really there to save human lives, as well as the lives of appliances and other electrical components.

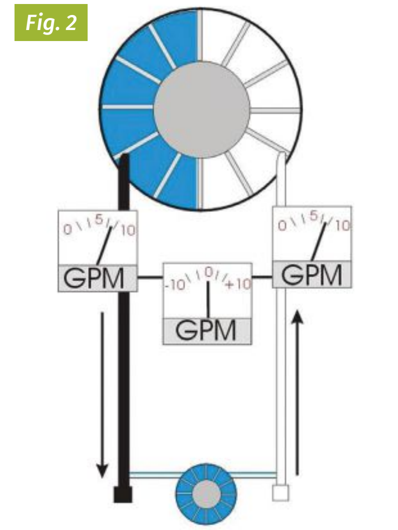
The most dangerous situation we typically find on a live stages is a performer getting between a backline guitar amplifier with a broken ground connection and a properly grounded microphone.

HOW DOES A GFCI WORK?

It's a very ingenious system that uses a small current transformer to detect an imbalanced current flow, so let's use a water pump analogy to review the typical current path in a standard electrical circuit. In **Figure 2**, let's imagine the pump at the top is pushing 7 gallons per minute (GPM) of water current around in a circle that the little turbine at the bottom is happily using to spin and do some work.

I've added flow meters at the bottom left and right of the illustration so we can keep track of these currents. Now since our pipes have no leaks, the current going out of the pump from the black pipe will exactly equal the return current coming back in the white pipe. This will be an exact balance since no water is lost in this closed loop. That is, if 7.000 GPM of water is flowing out of the black pipe, then 7.000 GPM will be returning to the pump via the white pipe. There are no water losses in this perfect system.

Fig. 2



KEEPING IN BALANCE

Let's add an extra meter in this system so we can keep track of the water flow a little easier. Notice there's now a center meter that will show the difference in flow between the other two meters. If the left and right meters show exactly the same water flow, the center meter will indicate 0.000 (zero) GPM of flow by centering its needle.

This is exactly what should happen in an electrical circuit that's working properly. That is, if a light bulb has exactly 1 amp of current flowing out from the black (hot) wire, then exactly 1 amp of current should be flowing back in the white (neutral) wire. And an electric griddle that has 10 amps of current flowing out the black wire should have exactly 10 amps of current flowing back in the white wire.

If there's nothing wrong in the light bulb or griddle circuit, this electrical current balance will be pretty close to perfect, out to at least 3 decimal places, in other words, 10.000 amps of current flow going out will equal 10.000 amps of current flow coming back in.

OUT OF BALANCE

In **Figure 3**, I've added a leak in the black outgoing pipe via the red pipe sticking out to the left. You can see from the red pipe's meter that 5 GPM of water is flowing out onto the ground. And since only 7 GPM of water is coming out of the black pipe on the pump, there can be only 2 GPM of water returning into the white pipe on the right.

Those 5 GPM of imbalance show up in the center balance meter, which alerts us to the fact that there's a leak somewhere in the system. Now, we really would like to know about small leaks as well, so that center meter will tell us about an imbalance down to very small drips, say less than 1/1000 of a GPM.

The same is true of an electrical circuit where we're interested in currents in the 1/1000 of an ampere range (1 mA). That's because just 10 to 20 mA of misdirected current flow is close to the danger level for stopping your heart.

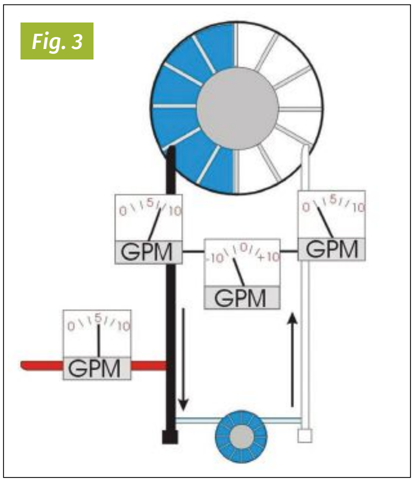


Fig. 3

TEETER TOTTER

In an electrical system, a similar type of detector is used at the center of the circuit that is acting like a balance beam. So if 7 amps of current shows up on both sides of the balance, then the beam will be exactly level. However, put 7 amps of current on the left side and 2 amps of current on the right side, and that 5 amps of imbalance will tip the scales, just like the teeter totter ride you took as a child (**Figure 4**).

In a GFCI circuit, this is a much more sensitive balance beam that only needs 5 mA (or 0.005 amps) of current imbalance to tip over, rather than the 5 GPM we've shown in the water pump illustration. The reason for needing this much sensitivity is that our hearts can go into fibrillation from just 5 mA of AC current flow, so we would like to detect and stop that flow before it gets to our heart.

THE BOTTOM LINE

So here's where it all comes together. Notice that our guy in **Figure 5** is unwisely touching a hot wire with a hand while his foot is in contact with the earth. And while the electrical outlet might have been supplying 7.000 amps of outgoing current to an appliance with exactly 7.000 amps of return current, there are now 7.005 amps going out and only 7.000 amps coming back.

Those extra 0.005 amps of current (5 mA) are taking a side trip from his hand to his foot via the heart. Meanwhile the current balance circuit inside the GFCI is sensitive enough to recognize that imbalance and trip the circuit open with as little as 5 or 6 mA of current flowing someplace it shouldn't be going.

The click that's heard when a GFCI trips is its spring-loaded contact opening up and interrupting the current flow in the circuit before it causes electrocution. That's the entire reason for the existence of GFCI – to save us from electrocution and keep our electrical systems safe from damage. Pretty cool, eh?

Also note that the GFCI doesn't really need a direct ground-bond connection via the ground wire to do its job. Yes, one is required to properly "earth" the entire circuit, but the current balancing act is only between the black and white wires going to the outlet.

If the current flow in the white wire exactly matches the current flow in the black wire to within 5 mA, the circuit

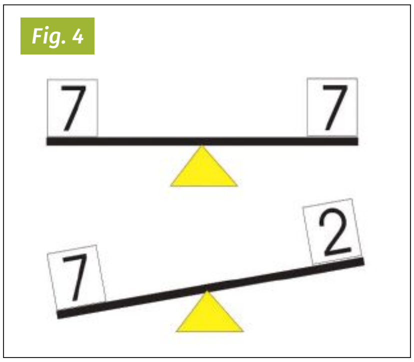


Fig. 4

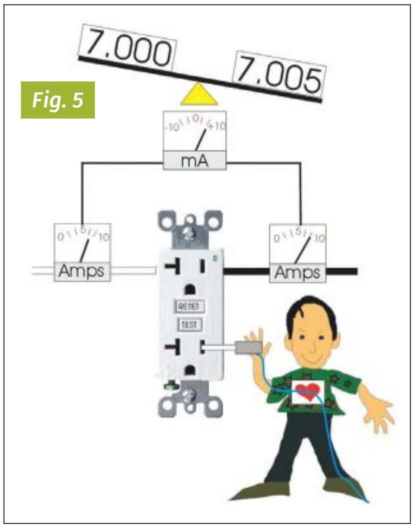


Fig. 5

stays activated. If the current flow is unmatched by any more than 5 mA, say by someone touching a live wire and the earth at the same time, then the trigger circuit inside trips a little switch and the current flow is stopped. It's that simple.

All of this means we should install GFCI breakers where required, and don't remove or bypass them on stage if there's false "nuisance" tripping. That so-called false tripping hints there's something else wrong in the sound system or backline that's leaking current to some place it doesn't belong. And fixing that electrical leak is important since getting a body in the middle of the current leak leads to shock or even electrocution.

Let's play safe out there... LSI

Mike Sokol *is lead trainer for Live Sound Co. in Maryland, and lead writer of the Live Sound Advice blog. He's a veteran audio educator and is also an adjunct professor at Shenandoah Conservatory in Winchester, VA. To ask questions and find out more about electrical issues, visit his AC Power & Grounding Forum on ProSoundWeb.*

Note: This article is provided as an educational assist in your gigs and is not intended to have you circumvent a licensed electrician or technician. If you even remotely believe you have a dangerous electrical condition on stage or on a project site, make sure to contact a qualified, licensed electrician to address it.

FRESH APPROACHES

New technologies in support of a range of applications. ***by Live Sound Staff***

Greg Price helping to make it happen for Black Sabbath with an Avid VENUE | S6L.

A NEW MIX PLATFORM FOR BLACK SABBATH'S FAREWELL

Veteran live engineers Greg Price and Myles Hale are utilizing Avid VENUE | S6L mix systems on Black Sabbath's farewell The End world tour at front of house and monitors, respectively. Price, who's worked with bands such as Van Halen, KISS, and Rage Against The Machine, has been mixing FOH for Black Sabbath since 1997.

I've been famous for Ozzy's 'vocal pyramid' – his vocal chain of plugins – but then I got my hands on the new console," Price notes. "Now I'm not using any plugins on Ozzy's channel, only a very modest channel EQ and an onboard brick wall limiter for when he screams into his mic. The sound is absolutely stunning. I've never heard Ozzy's vocal quality like this."

Hale adds, "As soon as the drummer [Tommy Clufetos] came in he said, 'Oh my god, what changed? These are the same speakers I've always had.' I explained to him that I wish I could say that I got better, but really it's this new console. It sounds like a badass analog console except with all the great user features that come with digital."

The S6L utilizes the same VENUE software as other Avid systems, enabling Price and Hale to quickly transition from the VENUE | Profile systems they've used on previous tours. Further, employing Pro Tools and VENUE's Virtual Soundcheck, they're able to fine-tune mixes FOH and monitor mixes during rehearsal. "In pre-production, I set up the S6L and Pro Tools and record everything the band does," explains Price. "After they play a song, they come straight into my room and we listen and we talk about everything. I cannot work without Pro Tools and the synergy that it offers my artists."

SUBWOOFER UPGRADE HELPS DAVID GILMORE RATTLE THAT LOCK

Clearwing Productions specified an L-Acoustics K1/K2 rig complemented by the company's new KS28 subwoofers in providing sound reinforcement for David Gilmour's Rattle That Lock North American tour. "KS28 is a marked improvement on the SB28, and I was not at all unhappy with the SB28," reports Gregg Brunclik, president and CEO of Milwaukee, WI-based Clearwing. The tour represented the official U.S. debut of the KS28, which extends frequency response down to 25 Hz, reduces

The L-Acoustics loudspeaker set in action during David Gilmore's North America tour.

weight and offers 3 dB greater output than its predecessor.

Gilmour is best known for the decades he spent with classic rock pioneers Pink Floyd, and his guitar work on tracks like "Money" and "Comfortably Numb" have been named as among the best in the history of rock music. Colin Norfield is the engineer tasked with bringing that sound to audiences, and he's also encouraged with the KS28.

"They are half-again as loud and 30 pounds lighter," Norfield states. "They have a tighter, deeper and richer sound, which helped to enhance David's show. The KS28 is great all-round and I'm looking forward to using them again on the next leg of the tour."

The subs were powered by the LA12X, a new amplified controller with a 4-in/4-out architecture that offers up to 3,300 watts per channel. So the sentence should read: It also includes open-standard AVB protocol for input of digital signals.

The full system complement for the tour included left and right main PA hangs of 14 K1 over four K2 per side, each backed by an array of 12 K1-SB subs. Out fill arrays were comprised of 10 K1 over eight K2 per side, with left and right extreme out fill arrays each made up of 12 Kara. Two dozen KS28 subs were ground-stacked double high across the face of the stage, while an LCR delay system of three hangs of six K2 covered the farthest seats.

MANAGING DIVERSE REQUIREMENTS FOR JUNUN IN LONDON

Radiohead's Jonny Greenwood and Israeli composer Shye Ben Tzur recently staged the first full UK performance of their new project, Junun, at the London Barbican, with front of house engineer Gavin McComb selecting an Allen & Heath dLive system to manage the diverse audio requirements. Junun comprises compositions performed by Ben Tzur and the 9-piece Rajasthan Express band, with Greenwood adding electronic elements.

"Having used iLive for many years, this was my first opportunity to use A&H's new dLive system," explains McComb. "The Junun act is a big sound with Indian drums, brass, guitars, harmonium, percussion and lots of vocals. For this gig there were 11 musicians on stage with a lot of open mics. There was a fantastic range of instrumentation to manage but these guys can seriously play, they're so tight as an ensemble. I also have complete confidence in the audio capabilities of A&H. Very little EQ was required for the instruments apart from having to contain the room a little where the overall GEQ couldn't without destroying large chunks of the sound."

The sold-out gig reunited the musicians for the first time since they recorded the album Junun last year in India with Radiohead producer Nigel Godrich. "I will most certainly be wanting to use the A&H dLive system on my next tour," McComb concludes. "It has all the best qualities required of a high-end console, and sounds even better."

CREDIT: PAUL WEBER

With a stage set for Junun *in the background, Gavin McComb at front of house with Allen & Heath dLive.*

LOUDER & CLEARER WITH IRON MAIDEN'S BOOK OF SOULS

Iconic British metal band Iron Maiden is traveling with a Meyer Sound LEO Family loudspeaker system on the Book of Souls World Tour. The six-month, 35-country tour began in February, with the system supplied by UK-based Major Tom and additional equipment support from Rainbow Production Services (Atkinson, NH).

"LEO's clarity and headroom help out a great deal when it comes to balancing the mix," reports front of house engineer Martin Walker. "Iron Maiden has three guitars, vocals, and occasional keyboards all vying for the same frequency bands in the mix. With LEO, the clarity cuts right through. That also allows me to mix louder than before."

For North American shows in "A-level" arenas of 15,000 to 20,000, a typical system configuration comprises dual hangs of 14 LEO-M over four LYON linear line array loudspeakers, 18 1100-LFC elements in a cardioid configuration, and a center stack of four 700-HP subwoofers. Out fill hangs are 15 LYON loudspeakers each, and twin arrays of six-each LEOPARD line arrays cover the rear stage. A Galileo Callisto loudspeaker management system supplies drive and optimization.

Walker notes that the band has been pleased with audience reactions to the sound, and appreciates the directed sound of

The Meyer Sound LEO Family awaiting the arrival of Iron Maiden at a tour stop.

the 1100-LFC low-frequency elements. "The band doesn't like to be soaked by sub-bass bleed back on the stage, and the low-end control I get from the 1100s takes care of that neatly," he says. "Out front, management has also noticed a difference this time out. They recently remarked to me, 'I don't know what this new rig is, but I love it.'"

QUICK SONIC TAILORING FOR PLACIDO DOMINGO IN MIAMI

The American Airlines Center recently provided the setting for Placido Domingo's first performance in the Miami area in over a decade, with locally owned Mix3 Sound providing the sold-out event with sound reinforcement headed by Anya and Anna Adaptive loudspeaker systems from EAW.

Jerry Eade, front of house engineer, notes, "It provided a really clean, pure vocal and a very natural, open sound. That's the mark of a great system. It was also easy to use – no fighting with EQ and masses of gain before feedback. I'd love to use it again."

John Ferlito, owner of Mix3 Sound, designed a main system that consisted of left-right columns, each made up of 12 Anya modules and six Anna modules for out fill. Twelve SB1002 subwoofers – 24 total – were stacked in a cardioid configuration under each column for additional low end.

Placido Domingo and friends performing in Miami, supported by EAW Anya and Anna.

"One of the reasons we invested in the Adaptive system was the flexibility it provides," explains Ferlito. "Because EAW's Resolution 2 software controls the processing of each acoustic cell individually, it is easy to modify the system for the unexpected. In this case, when the concert sold out, management decided, at the last minute, to open another seating section for 1,000 fans. Normally this would require physically reconfiguring the arrays to ensure coverage. In this case we merely modified the coverage parameters in the software and we were set."

POSITIVE MONITORING FOR HAMILTON ON BROADWAY

Hamilton – An American Musical, based on the life of one of the founding fathers of the U.S., is enjoying significant success on Broadway at the Richard Rodgers Theatre, with sound designer Nevin Steinberg opting to deploy Alcons Audio loudspeakers for monitoring.

The smash hit* Hamilton – An American Musical *on Broadway takes advantage of Alcons Audio for monitoring.

Specifically, Steinberg's design utilizes a dozen Alcons VR8 compact monitors and eight SR9 fill monitors. Both incorporate the company's proprietary and multiple-patented pro-ribbon transducer technology and are driven by Alcons Sentinel10 amplified loudspeaker controllers.

Winner of the 2016 Pulitzer Prize for drama and the 2016 Grammy for Best Musical Theater Album, the production's music includes sparse ballads, and via R&B and hip-hop, is completely "sung-through" – the lyrics provide the only plot information the audience receives. The show includes 21 cast members that are accompanied by a 10-piece orchestra.

"Dave Rahn at Alcons USA was incredibly helpful, making sure that I got a good listen to the products that I was interested in and then following through with the theatrical sound vendor, PRG Audio, to co-ordinate priority shipping and on-time delivery for our production schedule on Broadway," says Steinberg. "Audience response to the sound has been very positive, along with their response to the rest of the production. The show is quite special, and the people who have seen it are pretty overwhelmed by the experience. I feel very lucky to be a part of it." LSI

PRACTICAL SOLUTIONS

Lavalier and podium microphone design, applications and more.

by Gary Parks

Though not as "sexy" as handheld or headset microphones, lavalier and podium mics deliver practical solutions in sound reinforcement, especially for spoken-word applications.

Presenters often need to be hands-free, and in some cases are not comfortable with headworn mics or must have their faces completely unobstructed. In live theatre, the latest generation of tiny yet high-quality lavalier mics, combined with wireless transmitters, have rendered excellent audio capture that's virtually invisible. Some models are as small as 0.1 of an inch (2.5 mm) in diameter – not much larger than the cable itself.

Lavs by their nature are placed farther away from the voice than a headset or handheld mic – more distant from the direct output of the mouth, where the higher overtones of the voice emanate. To compensate for this less-than-ideal placement, they usually have a shaped frequency response that includes a high-frequency boost and sometimes a midrange dip to alleviate the effects of chest resonance.

Alternately, several have exchangeable caps which provide different levels of HF boost acoustically. Some will also offer a built-in or switchable low-frequency rolloff to dispel the pickup of handling and other noise sources.

Lavs are available in both omnidirectional and directional coverage patterns to suit a variety of applications. Omni models typically sound more natural when placed off-axis to the voice, and they have fewer problems with level and frequency response variations as the presenter's head moves. However, they're more prone to capturing ambient noise with lower gain before feedback, and are susceptible to cable and handling noise.

Directional models are much more sensitive to handling noise, breath, and plosive consonants, with the trade-off being that they're more selective – capturing the voice while attenuating ambient sounds with higher gain before feedback. When chest-worn, they should be aimed right at the mouth, and the presenter should try to minimize extreme head turns when speaking. Head movement is less problematic when the mic is fixed to the hairline or otherwise secured so that it moves with the presenter.

Some manufacturers will utilize the same miniature mic element, or a variation, in multiple applications – for example, as a lav, headset, and instrument mic. And they may be offered with different sensitivities and maximum SPL capabilities to increase their versatility. Detachable cables are available with some brands so that the same mic over time can be applied to wired or wireless uses, utilizing a variety of connectors and transmitters.

Podium (or gooseneck) mics can be attached to a lectern, table, or other surface, aimed in the basic direction of the presenter's mouth. They allow multiple people to come to the podium and talk hands-free, and offer easy adjustment for height and positioning. They're most common in church, government, corporate, and similar presentation settings.

A handful of the most recent generation of podium mics offer modularity, with elements offering a variety of polar patterns able to be mounted on goosenecks of different lengths, and also connected to permanently-installed or tabletop desk mounts. One manufacturer's model offers a choice of omni, cardioid, or hypercardioid heads, as well as a version with selectable polar patterns. Another has a mini shotgun-style head for very focused audio pickup at moderate distances. Other variants are available with longer goosenecks for choir and orchestral miking, offering wide frequency response, well-controlled polar patterns, flexible yet stable booms, and low handling noise.

Podium mics are designed to pick up the presenter's voice from a distance of several inches, and in many cases to present a wide enough polar pattern to allow the user some movement side-to-side. Goosenecks vary in length from under a foot to almost two feet, with typical sizes of 12, 18, and 24 inches; some models also provide a shorter option. Because they're typically used above a hard, reflective surface, the positioning of the mic element and its nulls relative to that surface and any nearby loudspeakers can make a difference in audio quality, intelligibility, and gain before feedback.

The roundup that follows presents some of the latest examples of lavalier and podium microphones from a variety of manufacturers. LSI

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

LAVALIER MICROPHONES

Audio-Technica AT898 | *www.audio-technica.com*

Transducer Type: Condenser
Polar Pattern: Cardioid
Frequency Response: 200 Hz – 15 kHz (LF rolloff at 80 Hz, 12 dB/octave)
Sensitivity: -43 dBV (7 mV/Pa)
S/N Ratio: 63 dB
Maximum SPL: 131 dB
Power Requirement: 11 – 52 VDC
Dimensions (L x D): 0.91 x 0.21 in (23 x 5.3 mm)
Weight: 0.03 oz (0.9 g)
Cable Length: 9.75 feet (3.0 m)
Connectors: TA3F
Notes: Battery power module offered; switchable LF rolloff; variety of accessory clips; AT899 omni and BP896 MicroPoint omni also available.

Shure Microflex MX185 | *www.shure.com*

Transducer Type: Condenser
Polar Pattern: Cardioid
Frequency Response: 50 Hz – 17 kHz
Sensitivity: -35 dBV (17 mV/Pa)
S/N Ratio: 66 dB
Maximum SPL: 124 dB
Power Requirement: 11 – 52 VDC
Dimensions (L x D): 0.85 x 0.46 in (22 mm x 12 mm)
Weight: N/A
Cable Length: 4 ft (1.2 m)
Connectors: TA4F
Notes: Available in black, tan, cocoa; model MX183 omni and MX184 supercardioid also offered; balanced, transformerless output for increased immunity to noise over long cable runs; also MX150 subminiature lav in omni and cardioid configurations.

Countryman B6 | *www.countryman.com*

Transducer Type: Condenser
Polar Pattern: Omni
Frequency Response: 20 Hz – 20 kHz
Sensitivity: 3 models: -36/-43 /-54 dBV (16/7/2 mV/Pa)
S/N Ratio: Equivalent acoustic noise – 24/29/39 dBA SPL
Maximum SPL: 120/130/140 dB
Power Requirement: 1 – 2 VDC
Dimensions (L x D): 0.23 x 0.1 in (5.8 x 2.5 mm)
Weight: 0.07 oz (2 g)
Cable Length: 5 ft (1.5 m)
Connectors: TA4F and other connectors available
Notes: Five colors available; moisture resistant; three end caps provided for HF enhancement (flat, +4 dB, and +8 dB); models with three different sensitivities; B2 directional, B3 omni, and EMW omni also available.

DPA d:screet | *www.dpamicrophones.com*

Transducer Type: Condenser
Polar Pattern: Omni
Frequency Response: 20 Hz – 20 kHz
Sensitivity: -34 dB (20 mV/Pa)
S/N Ratio: 71 dB, equivalent noise level – 23 dB (A)
Maximum SPL: 134 dB
Power Requirement: 5 – 50 VDC
Dimensions (L x D): 0.67 x 0.2 in (12.7 x 5.4 mm)
Weight: 0.26 oz (7.5 g), including cable and connector
Cable Length: 6 ft (1.8 m)
Connectors: MicroDot
Notes: Available in three sensitivities (models 4060, 4061, 4062); model 4071 has acoustic pre-equalization for presence boost; cardioid model offered, along with miniature omni and directional headset and instrument mics.

Point Source Audio CO-8WL | *www.point-sourceaudio.com*

Transducer Type: Condenser
Polar Pattern: Omnidirectional
Frequency Response: 20 Hz – 20 kHz
Sensitivity: 43 dB (1V/Pa)
S/N Ratio: N/A

Maximum SPL: 148 dB
Power Requirement: 1 – 10 VDC
Dimensions (diameter): 0.16 in (4 mm)
Weight: 0.6 oz (17 g)
Cable Length: 4 ft (1.2 m)
Connectors: Interchangeable X-Connectors
Notes: Available in white, beige, brown, black; IP57 waterproof rating; connector options include TA3F, TA4F, Hirose, Lemo, others; CR-8L cardioid also offered.

Mipro MU-55L | *www.mipro.com.tw*

Transducer Type: Condenser
Polar Pattern: Omni
Frequency Response: 40 Hz – 20 kHz
Sensitivity: -49 dBV
S/N Ratio: N/A
Maximum SPL: 135 dB
Power Requirement: +48 V
Dimensions (L x D): 0.39 x 0.18 in (10 x 4.5 mm)
Weight: 0.5 oz (14 g)
Cable Length: 5 ft (1.5 m)
Connectors: TA4F
Notes: Available in black or beige; headset also available.

Electro-Voice RE92L | *www.electrovoice.com*

Transducer Type: Condenser
Polar Pattern: Cardioid
Frequency Response: 40 Hz – 20 kHz
Sensitivity: -54 dBV (5.6 mV/Pa)
S/N Ratio: 64 dB
Maximum SPL: 135 dB
Power Requirement: 24 – 52 VDC
Dimensions (L x D): 0.95 x 0.41 in (24 x 10.5 mm)
Weight: 5.6 oz, including cable and inline module connector
Cable Length: 4 ft (1.2 m)
Connectors: XLR-3M
Notes: 12 dB/octave switchable high-pass filter; transformerless differential output to drive long cables; OLM10 omni lavalier also available.

Sennheiser ME-2 | *www.sennheiserusa.com*

Transducer Type: Condenser
Polar Pattern: Omni
Frequency Response: 30 Hz – 20 kHz
Sensitivity: -34 dB (20 mV/Pa)
S/N Ratio: 36 dB equivalent noise level
Maximum SPL: 130 dB
Power Requirement: 7.5 V
Dimensions (L x D): 0.35 x 0.26 in (9 x 6.5 mm)
Weight: N/A
Cable Length: 5.25 ft (1.6 m)
Connectors: Locking mini-jack (for Evolution Tx)
Notes: Available in black; used with D1 digital and other Sennheiser wireless.

AKG MicroLite LC81 MD | *www.akg.com*

Transducer Type: Condenser
Polar Pattern: Cardioid
Frequency Response: 400 Hz – 18 kHz (+/-6 dB)
Sensitivity: -39.2 dBV (11 mV/Pa)
S/N Ratio: 54 dB
Maximum SPL: 118 dB
Power Requirement: 5 VDC
Dimensions (L x D): 0.39 x 0.19 in (10 x 4.8 mm)
Weight: 0.07 oz (2 g; 4.9 g including cable/connector)
Cable Length: N/A
Connectors: Microdot for connection to wireless Tx
Notes: Moisture-resistant design; colors include black, white, beige, cocoa; LC82 omni model also available; three mounting styles for capsule (lav, ear-hook, headset).

beyerdynamic TG L34c
www.north-america.beyerdynamic.com

Transducer Type: Condenser
Polar Pattern: Cardioid
Frequency Response: 200 Hz – 18 kHz (30 Hz close miking)
Sensitivity: -40 dBV (10 mV/Pa)
S/N Ratio: 56 dB
Maximum SPL: 136 dB
Power Requirement: 3 – 9 VDC
Dimensions (L x D): 1.34 x .43 in (34 x 11 mm)
Weight: 17.3 g (with cable and connector)
Cable Length: 4.76 ft (1.5 m)
Connectors: 4-pin Tiny QG
Notes: Optional power adapter for wired operation; TG L55c omni with moisture-resistant design also available, choice of black or tan.

Sanken COS-11D
www.sanken-mic.com

Transducer Type: Condenser
Polar Pattern: Omni
Frequency Response: 50 Hz – 20 kHz
Sensitivity: -35 dB (17.8 mV/Pa)
S/N Ratio: 28 dB (A) equivalent noise level
Maximum SPL: 127 dB
Power Requirement: +48 V
Dimensions (L x D): 0.63 x 0.16 in (16.1 x 4 mm)
Weight: 2.9 oz (82 g), including cable and connector
Cable Length: 10 ft (3.0 m)
Connectors: XLR-3M
Notes: Available in white, beige, gray, black; moisture resistant; also offered with connectors for wireless transmitters.

Sony ECM77B
pro.sony.com

Transducer Type: Condenser
Polar Pattern: Omni
Frequency Response: 40 Hz – 20kHz
Sensitivity: -36.5 dBV (15 mV/Pa)
S/N Ratio: 64 dB
Maximum SPL: 120 dB
Power Requirement: +48 VDC
Dimensions (L x D): 0.5 x 0.25 in (12.7 x 6.4 mm)
Weight: 4.3 oz (including cable/power supply/connector)
Cable Length: 9.8 feet
Connectors: XLR3M
Notes: Power with AA battery or phantom; ECM44B omni also available.

Audio-Technica ES915SML | *www.audio-technica.com*

Transducer Type: Condenser
Polar Pattern: 90 degrees MicroLine, interchangeable elements for omni, cardioid, hyper
Frequency Response: 30 Hz – 20 kHz
Sensitivity: -35 dBV (17.7 mV/Pa)
S/N Ratio: 70 dB
Maximum SPL: 133 dB
Power Requirement: 11 – 52 VDC
Dimensions: 16.7, 19.7 and 22.7 inch lengths
Weight: 4.6 – 4.9 oz (depending on model)
Connectors: XLR-3M
Notes: Power module at gooseneck base provides mute and LF rolloff switches; universal isolation mount and stand adapter included; optional ESE interchangeable mic elements for omni, cardioid, hypercardioid.

Shure Microflex Series | *www.shure.com*

Transducer Type: Condenser
Polar Pattern: Cardioid, supercardioid (modular design)
Frequency Response: 50 Hz – 17 kHz
Sensitivity: -35 dBV (17.8 mV/Pa)
S/N Ratio: 66 dB
Maximum SPL: 122 dB
Power Requirement: +48 VDC
Dimensions: 5, 10, 15 inch lengths
Weight: N/A
Connectors: XLR-3M
Notes: Standard 12 and 18 inch goosenecks also available, as well as models with attached or in-line preamp, or desktop base; also Centraverse cardioid goosenecks in same standard lengths.

Countryman A3 Series | *www.countryman.com*

Transducer Type: Condenser
Polar Pattern: Cardioid, hypercardioid, omni, or switchable pattern
Frequency Response: 70 Hz – 16 kHz (cardioid)
Sensitivity: -35 dBV (17.7 mV/Pa)
S/N Ratio: 14 dB(A) equivalent input noise
Maximum SPL: 130 dB
Power Requirement: +48 VDC
Dimensions: 12, 18 and 24 inch lengths
Weight: N/A
Connectors: XLR-3M
Notes: Dual-element mic design; Active Plus Vibration Cancellation technology; fixed or switchable polar patterns; optional mounting flange available.

Earthworks FlexMic Series
www.earthworksaudio.com

Transducer Type: Condenser
Polar Pattern: Cardioid (or hypercardioid)
Frequency Response: 50 Hz – 20 kHz
Sensitivity: -40 dBV (10 mV/Pa)
S/N Ratio: 22 dB equivalent input noise
Maximum SPL: 145 dB
Power Requirement: +48 VDC
Dimensions: 13, 19, 23, and 27 inch lengths
Weight: N/A
Connectors: XLR-3M
Notes: High Definition series with 50 Hz – 40 kHz frequency response also available, in addition to IM installation series with 3, 6, 10, and 12 inch goosenecks and LumiComm touch ring sensor.

Electro-Voice RE90P Series | *www.electrovoice.com*

Transducer Type: Condenser
Polar Pattern: Cardioid
Frequency Response: 100 Hz – 15 kHz
Sensitivity: -47 dB (4.5 mV/Pa)
S/N Ratio: 28 dB equivalent output noise
Maximum SPL: N/A
Power Requirement: 9 – 52 VDC
Dimensions: 12 and 18 inch
Weight: 6.4 and 8.6 oz
Connectors: XLR-3M
Notes: Flange or shock mounts available; 200-ohm output for long cable runs; dual-element PolarChoice selectable polar pattern series (switchable omni, cardioid, super and hyper patterns) also offered in 12 and 18 inch lengths with integral mounts.

Sennheiser ME/MZH Series | *www.sennheiser.com*

Transducer Type: Condenser
Polar Pattern: Cardioid, supercardioid or hypercardioid (modular design)
Frequency Response: 40 Hz – 20 kHz (cardioid)
Sensitivity: N/A
S/N Ratio: 26 dB (A) equivalent noise level
Maximum SPL: 130 dB
Power Requirement: 12 – 48 VDC
Dimensions: 5.9 and 15.7 inch lengths
Weight: N/A
Connectors: XLR-3M
Notes: Modular design, with mating goosenecks and mic elements with different polar patterns; ME36 mic element is a mini-shotgun style.

AKG DAM+ | *www.akg.com*

Transducer Type: Condenser
Polar Pattern: Cardioid (also super and hypercarioid)
Frequency Response: 60 Hz – 22 kHz
Sensitivity: -46 dBV (5 mV/Pa)
S/N Ratio: 64 dB
Maximum SPL: 142 dB
Power Requirement: +48 V
Dimensions: 6.5, 12, 20 and 60 inch lengths
Weight: From 2.8 – 4.2 oz (6.5 – 20 in models)
Connectors: XLR-3M
Notes: Cardioid, supercardioid, and shotgun mic heads; modular connector bases and mounting options.

Audix MG Series | *www.audixusa.com*

Transducer Type: Condenser
Polar Pattern: Cardioid (or hypercardioid)
Frequency Response: 60 Hz – 19 kHz
Sensitivity: -29.9 dBV (32 mV/Pa)
S/N Ratio: 72 dB
Maximum SPL: 130 dB
Power Requirement: 18 – 52 VDC
Dimensions: 16.3, 18.9 and 22.4 inch lengths
Weight: 4.2 – 4.7 oz (depending on model)
Connectors: XLR-3M
Notes: Mounting flange supplied; optional table stand, shock-mount; optional replacement capsules for hypercardioid, supercardioid, omni.

beyerdynamic Classis GM Series
www.north-america.beyerdynamic.com

Transducer Type: Condenser
Polar Pattern: Cardioid
Frequency Response: 50 Hz – 19 kHz
Sensitivity: -35.4 dBV (17 mV/Pa)
S/N Ratio: 69 dB
Maximum SPL: 107 dB
Power Requirement: 8 – 52 VDC
Dimensions: 8, 12, 16, 20 and 24 inch lengths
Weight: N/A
Connectors: XLR-3M
Notes: Versions with mute switch available; mounting accessories available.

DPA d:screet SC4098
www.dpamicrophones.com

Transducer Type: Condenser
Polar Pattern: Supercardioid
Frequency Response: 20 Hz – 20 kHz
Sensitivity: -36 dBV (16 mV at 1 Pa)
S/N Ratio: 71 dB (A)
Maximum SPL: 135 dB
Power Requirement: Phantom power
Dimensions: Various lengths available
Connectors: XLR or MicroDot
Notes: Available in several variants – comes in either black or white, with either an XLR or a MicroDot termination and in five lengths ranging from 6 to 48 inches; several table mount options available.

CAD Audio 2800VP | *www.cadaudio.com*

Transducer Type: Condenser
Polar Pattern: Continuously variable (remote control)
Frequency Response: 40 Hz – 20 kHz
Sensitivity: -29 dBV (35mV at 1Pa)
Maximum SPL: 110 dB
Power Requirement: P12, P24, P48, 4mA
Dimensions: 18 inch length
Connectors: TA3F type and XLRM type
Notes: Variable polar pattern with remote control allows users to adjust the shape of the pick-up pattern from within a companion DSP unit.

Milab VM-44 | *www.milabmic.com*

Transducer Type: Condenser
Polar Pattern: Cardioid (omni, supercardioid also available)
Frequency Response: 20 Hz – 20 kHz
Sensitivity: N/A
Maximum SPL: 128 dB (140 dB w/12 dB pad)
Power Requirement: 12 – 52 V
Dimensions: 8.3 inch length
Connectors: XLR
Notes: XLR-Amp fits direct to mixer; three capsule options, interchangeable; includes 12 dB attenuation pad; phantom power supply and table stand offered as options.

Chris Humphrey has joined **QSC** as vice president, marketing, where he is overseeing all of the company's global marketing activities and initiatives for the professional, systems and cinema divisions. He brings a career of global marketing experience in technology-oriented businesses to QSC, with more than 20 years of executive marketing roles at companies which focus on IT related services, including enterprise software, embedded computing, storage networking, scientific instrumentation and security. Humphrey is based at QSC headquarters in Costa Mesa, CA.

Adamson Systems has expanded its North American support team, with **Richard Woida** (pictured left) named to the U.S applications group and **Scott Shields** (right) now serving as sales and marketing coordinator. Woida, who is based in Minneapolis, spent the previous six years in Dubai, UAE, working as a systems specialist for Clair Solutions through NMK Electronics and also heading the audio department at Lighthouse Production Middle East. Shields, who holds a degree in mechanical engineering/product design from Georgian College, comes to Adamson after serving as PA rental administrator at The Arts Music Store located in Newmarket, Ontario.

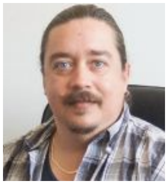
"We're a company that's driven by the love of audio and not by financial investors," says **James Oliver**, Adamson director of marketing and sales. "We've grown organically by bringing in individuals of the highest character and technical skill. Both Richard and Scott fit the mold and they're perfectly suited to support and continue to grow our rapidly expanding user base."

Jamie Griffin has joined **Shure** as vice president of operations, responsible for all of the company's global manufacturing and plants around the world; global supply chain; operations program management; and process, tool, and automated test engineering.

Originally from Ireland, he has lived and worked in several countries across Asia, Europe, and the U.S., and comes to Shure from Lenovo, where he was director of operations. Griffin also he spent a decade with Dell, holding senior positions in operations and engineering while designing and launching manufacturing facilities in China, Malaysia, and Brazil.

Meyer Sound announces the expansion of its worldwide sales team, with **Andy Willcox** (pictured left) serving as sales manager, residential systems and **Daniel Rivera** (right) onboard as sales specialist, house of worship market.

In addition, the company has named **Antonio Manzo** as sales director, Mexico. He brings more than 25 years of experience in the music and pro audio industries to the position, and for nearly 22 years, worked at Yamaha Corporation of Mexico, taking on new market development in the company's sales and marketing department.



Joe Fustolo has joined **Renkus-Heinz** as application engineer. An experienced engineer with roots in live and installed sound, he has worked extensively with loudspeaker technology, including more than a decade providing applications support for EAW. He's also held technical positions with FBT and Outline as well as production roles with major concert and theatrical companies, along with live venues.

Fustolo is working from his offices in Massachusetts, offering support to R-H clients east of the Mississippi.

M&L Sound (Knoxville, TN) has joined the **L-Acoustics** Rental Network as the manufacturer's latest Certified Provider (CPr). Headed by president **David Akers**, the company has provided concert sound reinforcement, rental and installation services throughout the mid-southern U.S. for more than 35 years.

"When grading all of the major loudspeaker brands and systems on the market by a specific set of criteria – sound quality, weight-to-output ratio, phase coherency, vertical splay angles, horizontal coverage, ease of deployment, rental network strength and quality of customer service– L-Acoustics and its K2 clearly emerged as the winner on all fronts," notes M&L production manager **Justin Slazas**, who adds that the company's initial investment in the brand comprises 24 K2 and 12 ARCS WiFo enclosures, 16 SB28 subs and 15 LA8 amplified controllers housed in five LA-RAKs.

Amazing Audio (Ann Arbor, MI) recently utilized nine **Radio Active Designs** UV-1G wireless intercom systems for press events held before and during the annual North American International Auto Show (NAIAS) in Detroit. Amazing Audio's

Jeff Jones provided RF coordination for the event, managing 603 frequencies during press week, while seven auto manufacturers also hosted press events that utilized the RAD intercom systems.

"Given how crowded the RF spectrum was, it would have been impossible to deploy enough RF intercoms," explains Jones. "Fortunately, RAD systems operate primarily in VHF with the base stations taking up only a sliver of UHF bandwidth. So we were able to use nine base stations and 54 RAD packs, which helped tremendously when coordinating press events of this size."

Waves Audio has announced that **Plugin Alliance** will now be SoundGrid-compatible with the release of its 100% SoundGrid Bundle V1.0. The package delivers five processors from four respected entities, with each plugin tuned and tested to run on SoundGrid to optimally meet the needs of front of house and monitor engineers.

"We're really pleased to be able to bring products from key Plugin Alliance member companies like Brainworx, SPL, elysia and Mäag Audio to the Waves SoundGrid platform," says Plugin Alliance CEO **Matt Ward**. "We've had many users from the enormous SoundGrid user base ask us when we will have plugins for them, and we couldn't be happier to be able to say, 'now.'"

Anderson Audio (Harrisburg, PA) recently added an **EAW** Anna Adaptive system to its inventory. Specifically, 24 Anna modules and eight companion Otto subwoofers join a host of other EAW gear already owned by the company, including KF730, KF740 and NTL720 line arrays, SB1000, SB2001 and NTL250 subwoofers, and JF and Redline Series loudspeakers.

"This is a flagship system for us," states Chris Anderson, owner of the Anderson Audio. "The new Anna PA won't replace our current systems; instead it will give us the opportunity to build a new client base and get work that we haven't won before." (Anderson is pictured second from left with EAW's Jim Newhouse, Rusty Waite and TJ Smith joined by some of the new inventory.) LSI

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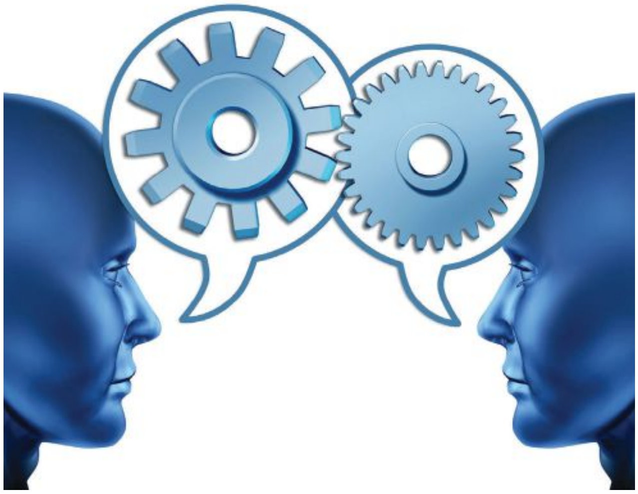
MODULAR GENIUS

Using existing tools to do cool things. *by Jonah Altrove*



Every time a new audio innovation arises, it's met by a certain amount of resistance. This is ironic and puzzling to me, seeing as our field is driven almost entirely by technology. What motivates this response?

Perhaps fear that **technology will replace us**. Are system techs quaking in their boots, fearful of being replaced by a system processor with auto-EQ abilities? How many engineers have become unemployed as a result of automixer technology?

More likely, I think it's fear that the **technology will make us lazy**. I know an engineer who won't use a dynamic equalizer because it's "too easy." He also thinks that RTAs and FFT programs such as Smaart (not to mention consoles with built-in analyzers) make it possible for "anyone to be a monitor engineer" since there's now no need to "learn your frequencies." While this may be true at the low end of the spectrum (no pun intended), I doubt that any professional-caliber monitor engineer feels that his livelihood is threatened by a $200 feedback suppressor unit.

This is an interesting point, though. *Does* technology make us lazy? Does the prevalence of Spell Check mean that today's generation of kids don't have to learn to spell? Do our ever-present smartphones absolve us of the responsibility to memorize important phone numbers or do mathematical calculations by hand?

With audio technology advancing at such a rapid rate, perhaps too much emphasis is put on workflows that are *changed* as a result of new tech, and not enough emphasis on brilliant new uses of the new technology to enhance our jobs… in other words, finding the art among the science.

The excitement lies, for me, in discovering new uses for existing technology to do cool things. GPS is nothing new, and neither are camera phones or online restaurant reviews. But when you combine the three, you get a smartphone app that will offer up dining suggestions simply by panning your phone's camera across the storefronts.

Likewise, online banking has been around for decades, and Optical Character Recognition is coming into its own… nothing new or exciting there, right? But what about the bank apps which let you photograph a check with your phone and deposit the funds into your account? *That's cool!*

These are not new inventions, but rather "cobbling together" chunks of existing technologies to allow us to do new, creative, amazing things.

In my mind, one of the figures at the forefront of this "modular genius" is Dave Rat, whose "Rat-isms" include obtaining an accurate idea of a device's latency by listening to its signal in one ear while the "dry" signal panned to the other ear is increasingly delayed. When it sounds "in the middle," the added delay is the device's latency.

According to Dave, this resulted, in some cases, in a measurement that was more accurate than the manufacturer's stated values. He also used water and bags of salt to ground the stage at a Red Hot Chili Peppers show and saved the band from problematic RF. (We all know that saltwater is electrically conductive, but who would have thought to apply that knowledge in such a way?)

The underlying principles of these examples are basic, known by any engineer worth his or her salt (OK, that pun was intended). The brilliance, to me, lies in the application of astoundingly simple principles to find ways to do really cool stuff.

That feeling of "Wow! That's so cool!" is the audio engineer's equivalent of watching a magic show. We've all experienced that moment of excitement when we learn of a new technique or application that's just plain neat. As our technologies and gear continue to advance, let's not feel threatened and "disarmed." Instead, let's flip it on its head and blow people's minds. LSI

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