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

LIVE SOUND INTERNATIONAL

INSIDE

THE MODERN MIX EXPERIENCE

Applications of digital console platforms.

INSIDE DIGITAL & ANALOG INTERFACES

THE LATEST CAPABILITIES OF DI BOXES

DYNAMIC & CONDENSER MIC SENSITIVITY

25th ANNIVERSARY


NO COMPROMISE.
DIGITAL 6000.
HEADPHONES
VOLUME
SYNC
SKM 6000
510.000 MHz
4:34
ESC
SAVE
SYNC
2000 | 5000 | Digital 6000 | Digital 9000
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AXIS
DIGITAL MIXING SYSTEM

Anticipating the needs of the mobile production and system integration markets takes a terrific amount of user research and a delicate understanding of the multi-faceted problems that mid-size companies face in this economy. Simply put, they are doing more with less. Budgets are stretched and staffing can be slim.

So, when embarking on a joint project to help these companies succeed, EAW and Mackie focused on the number one pain point, knowing the right products could make a tremendous difference in day-to-day operations for countless hungry production and install companies. With the EAW RADIUS system, setup time and effort is dramatically reduced.

RADIUS™ with EAWmosaic
Smarter sound. Made simple.

- Compact array featuring OptiLogic™ array detection and optimization
- Powerful EAWmosaic™ control app for wireless system control with smart prediction and room design tools
- Hallmark EAW sonic quality featuring Focusing™ and DynO™ DSP
- Complete Dante™ integration

The RADIUS compact line array features OptiLogic™ array detection and optimization. A clever combination of integrated infrared transceivers and tilt sensors allows each array module to know its exact position and splay angle within the array. The modules are grouped accordingly and DSP optimizes the systems acoustical output to compensate for array size, audience geometry and throw. This can also be controlled using the EAWmosaic™ control app, featuring complete wireless system control plus highly approachable prediction and room design. All of this equates to quicker setup and the ability to dial in amazing sound with very little effort. Having the entire AXIS + RADIUS system on the Dante™ network also saves setup time and drastically simplifies routing.

At FOH and beyond lies the Mackie AXIS Digital Mixing System, a modular 32-channel design featuring the sleek DC16 control surface. With AXIS, Mackie concentrated on creating a product that delivers an incredibly fast workflow through visual feedback and customization to reduce setup time, allowing engineers at any level to dial in their mix quickly. The control surface layout is extremely intuitive and features tons of high-resolution, full-color channel ID screens. Having a multitude of small screens delivers information right where it's needed, so the engineer is never looking for anything, it is always there. Adding to the AXIS system flexibility is the SmartBridge™ design which allows up to three iPads® to dock in the control surface, allowing

MACKIE.
Mackie.com

"With the AXIS and RADIUS systems, Mackie and EAW have provided us with a complete set of tools to serve our clients with excellence."

APPLICATION PROFILE: BALLARD SEAFOODFEST

For more than 42 years, the Ballard SeafoodFest has celebrated one of Seattle's unique neighborhoods with great food, beer and (of course) tons of live music.

"With 75,000 visitors over the course of a two-day weekend, this is a pretty complicated setup," remarks the event's Executive Director, Mike Stewart. "It's critical that the system can get setup quickly and efficiently in a dense urban environment."

To support the wide range of musical performances throughout the multi-day festival, EAW and Mackie debuted their new end-to-end audio solution made up of EAW RADIUS loudspeakers and Mackie's AXIS Digital Mixing System.

The EAW RADIUS PA consisted of left-right hangs of six RSX208L line array modules reinforced by 12 RSX18 subwoofers stacked six per side. Front fill was managed by two RSX89 loudspeakers. Each module provides up to 125 dB whole space SPL ensuring more than enough bandwidth to handle any performance type. The RSX18 delivered driving low end compliments of an 18-inch cone (3-inch voice coil) loaded in a vented enclosure.

And, at FOH, Mackie harnessed the power of their new modular AXIS system, combining the power of the 32-channel DL32R Rackmount Digital Mixer and innovative DC16 Control Surface to deliver a live sound solution with the efficiency fast-paced festival stages demand. Thirty-two remote-controllable Onyx+ mic preamps and 18 outputs with built-in DSP can handle the most rigorous schedules.

"This year everyone noticed the sound quality – which was pretty incredible," concludes Stewart. ■

control over multiple channels at once. AXIS intelligently senses when an iPad is in place and its operation adjusts accordingly. A user can grab an iPad and walk the room for system tuning or dial in monitors. When they return to FOH and dock the iPad, it reverts to the previously chosen state. These are just a couple of the innovative features that help AXIS drive efficiency for any application.

AXIS

Better workflow through innovation.

- Flexible modular system with unmatched speed, visibility and customization
- Intelligent surface-to-wireless mixing via SmartBridge™ and Master Fader™ control app
- Flexible 32x32 recording/playback
- Complete Dante™ integration

The combination of AXIS + RADIUS is a powerful solution for any medium-sized mobile production or system integration company, delivering world-class sound and unmatched workflow speed to allow you to do more and do it all more easily and quickly. ■

~ Corner Audio & Video,
Nashville full-service design & build company

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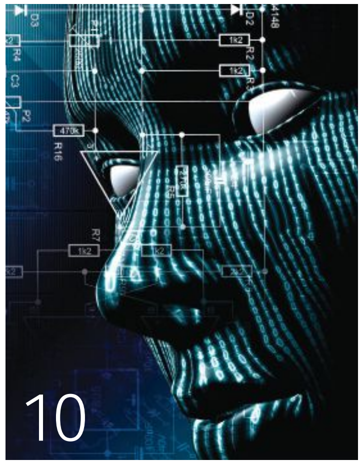
EAW.com

In This Issue

DECEMBER 2016 | VOL. 25, NO. 12

25th ANNIVERSARY
LIVE SOUND INTERNATIONAL

CREDIT: RALPH LARMANN



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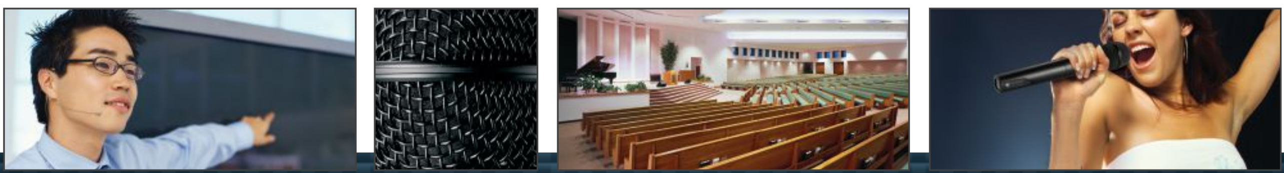
DIGITAL 2.4 GHz



From the Editor's Desk

THE PHRASE "DO THINGS THE RIGHT WAY" is well understood, but making the "right way" happen can come from taking different approaches that end up in the same place. In other words, there can be several "right ways" to do something the "right way." The expectation is excellence regardless of the path taken.

I mention this because in virtually every issue of this publication for more than five years, Craig Leerman has presented a range of insights on performing the role of audio professional at its highest level; a.k.a., the right way. What I find most interesting is that he often focuses on the "little things" that add up to a successful big picture.

On the Back Page of this issue, Craig's at it again, noting a range of "must haves" that, if overlooked, can lead to some pretty significant consequences on gigs. He also checks in with a look at direct boxes as well as some handy information on miking acoustic instruments.

Elsewhere in the issue, enjoy Merlijn van Veen's take on an interesting performance facet of condenser and dynamic microphones, as well as Jonah Altrove detailing his sound check process. Also don't miss Pat Brown's in-depth focus on the differences between analog and digital interfaces. Talk about doing something the right way... (sorry, couldn't resist).

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

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

ON THE COVER: Front of house engineer Garry Brown with the Yamaha RIVAGE PM10 console he's utilizing with Phish on tour.
(Photo by René Huemer)

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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: Pat Brown | Kevin Young | John Murray
Jonah Altrove | Gary Parks

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
CREATIVE SERVICES DIRECTOR Manuela Rosengard
mrosengard@ehpub.com | 508.663.1500 x226
AD PRODUCTION MANAGER Jason Litchfield
jlitchfield@ehpub.com | 508.663.1500 x252
JR. PRODUCTION DESIGNER Amanda Winitzer
awinitzer@ehpub.com | 508.663.1500 x478
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
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(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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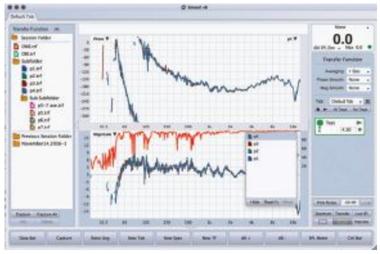
Sennheiser Digital 6000 Series

A digital 2-channel wireless receiver in two different versions, along with a bodypack and a handheld transmitter as well as a rack-mount charging unit. The receiver works across a switching bandwidth of 244 MHz (470 to 714 MHz), which is covered by three transmitter versions (470 – 558 MHz, 550 – 638 MHz, and 630 – 718 MHz). True bit diversity enhances reception. Digital 6000 also incorporates AES 256 encryption for data security, and works with standard active and passive UHF antennas. A Dante version of the receiver is also available. ***www.sennheiserusa.com***

Roland Pro AV VR-4HD

Integrates a digital audio mixer, video switcher, multi-viewer touch screen, and USB video/audio streaming interface into a stand-alone device. The 18-channel audio mixer includes the company's DSP, offering 3-band parametric EQ, reverb, a compressor/gate on the microphone inputs and level/multiband EQ on the master mix. It also supports auto-mixing, echo cancel, audio follow, and up to 500 ms of delay. It can mix the four XLR mic inputs, unbalanced stereo inputs on RCA, and 1/8 mini connectors, along with the audio embedded in the four HDMI inputs. The final mix can be output via XLR or RCA and also embedded into the HDMI output. A headphone output facilitates monitoring. Loop-back capability via the USB 3.0 port can bring in audio from a connected PC using conferencing software such as Skype and return audio to the PC without feedback due to an echo cancellation feature. ***proav.roland.com***

Rational Acoustics Smaart v8.1



A free update recommended for registered Smaart v8 users includes a rework of data handling, the addition of a multi-spectrum plot view, a built-in program updater, automatic broadband meter configuration based on selected inputs, password protection for the API, and the ability to run measurement engines in the background when switching between tabs. A new component within data handling is the Session Folder for improved day-to-day and gig-to-gig data management. v8.1 can be downloaded by users by logging into their license management accounts at the company website. ***www.rationalacoustics.com***

Allen & Heath dLive iPad Apps

Two apps for the company's dLive digital mixing platform. dLive MixPad provides remote control of the live mix while dLive OneMix focuses on personal monitoring.

MixPad offers control capability over channel processing, including filters, gates, parametric and graphic EQ, compressors and input/output delays, and it also supplies instant access to any of the mixer's channel faders and mutes, DCA faders and mutes, pan, aux sends and assignments, as well as mic preamp control and metering. In addition, it includes a real-time analyzer (RTA) to assist in the EQ process. OneMix locks control to a single aux monitor mix to give musicians customized personal monitor control without affecting the other monitors or the front of house mix. Multiple iPads can be set up by an admin user to give the artists a custom set of controls. Both apps run on the current dLive firmware, V1.3. ***www.allen-heath.com, www.americanmusicandsound.com***

RCF M18 Upgrades

An update for the tablet mixer that makes it functional with Android operating systems in addition to Apple iOS, joined by a new firmware version that allows for updating the M18 with the MixRemote 2.0 app that provides added functionality. Enhancements include the expansion of amplifier modeling simulations (licensed from Overloud), an acoustic simulator, as well as four additional stomp box style effects. Other additions include an audio metronome, DCA and mute groups, backup and restore functions via USB, more mute controls, and expanded MIDI implementation. The stereo file player will also be expanded to allow 4-channel playback. ***rcf-usa.com***

Lectrosonics DBa

A digital belt pack transmitter that's part of the DSW (Digital Secure Wireless) microphone system. It provides 24-bit/48 kHz digital audio and also includes AES-256-CTR (Advanced Encryption Standard) technology for situations where privacy is a concern. Also included are wideband tuning (470 to 698 MHz), a linear RF output stage for reduced intermodulation distortion, and 50 mW transmission RF power. The TA5M mic/line input accepts all lavalier and headworn microphones wired for the company's servo-input transmitters. DBa uses two AA batteries for a 5-plus hour (alkaline) or 9-plus hour (lithium) run time. ***www.lectrosonics.com***

L-Acoustics Soundvision 3.0.6 & LA Network Manager 2.4.4

Updated versions of both platforms are available for download from the company website. Soundvision 3.0.6 includes updated Kiva II line array and SB15m subwoofer presets, with gain calibration set for default headroom of 8 dB, along with a minor bug fix for Kara/SB18 arrays. LA Network Manager 2.4.4 includes support of Kiva II and an enhanced setup page. ***www.l-acoustics.com***

Shure SystemOn

Audio asset management software that provides centralized IT support and real-time troubleshooting for Shure Microflex wireless and ULXD wireless systems as well as the SCM820 automatic mixer. It supplies hardware status view from one central portal that identifies issues such as dead batteries and missing equipment, and it also enables remote adjustment of channel level, mute status, and other parameters. ***www.shure.com***

Klark Teknik 1176-KT

An FET-style compressor that's an homage to the 1176LN, outfitted with a discrete signal path utilizing custom Midas input and output transformers as well as a user-friendly interface. Several ratios are provided: 4:1 for moderate compression; 8:1 for severe compression; 12:1 for mild limiting; and 20:1 for hard limiting. An "all-button" mode is included for aggressive vocals, and it can also be applied to drums, bass, guitar and room microphones. An attack knob adjusts the time it takes for the compressor to respond to audio that exceeds the threshold, while a release knob adjusts how long the compressor remains engaged after incoming audio falls below the threshold. An illuminated vintage-style VU meter indicates gain reduction and output level based on which meter button is selected. The 1176-KT comes with a universal power supply with auto-voltage sensing for worldwide usability. ***www.klarkteknik.com***

Crown CDi DriveCore Series

Power amplifiers utilizing the company's proprietary DriveCore technology that eliminates hundreds of components, reducing the variability of component values and increasing reliability. Six models are available, three with BLU link (Harman's 256-channel digital audio bus) and three without. Onboard DSP provides input and output parametric EQ, input and output delay, crossover, proprietary LevelMAX limiting, and more. 2-channel models are available at a specified 300 or 600 watts per channel, while 4-channel models are rated to provide 300 watts per channel. Each output channel is capable of providing either 70 or 100 volts for high-impedance applications. The amplifiers can be controlled, configured, and monitored on a standard TCP/IP network via Harman HiQnet Audio Architect software. ***www.crownaudio.com***

Martin Audio Display 2.2

An update of the software "brain" of MLA Series systems. Acoustic data has been re-measured using more advanced techniques and re-calibrated to deliver accurate prediction and optimization. In addition, the Hard Avoid feature is optimized more effectively for varying distances from the array. Also, elemental EQ is now full-band for all systems. Display 2.2 has been released in tandem with VU-NET 2.0 control and monitoring software, which allows for the input EQ to be adapted automatically to the new output from both platforms. It also supports new CDD-LIVE! loudspeakers. ***martin-audio.com***

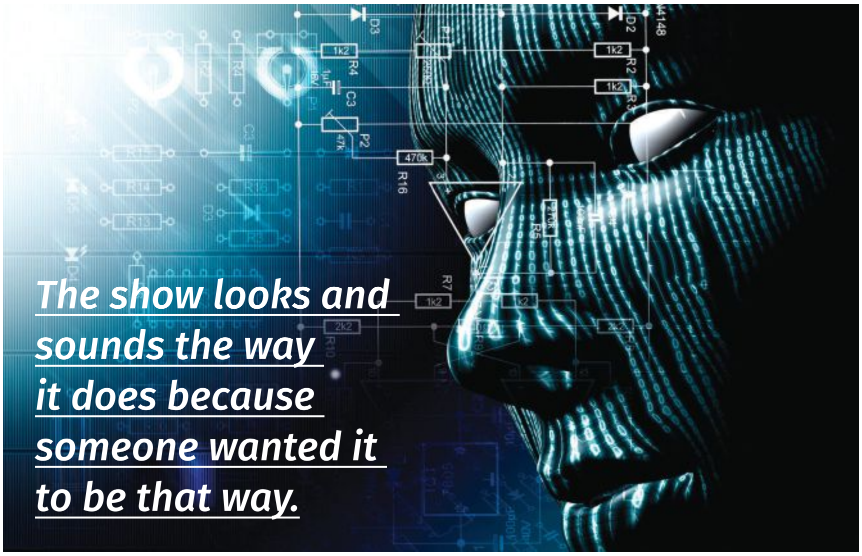
Perspective

WAITING ON THE WORLD TO CHANGE

Musings on the future of professional audio… *by Jonah Altrove*

"Any industry founded on a particular technology faces the danger that a new invention will render it obsolete." *–Tom Standage, "The Victorian Internet"*

When I came across this quote in Standage's fascinating book about the history of the telegraph, it struck me with such veracity that I just put the book down and stared at the wall for a few seconds. This is something I've occasionally found myself wondering – will the inevitable advance of technology eventually eliminate my job(s) as a system tech and/or front of house engineer?



If you're paying attention, the answer might appear to be "yes." It could certainly be argued that developments like the advanced beamsteering DSP that drives certain high-end sound reinforcement systems will turn the tech's ability to determine array splay angles an archaic skill. "When I was your age, we had to put the angles in by hand! One pin at a time! Uphill! Both ways! With no shoes on!" Meanwhile the July/August 2016 issue of the Journal of the Audio Engineering Society describes the work of researchers who are designing an algorithm that can dissect and analyze the creative decisions of human mix engineers.

So it's not hard to imagine that 20 years from now a reference-mic-equipped drone will measure the system-room response, optimize the DSP, and then turn the system over to a 1U rack-mount Wi-Fi-enabled automixing processor in a seamless brushed aluminum package with a proprietary power connector. Oh wait, it'll be powered wirelessly via induction coils. Proprietary induction coils.

Here's the counterargument: music isn't necessary. Put down your pitchfork – of course it's necessary and valuable in a cultural, societal sense. What I mean is that it's not necessary in the way that food, water and sleep are necessary in order to continue living. So the fact that we, as a species, continue to create and enjoy music tells us all we need to know.

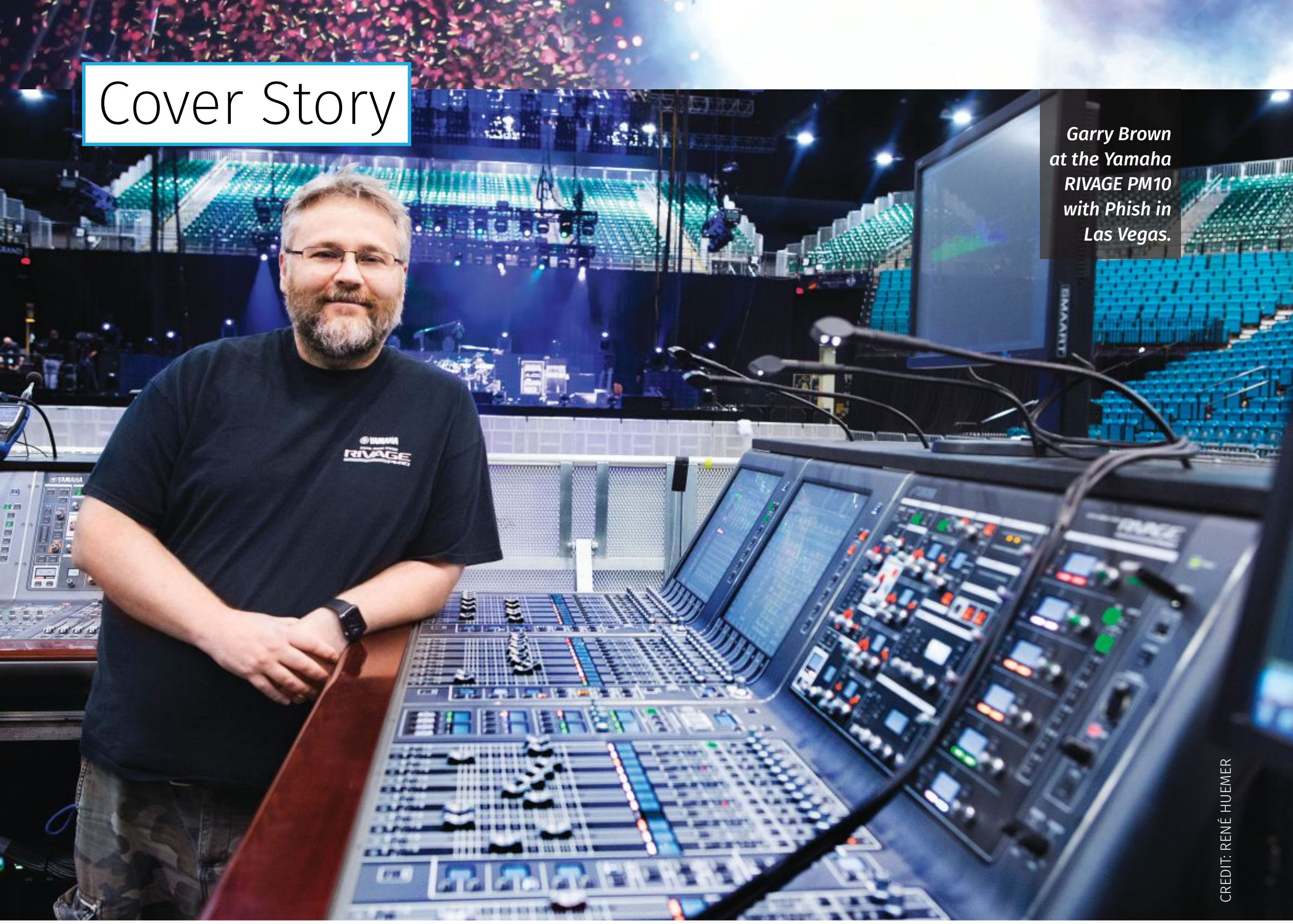
Ever since the phonograph replaced the saloon pianist, there's been a fear that technology will put live musicians on the rocks. Over a century later, live music hasn't gone anywhere, and in some senses, it's bigger than ever. In a purely utilitarian sense, this doesn't compute.

Let's optimize everything – completely automated concerts with algorithmically automixed sound and computer designed lighting and video supporting androids and robots playing procedurally generated music. You might go once, as a novelty, but this is not going to supply the potentially life-changing concert experience that's the reason each of us got into this business.

In terms of efficiency, the most efficient form of a concert is not to have it, so it's not about that – it's something we do simply because we love it. It makes us feel. Therefore, pro audio is not based on any potentially fleeting technology, but on the basic desire of people to enjoy artistic expression, which is a deeply, profoundly human thing that can't be replaced by technology.

Technology broadens our toolset every day, and these innovations are to help us, not replace us. The way in which we do our jobs – and the way the musicians do theirs – is undeniably shaped by advances in technology. But in order to have that soul of expression, there's always got to be someone pressing the buttons. The show looks and sounds the way it does because someone wanted it to be that way. Otherwise, it'd just be noise. LSI

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



Garry Brown at the Yamaha RIVAGE PM10 with Phish in Las Vegas.

CREDIT: RENÉ HUEMER

MIX EXTRAVAGANZA

Applications of a wide range of digital console platforms. ***by Live Sound Staff***

A NEW DIRECTION FOR AN EVER-EVOLVING GROUP

Reunited in 2009 after a five-year hiatus, Phish is still hard at it, delivering truly unique live shows marked by improvisation, extended jams, and the blending of musical genres that include progressive and psychedelic rock, folk, funk, blues, country, bluegrass and more. Band members Trey Anastasio (guitars, lead vocals), Mike Gordon (bass, vocals), Jon Fishman (drums, percussion, vacuum, vocals), and Page McConnell (keyboards, vocals) tour consistently each year, making long seasonal runs that pause for an occasional break before resuming again.

Front of house engineer Garry Brown has worked with the band since it reunited, following previous efforts with Anastasio since 2005. The veteran mixer sought a new console platform prior to this year's dates, working closely with the band's long-time sound company Clair Global in sorting through options and ultimately selecting a Yamaha RIVAGE PM10 joined by an RPio622 I/O rack.

Brown had been utilizing the same digital console for six years, and the platform itself was more than a decade old, so he was in search of a new direction, including entertaining thoughts of returning to the analog mixing realm. He went through a thorough, week-long evaluation process of six viable models at the Clair Global facilities in Lititz, PA, with the PM10 emerging as his choice.

“These days it’s much more about sound quality than some of the functionality aspects of consoles,” Brown states. “The key is, does the thing sound good? In comparing the PM10 to the other models, this is where it clearly distinguishes itself.

“On top of that,” he continues, “it’s easy to get a mix going. I found that in working with the PM10 in just one morning period that it was quite simple to get it together and working completely in a relatively short amount of time.”

He runs about 80 inputs for a standard show, but that count can range up to 120 inputs depending on the goals of a specific performance. “My old console maxed out at 96 channels, so if we went past that, it got complicated,” he notes. “But with 144 inputs, the PM10 makes it much easier from the get-go for me to be flexible with what the band needs to do.”

Clair Global provides full house and monitor systems for Phish, including a main PA headed by Cohesion 12 main arrays with CP-218 subs and 12AM monitors on stage receiving mixes from monitor engineer Mark Bradley via a Yamaha PM5D console.

Brown adds that the PM10 has also proven to be a good fit in meeting his specific mix needs with the band, which in addition to being highly improvisational has never done the same set list twice in its entire career. In fact, for a recent four-show engagement at the MGM Grand in Las Vegas, the band didn’t even play the same song on any of the nights. (That’s 12-plus hours of music without a repeat.)

Jon Lemon mixing Sia’s Nostalgic for the Present tour on a DiGiCo SD5 with Stealth Core 2.

“It’s a challenge for the band and it’s a challenge for me,” he says. “They’re a four-piece band, so it’s not that hard to keep it together, but there’s a whole lot of inputs, they have a lot going on, and they have a lot of flexibility in what they want to play. It’s also the most fun part about working with them – you have no idea what they’re going to play, so it’s not about getting your scenes done and then going on to the next scene, and then the next scene, and so on – you wait and see what comes out of the gate and then just start mixing the show.

“Yes, it keeps me on my toes. But it would be very tough to go back to a more standard tour – it might seem just a bit too mundane.”

A RECENT UPGRADE COMES INTO PLAY

The recent Nostalgic for the Present 23-date North American tour by Sia employed a pair of DiGiCo consoles, including an SD5 loaded with the company’s new Stealth Core 2 at front of house and an SD10 for monitors.

“I picked up the SD5 right after we finished the European festival segment of the tour,” says front of house engineer Jon Lemon, whose lengthy portfolio includes work with Beck, Smashing Pumpkins, Nine Inch Nails, The Cure and others, and who had one of the first SD5 consoles on the market in 2012. He notes that the lighter SD10 suited his needs for the tour’s hectic European flight schedule but finds the SD5’s three full-color touch-sensitive TFT LCD screens offer the best interface for Sia’s kinetic show.

Monitor engineer Adam Jackson then switched over to the SD10, exchanged for the SD11 he had been using. Both consoles were supplied by Sound Image (Escondido, CA), the tour’s SR provider, which also provided Adamson line arrays and system technician Vic Wagner.

“The SD5 is a fabulous desk; I can configure it any way that I want,” Lemon states, detailing that he splits various input banks to either the left or right TFT monitors and can mirror them on the third screen. “Literally, in terms of workflow, everything’s at my fingertips. With more than 60 inputs overall, that's a huge help. I find that the visual feedback they provide really makes a big difference."

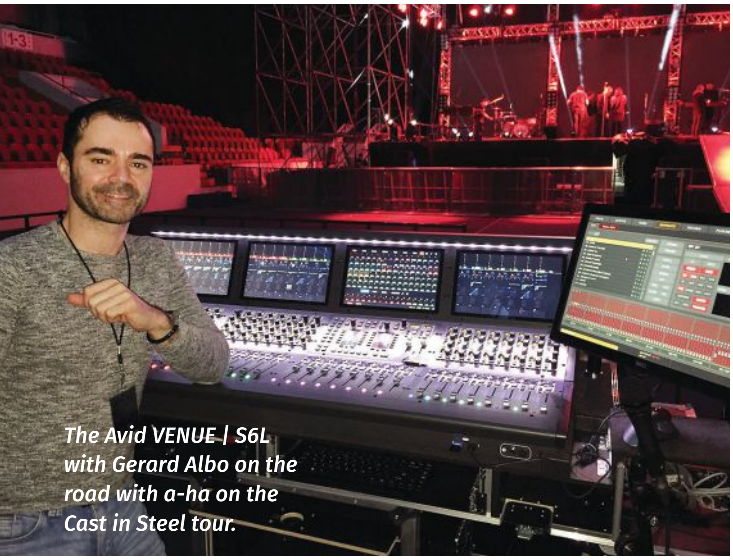
He adds that the upgrade to new Stealth Core 2 is also of considerable help, with its complement of dynamic EQ, multi-band compressors, DiGiTuBe emulators and expanded MADI connectivity. On the tour, the SD5 is connected to an SD-Mini Rack on an Optocore loop that offers Lemon a combination of digital and analog I/O, and he has two Waves SoundGrid Extreme servers that can run over 500 instances of Waves stereo SSL E-Channel or C4 Multiband Compressor plugins.

CRAFTING A TRANSITION WITH A FAVORITE

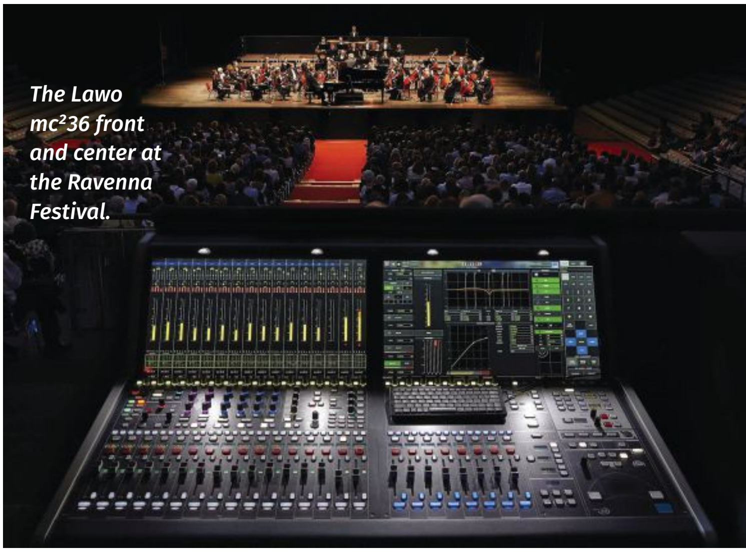
An Avid VENUE | S6L console was on the road earlier this year with a-ha on the Cast in Steel international concert tour, with stops in Russia, Sound America, Europe and the UK before wrapping up in the band's home country of Norway. Front of house engineer Gerard Albo, who has toured with artists such as Amy Winehouse, Patti Smith, Anastasia, and Tom Jones. purchased the console before heading out for the tour based on his long history with VENUE models.

"I've always worked with the Profile, it was my favorite desk," Albo states, "so I decided to buy this one for the future. I'm extremely happy with the console quality, it's very robust and built like a tank."

The S6L shares a common VENUE software platform with Profile, making his move to the new console straightforward. "The S6L is so powerful, yet so simple to use," he says. "The transition from Profile was simple, and at rehearsals, I felt confident to take it out on the road after just a few hours. The touch screen workflow is much faster and the sound of the preamps is first class, as are the onboard EQ and dynamics. Overall, it's a delight to mix on – I love it."



The Avid VENUE | S6L with Gerard Albo on the road with a-ha on the Cast in Steel tour.



The Lawo mc²36 front and center at the Ravenna Festival.

CHANGING THINGS UP WITH MULTIPLE ELEMENTS

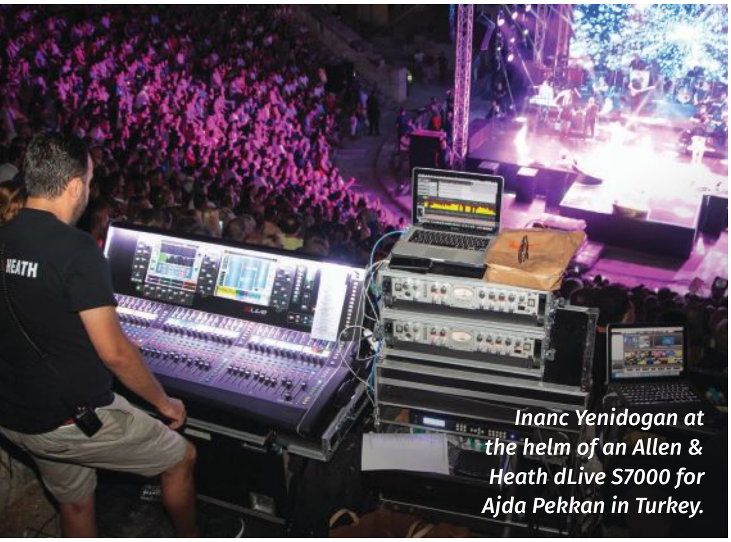
Staged across the city in Italy that bears its name, the annual Ravenna Festival presents a range of live performances. At one of the event's primary venues, the Dante Alighieri theatre, the festival's long-time audio provider BH Audio and newcomer Mediacare Audiovisuals (both based in Italy) deployed a Lawo mc²36 console joined by a compact I/O Stagebox and RAVENNA networking.

"After using more or less the same set-up for the past three years at the venue, the first thing that struck me was the sound," explains BH Audio partner Massimo Carli. "We'd used a top-grade analog desk in the past, so the comparison wasn't between products of a different level, and we hadn't changed PA or microphones, but we immediately noticed the difference in timbre."

The choice of the mc²36 console also triggered the application of RAVENNA networking. "The use of RAVENNA was not so complicated," Carli notes. "I needed just one compact I/O interface and Cat-6 cabling for the RAVENNA link. The great thing was being able to have 128 inputs and 128 outputs on a single Cat-6 cable.

"There are many reasons for choosing Lawo; two are particularly important," he continues. "One is the company's philosophy – they choose to work with the RAVENNA open protocol. I've been a fan of open source technology for a long time, not for financial reasons but for philosophical reasons. Sharing our knowledge allows us to grow faster and better. The second reason is the fact that Lawo systems are really stable and reliable."

"During rehearsals for the concert conducted by Riccardo Muti, his wife Cristina came to the front of house platform and asked me what we'd changed, as the system seemed to sound better than usual, and when Muti asked how the sound was, Cristina told him it was because I was using a new audio console."



Inanc Yenidogan at the helm of an Allen & Heath dLive S7000 for Ajda Pekkan in Turkey.

MAKING THE MOST OF POSSIBILITIES

A recent concert series by Turkish pop star Ajda Pekkan in her home country, backed by a 30-piece band and appearing at venues including the Harbiye Cemil Topuzlu Amphitheatre in Istanbul, Bodrum Amphitheatre and Izmir International Arena, utilized Allen & Heath dLive mix systems. Specifically, two S7000 surfaces and DM64 MixRacks served as the pivot points at front of house and monitors, as well as for live recording.

Pekkan production manager and front of house engineer Inanc Yenidogan specified dLive for the tour. The band utilized stereo in-ear monitors plus side fills and mono sub monitor mixes for the drummer and bass player, requiring up to 60 mono auxes. Additional auxes were routed over Dante networking to the front of house DM64 MixRack as an output expander. All the shows were recorded via M-Dante on a Mac DVS.

"We made the most of the interlinking possibilities of dLive," states monitor engineer Wayne Nigel Gittens. "With 30 musicians on stage with high monitor requirements, the front of house rack was partly used as an output expander via Dante."

WORKING FROM ANYWHERE IN THE HOUSE

When he's not bouncing between tours with Capital Cities, Atlas Genius, or The Ting Tings, Jason Stiegler is in the studio producing an eclectic range of projects, while in his spare time, he's out spinning tracks at venues and parties across Southern California. More intimate clubs and parties in particular provide a way to test out new tracks and gauge audience reaction, he explains, and for these gigs, he carries a system headed by Mackie DL32R 32-channel wireless digital mixer joined by SRM450 loudspeakers and SRM1550 subwoofers.

"What's great about the DL32R is that it can be set up anywhere in the house," Stiegler offers. "We've used it on tour as a monitor desk, and as a front of house desk on smaller shows. Tonight we've got it set up under my DJ rack. I can walk around the room before the show, EQ and time-align the speakers, and check the sound from anywhere in the venue. It allows me to do all the things I can do on a larger scale with a larger system."

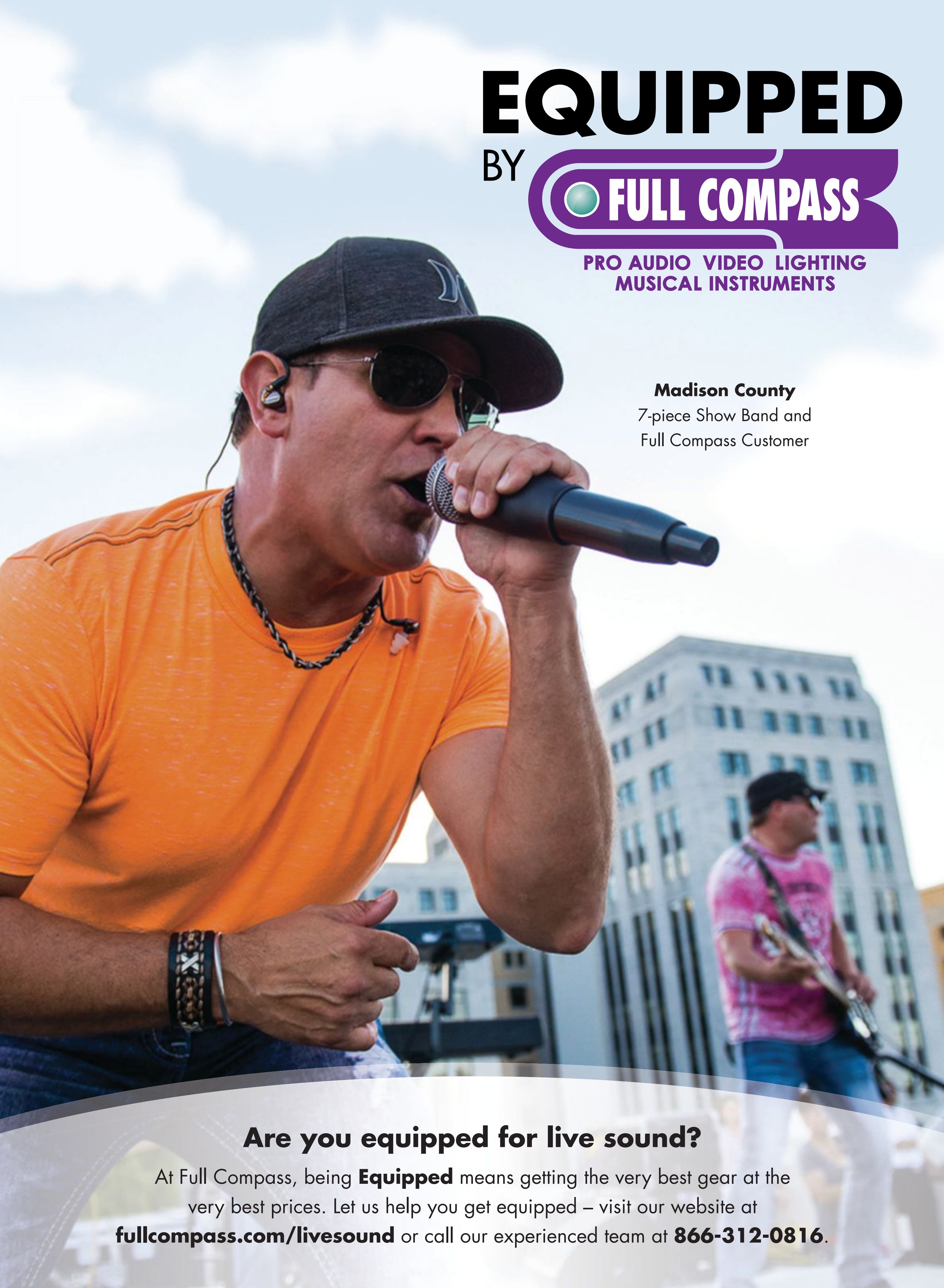
Jason Stiegler dialing things in on his Mackie DL32R via an iPad.

He also calls out another facet of the DL32R: "The ability to use it not just as a mixer but as a multi-track recorder, to take those recordings and load them into your favorite DAW, is something you just can't do with most mixers."

TRAVELING LIGHT WHILE GETTING RESULTS

Canadian hard rock band Kobra and the Lotus just finished a 35-date tour of Europe that saw the 5-piece group visit 18 countries. Helping the band travel lightly on the road was a Midas M32C digital rack mixer and DL32 stage box handling in-ear monitor mixing. Production manager Andrew Peters, who also recently completed a stint of shows in the U.S. with

Lead singer Kobra Paige utilizing the app to adjust her own monitor mix on the Midas M32C.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metal band DragonForce, also made use of the M32C, stating, "Ease of transportation is the major selling point for me and the bands I work with, so this is the ultimate mixing system."

Lead singer Kobra Paige adds, "The result for us has been incredible; this is the first time we've been able to get hands-on with our in-ear mixes and take full control of the show." Inputs and outputs were run via the DL32, with an AES50 port from the stage box feeding the M32C and a router connected via Ethernet, allowing for internet connectivity to the unit. Each member of the band then used the M32 Edit App to adjust their own mixes.

"As we travel light, I make use of each venue's front of house console," Peters notes. "The highlight has been mixing on the flagship Midas PRO X console at Backstage Munchen in Germany. Also, getting to mix on the M32 at Yo-Talo in Helsinki was sweet, especially to have our entire sound run on the Midas M Series."

COMFORTABLE WITH A NEW ACQUAINTANCE

Front of house engineer Tomas Wolfe utilized a Roland Pro AV M-5000 OHRCA console on both the U.S. and European legs of the recent tour by The Neighbourhood.

Wolfe, a native of Los Angeles, worked his way up from the bottom in live audio, with two years on the area's club circuit leading to touring with a series of artists that includes BØRNS, Belly, Everlast, and the Mowgli's. The stint with The Neighbourhood marked the first time for Wolfe on tour with the Roland console.

"I'd read a bit about the M-5000," he recalls, "so I went to the Roland offices in L.A. and checked it out, and ended up doing a couple of shows with it in L.A. I found that I really liked the sound and the flexibility of the workflow. It's very customizable to my needs. The user layers and the user-defined sections are just that – designed for the user to configure it, so I can set up any show for the best way for me to work. Anywhere I want to go on the console is no more than a click or two away."

Tomas Wolfe touring with The Neighbourhood, joined by his Roland Pro AV M-5000.

Wolfe adds that the M-5000 is sonically transparent. "What you put in is what comes out," he says. "There's no inherent coloration to the sound. The onboard EQ and dynamics are great. I'm a huge fan of the sound and smoothness of the dynamic EQ, and the multiband compressor. Then there are the onboard effects. I was a huge fan of the classic Roland [RE-201] Space Echo and I grew up on the SDE-3000 reverb and delay, and it's so great to have all of them and more on the console."

Miguel Angel Tapia Trujillo and the SSL Live L500 supporting Pope Francis in Ciudad Juárez.

STAYING INSIDE THE BOARD

During a recent week-long tour of Mexico, Pope Francis addressed huge audiences at five venues around the country, accompanied by Solid State Logic Live L500 and L300 consoles in a variety of roles and supported by Audio Acústica, SSL's platinum partner in the country. Miguel Angel Tapia Trujillo, who served as front of house engineer for the appearance at the Parque Chamizal in Ciudad Juárez, notes that his L500 console used the SSL Blacklight II high bandwidth multiplexed MADI connection for around 90 inputs from the stage including choir, orchestra, and podium microphones, plus the stage feeds from five opening bands.

"We did that all in one console," Tapia says. "I set it up so every tile on the console could access any channel – inputs, VCAs and stems." The stem group path type, an audio group available with full processing and extensive routing options, was used extensively for the show. He utilized them for all effects (rather than auxiliaries) and to manage the large number of orchestra microphones.

"I had 14 fully processed string section stems alone," he notes. "Then every section of the orchestra had its own VCA, and I had several main VCAs on top, including the orchestra,

the choir, and the pope." He adds, "I used the internal FX quite a lot as well. Every stem had multiband compression on it, and I used doubling and chorus on the strings, plus some small room reverbs. I also engaged the 'cathedral reverb' preset on the choir, which sounded great."

Yerye Marrun, monitor engineer at the event in Ecatepec (audience of approximately 400,000), utilized an L300 for his work. "It was my first time with the console, and I really enjoyed it," he states. "The sound is great, of course, but so was the ease with which we ran the sound check, making sure everything worked for both the stage and the broadcast. It made me feel confident, and in turn it was easy to run the show with confidence."

SMALLER CAN BE JUST THE TICKET

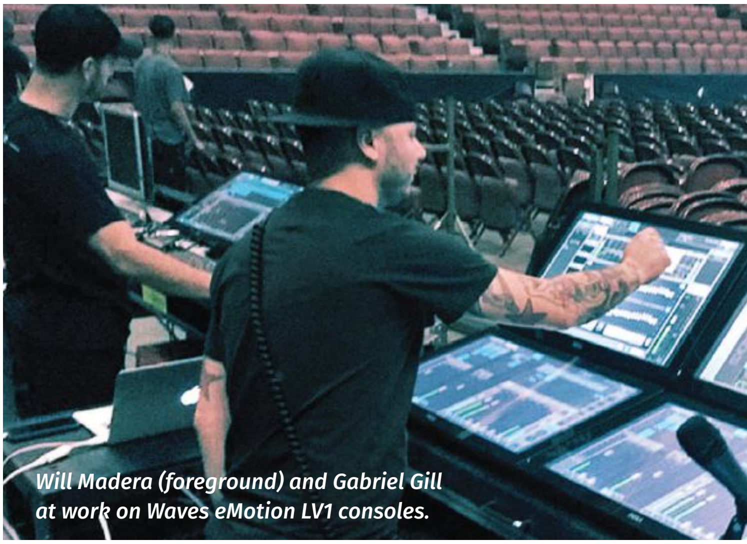
A Soundcraft Si Performer 3 compact console proved the right fit for British recording artist Gabrielle Aplin's small venue tour throughout the UK earlier this year, fitting well in smaller venues that ranged in capacity from 200 to 900 people while also serving an 8-piece band. Independent sound engineer Darryl Walsh worked with Aplin on the tour, helping in the selection of the Si Performer 3 after it was decided the group would bring its own system to each of the smaller venues.

"Having never used a smaller mixer, I admittedly had my reservations but these were quickly dispelled after the first day in rehearsals," Walsh states. "I was so impressed with the sound quality. The in-ear monitors sounded great and all of the band members were happy."

In fact, Walsh purchased his own Si Performer 3, putting it to work on monitors for British alternative rock band Maximo Park's 10th anniversary tour. He specifically cites the layout and usability of the console, noting that any fader can be assigned anywhere, with four banks and a dedicated button to send to every mix.

"The feedback from the bands has been excellent," Walsh says. "Both Maximo Park and Gabrielle Aplin claimed it was the best monitor sound they'd experienced, especially regarding how clean and smooth the sound was."

The Soundcraft Si Performer 3 stage side with Darryl Walsh for Gabrielle Aplin.



Will Madera (foreground) and Gabriel Gill at work on Waves eMotion LV1 consoles.

EVOLVING TO NEW SONIC PLACES

Will Madera, front of house engineer for Pitbull, and Gabriel Gill, front of house engineer for Farruko, deployed Waves Audio eMotion LV1 consoles on their recent joint tour.

Madera notes, "The state of music is constantly evolving, and when the Waves eMotion LV1 mixing console burst onto the scene, there was no question in my mind that it would change the future of live sound. It's innovative touch screen console surface and superior sound quality with incredible headroom have made it my clear choice. It's also perfect for a true lover of plugins: the ability to use eight plugins on a channel strip is phenomenal, and it makes the LV1 a true game-changer. It was the number one choice for our U.S. tour."

Gill remarks, "I'd been looking at the eMotion LV1 mixing console in January 2016 at the NAMM convention. After a day spent in New York at a Waves seminar, I knew that would be the next board on my rider. The portability, ease of use and integrity with my Waves plugins are very important things for me. But above all, the sound quality of the board was the determining factor. Being able to integrate my Waves servers, and computers with Waves TracksLive together with the DiGiGrid IOX in such an easy way makes my setup time a lot quicker.

"With shows every day, spending less time in setup allows me to spend more time for sound check, or for Virtual Sound Check and recording the show using TracksLive," he continues. "Also, having used Waves products in the studio for over 11 years, bringing them over to the LV1 is very beneficial. Knowing what the producers use in the studio lets me get my artist's vocals to what the audience is used to hearing on the recording. It also lets me experiment with new sounds in the studio before I try them in a live situation." LSI

AN EYE ON DI

Principles and the latest capabilities of direct boxes.

by Craig Leerman

The new Radial 4-Play multi-output DI.

In its most basic form, a DI – also known as a direct box and short for "direct injection" and/or "direct insert" – takes an unbalanced high-impedance signal (i.e., from an electric guitar or keyboard) and converts it to a balanced low-impedance signal.

This process is needed when running high-impedance signal at long distances, such as down a snake to a front of house console. Unconverted, the unbalanced signal can pick up noise; also, sending high-impedance signal at longer distances can attenuate higher frequencies. While there are many basic DI models on the market, there are also several feature-laden versions that do quite a bit more.

INCREASING RANGE

DIs come in two basic varieties: passive and active. Passive designs use a transformer that performs impedance matching and balancing, essentially creating a "magnetic bridge" for audio to pass. They don't require a battery or phantom power. Sound quality basically depends on the quality of the transformer – good ones sound far better than cheap ones. When faced with a hot input signal, passive designs saturate instead of clipping like their active counterparts, resulting in a warmer sound.

In addition, higher-end passive units may use a shielded transformer to help reject any interference from entering the unit. Speaking of interference, a passive DI really shouldn't be placed atop an instrument amplifier because the amp transformer's magnetic field can interfere with the magnetic field of the DI's transformer. Numerous DI models have a ground switch and an output or loop-thru that facilitates connection of the instrument to the stage amplifier.

Active DIs, on the other hand, utilize electronic circuitry that requires power, provided via a battery or more commonly via phantom power from a mixing console. Some studio-type units may require AC power (usually via a "wall-wart"). Active DIs are much like preamplifiers and can also offer features like ground-lift switches, high-pass filters, mono summing, polarity switches, and equalization circuits. Solid-state electronic designs are common in live audio production, but there are some tube DI models that are popular, particularly in studio applications.

When I was a young soundman, I learned "The Rule" with DIs: Use a passive unit for high-output instruments like electronic keyboards or guitars with powered pickups, and use an active unit for low-output instruments. It's still a pretty good rule of thumb but is not absolute, as many active models now have input pads to reduce hot signals while many modern passive models work quite well with low-power signals.

Most DIs convert an instrument signal to a balanced mic output, but there are passive and active models that can also convert a loudspeaker output to a low-impedance balanced microphone signal. Some also include things like cabinet modeling and EQ. Newer on the scene are "media-type" DIs designed specifically for use with playback devices like PCs, phones, tablets, and CD players, converting a headphone speaker output or line output with RCA jacks into a balanced mic signal. Some can take a stereo input and sum it to mono, while others forgo the wires altogether and connect via Bluetooth.

DIs also come in many form factors. Single-channel versions are the most popular, but stereo models and even multi-channel rack-mount units are great when dealing with multiple inputs onstage.

Concert-1 and Concert-2 DIs from Jensen. (Want to bet on what type of transformers they use?)

CLASSICS & INNOVATIONS

When I first started working in pro audio more than 30 years ago, sound companies had to build their own DIs because there weren't many commercially available units. But then as now, Jensen transformers were highly coveted in this application due to a number of factors, chiefly that they sound great.

Today, Jensen – acquired by Peter Janis and the gang at Radial Engineering a couple of years ago – offers DI boxes built around it's own transformer technology. These transformers are also at the heart of Radial DI boxes and a host of others. The Jensen Iso-Max line includes the single-channel Concert-1 and the stereo Concert-2 that are passive models using internal shielding to help protect against interference from RF. Both have a rugged extruded aluminum enclosure and steel slide-in inner tray with recessed zones at each end to protect the switches and connectors. Jensen also offers a selection of media-type DI boxes.

Countryman Type 10 and Type 10S (stereo) active DIs.

The Countryman Type 85 was (and still is) so common in production circles that many performance riders simply say "Countryman" instead of DI on the input list. The Type 85 set a standard for rugged construction, often featured in ads that showed trucks driving over the unit with no damage. It has a single-piece extruded case with thick walls and recessed connectors and switches.

The Type 10, which joined the lineup more recently, delivers increased performance with the same rugged construction, with the company stating that it offers the lowest distortion and noise floor of any commercially available DI. Dual-channel versions of both Countryman units are available, each outfitted with a stereo 1/8-inch TRS jack and a pair of RCA inputs in addition to 1/4-inch inputs and thru jacks. They operate from phantom power or a 9-volt battery. While the lineup is limited, the performance is not, with all units designed to work with any input, including loudspeaker level, while the stereo units can perform double duty as media-type DI boxes.

With tube models, rack-mount units, Dante and Bluetooth versions, to say that Radial has a DI for any application is not hyperbole, but given the company's track record, they'll probably continue to turn out innovative designs for an increasing array of uses. Classic models like the passive JDI and active J48 are ubiquitous on live stages. One unit (of many) that's more specialized is the active PZ-DI that's specifically designed for acoustic and orchestral instruments. It allows users to properly match the input impedance to the source to optimize frequency response, dynamics and "feel."

A cool newer model is the Di-NET DAN-TX Dante-enabled DI. Equipped with 1/4-inch, RCA and stereo 3.5 mm input jacks, instruments and line level sources can be directly patched to networked audio systems using the Dante protocol. The DAN-TX has stated 24-bit/96 kHz analog-to-digital conversion to provide high audio quality, and a local 3.5 mm headphone output provides the means to quickly test audio.

The iconic IMP 2 from Whirlwind just passed a milestone with over 1 million units sold. This passive DI has parallel-wired 1/4-inch input/output jacks,

Whirlwind Di2 stereo DI that's hard-wired to a 15-, 25- or 50-foot stage snake.

a ground lift switch, and a tough metal enclosure. Whirlwind's TRHL transformer is at the heart, and it's riveted – not glued – to the chassis so it will stay in place when the box gets tossed around (and you know stuff gets tossed around at gigs).

Whirlwind also has a large selection of

Recently introduced TDA-1 and TDA-2 active direct boxes from Telefunken.

active and passive DIs, stereo units, and multichannel rack-mount configurations. One that I absolutely love is the Di2 stage snake stereo DI, a version of the Director 2. It's a 2-channel unit hard-wired to a 15-, 25- or 50-foot stage snake. Available in a selection of colors, the Di2 makes it easy to wire up a band when several DIs are needed while also eliminating a pile of XLR cables for a clean stage. In addition, Whirlwind offers Medusa Series snakes with XLR inputs along with DIs built into the same stage box, great for situations such as working with a band with a singing instrumentalist who needs a DI as well as a mic.

NEW DIRECTIONS

Telefunken recently entered the DI business, launching several models at the 2016 AES show. The range comprises four different units, including mono and 2-channel versions of both active and passive designs. The Telefunken TDA-1 (mono) and TDA-2 (dual) are active FET direct boxes that employ discrete Class-A FET circuitry coupled with a transformer. According to the company this design provides "the perfect balance between clean, high-headroom performance and warm, saturated tone." Meanwhile, the Telefunken TD-1 (mono) and TD-2 (dual) are the passive models.

At the heart of each of these new designs are custom-wound output transformers, and the circuit boards are hand-assembled. The electronics are housed in a durable, extruded aluminum enclosure with recessed heavy-duty toggle switches for the -15 dB pad and ground lift. Telefunken notes that these new boxes are virtually indestructible, and by the rugged looks of them, that's likely not an overstatement in the least.

Noted for its connectors, Switchcraft also presents a DI lineup of both passive and active units with solid construction and recessed switches. On a recent corporate show, I had a chance to use the new 318, a purpose-designed media DI. Part of the AUDIOSTIX compact interface line, the 318 is the width of a D series connector and less than four inches long, and is equipped with an XLR output on one end and a TRS input jack with recessed ground lift switch on the other. A volume control resides on one side.

Both ends of a Switchcraft AUDIOSTIX 318.

The unit takes a TRS input (cable included) from the headphone jack of a laptop, tablet, or phone and provides transformer isolation while summing the signal to a balanced mic level mono output. The miniscule size allows placement on a cluttered podium, making it easy to hook up when the presenter brings an unannounced laptop at the last minute. An install version called the 319 replaces

The appropriately named ARX Blue DI that serves as a Bluetooth interface.

the XLR connector with a terminal block set for line level output.

RapcoHorizon has a variety of well-built DI boxes in mono and stereo configurations, as well as a 4-channel rack-mount version called the SL-4. It's basically four SL-2 single-channel DIs mounted in a single rack space. Each channel offers a -20 dB pad switch, ground lift switch, and both front and rear inputs/outputs to suit any wiring scheme. In addition, the company's BLOX Series presents some very compact units, including the LTIGLBLOX with a built-in 2-foot cable terminated with a 1/8-inch TRS male plug that makes connections easier.

ARX, based in Australia, has introduced the Blue DI, a stereo interface that can pair with any Bluetooth audio device, providing balanced low-distortion outputs in a heavy-duty steel chassis. It can be driven via phantom power or 12-volt DC. In addition, the USB DI plugs into a computer and

The Peavey USB-P that connects to the USB port of computers, appearing as the stereo output device.

allows the use of balanced audio outputs instead of relying on the headphone jack. And, ARX also offers a range of different signal devices, including single channel and stereo DI boxes that all use the same size chassis, with up to four units able to be rack mounted together in one space using the optional RMK-1 kit.

Recently at a demo, I spotted a loudspeaker manufacturer using a Peavey Electronics USB-P interface between a laptop and the system processor. The USB-P is a playback-only interface that connects to the USB port of a computer and appears as the stereo output device. Offering transformer-isolated balanced stereo outputs and a ground lift switch, the unit makes it easy to connect a computer to a sound system. It can also sum stereo signal to mono. Peavey also makes more standard DI boxes, including the ADI-Q active direct box with 3-band EQ.

Many other manufacturers offer quality DI choices as well, with the list including ProCo, Rupert Neve, Pyle, Hosa, CBI, Rolls, ART and others. The key to finding the "right" DI is giving it a listen to make sure it sounds good for your particular application. That's why production companies (like mine) stock a wide variety of DIs. LSI

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

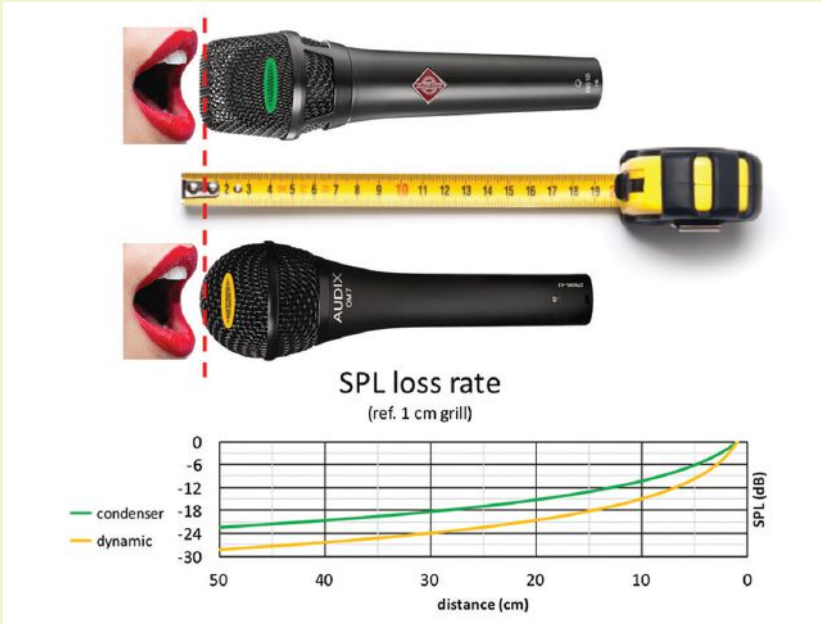
SENSITIVE MATTERS

An interesting performance facet of condenser and dynamic mics. *by Merlijn van Veen*

Many believe that condenser microphones are more sensitive than dynamic microphones (moving coil) and therefore pick up "everything," e.g., stage wash and noise. Sensitivity, however, is nothing but a constant conversion rate from pressure to voltage and more important, it is distance independent. Contrary to popular belief, it's not the reason for picking up "everything."

Most condenser mics are indeed more sensitive. A Neumann KMS 105 condenser with a sensitivity spec of 4.5 millivolts at 1 Pascal (4.5 mV/Pa) is 15 dB more sensitive than an Audix OM7 dynamic with 0.8 mv/Pa.

Many artists judge their distance to the mic based on the grille, not the actual position of the (often hard to see if not invisible) capsule, which in most dynamics is typically closer than in condensers.



At 1 centimeter (cm) to the grille, the KMS 105's capsule is approximately twice as far (-6 dB) in comparison to the OM7. Doubling the distance to the grille (2 cm) translates to a 5/4 or 120 percent (-1.9 dB) increase for the KMS 105 but a 3/2 or 150 percent (-3.5 dB) increase for the OM7. Doubling the distance again (4 cm) translates to a 7/4 or 175 percent (-4.9 dB) increase for the KMS 105 but a 5/2 or 250 percent (-8 dB) increase for the OM7!

The accompanying chart showing SPL loss rates clearly indicates that the greatest rate of change occurs within the first 10 cm. This makes dynamics much more responsive to a change in distance, which is the reason why artists need to maintain their distance and stay on top of them.

Inherently, this increased responsiveness also makes dynamics more prone to tonal changes caused by proximity effect. So in a way, one could argue that condensers are actually less "sensitive" to these issues.

The caveat, however, is that all things being equal, accounting for the increased offset in capsule depth between dynamic and condenser mics by means of gain results in an inversely proportional increase in terms of stage wash or noise for condensers. Due to the KMS-105's capsule being removed twice as far (-6 dB) in comparison to an OM7, electronic compensation of the SPL loss due to increased distance not only raises the level of the direct sound but also the level of the indirect sound, e.g., "everything" else.

These relative differences between dynamic and condenser mics are most notable in very close proximity to the grille, where the offset in capsule depth is still relatively great. At greater distances the offset becomes trivial, which is apparent from the way that the loss rates in the chart start to run parallel to each other.

Capsule depth is the reason why a dynamic makes sense on a loud and/or noisy stage, provided the artist stays on top of it. Otherwise the advantage is quickly lost.

The actual mic sensitivity has nothing to do with this behavior, which is why I would like to suggest avoiding the word "sensitivity" in this particular context and exchange it with something along the lines of "responsiveness" – for lack of a better term... LSI

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

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GIMME THREE STEPS

An alternative approach to sound check.

by Jonah Altrove

Like many sound engineers, I have a background in music, so I've been on both sides of sound checks and know how annoying a poorly-run check can be for musicians. These experiences have led to a paradigm shift in the way I approach sound checks: they're for the artists, not the engineers.

On the road, practice time is in short supply, so many artists prefer to use their stage time to rehearse new material, tweak the backline, and tighten up the performance. It's very helpful to spend some time dialing in monitor mixes, but standing around while the front of house engineer listens to Rack Tom 2 for five minutes straight? Not so much.

Bringing this mindset to the other side of the console, I developed a system that helps me get up and running at a gig as quickly as possible, catch potential problems early, and dial in a good mix, all without bugging the artists. (Although I came up with this process for myself over the years, I'm sure that I'm not the only person to have done so, and as a result I'm not claiming originality or innovation here, just sharing what works for me.)

The process that laypeople call "sound check" is actually a conglomerate of several tasks. My approach divides the process into three steps: system check, line check, and sound check.

STEP ONE: SYSTEM CHECK

I start every gig by examining the relationship between the sound system and the room. There are the physical and acoustic properties of the space itself (resonant low-frequency modes, wall reflections, reverb decay, rattles in ductwork, HVAC noise, etc.), but my biggest priority at this step is the system's variance, or how its coverage changes throughout the space. Ideally, my mix should be related to every seat with the same tonal balance (low spectral variance) and at the same volume (low level variance).

Using a playlist of reference tracks, I move around the space, listening from every section. Maybe the level at the back of the balcony is down 8 dB, or the front fill coverage area is too hot above 12 kHz. Maybe the system's HF coverage drops off at the end of the rows, or there's a "power alley" of coupling-zone subwoofer energy that passes through the mix position. If what I'm hearing at the mix position is not representative of what other audience members are hearing, I want to know how, why, and where.

It's important to note that I might not be able to address these issues. If it's a one-off on an installed rig, the venue is not going to appreciate me tinkering around with the DSP.

But even if I can't rectify the problems, I need to know they exist so I can mix accordingly. I learned this the hard way after realizing halfway through a club gig that my mix sounded good at FOH and pretty lousy everywhere else. Once I determined that there was far more sub energy and less HF at the console, I altered my mix to correct for it, and the audience was noticeably more pleased with the second half of the show.

Since that experience, walking the space to my playlist (see sidebar) is the first thing I do when I get into a venue. By the way, I try to use the same tunes every night, which more readily reveals differences gig-to-gig. At a festival, I wander around during another band's set to scope out the rig's coverage variance before it's my turn.

STEP TWO: LINE CHECK

This is exactly what it sounds like: checking the lines. While Step One was focused exclusively on everything downstream of the mixer (post-console, if you will), Step Two deals strictly with everything upstream — microphones, direct boxes, cables, and snakes.

I'm checking to ensure each input is coming up in the right spot (correct patching) and that there are no hums, buzzes or bad cables in play. I rough in some input gains and high-pass filters, but this is not the time to spend six minutes EQ-ing the floor tom. As soon as I verify that I'm getting signal and the input is behaving as expected, it's time to move on. This takes maybe five to 10 seconds per input, so I can line check a 24-input show in a couple of minutes.

The secret to line check is to not grind to a halt for a non-show-critical issue. Note the problem, move on, and come back to it later — otherwise you risk burning all of your time tracking down something relatively mundane only to discover serious issues with a "show stopper" input like lead vocal. Get through the process then attack the remaining issues in order of decreasing priority.

STEP THREE: SOUND CHECK

Once the inputs (Step Two) and outputs (Step One) are checked out, I can be confident that everything is entering and leaving the desk properly. All that remains is to dial in a good mix. I tell artists, "OK, this is your time. You go ahead and rehearse, run some tunes, jam, whatever you need to do. If there's something specific I need to hear, I'll let you know." Artists tend to be more relaxed, and their playing is more representative of how it will be during the show, which allows me to dial in a mix pretty quickly. (We'll get to that in a minute.)

When the situation permits, I like to walk up on stage after a couple of songs and find out how everyone is feeling, face to face. Since I do a lot of one-offs, the artists and I may not know each other, and this is a bit more personal than grunting into a talkback mic. A typical comment might be something like, "Everything is sounding good out front, but I'm getting a lot of that guitar amp off the stage. It will sound a lot more even in the house if we can turn it down a bit or re-aim it."

Your mileage will vary depending on your rapport with the particular artist, but the approach has a lot to do with it as well. If I need to recommend an adjustment, I always explain why it will help them sound better, and that's often enough. (Sometimes there is no practical solution. During a one-off for a pop artist with a stage volume issue, I told her TM, "She really needs to be on IEMs." He agreed, but added that it was a "next tour" solution not a "tonight's show" solution. In these cases we simply do the best we can.)

Selections From The Author's Reference Track Playlist

- *"Domino,"* Jesse J. A clean, dry mix that tends to reveal time alignment issues in sub arrays.
- *"Neon,"* John Mayer. On systems with excess energy in the 2 - 8 kHz region, this track will sound overly harsh in the vocals.
- *"The Real World,"* Owl City. There's a nice bass drop that's very revealing of LF room issues, modes, and rattles.
- *"117,"* Halo 4 Soundtrack. A cool orchestral piece that illuminates issues in the 1 kHz region quite readily.
- *"Breakeven,"* The Script. This recording tends to be representative of how a live band will sound in the space, particularly the drums.

MIX IT UP

Since larger-venue artists tend to have their own engineers, most of my one-off gigs are relatively small rooms, 1,000 seats or less, which means stage sound is a significant factor. My approach as to the best way to get a quick mix in a small venue is to start with all non-vocal channels completely out. As the musicians play, I listen to what's already coming off the stage (probably a lot of guitar, bass, and drums), and simply add what's needed. It's called live sound reinforcement for a reason.

With vocals at a comfortable level, I bring up everything else around them. In small rooms, low frequencies warrant special attention. As the bass guitar comes up in the subs, it starts to sound "wider" before it gets appreciably louder, and I find that this is often the "sweet spot" in many rooms.

Bringing drum channels up, rather than down from 0 dB, ensures only adding what's needed, which is important as your drum levels will dictate the level of the mix as a whole. The specifics vary quite a bit with genre, but the point is to work with the room sound, not fight it — that's a losing battle.

OFF & RUNNING

From here, I tighten up EQs (cutting what I don't want rather than boosting what I do), and tweak compressors and reverbs. At first it's a bit unnerving to launch into a tune with only high-pass filters in place, but I've found that initial attention spent on Steps One and Two really pays off here in a big way, as I'm already heading down the path to a good sound.

This is why the order of the steps is important: boosting HF on a muddy vocal won't do a bit of good if the culprit is a burned-out HF driver in the mains (which I should have found in Step One) or a "spit-out" mic diaphragm (ditto, Step

Two). Due diligence during system and line checks means that virtually any problem I run into during sound check is fixable at the console.

Many artists are pleasantly surprised when I don't ask them to stop what they're doing and give me kick... kick... kick... OK... snare... snare... As any studio engineer will tell you, each instrument can be sounding great on its own but there can be a horrible mess when they're mixed together. So I skip the tedious one-at-a-time approach and just let them play as I build the mix as a whole.

There are, of course, some caveats to this approach. It's best tried with an experienced pair of ears, as a seasoned engineer is more easily able to hear something funky in the mix and identify which channel is responsible. It's also necessary to be able to identify frequencies by ear, a skill I call "hand-ear coordination" – knowing what sound you want and which knobs to turn to get there.

The secret to line check is to not grind to a halt for a non-show-critical issue.

This applies both to feedback ("That acoustic guitar is ringing at 125 Hz") and tonal adjustments ("I have too much 400 Hz in that vocal"). Also required is a high degree of fluency with console operation, as the "all at once" sound check requires making a bunch of adjustments in rapid succession.

Not infrequently, I have to mix artists without a sound check. It's not optimal but it is reality. Experience helps, but the main reason I'm able to do this is because I strictly adhere to the process described here. For novice engineers, sound checks can be pretty stressful, but each one makes us a little faster, a little more effective, and a little cooler under fire. Over the course of many checks, we can develop a sort of "rhythm" that grows into an effective routine. LSI

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


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AIMING FOR CONSISTENCY

Getting the electric guitar "monster" under control.

by M. Erik Matlock

On a recent quest to explore Belize, I came across the last thing I expected to find: A professional guitar player. He was actually the manager of a small retreat where we spent a few nights. Between admiring crocodiles and enduring a hurricane, I ended up discussing the "sound mixer versus guitar player" topic with him.

For an incredibly talented player who's been at it almost as long as I've been alive, he was still missing a few pieces of the puzzle. "They can never hear me in the audience," he griped. "We don't have anyone here who is good with sound."

That pushed a few of my buttons, and when he offered to show me a few things, I quickly saw the issue(s). He was having three primary problems, and they're common to players who struggle to perform with an audio system.

FIRST…

The reverb and effects that sound so awesome from two feet away were not helping in the mix. Another young player I worked with did the same thing. (I called him the electric hairball.) His effects included heavy distortion, weird reverbs and random delay. From front of house, I couldn't tell one chord from another. It was one big washed-out mess.

My Belizean friend was creating a similar situation. His choice of effects generated a tone that was virtually impossible to amplify. I explained that the room was already creating delay and reverb issues, and that his effects were making it worse. From there, we tweaked his effects down to a point where the instrument was still "nasty" enough for him to be happy but clean enough that I could pull it into a mix with some solid definition.

NEXT…

The whole time he was showing off his Satriani-style skills, I couldn't stop staring at the flashing tap delay on his Fender Champion amp. The song was around 85 beats per minute, but the delay light was flashing closer to 100. He had no idea what the tap delay was for.

After showing him how to follow the song tempo by tapping that little button at quarter, half and whole notes, everything changed. His effects began to blend with the song instead of working against it. Simple issue, but too many guitar players seem oblivious to it.

AND FINALLY…

There was the age-old microphone in front of the cabinet versus line output decision. He'd been insisting on giving the sound crew a direct feed instead of letting them mike the cabinet. That might not have been an issue except for the crazy settings he was force-feeding them.

My suggestion? If there are enough inputs available, use both. It gives front of house folks a choice of the better signal or the option of blending the two. Given the option, I prefer to set an SM57 on axis, tight to the grille, aimed about halfway from the dust cap to the surround. It just usually seems to produce an accurate representation of the overall tone.

I also can't begin to tell you how many times I've watched a player move the mic, intentionally or by accident. Off-axis positions, distance from the grille, amp settings too loud or too quiet – it all adds up to problems in the house. My tips were simply based on years of being dropped into one church, club, and festival situation after another where we needed consistent techniques that always worked.

I also firmly believe that mixing requires as much diplomacy as skill. This point was verified when my new buddy aggressively shook my hand and thanked me. Advice received with zero resistance.

The battle between the crew and musicians is manageable when we're legitimately concerned about them and helping them to be their best. Give it a try the next time the electric hairball attacks. LSI

Senior editor ***M. Erik Matlock*** *has worked in professional audio for more than 20 years in live, install, and recording.*


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IN & OUT

The differences between analog and digital interfaces. *by Pat Brown*

It's all about inputs and outputs (I/O). How do I get an audio signal from one to the other? The ongoing evolution of professional audio has produced a number of viable digital interfaces to complement legacy analog I/O practices. The choices may seem confusing at first, but when you break them down the strengths and weakness of each become apparent.

In this overview, I will start with analog since it is familiar to most readers and serves as a reference for the discussion of digital formats. I will focus on professional interfaces only. While similar in many ways to consumer I/O, professional interfaces are more robust against electromagnetic interference (EMI) and allow much longer cables – both requisites for large sound systems.

A professional analog interface is a point-to-point connection between an output and an input (**Figure 1**). It is electrically balanced, which provides strong immunity to EMI. Cabling is shielded twisted-pair. The interface is impedance mis-matched, where a low output impedance (typically < 200 ohms, or Ω) drives a high input impedance (typically > 2000 Ω). The impedance mismatch simplifies the interface by (usually) eliminating the need to consider specific impedance values.

For example, a 100-ohm output would produce the exact same signal level into any high impedance input (10 kΩ, 20 kΩ, 30 kΩ, etc.). An "output" simply connects to an "input" – end of story. It is often permissible to passively split an analog output to drive several inputs. It is not permissible to "Y" multiple outputs together.

The signal is in the form of a time-varying analog voltage that can span a level range of over 100 dB. Signals are classified by the magnitude of this voltage (e.g., microphone level, line level, loudspeaker level). The signal flows in one direction only – from output to input. Cable lengths are limited by cable capacitance, and can approach 300 meters (1000 feet) in some applications.

Analog connectors are classified by the number of electrical contacts. The impedance of cables and connectors is not a consideration, since analog audio signals are low in frequency in terms of the electromagnetic spectrum (**Figure 2**).

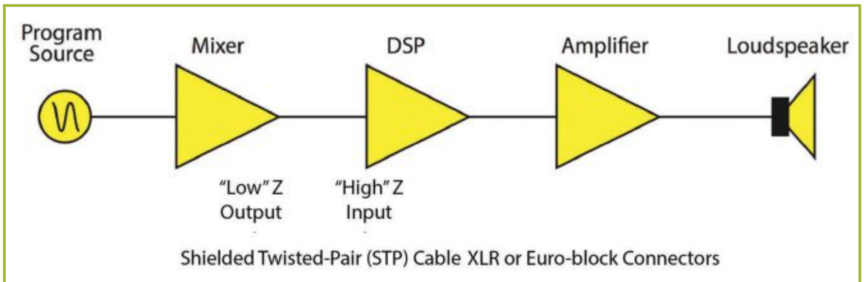


Figure 1: An analog interface is point-to-point, with no exact requirements with regard to cable type, connector type, and cable.

PROS:

- Simple and mature.
- Easy to troubleshoot.
- Ultra-reliable.
- Does not require impedance-matching between output and input.
- Signal propagation through a device or down a cable is practically instantaneous, with no practical delay or "latency" that must be considered.

CONS:

- The electrical properties of the cable can degrade the signal.
- Multiple audio connections, along with electrical "grounds" can produce ground loops, which may in turn cause hum and buzz problems.
- Multi-pair cables are heavy and expensive.

DIGITAL I/O – PRO AUDIO-SPECIFIC FORMATS

The professional audio industry has standardized several point-to-point digital audio formats. These are designed to largely insulate the user from the complex inner workings of the interface and the complexity of the data. The most common are AES3 (USA) and AES-EBU (Europe). I'll use AESx to refer to both. They are mostly identical except for a few details.

Like analog, AESx (**Figure 3**) is a one-way connection between an output and an input. AESx was designed to allow the use of balanced analog cabling and connectors, most often an XLR male output driving and XLR female input via a shielded twisted-pair cable. For short runs (< 30 meters, or 100 feet), one may use the same cable as for a balanced analog interface. Low-capacitance cable designed specifically for digital I/O can extend the cable length to 100 meters (300 feet). As always with cabling, your mileage may vary depending on the specifics of your application.

Other cabling options exist including coax (AES3id – essentially a video interface) and category cabling (e.g., Cat-5e). AESx can also be deployed using space-saving DB25 connectors on multi-

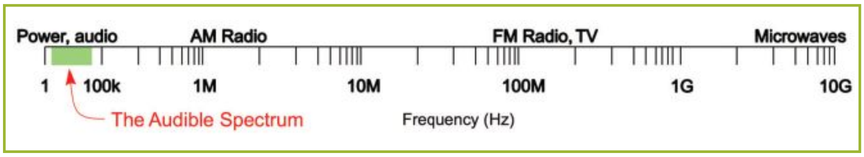


Figure 2: Analog audio (sometimes called baseband audio) is below 100 kHz in terms of spectral content. This is "low" frequency in terms of the entire electromagnetic spectrum.

pair cable (up to 16 channels). The exact implementation is product specific, and there are passive methods for converting between each. The manufacturer determines the specifics based on the target market for the product.

An important difference between analog and AESx is that an AESx connection carries two audio channels over a single twisted-pair. There is nothing special about the XLR connectors, and some products allow the user to select between analog and digital using the same input or output connector, saving space on the chassis (**Figure 4**). The high compatibility with analog I/O cabling and connectors is a major strength of AESx.

DIGITAL AUDIO QUALITY

The objective of digitization of the analog signal is lossless encoding. This requires a sample rate greater than two times the analog signal bandwidth, along with a dynamic range that approaches 100 dB. So, the main attributes of the data are the bit depth and sample rate. Most devices default to "24/48" which means 24 bit words flowing at a 48 kHz sample rate. Greater sample rates are possible, and are sometimes used for special applications.

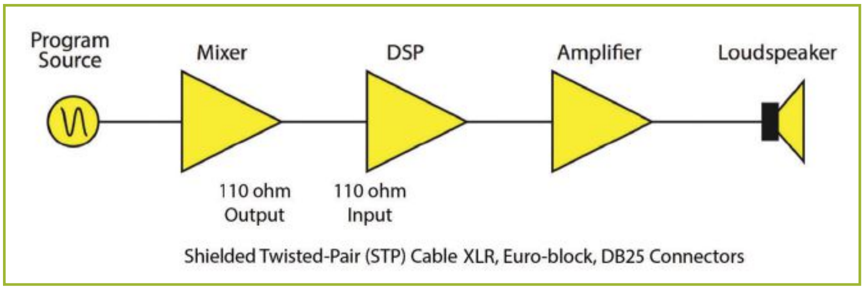
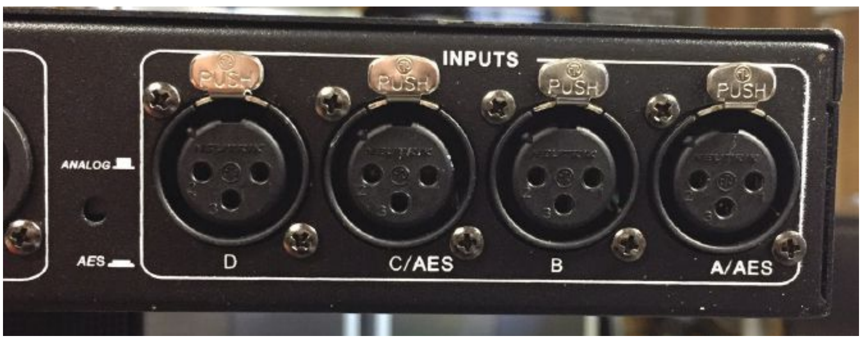


Figure 3: A digital audio interface has more exacting requirements than an analog interface. This includes an (approximate) impedance match, and there are less connector options.



COURTESY MARANI

Figure 4: The input connectors for a four-input AESx processor. Note that four XLRs are required for analog, but only two are required for four channels of AESx (two channels on each connector). There is a switch to select between analog and AES.

The sample rate, bit depth, and number of channels can be multiplied to yield the "data rate." This gives us a simpler, one number way to describe the digital audio resolution. **Figure 5** shows the minimum data rate for one channel of full-range audio. A strength of digital audio in general is that the data rate can be reduced using lossy or lossless compression schemes. AESx signals are usually not compressed to reduce the data rate, since the minimum requirements for full resolution are easily met by the current technology.

The bit stream contains the audio samples (or payload) along with metadata that carries information about the signal required for decoding. This "protocol" must be adhered to by both the output and input circuits, or no audio will flow.

Since the data rate approaches 10 MHz (a 1.5 MHz fundamental plus odd harmonics), the interface must be impedance-matched (110 Ω) to prevent degradation of the signal traveling down the cable by reflections and standing waves. Like all impedance-matched topologies, AESx is a one-to-one connection – one output drives one input.

Multi-channel versions include AES10 (MADI) and AES50 (HRMAI). While based on AESx, the details regarding clocking are quite different. These multi-channel interfaces are popular for connecting digital mixers to their respective stage boxes.

IT'S ABOUT TIME

All digital signals require a clock signal to keep multiple components in synchronization. AESx has the clock signal embedded in the data stream, so a dedicate word clock connection is often not needed for simple systems. AESx components often have word clock I/O for more complex systems.

The output of a digital component is always latent relative to the input. There is an unavoidable delay. Latency is cumulative, and system designers have a "latency budget" that must be observed to avoid timing issues.

PROS:
- "Analog-like" regarding I/O.
- Two channels on one cable.
- High immunity to electro-magnetic interference and ground loops.
- 24/48 resolution is lossless with regard to analog I/O.

CONS:
- Potential clocking issues.
- Requires special instrumentation to troubleshoot.
- Due to the high frequency nature of the signal, an impedance match is required between an output and an input.
- Output is always latent – the only variable is "by how much?"

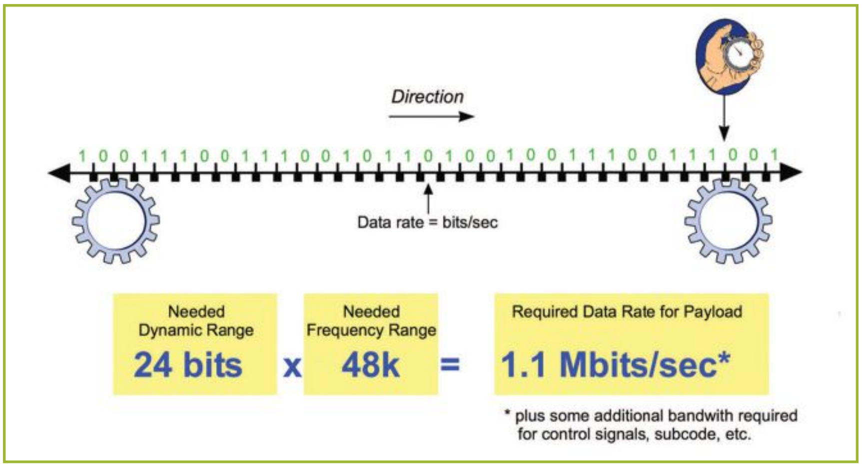


Figure 5: Sample rate and bit depth can be expressed as a data rate for the digital bit stream, a sort of conveyor belt for the one's and zero's that make up the digital signal. AESx digital audio consists of two channels, with a data rate of about 6 Mbps.

DIGITAL I/O – DATA NETWORKS

Once the analog audio signal is digitized, it's data. Technology has provided some very efficient means of data transport between computers, the most widespread of which is the Ethernet network. Audio-over-Ethernet (AoE) exploits the low cost and ubiquity of data networks to transport audio data. While AESx and its siblings are audio industry-specific, AoE utilizes technology from the Information Technology (IT) industry as a means of audio transport (**Figure 6**).

The analog waveform is first sampled, and then "packet-ized." A packet consists of a few audio samples and some additional "meta" data necessary to traverse the data network. The AoE protocol (e.g., CobraNet, Dante, Q-Sys, Ravenna – there are others) assures that the data gets "on" and "off" of the network to produce a contiguous waveform at the receiving end. High-speed networks (e.g. 1000BaseT or "Gigabit") can transport hundreds of channels. Unlike the "one-way street" AESx formats, a single cable can carry signals in both directions.

IT'S NOT AUDIO. IT'S DATA!

Regarding I/O, it's a data network and the rules are rigid and established by the IT industry. The user is largely insulated from the complex inner workings of the network, and deployment in smaller systems is relatively plug-and-play.

Audio practitioners must acquire IT skills to deploy large AoE networks. Ideally, an audio-only data network is established using dedicated network switches. Increasingly, AoE may reside on the venue's backbone with other data traffic such as email, web browsing, point-of-sale, etc. – the so-called converged network. On a converged network, the audio packets must have higher priority than other data traffic. This is called Quality of Service (QoS). A chunk of a large network can be reserved for audio data through use of a Virtual Local Area Network (VLAN). A VLAN requires a more sophisticated "managed" switch than a simple, dedicated network.

Due to the simple wiring, quantity of channels, and low-cost network hardware, AoE has become a very popular digital audio transport. A major difference between AoE and AESx (and its siblings) is that AESx, like analog, is point-to-point, requiring a direct connection between the output and input. AoE is a network, so signals can be routed between any network devices, regardless of where they reside on the network. This provides extreme versatility regarding signal routing for larger venues, campuses, studios, etc.

The audio practitioner must learn the specifics of the AoE "flavor" that they are using. This includes running a control application on a PC to route audio signals, and configuring network switches for the required QoS.

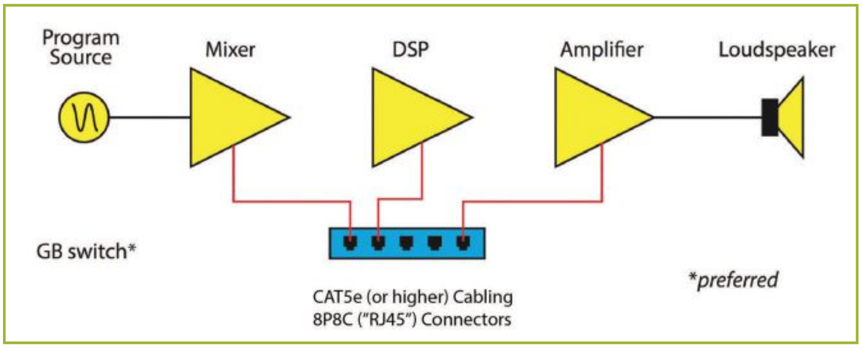


Figure 6: AoE requires the connection of each component to a network switch. The I/O connections are made using a configuration program (e.g., Dante Controller or CobraNet Disco).

WHICH AoE "FLAVOR?"

There are competing AoE formats. As of this writing, Audinate Dante enjoys the most widespread use. This can (and probably will) change with time, and competing formats such as Audio Video Bridging (AVB) (now referred to as Time Sensitive Networking (TSN)), will emerge as alternatives. AES67 is an initiative to produce inter-operability between the various AoE formats, so it may eventually be possible intermix AoE formats.

It's important to point out that AoE is just a transport. The quality of the digital audio is determined by the sample rate and bit depth. As such, all AoE formats should sound the same.

PROS:

- 24/48 digital audio.
- Hundreds of channels on a single Cat-x cable.
- Ubiquitous cabling and connectors.
- Low-cost hardware infrastructure.

CONS:

- Data networks can get complicated.
- Problems can be difficult to troubleshoot.
- Converged networks require cooperation with the IT department.
- No intuitive relationship to analog audio.
- QoS issues.
- Multiple AoE protocols exist, and no manufacturer supports all of them.
- IT is a different skill set, and requires specialized training.

SPECTRAL CONTENT & NEED FOR "EXACTNESS"

One way to frame the differences between the formats is in regard to the requirement for "exactness" of the interface variables. This, in turn, is based on where the signal spectrum falls in the overall electromagnetic spectrum.

Figure 7: A specialty tester for AESx interfaces. COURTESY WHIRLWIND

As with most things audio, it is a gradient, with no clear "right" or "wrong" answer.

Since audio by definition is low in frequency, analog audio interfaces are fairly permissive with regard to the interface details. One can sort of "get away with murder" and still get audio. Many wire and connector types will "work" but it is good practice to use twisted-pair cabling, ideally with a shield. Cable testing is basically a simple continuity test. The signal pair can be "bridged" with a butt set to detect the signal.

Digital I/O moves further up the spectrum. The interface details become more black and white. For example, analog audio simply requires a low impedance output be connected to a high impedance input. All digital I/O formats require an impedance match, but AESx is more permissive than AoE. This explains why we don't need "special" XLR connectors for AESx signals. It's also why Euroblock and DB25 connectors can work. Even analog cabling can work at short distances.

Figure 8: A stand-alone tester for network troubleshooting.

Cable testing is often performed by the audio gear itself, and whether the AESx input can "lock" to the AESx output. Special testers are available that allow the lock signal to be detected, and the audio decoded and listened to (**Figure 7**).

Higher up the spectrum (AoE on a gigabit network) the interface details are so stringent that they can't be left to chance. The interface, cables, and connectors have an exact specification and no decisions are left to the end user. While continuity testers can work at the most basic level, the testing can (and should) be much more sophisticated (**Figure 8**).

Along with cable testing there is a need to verify the system bandwidth, component addressing, Power-over-Ethernet (PoE) requirements, and more. Instrumentation for troubleshooting can be PC-based or stand-alone.

THE INEVITABLE METAPHOR

If you read my stuff then you knew this was coming. Here's a way to describe the various interface methods to the lay person.

Analog is like driving a jeep on a one-way dirt road. In spite of the bumps, the vehicle can weave all over the place and still get to its destination intact. AESx is like a Formula One car on an oval track. The higher speed requires a smoother surface and some boundaries. AoE is like a bullet train. There is an exact path and the margin for error is razor thin. At those speeds, any small discontinuity can have disastrous consequences.

This also explains why there are so many poorly performing analog systems in use. Impaired analog interfaces can still pass audio and no one may even realize they are impaired unless there is a direct comparison to a more ideal interface. So, as we move up in the spectrum from analog, to AESx digital, to AoE the interfacing details become more exact and there is less room to bend the rules. What we get in exchange for increased complexity are more channels and greater versatility – valuable benefits for many system types.

Digital audio has not obsolesced analog, and AoE has not obsolesced AESx and its siblings. All have their place. Analog still remains the standard by which to judge the fidelity of a digital system. Sometimes the simplest approach is the best, and sometimes it isn't. LSI

***Pat & Brenda Brown** lead SynAudCon, conducting audio seminars and workshops online and around the world. For more information go to www.prosoundtraining.com.*

SUBTLE MATTERS

Microphone selection and techniques for a range of acoustic instruments.

by Craig Leerman

Acoustic performances, and the instruments themselves, usually need to be handled differently than the electronic side of things. Thumping kick drums, compressed bass guitars and the like are not the norm for performances intended to be more musically subtle.

A more natural sound is desired and it starts with microphone choice and deployment. Fortunately, most instruments can be picked up quite nicely with just three types of mics that most sound folks already have in their inventories.

A mic locker made up of small-diaphragm condensers, standard dynamics, and a few large diaphragm dynamics can get the job done in most cases, combined with appropriate techniques. Here are some approaches that have proven to work well in my work with acoustic performances over the years.

Acoustic Guitar. While many modern acoustic guitars have built-in pickups that sound quite good, some performers prefer older models without them (or they simply don't like to use pickups). This entire article could be devoted to the different ways of capturing acoustic guitars, but here are two of my favorites.

First, employ a small-diaphragm condenser like a Shure SM81, placed about 8 inches away from the guitar and pointed between the sound hole and the neck. This location captures an accurate overall representation of the instrument.

An Electro-Voice ND66 small-diaphragm condenser capturing an acoustic guitar.

CREDIT: BLOGS.CHAPMAN.EDU/MAGAZINE

A bluegrass group practicing the single-mic technique for both instruments and vocals, utilizing a Shure wireless transmitter.

Second, change to a large-diaphragm condenser like an AKG C414, and set it to a cardioid pattern with the same positioning. To my ear, at least, the larger diaphragm sounds better, but the downside is that it's more bulky.

Banjo. As the old joke goes, "What's the definition of perfect pitch? When the banjo lands squarely on the accordion after you toss it in the dumpster." (Sorry, couldn't resist.) And actually, I like the sound of banjo. Several clip-on mic systems capture the instrument well, but since I don't own any, instead I opt for a dynamic instrument mic. Recently I pointed a CAD D89 at the head, just below the bridge, and it worked quite nicely. I've also attained good results with condensers as well

Dobro. For the unfamiliar, dobro is a generic term for a resonator guitar played with a slide. The strings pass over the bridge, which is attached to one or more resonator cones, producing a distinctly metallic sounding guitar tone. There's no center sound hole, and the mic needs to be pointed at the resonator for best effect. An Audix D2 dynamic sounds very natural for this application, in my experience.

Accordion. A good accordion player is magical to me, an important part of music like Zydeco and traditional folk. Accordions come is three basic sizes that I refer to as Button, Cajon and Full. Button is the smallest, utilizing buttons instead of piano keys. There are still reeds on both sides. Cajon units are "medium" in scale, with piano keys as well as buttons on the other side. Full, of course, is the largest, with piano keys on one side and bass/chord buttons on the other.

There are some great clip-on systems for Full accordians, like the DPA d:vote stereo kit. My company's inventory doesn't have accordion mounts so we opt for a stand-mounted large-diaphragm cardioid dynamics like the Electro-Voice RE20 or RE320. Use two mics, one pointed at the reed opening on each side of the instrument. For Button and Cajon, a single cardioid mic placed far enough away to pick up the reeds on both sides can do the trick in capturing a full, honest sound.

Violin/Fiddle. If possible, I prefer to clamp a small condenser right on the instrument. In addition, Audio-Technica, DPA, and others offer high-quality small mics on goosenecks with clamps. If clamping isn't feasible, try setting up a boom stand with a small-diaphragm condenser about a foot away from the instrument, pointing forward of the bridge. Just make sure the mic and stand aren't in the way of the bow.

Violas, cellos, and upright basses (discussed next) are also in the string family, and in smaller ensembles, place a mic on each one. If the performers are playing in a section and individual mics can't be used, place mics on the section leaders (called first chairs in an orchestra) and then deploy area miking for the sections with small condensers, preferably at a one mic/two player ratio. The mics should be about 4 to 5 feet above the seated performers, and raising the stands up a little higher allows pick-up of three performers (if you're short of mics or channels).

An Audio-Technica ATM350U cardioid condenser with clip-on mount and gooseneck on a violin/fiddle.

A quick note before moving on: Acoustic orchestras are actually relatively easy to capture, but the trick is getting the right balance. Provided that the audience is in the "sweet spot" of a stereo PA, pan out the instruments in relation to where they're positioned on stage to give the mix some room. The conductor sets the balance and volume of each section. The PA should just reinforce the orchestra in the venue.

Upright Bass. Everything from a dynamic like a Shure SM58 wrapped in foam stuffed under the bridge to small clip-ons can produce good results with an upright bass. For players that don't move around, I prefer to stand-mount a large dynamic like an AKG D112 or Shure BETA 52 pointed at one of the "F" holes. Earlier this year in a Road Test of Lewitt Audio's Beat Kit Pro 7 drum kit, I put an DTP 640 dual-element mic on an upright. The two elements (large-diaphragm dynamic and small-diaphragm condenser) provided an excellent, very rich tone.

A DPA d:vote 4099T clipped to the bell of a trumpet.

Bluegrass Band. Many of the individual techniques noted here can be used with bluegrass (and other acoustic) groups, of course, but the single mic technique is also quite effective. In the "old days" many musical groups would opt for a single mic to capture live performances (and recordings). The mic is placed center stage, with the musicians spaced around it. Musicians get closer or farther away from the mic as needed for leads and solos.

This technique is popular again and a single mic onstage allows for good gain before feedback as there are no other open mics onstage. A large-diaphragm cardioid condenser like the Baby Bottle from Blue does a quality job, with a traditional look, for small bluegrass and acoustic ensembles. I've used a single mic for up to four performers, and if it's larger than that, split the group around two matched mics. Note that there still might be need to mic an upright bass with this technique in order to capture it clearly.

Brass. Trumpets, cornets, trombones, tubas and the like can get quite loud, so it's best to utilize mics designed for high SPL. For higher pitched instruments (i.e., trumpets) place a dynamic like a beyerdynamic M88 a few inches in front of the bell, and ask the performer not to inset the mic into the horn. Another solution is clipping a small condenser such as a DPA 4099 to the bell. For larger instruments (i.e., tuba), do the same or try placing the mic inside the bell. A sousaphone performer I recently worked with likes to drop an SM58 inside his instrument and walk around the stage. It actually sounded quite good.

French horns are also brass, but because they're usually played in orchestras or chamber ensembles with the performers hand inside the bell, they require different treatment. An RE20 or RE320 about 8 to 12 inches away, pointed at the bell, captures the unique sound of this instrument.

Woodwinds. Reed instruments are a bit different that horns in that the sound can emanate from the open keys in addition to the bell. Because of this, try placing a single mic directly in front of a clarinet or oboe to pick up the whole instrument. A larger diaphragm Sennheiser MD421 is my top choice here.

Saxophone. Also a woodwind instrument, but many sax players are used to a single mic at the bell. If a player prefers wireless, we stock some Shure SM98 and BETA 98 clip-ons that work with our systems. An Audio-Technica ATM350, DPA 4099 and AKG C519 also work great. When the performer is station-

The VP2 from AMT, a two-microphone setup for vibraphone and mallet instruments.

ary, an MD421 is another option. A few years ago I worked with a group called Saxophobia that featured the "world's largest" saxophone, with an SM58 on a gooseneck stand picking up the low notes from that sizeable instrument very well.

Flute. A common approach is placing a mic in front of the player's mouthpiece. The resulting wind noise into the mic can make every note sound like Jethro Tull. My method now is placing a Countryman E6 earworn directional mic on the musician so the breath is moving away, so all that's picked up is the instrument. This technique also works well with piccolos and pan pipes.

Hand Percussion. For most hand percussion instruments, dynamics like a Shure SM57, Blue 100i, AKG D40, Heil Sound PR20, and so on work well, but if the percussionist is playing softer instruments like shakers, consider opting for a large diaphragm condenser like an AKG P220 or Joe Meek JM37. Some manufacturers also offer smaller condensers with goosenecks that can clip on instruments like bongos. Another approach that works well is to clip an omnidirectional lavalier like a Point Source Audio CO-8WL to a performer's chest. The body acts as a boundary and the percussion sound is picked up naturally. This works particularly well in an acoustic setting where you don't want a mic and stand blocking the performer.

Cajón. Pronounced "Ka-Hon," it's a wooden box that's played like a drum. The percussionist sits on the top and hits different area to different (lower through higher) tones. Mostly the striking is done with hands, but sometimes the musician will use sticks, brushes or even a foot pedal and play it like a kick drum. Some cajóns have metal inside to give a snare-like snap to the upper tones. Deploy two mics on a cajón if possible, one placed at the rear pointed at the sound hole for the lower tones and the other pointed at the face where the player hits. I think of it as any other drum and use dynamic mics like SM57s, MD421s and D4s, sometimes opting for a larger diaphragm dynamic for the rear sound hole. If only one mic is available, place it at the front, aimed at the playing surface.

Tympani. These big drums can get quite loud so choose a mic that can take it. I usually capture them individually, placing a large dynamic like an AKG D112 or BETA 52 about a foot away from the head, pointing at the rim.

Chimes. Try to capture them from the rear so as not to get in the way of the player. A large-diaphragm condenser is my first choice.

Mallets. This family is comprised of wooden key instruments like xylophones and marimbas as well as metal key instruments such as bells and vibraphones. I use the same technique no matter the instrument – position two mics spaced about one-third apart along the instrument length, 2 feet above and 1 foot behind the keys, and pointing at the lower row (whole notes) of keys. Dynamics can work well but I usually opt for condensers such as Joe Meek JM27s.

Piano. There are dozens (and dozens) of fine ways to mic a piano. Earthworks has a system I'd like to try out called the PianoMic PM40. It consists of a telescoping bar with two attached mics. The bar can adjust to any piano size and the lid can be closed. No stands are required and the mics can be adjusted close to or away from the dampers for a natural sound.

My usual method is a pair of condensers spaced about one-third across the strings, positioned 6 to 12 inches away from the dampers toward the back of the piano. This technique requires an open lid or one that's close-sticked because the mic stand booms will interfere with closing the lid.

The telescoping bar and two mics in an Earthworks PM40.

If the lid must be closed, I capture the piano with Audio-Technica AT871R boundary mics (now discontinued), but any good boundary mic will work. A single boundary placed in the center of the lid about 12 inches away from the dampers sounds nice, and two boundaries taped to the lid (positioned as above) sounds even better. LSI

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

BLANKET COVERAGE

Loudspeakers in many forms deliver diverse functionality. ***by Live Sound Staff***

RCF line arrays flying in support of 4th Street Live in downtown Louisville.

SATISFYING INCREASING DEMANDS IN LOUISVILLE

A system upgrade at the live performance stage at 4th Street Live, an open-air mall-style entertainment complex covering a two-block area in downtown Louisville, is headed by new RCF line array technology. An A-frame glass roof encloses the area, with the stage situated on the street going to an expansion of event programming to include more national artists.

Previously the venue had been leasing gear to meet touring requirements that can now be met by the upgraded system. Specifically, it incorporates 16 HDL50-A 3-way cabinets flown in left-right arrays flanking the stage, joined by eight SUB9007-AS dual-21-inch subwoofers.

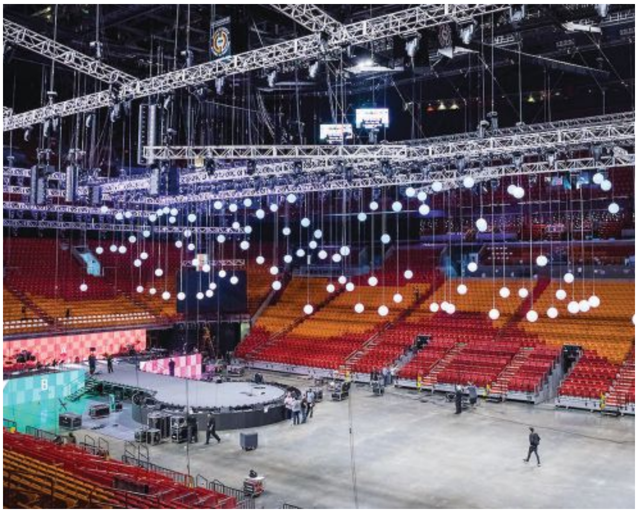
"We knew of the quality and customer service of RCF, and with the introduction of the new HDL50-A, knew we could satisfy both local productions and any national act we schedule to come in," says 4th Street Live production manager Dave Moskowitz. "The value RCF offers in the HDL50-A certainly was a factor in the decision," he adds, noting that the acquisition will pay for itself in about one season of events.

"And with the functional ease of use of RCF's RDNet control software, we're able set up the system for various program formats," Moskowitz continues. "Plus, being outdoors (even though under roof), with the wind and Ohio Valley heat and humidity, we can continuously monitor all system functions and components."

CONSISTENCY ON TOUR WITH SHAWN MENDES

Canadian singer-songwriter Shawn Mendes recently toured North America with Adamson E-Series loudspeakers provided by Eighth Day Sound (Cleveland). "I 'd not used the rig before, but had heard it many times and really liked the sound quality," explains Tom Wood, front of house engineer for Mendes.

Adamson E-Series full-range and sub arrays supporting Shawn Mendes on the road in North America.



More than 100 EAW Adaptive modules flown throughout the American Airlines Arena in Miami.

"When I talked with Eighth Day Sound and they explained how flexible the system was, I knew it would work, and it was exceptional."

Larger arena shows used left-right arrays made up of a dozen E15 and three E12 enclosures accompanied by 16 E119 subwoofers flown in two arrays of eight. Twelve S10 enclosures per side were selected for out fill when necessary, with further low end provided by 16 E119 ground-stacked subs. Eight more S10s delivered front fill.

The tour's final concert, at Madison Square Garden, required 270 degrees of coverage, with additional E-Series enclosures brought in to cover the larger venue and two more arrays, each made up of eight S10 enclosures, extending coverage even further.

"The bar has been raised for what people expect sonically at concerts," Wood states. "Making sure the people in the front are not having a radically different show from those in the back can be difficult to achieve. With the Adamson system, we consistently found a happy medium."

ADAPTING AT THE AA ARENA IN MIAMI

In partnership with Miami-based Mix3 Sound, EAW recently brought its largest Adaptive system ever to the 2016 Apostolic and Prophetic Conference (CAP) in Miami to deliver sound reinforcement to an audience of more than 20,000 at American Airlines Arena. The program included live and recorded music as well as a range of presentations.

The system incorporated 116 full-range modules across 16 column arrays. Each acoustical component within a module is independently powered and processed. Due to the 360-degree seating and the in-the-round staging of the event, the system needed to be flown very high. Because the vertical coverage of the arrays is adaptable from 90 degrees down to 90 degrees up, supplemental down fill or up fill enclosures were unnecessary.

"Having this level of control allows us to easily make any changes necessary without having to lower the system," states James Bamlett of EAW. "The video walls were not on any of the original drawings, but with the Adaptive system we had the ability to change the coverage to avoid hitting the walls without having to re-hang the arrays. This would not have been possible with a traditional line array system. Just through our software, we were able to change the pattern of the system and avoid reflections."

AN EVENT AS BIG AS THEY COME IN POLAND

GMB Pro Sound (Warszawa, Poland) deployed more than 600 d&b audiotechnik loudspeakers, nearly 150 D80 amplifiers, and 29 delay towers for one stage to deliver reinforcement at World Youth Day held at Blonia Park, near Poland's ancient city of Kraków. The event presented an opportunity for Pope Francis to engage with hundreds of thousands of young people, who were also treated to a music festival with artists of a wide range of different backgrounds and genres.

The widespread system was both designed and deployed by GMB. The mains included two long J-Series arrays left and right, in fills comprising V-Series, and two pairs of out fills realized with hangs of J-Series and V-Series. The rest of the J-Series cabinets were used on the delay towers. These loudspeakers

Just a portion of the massive coverage area handled by a range of d&b audiotechnik systems in Poland for World Youth Day.

were joined by almost 50 B22-SUBs placed near the stage and more than 30 J-SUBs hung on the first ring of towers.

The system was controlled via a d&b R1 Remote control network from a central control container located behind front of house. Audio and network distribution was realized via Optocore, with a d&b DS10 audio network bridge in every tower serving as a switch and multicast filter.

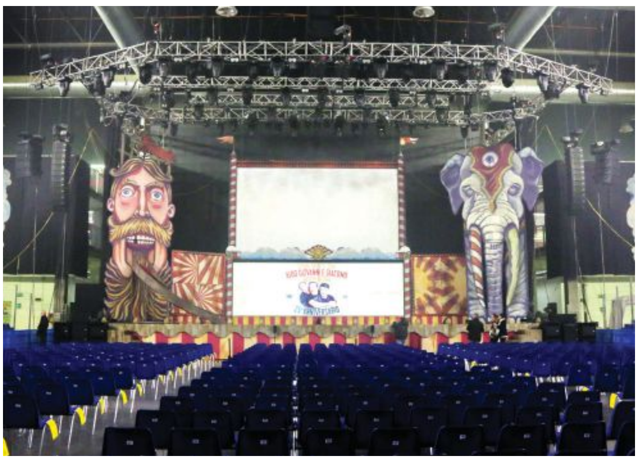
Forward planning was a key element, as GMB's Remigiusz Kasztelnik explains: "Naturally, it all began with not just one but several ArrayCalc project files, as we could not fit the whole system into one. It was absolutely crucial in designing the system, making sure the general tonality of all towers was the same, and presetting delay times. Laminated print-outs from ArrayCalc, detailing the mechanical configuration of the respective array, accompanied the equipment set for every delay tower, so that the numerous crews that were working on rigging the system did not always have to consult back with the project management staff."

REPRODUCING COMEDY & MUSIC IN ITALY

For the 2016 tour by "Aldo Giovanni e Giacomo," comprised of three popular Italian comic performers and also featuring live orchestral music, sound company Joint Rent (Ravenna, Italy) chose to invest in NEXO STM Series modular arrays. The inventory includes the new STM M28 serving as main cabinets alongside B112 bass and S118 sub modules.

In bolstering its extensive range of NEXO cabinets, Joint Rent was able to reconfigure systems to suit the different sized venues on the itinerary. The PA typically incorporated 12 STM M28s per side, expanded to 16 per side in the largest venues. Six B112 bass modules and eight STM S118 subs delivered the low end. GEO D10 side fills were exchanged for GEO M6 cabinets in smaller theatres, with NEXO's new ID24s for monitors and front fill.

"From the beginning, we believed in the potential of the M28 and the STM concept. For a tour like this, it was definitely the right PA product, offering complete modular assembly, very fast rigging and ease of set up," says Gianni Fantini, CEO of Joint Rent. "We won our bet with the crew and the production. They couldn't believe that such a small array could push out all that volume, and that we would be able to get coverage in all the different venues (35 meters in the smallest, 75 meters in the Assago Forum) without a delay system. We just varied the number of M28 main cabinets, and enjoyed the looks of surprise on their faces."



NEXO STM Series modular arrays set to deliver the laughs (and orchestral music) in Italy.

PHOTOGRAPHY BY RALPH LARMANN

Celine Dion performing in French-speaking lands with help from the Meyer Sound LEO Family.

A FAMILY AFFAIR FOR UNIQUE DATES WITH CELINE DION

Solotech (Montreal) utilized Meyer Sound LEO Family loudspeakers on Celine Dion's recent arena-scale, 28-show tour of French-speaking lands on both sides of the Atlantic. For the Paris run, the system was configured with 52 LEO-M and 12 LYON loudspeakers in the four main front arrays, with dual arrays of 16 LYON-W each covering the far side seating and eight LEO loudspeakers for rear delays. Sub-bass was supplied by end-fired arrays of six each 900-LFC low frequency control units flown behind the left and right arrays, augmented by 24 1100-LFC elements floor-stacked as three cardioid arrays.

The systems for the 10 Montreal shows, as well as five performances at the Videotron Centre in Quebec City, deployed most of the same loudspeakers but in slightly different configurations as suited the acoustics and layout of the venues. "When you looked at the IMAG screens and listened to the system, everything seemed close and intimate, even when you were 200 feet from the stage," remarks system designer/engineer Francois "Frankie" Desjardin, who has prepped the audio system for every Dion show since 1992. "The 'proximity effect' you get from the LEO system is remarkable. It removes the gap between performer and audience.

"The system is very flexible and easy to configure," he continues, "and it worked great for Celine. The dynamic characteristics of LEO are really outstanding. It goes from really quiet music through moderate pop right up to rock. The intelligibility is also very high, which helps us to keep everything absolutely transparent."

AN AUDIO THING HAPPENED ON THE WAY TO THE FORUM…

The Forum, a 370-seat multi-use facility that serves as an entertainment space, lecture hall and classroom inside the new Taylor Institute for Teaching and Learning at the University of Calgary, offers a flexible sound reinforcement system headed by Renkus-Heinz VARIAi modular point-source line arrays.

The room's overall ceiling height is 40 feet, too high to use a ceiling loudspeaker approach. "Given the room's complexity and high ceiling, we decided on a line array for the main system," confirms senior systems designer Travis Seibel of The Sextant Group, which provided the design. "But because of the layout, we had to overshoot the location where the presenters stand. We could use modular point-source arrays as long as they were highly directional, and Renkus-Heinz VARIAi provides that."

The mains are deployed in a left-center-right (LCR) configuration, concealed in wall enclosures. Left and right arrays employ five VARIAi loudspeakers, with the top three delivering 7- by 90-degree coverage, the next one down at 15 by 90 degrees and the bottom cabinet at 22 by 90 degrees. The center array consists of six cabinets, with four 7- by x 90-degree boxes on top and two 15- by 90-degree boxes below. The center array is primarily used for speech, although it also supports program music if desired. Sub-low energy is supplied by four R-H CF15S subwoofers set on the floor within cavities.

Renkus-Heinz VARIAi arrays, heard but not seen because they're concealed, deliver coverage at The Forum at the University of Calgary.

The client also wanted to provide surround sound for films, with six left and right wall-mounted R-H TRX61 2-way loudspeakers handling this role, joined by two TRX82 loudspeakers mounted in the back. And, the mezzanine is served by an R-H IC7-II mechanically steerable line array on each side, hidden behind fabric.

Alcons Audio QR24/QM24 thin-profile loudspeakers flanking the stage at Norway's grand old Trøndelag Teater.

WHAT'S OLD IS NEW AGAIN IN NORWAY

First opened in 1816, Trondheim, Norway's Trøndelag Teater is the oldest Scandinavian stage in continuous use. A 1990s rebuilding project saw the original auditorium, the Gamle Scene (The Old Stage), incorporated into a modern complex which added three further stages, and this year has seen Alcons Audio QR24/QM24 pro-ribbon loudspeakers installed in the Gamle Scene to provide an audio upgrade.

Specifically, the new system comprises five Alcons QR24 and QM24 modular, pro-ribbon column arrays per side powered by two Alcons Sentinel 10 amplified controllers. Supplied, installed and commissioned by Trondheim Lyd, the gear replaces an existing point source PA.

"The room has a unique acoustic quality, while the other main factor was that we needed a system that could adapt to the style and be a modern part of a listed baroque stage from 1816, which is also used without speech amplification," explains Jomar Johansen, head of production at Trøndelag Teater.

"The layout of the room is acoustically challenging and being able to effectively steer the sound around the balcony front, without losing the high frequencies, is not something all systems can do," adds the theatre's Erlend Aune. "The arrangement of an array of five Q-series per side avoids throwing high frequencies at the walls and balcony front, providing a complete array which doesn't reflect too much sound back to the stage." LSI

WHAT'S THE MEASUREMENT?

Understanding and properly using RTA and FFT.

by John Murray

EDITOR'S NOTE: This fine article was featured in the June 2004 issue. We reprint it here in celebration of our 25th anniversary.

What's the best way to equalize a sound system – by ear or by measurement? The short answer is both. Each method compliments the other. The ultimate qualification for sound quality is the ear. If it doesn't sound right, nothing else matters.

I once spent two-plus hours tuning a church sound system using the same methods I've employed on more than 100 systems, and the resulting sound was terrible. Rather than trying to convince my ears that the very good curve on the laptop screen sounded just wonderful, I had to trust my ears that something was wrong.

It turned out that my calibrated microphone was damaged. Once the process was repeated with an undamaged mic, the sound quality matched the curve.

The ear is the final judge. No matter how enamoring the technology, common sense must prevail. However, due to things like illness, drug effects, fatigue, poor acoustic memory (from which we all suffer) and hearing deterioration due to age, tuning a system purely by ear will not produce consistent sound quality nor the absolute best a system can do.

Tuning by measurement will uncover problems that the human ear just isn't very good at detecting, but they are problems that make the difference between sound that's "O.K." as opposed to spectacular (and consistent). I challenge anyone to find the one out-of-polarity transducer in a large system by ear. You might be able to tell something is wrong at a particular listening position, but you just can't tell exactly what. Measurement pinpoints that type of problem exactly.

Don't get me wrong – the ear is ideal for certain things. For example, balancing levels between separately amplified multi-way loudspeakers. Level matching two jagged response curves of a woofer and high-frequency section on a computer screen is much more difficult to get right than by ear. This is particularly true when changing just 1 dB of relative level can completely change the character of the speaker system.

And there are many systems that have been "TEF'ed, SIM'ed and Smaart'ed" that plain just don't sound good. Could this be a big reason some sound folks think that measurement systems just don't work? I think so.

But the issue lies not with measurement systems, which are improving steadily. No, the largest problem, by far, is operator error. Rather than continue the debate, let's get busy with addressing the errors that commonly plague us as we tune by measurement.

BIG THREE

Operator errors fall into three general categories:

1. The flat RTA response misnomer
2. Improper measurement-mic placement, and
3. Attempted equalization of multiple-source or multiple-reflection-contaminated, FFT measurements

It's been my experience that 95 percent of all sound systems are equalized improperly due to these three errors, and this is why some "road dog types" thoroughly mistrust measurement geeks.

An RTA (Real-Time Analyzer) is a two-dimensional measurement system that displays energy in dB SPL or volts versus frequency in hertz. Meanwhile, TEF, Smaart, SIM and the like are all 3-dimensional (3-D) measurement systems that display energy vs. frequency vs. time. Therefore an RTA, unlike FFT (Fast Fourier Transform) based 3-D measurement systems, is time blind and lumps all energy occurring within a fraction of a second together. A fraction of a second is an eternity to a 3-D measurement system.

When measuring an electrical voltage signal, like pink noise at the output of an analog mixing console, the mixer's electronics have very little propagation delay. Electrical signals on an RTA display will very closely match what 3-D FFTs display. This is because electronics do not time-smear the original signal. Therefore, if an electrical pink-noise signal is flat on an RTA, it will also be flat on a 3-D measurement system as well.

However, once an electrical signal is converted by a loudspeaker to an acoustical one and reflected around a room, the time smear is substantial. All the energy is not present at the same point in space at a single point in time, nor is it all dispersed from the speaker uniformly with respect to frequency.

The direct sound signal that travels straight from a loudspeaker to a measurement mic will be the shortest path and travel time between the two. Energy that first reflects off a side wall, then a back wall, then the floor, then to the mic, will take many milliseconds more. The later energy appears to arrive simultaneously with the direct sound on an RTA display and will be summed with it. Note that this reflected energy can be ignored by the display of a 3-D measurement system, and it is this characteristic that makes it a superior measurement system.

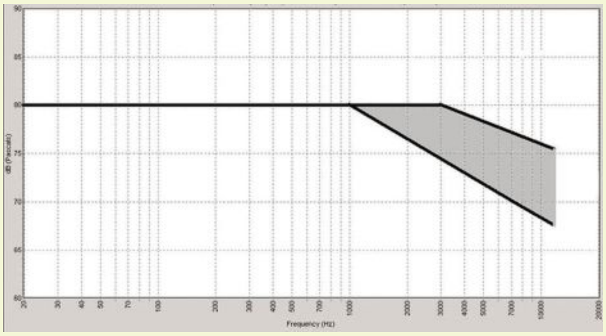


Figure 1: The ideal-room curve, or preferred-listening curve with its range of high-frequency variation in gray. Note that the lower limit is the original standard for cinema sound systems with beaming, radial-derivative, high frequency horns. The upper limit fits better for more-recent, constant coverage horns.

If you equalize a loudspeaker to be flat on an RTA display with the measurement mic in the middle of the listening area,

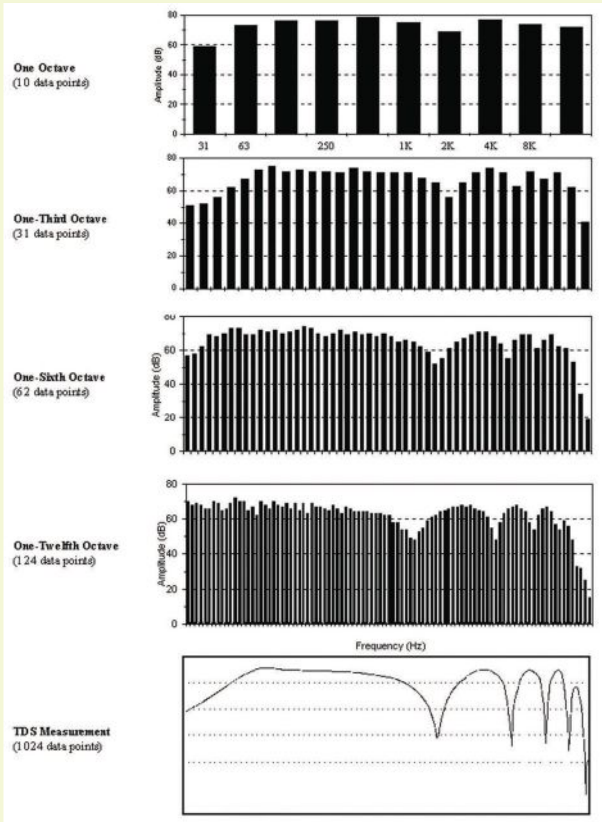


Figure 2: Various measurement-resolutions of a comb-filter caused by a 3-inch signal delay. Even the 1/3-octave resolution cannot clearly show that the frequency-response problem is non-minimum phase, time-oriented, and therefore cannot be equalized.

you'll be unpleasantly surprised by the resulting bad sound quality. This is not a similar measurement to the electrical one by any means.

TYPES OF CURVES

Probably the first person to recognize this difference was Dr. Charles Boner, the godfather of audio consultants who was one the first to practice equalization. He developed what has been called the ideal-room curve or preferred-listening curve. It's largely the acoustical power response of the loudspeaker system, as modified by air and surface absorption within the room (**Figure 1**).

What is the power response of a loudspeaker system? Other than being one of the most over-used and least-understood terms used today in audio, it is the sum total acoustic power that a loudspeaker produces.

For example, let's measure a two-way loudspeaker system in a large room, with the measurement mic in front of the speaker in the middle of the listening area, in the reverberant field beyond critical distance, where the direct sound is lower in level than the reverberant field energy.

Most of the high-frequency energy is aimed in the general direction of the mic due to the directional effect of a horn on the high-frequency (HF) driver. However, most of the low-frequency energy is not aimed at the mic because the low-frequency (LF) driver is omnidirectional for most of its passband. Therefore, if a flat anechoic or direct sound is desired, much more energy must be generated into the room by the LF driver to equal the sound pressure level (SPL) of the HF driver at the mic's position.

On an RTA display, this will look like the LF is a big haystack, and the HF gradually rolls off toward the higher frequencies where its horn exhibits better dispersion control. Obviously, this is Boner's curve. Where the exact hinge-point of the HF roll-off begins – and just how steep the roll-off is – depends on the dispersion of the HF horn, the number of devices, and whether the LF section has any directional control or not.

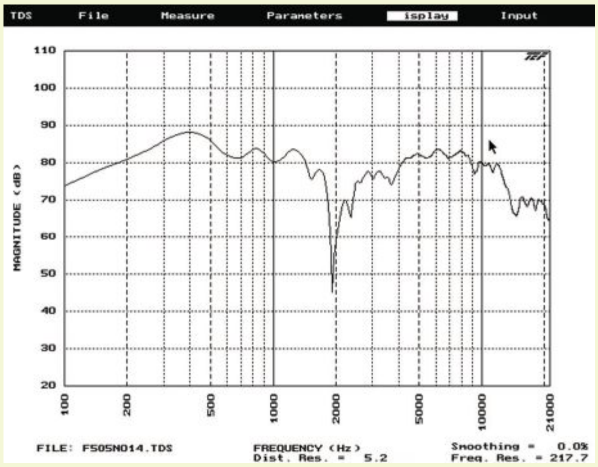


Figure 3: Non-synchronized LF and HF drivers cause the notch at 2,000 Hz. Since this problem is due to the later arrival of the signal from the system woofer, it is a time-related, non-minimum-phase event, and is also not equalizable. Viewed by an RTA in the reverberant field, this notch would not be revealed.

In a movie theater, where the size and absorption characteristics of the room, number, location, and specs of the loudspeakers are all fixed, a tightly defined curve can be used. For most other sound reinforcement applications, where every room and loudspeaker system is different, the modified power-response curve that produces a flat direct response can vary a lot.

In the days before 3-D measurement systems, one had to vary the hinge-point and roll-off characteristics of the curve until everything sounded right. This took a lot of time to get a satisfactory result. It had to be done with each individual system until manufactured one or two-box systems (mid-high packs and subs usually) came along.

TIME ORIENTED EVENTS

Let's shift focus to "non-minimum-phase anomalies." This $10 phrase describes time-oriented events that cannot be equalized. Examples of this are other delayed sources, like reflections or more distant loudspeakers, which are delayed enough in time to cancel the direct-sound energy from a loudspeaker at particular frequencies (**Figure 2**) .

Another is the notch at the crossover frequency of a loudspeaker system when the drivers are not time-synchronized (**Figure 3**). Neither of these frequency-response problems can be remedied by equalization. These are non-minimum-phase events and cannot be fixed with EQ.

Yes, the EQ had to be changed, but it was not due to reduced reverberation in the room.

This also applies for reverberation or echoes. Even a change in the reverberant nature of a room, due to a change in its acoustical absorption characteristics, is not an equalizable situation. I can already hear the protests to this statement: "But I've had to change the 'room EQ' numerous times when it was equalized empty early in the day, and then didn't sound right when the room filled with people because it was less reverberant."

Yes, the EQ had to be changed, but it was not due to reduced reverberation in the room. What had to be accommodated was the effect of temperature and humidity changes on the direct sound from the loudspeakers, not the reverberation. These effects on the direct sound are not time-oriented; rather, they are minimum-phase and can be equalized. Why else would a BSS Omnidrive include a meteorology probe? LSI

__John Murray__ is a 30-plus-year industry veteran who has worked for several leading manufacturers, and has also presented two published AES papers as well as chaired several SynAudCon workshops.

BASSBOSS SV8

Evaluating a compact 2-way self-powered loudspeaker.

by Craig Leerman

Having used a variety of excellent subwoofers from BASSBOSS over the years, I was eager to get my hands on one of the company's recently released full-range loudspeakers. The SV8 is a compact, 2-way design with a single 8-inch neodymium woofer (with 2-inch diameter copper winding voice coil) and a compression driver (1-inch throat, 1.7-inch diaphragm) mounted on a 90- by 60-degree constant directivity horn that's rotatable. A large venting port located between the woofer and horn is designed to deepen response.

The SV8 has a stated frequency response of 50 Hz to 19 kHz (+/-3 dB), with specified maximum sustained output of 112 dB and a short-term peak output of 118 dB. It's a self-powered box with a 700-watt, 2-channel Powersoft class D amplifier with built-in DSP for high- and low-pass filtering, phase alignment, EQ, and limiting.

The processor has four presets: preset 1 is full-range mode, offering frequency response to 40 Hz with limiters engaged to prevent overdriving; preset 2 offers slightly more SPL and response to 60 Hz; preset 3 provides another step up in SPL with the high-pass filter moved to 80 Hz; preset 4 offers the highest peak SPL, with the high-pass filter set at 100 Hz. Presets 3 and 4 are recommended when the SV8 is used with a subwoofer.

The enclosure is constructed of 15 mm Baltic birch plywood and is equipped with two handles built into the rear for easy transport, as well as a 35 mm pole socket

The BASSBOSS SV8 loudspeaker.

for stand mounting. Four optional internally braced fly points can also be ordered.

The cabinet is finished in black weatherproof bonded high-pressure polyurethane, and a white textured epoxy paint finish is an option. A rugged, perforated, powder-coated steel grille protects the woofer. The SV8 is indeed compact, with the slightly trapezoidal unit measuring just 18 inches high, 9.5 inches wide and 14 inches deep, and weighing in at 26 pounds. Options include a heavy-duty padded nylon transport cover and carrying bag as well as shoulder eye bolts for flying.

FIRING IT UP

Taking the two SV8s provided for this evaluation out of the box, I was impressed at just how small and light they really are. I like the built-in rear carry handles, one at the top and one at the bottom. The top handle is positioned so an upright cabinet is easy to grab with a single hand.

The handles also provide some protection for the rear-mounted amplifier module, which houses all of the connections and controls. There's an XLR input and XLR output/pass-thru, a detented volume knob, a preset switch, four indicator lights for the presets, Limit, Temp, Signal and Ready lights, a power switch, and an IEC connector for the power cord. Rows of heavy duty cooling fins help keep the amp module running at the proper temperature.

As with every Road Test I do, the first stop is my test bench where I can run signal through the gear, take a listen, and get familiar with controls. With a quick look at the manual to figure out what the various DSP settings did, I plugged the SV8s into my test console, played a few tracks, and listened. They sound great and have a lot of bottom end for such a compact footprint. In fact, the sonic quality equates to the older "studio" monitors I regularly use at the bench.

INTO THE WORLD

Satisfied that the loudspeakers were working properly, it was time to use them at some gigs. The first was a government advisory board meeting that my company has served every other month or so for the last four years. The configuration consists of a large U-shaped table with 14 to 20 table microphones, along with wireless earsets and lavaliers for the moderator and presenters and a wireless handheld for an audience mic.

The main loudspeakers are positioned facing the "arms" of the U-shaped table so that the board members can hear each other as well as the presenters and any media audio. This particular meeting was held at a local government building in a room unfamiliar to me. Normally we bring in a pair of self-powered, 2-way loudspeakers with 15-inch woofers because that's what's in our inventory, so it was nice having smaller units that could get the job done while also being easier to transport and set up.

During a typical meeting there may be a

high number of table microphones open at the same time when the board is discussing an item, so tuning the PA is crucial to the event's success. The SV8s were configured at full range, and I found that it took way less time for me to "ring out" the room than it normally does. The overall sonic signature was right on, and the amount of bass delivered when playing music was impressive from such small cabinets.

Next up was a corporate meeting in a hotel convention space, this time a more typical setup in a breakout room that seated a few hundred people. The inputs were a podium mic, a few wireless lavaliers on stage, video playback, some walk in/out music and a pair of Q+A (question and answer) handheld mics on stands located in the audience. At each side of the stage, we put the SV8s on poles atop the small self-powered subs I'd tested them with at the shop. Full-range coverage to the entire room was nothing short of impressive, and again, the system took little time to tune and was easy to set up. An in-house A/V tech handling the lighting and video playback commented on the sound quality and also on the compactness of the SV8s.

DIVERSIFIED USES

A week later we were back at the same hotel, but in a much larger room, and needed some front fills as the stage and room were very wide. In five minutes and using only a screwdriver, I easily rotated the horns so that the cabinets could be positioned on their sides. While the cabinets have a trapezoidal shape (the rear is narrower than the front), it's relatively slight, so they position just fine in fill applications like this one.

The power amplifier module, DSP controls, and I/O.

A day later, I received a call from a local DJ that my company partners with in supplying larger systems when needed. He had a large wedding in a spacious ballroom coming up and needed some of the "big stuff" from us, so we showed up with a pair of powered band-pass subs and the pair of SV8s. He was initially dismayed, thinking they weren't enough to handle the job, but he quickly changed his mind as soon as we fired up the system, telling me "O.K., you win." The SV8s easily covered the entire space populated by a few hundred guests, and they sounded great.

The final application was a large corporate meeting where I freelanced as an A2. As usual for these types of gigs, my work area was behind the stage. The A/V provider for the event had set up a small backstage monitors, each with a 5-inch full-range driver, at several locations so that the production staff (including me) could hear the show.

My location got a busted unit that sounded distorted no matter what I tried, so the next day for rehearsals I brought in an SV8 and used it instead. The small size worked well as a local monitor, and during the CEO's rehearsal, many of his staff gathered around my backstage table to listen because my audio feed easily sounded the best.

The BASSBOSS SV8 is a versatile unit, capable of easily handling full-range "on a stick" and "speaker with sub" duties in medium- to larger-sized venues. The compact size combined with stellar audio also make it well-suited for fill and delay needs, as well as studio monitor applications. LSI

U.S. MSRP/MAP: $1,495

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

KEY COMPONENTS

Cone and compression driver design parameters, along with recent models.

by Gary Parks

Inside every loudspeaker, from the practice room to arenas and everything in between, there are cone drivers reproducing low to mid frequencies and (mostly) compression drivers on horns for the upper range. Though do-it-yourself cabinets are less prevalent than they were two to three decades ago, there are still several manufacturers that offer high-performance professional components that may be used for replacements, upgrades, custom installations, or to build full-range loudspeakers of any scope.

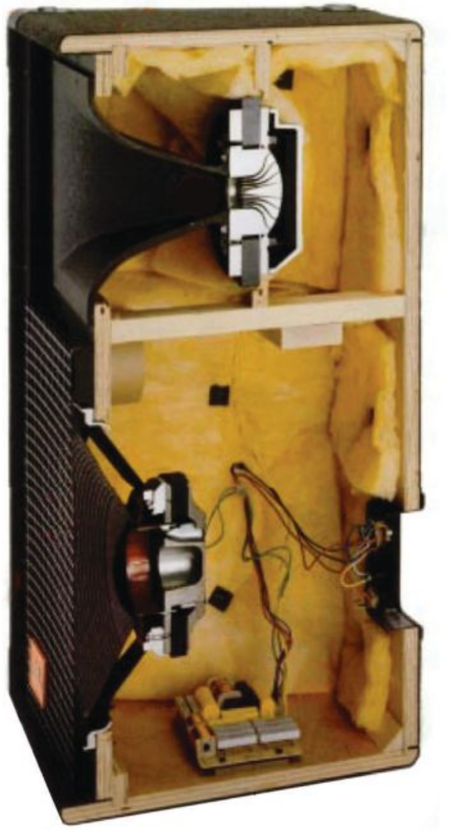
Almost all of these companies specialize in designing and marketing "raw" components rather than building systems. A couple do both, though with the emphasis toward the system side. In preparing this report I was surprised that one of the main U.S. players, which had been a pioneering force in providing components and expertise to individuals and audio companies to build their own loudspeaker systems, has completely shifted its considerable expertise from offering both components and complete systems to promoting many lines of powered and unpowered loudspeakers.

The latest generation of components – both within finished systems and available separately – incorporate numerous innovations that have led to more consistency, greater durability, and better overall performance.

SIMILAR YET NOT EQUAL

However, even though they share the same basic form factor, not every 15-inch cone woofer (as an example) will provide the same response characteristics when given the identical audio input, or be blindly interchangeable. Each has its own characteristic response, based on the materials used in its construction, the stiffness and density of the cone, the surround and spider compliance and resistance, the voice coil diameter and excursion, the movement potential provide by magnetic flux, coil layers and weight, gap width, and so on.

Some may be designed to strongly reproduce the lowest octaves in subwoofer applications, while others are better used for electric bass guitars – providing both fundamentals and the strings' brighter upper harmonics. Look at the manufacturer's description of suggested applications as well as the detailed Thiele-Small parameters that most data sheets provide.



A nifty cutaway view of a JBL SR Series II loudspeaker (circa 1997) that shows the placement of the cone and compression drivers.

POWER HANDLING MEETS SENSITIVITY

In addition, the power handling capability listed on the specification sheet does not automatically correlate with how loud a given driver will be. This figure interacts directly with the sensitivity (typically given as a 1 watt/1 meter SPL measurement) to offer a basic sense of the average and peak "loudness" that can be expected over a specified frequency range.

In simple terms, a driver with a stated power handling capability of 300 watts RMS or AES with a sensitivity of 100 dB, and another with power handling of 600 watts and sensitivity of 97 dB, will be virtually identical in loudness – though with the latter driver requiring twice the power

for the same loudness. Also, look at the details of the specifications to compare apples with apples, since the headline might say "3,200 watts" but could be a peak rating rather than a long-term rating.

INNOVATIVE FEATURES

When I began working with a microphone and loudspeaker manufacturer many years ago, neodymium had just been introduced to the pro audio world, first utilized in high-output dynamic mics. The concept of lighter weight compression drivers with neodymium magnetic structures was still an engineering dream that took another two or three years to reach fruition – let alone re-designing cone drivers with the material. Now, almost all component manufacturers offer drivers with neodymium magnetic structures, and others who exclusively design and build complete systems with proprietary components also use it to lighten the load.

To accurately track the impulse of a kick drum or percussive bass, voice coils are often extended to allow greater linear travel.

The precision brought by computer-enhanced design, tooling, and manufacturing, in combination with strong new adhesives and surround materials, has helped increase the durability of drivers, as well as the linear control of their voice coil's excursion. Power handling has also increased in many cases, propelled by these materials along with efficient methods of ventilating the motor structure and dissipating heat. The cones are also more advanced, containing light yet extremely strong materials like Kevlar, carbon fiber, and other additives and coatings.

Compression driver diaphragms use a variety of materials, ranging from Mylar and aluminum to titanium and even beryllium. Some have compliant suspension systems of different materials, and others will emboss the edges of the metal dome with 3-dimensional patterns designed to control the diaphragm's movement.

FREQUENCY RANGES

The lowest octaves have traditionally been covered using cabinets housing single or dual 18-inch woofers. Several manufacturers have also added 21-inch designs to the mix, providing an even larger cone area. To accurately track the impulse of a kick drum or percussive bass, voice coils are often extended to allow greater linear travel. Subs are usually rolled off at a fairly low fundamental frequency, so in this application, their ability to reproduce midrange frequencies is not used.

Full-range cabinets typically use 12- or 15-inch cones (although some go much smaller), and line arrays often have 10-, 8- and even 6-inch cones to minimize the height of the individual boxes. Midrange drivers in 3-way systems are usually 6 to 8 inches. The companies represented in the following round-up are likely to offer most or all of these sizes.

High-frequency drivers are also available in several sizes, and selecting which one to use will depend on the chosen crossover point of the low-to-mid driver in a 2-way configuration or between the midrange device and the HF driver in a 3-way system. A typical larger-format compression driver may feature a 3-inch voice coil bonded to a diaphragm with similar dimensions, and have a throat exit of perhaps 1.4 to 2 inches going into a horn (and be crossed over between, say, 500 Hz to 1 kHz). The smallest drivers typically have a 1-inch throat, a diaphragm with a diameter of 1.25 to 1.75 inches, and a crossover point of 1.5 kHz or higher.

Coaxial drivers present another option, incorporating both LF and HF drivers that radiate from the same point. They're utilized in both full-range loudspeakers and stage monitors, and when designed correctly, they can offer additional coherence and a more even sound field.

WHO MAKES 'EM?

I was amazed that of the pro audio loudspeaker component companies represented, half are based in Italy and Spain. The remainder split between the U.S. and U.K. While there are more companies manufacturing components, their output seems is primarily targeted toward car audio, ceiling- and wall-mounted distributed systems, or home and computer audio.

While traveling in Central America, I've run across loudspeaker systems made in China, primarily used in clubs and by local sound reinforcement companies. However, I found no pro audio woofers or compressions drivers offered for sale via a Google search, and only a Facebook page highlighting a major manufacturer of line arrays and other loudspeaker systems based in China. Manufacturers based in Japan, South Korea, and elsewhere also offer a variety of full-range loudspeakers and subwoofers for professional applications, but electronics and wireless microphone systems seem to be what make it to the pro audio dealers.

In our tour of newer models presented here, specifications include sensitivity, power handling, impedance, frequency response, and more, while the notes point out key design features as well as the breadth of company product offerings.

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

Celestion FTX Series | *celestion.com*

Coaxial loudspeakers act as a single source, concentrically aligning low and high frequency drivers which can provide improvements in signal alignment compared to traditional 2-way systems. Celestion's FTX coaxial range features fully combined LF and HF components powered by a Common Magnet Motor Assembly (same magnet for both elements), enabling the voice coils (the acoustic centers) of the two drivers to be closer together. This delivers improvements in coherence and time alignment for more natural sounding audio reproduction.

FEATURES:
- *The LF cone is used as an HF horn for superior off-axis response*
- *Demodulation rings minimize the effects of power compression and substantially reduce harmonic and intermodulation distortion*
- *The single magnet assembly means lighter weight and a more compact driver profile compared to conventional dual motor coaxial designs.*

OF NOTE: *The cast aluminum chassis FTX Series common motor coaxial drivers are available in 6-, 8-, 10-, 12- and 15-inch formats. Celestion also offers a range of pressed steel chassis coaxial drivers utilizing the same common motor technology.*

KEY SPECIFICATIONS:
Model: TF1225CX (TF1225 LF driver with modified CDX1-1730 HF driver mounted to the rear)
Size: 12 inches
Frequency Response: 50 Hz – 20 kHz
Sensitivity (1 W/1m): 97 dB
Impedance: 8 ohms
Voice Coil: 2.5 inches (LF)
Excursion/Xmax: 3.25 mm (LF)
Magnetic Structure: Ferrite
Power Handling: 300 watts RMS
Weight: 13 pounds

Eminence DEFINIMAX 4018LF
www.eminence.com

Size: 18 inches
Frequency Response: 31 Hz – 2 kHz
Sensitivity: 94.9 dB
Impedance: 8 ohms
Voice Coil: 4 inches
Excursion/Xmax: 8.5 mm
Magnetic Structure: Ferrite
Power Handling: 1200 watts (per EIA 426A standard)
Weight: 24 pounds
Notes: Designed as a high-power, low-distortion subwoofer in single or multi-driver systems; company offers a wide variety of ferrite and neodymium magnet structure cones, instrument and sound reinforcement loudspeakers, and compression drivers.

FaitalPRO 18XL1800
www.faitalpro.com

Size: 18 inches
Frequency Response: 30 Hz – 1.6 kHz
Sensitivity: 95 dB
Impedance: 8 ohms
Voice Coil: 4 inches
Excursion/Xmax: 20.2 mm
Magnetic Structure: Neodymium
Power Handling: 1600 watts AES
Weight: 22.5 pounds
Notes: Copper coil on glass fiber former; triple-roll surround; aluminum demodulating ring; vented magnetic structure; a variety of woofers, mid-bass, coaxial, and compression drivers are available.

RCF LF18N405
rcf-usa.com

Size: 18 inches
Frequency Response: 25 Hz – 1 kHz
Sensitivity: 97 dB
Impedance: 8 ohms
Voice Coil: 4 inches
Excursion/Xmax: 14.2 mm
Magnetic Structure: Neodymium
Power Handling: 1500 watts AES
Weight: 19 pounds
Notes: Optimized T-pole magnetic structure design; triple-roll surround; proprietary fiber-loaded cone; dual forced-air venting; edge-wound flat wire voice coils; a variety of woofers (including a 21-inch) along with mid-bass, coaxial and compression drivers are available.

REAL WORLD GEAR: CONE DRIVERS

JBL Selenium Pro 15SWS800
www.jblpro.com

Size: 15 inches
Frequency Response: 35 Hz – 1.5 kHz
Sensitivity: 93 dB
Impedance: 8 ohms
Voice Coil: 4 inches
Excursion/Xmax: 6.5 mm
Magnetic Structure: Barium ferrite
Power Handling: 600 watts AES
Weight: 22.9 pounds
Notes: Four-layer copper voice coil on fiberglass former; proprietary triple cooling system; dual, counteracting poly-cotton-fiber spiders; a variety of woofers, instrument models, and compression drivers are available, with standard and neodymium magnetic structures.

B&C Speakers 21DS115
www.bcspeakers.com

Size: 21 inches
Frequency Response: 30 Hz – 1 kHz
Sensitivity: 99 dB
Nominal Impedance: 8 ohms
Voice Coil: 4.5 inches; aluminum wire on glass fiber former
Excursion/Xmax: 60 mm peak-to-peak
Magnetic Structure: Neodymium
Power Handling: 1700 watts nominal; 3400 watts continuous
Weight: 32.6 pounds
Notes: Ventilated voice coil gap for low power compression; 4-layer voice coil; double silicone spider; aluminum demodulating ring; MBX-series mid-bass woofers and other cones also available.

Beyma 15P1000Nd
www.beyma.com

Size: 15 inches
Frequency Response: 30 Hz – 2 kHz
Sensitivity: 97 dB
Impedance: 8 ohms
Voice Coil: 4 inches
Excursion/Xmax: 52 mm peak-to-peak
Magnetic Structure: Neodymium
Power Handling: 1000 watts AES
Weight: 13 pounds
Notes: Forced-air convection for low power compression; double spider design; optimized under-pole magnetic topology to maximize flux density; a variety of woofers, coaxial models, mid-bass and compression drivers are available.

Eighteen Sound 18LW2500
www.eighteensound.com

Size: 18 inches
Frequency Response: 30 Hz – 1 kHz
Sensitivity: 95 dB
Impedance: 8 ohms
Voice Coil: 4 inches
Excursion/Xmax: 70 mm peak-to-peak
Magnetic Structure: Ferrite
Power Handling: 1600 watts AES
Weight: 36.8 pounds
Notes: Proprietary Double Silicon Spider (DSS); curved, ribbed, fiber-loaded cellulose cone; a variety of woofers (including a 21-inch), compression drivers, and neodymium magnetic structures are available; Eighteen Sound acquired driver manufacturer Ciare in late 2015.

Radian Audio 2216 Neo
www.radianaudio.com

Size: 15 inches
Frequency Response: 28 Hz – 2 kHz
Sensitivity: 101 dB
Impedance: 8 ohms
Voice Coil: 4 inches
Excursion/Xmax: 21.3 mm
Magnetic Structure: Neodymium
Power Handling: 1400 watts AES
Weight: 14 pounds
Notes: Ribbed, composite cone; double-wound coil; ventilated magnetic structure; a variety of woofers, coaxial speakers, and compressions drivers (including beryllium dome) are available.

Fane FC-185F01
www.fane-international.com

Size: 18 inches
Frequency Response: 30 Hz – 2 kHz
Sensitivity: 96 dB
Impedance: 8 ohms
Voice Coil: 5 inches
Excursion/Xmax: 52 mm
Magnetic Structure: Ferrite
Power Handling: 900 watts AES
Weight: 28.8 pounds
Notes: Cast aluminum chassis design; high total Q (Qts of 0.530) for larger enclosures; double-spaced suspension system; suitable for bass reflex and horn loaded designs; series includes models as small as 5 inches.

REAL WORLD GEAR: COMPRESSION DRIVERS

Eminence N320T-8
www.eminence.com

Throat Size: 2 inches
Frequency Response: 800 Hz – 20 kHz
Sensitivity: 110.4 dB
Impedance: 8 ohms
Voice Coil: 3 inches
Magnetic Structure: Neodymium
Power Handling: 100 watts AES
Weight: 5.5 pounds
Notes: Titanium diaphragm, ribbed for stiffness; polymer surround; N314T-8 with 1.4-inch throat are available.

FaitalPRO HF2000
www.faitalpro.com

Throat Size: 2 inches
Frequency Response: 500 Hz – 18 kHz
Sensitivity: 110 dB
Impedance: 8 ohms
Voice Coil: 2.9 inches
Magnetic Structure: Neodymium
Power Handling: 100 watts AES (above 900 Hz)
Weight: 5.9 pounds
Notes: Titanium diaphragm; aluminum voice coil on Kapton former; a variety of compression drivers with 1- to 2-inch exits are available.

RCF ND950
rcf-usa.com

Throat Size: 2 inches
Frequency Response: 500 Hz – 20 kHz
Sensitivity: 110 dB
Impedance: 8 ohms
Voice Coil: 4 inches
Magnetic Structure: Neodymium
Power Handling: 140 watts AES (above 800 Hz)
Weight: 7.8 pounds
Notes: Titanium diaphragm with polyimide suspension, on Kapton former; edge-wound aluminum voice coil; 4-slot phase plug; also available with a 1.4-inch throat.

JBL Professional 2450H/J
www.jblpro.com

Throat Size: 2 inches
Frequency Response: 500 Hz – 20 kHz
Sensitivity: 111 dB
Impedance: 8 or 16 ohms
Voice Coil: 4 inches
Magnetic Structure: Neodymium
Power Handling: 100 watts continuous (150 watts above 1 kHz)
Weight: 10.5 pounds
Notes: Titanium diaphragm with 3-dimensional diamond pattern surround; edge-wound aluminum voice coil; Coherent Wave phase plug; compression drivers with 1- to 2- inch exits are available.

Celestion CDX20-3020
www.celestion.com

Throat Size: 2 inches
Frequency Response: 500 Hz – 18 kHz
Sensitivity: 107 dB
Impedance: 8 ohms
Voice Coil: 3 inches
Magnetic Structure: Ferrite
Power Handling: 100 watts AES
Weight: 10.8 pounds
Notes: CDX Series models are available with 1- to 3-inch diameter voice coils and 1-, 1.4- and 2-inch-exit throat sizes; the range also offers titanium, PETP or polyimide film diaphragms, as well as flange or screw mounting options.

B&C Speakers DE910TN
www.bcspeakers.com

Throat Size: 1.3 inches
Frequency Response: 500 Hz – 18kHz
Sensitivity: 108.5 dB
Impedance: 8 ohms
Voice Coil: 3 inches
Magnetic Structure: Neodymium
Power Handling: 110 watts AES
Weight: 5.3 pounds
Notes: Titanium diaphragm; aluminum wire voice coil; a variety of compressions drivers, with 0.75- to 2-inch exits, and titanium, mylar, or polyimide diaphragms are available.

Beyma SMC225/ND
www.beyma.com

Throat Size: 1 inch
Frequency Response: 800 Hz – 20 kHz
Sensitivity: 108 dB
Impedance: 8 ohms
Voice Coil: 1.75 inches
Magnetic Structure: Neodymium
Power Handling: 40 watts AES (above 1.5 kHz)
Weight: 1.5 pounds
Notes: Edge-wound aluminum flat wire coil; polyimide former; mylar diaphragm; drivers with larger throat and voice coil diameters are available.

Eighteen Sound ND4015Ti2
www.eighteensound.com

Throat Size: 1.5 inches
Frequency Response: 800 Hz – 20 kHz
Sensitivity: 113 dB
Impedance: 8 ohms
Voice Coil: 4 inches
Magnetic Structure: Neodymium
Power Handling: 160 watts continuous
Weight: 7 pounds
Notes: Titanium diaphragm; 4-slot phase plug; edge-wound aluminum voice coil; also available in 1.4- and 2-inch exit versions.

Radian Audio 950PB
www.radianaudio.com

Throat Size: 2 inches
Frequency Response: 500 Hz – 20 kHz
Sensitivity: 111 dB
Impedance: 8 or 16 ohms
Voice Coil: 4 inches
Magnetic Structure: Neodymium
Power Handling: 125 watts AES
Weight: 10.4 pounds
Notes: Aluminum diaphragm with Mylar suspension; 5-slit phase plug; a variety of other compression drivers with 1- to 2-inch throats are available. LSI

In Memoriam

JACK MAXSON, 1940-2016

Marking the life of a noted mix engineer and production entrepreneur.

by Kevin Young

Jack Maxson passed away in Dallas in late October at the age of 76, leaving an enduring legacy in the entertainment and production industry.

"Jack was an expert mixer," says Rusty Brutsche, who co-founded leading sound company Showco with Maxson and Jack Calmes. "He ended up being, I think, the preeminent audio man in the 1970s because of his experience and the level of bands we were servicing."

Brutsche first met Maxson in the mid-1960s at a recording studio Maxson had built in Fort Worth. "I was in a rock band, but had a real deep interest in audio," he notes. "We went to the studio to record an album and had an immediate bond because we both loved audio so much. We became really close friends."

FORMATIVE YEARS

Born John DeGolyer Maxson on March 11, 1940, in Dallas, he was fascinated by both music and electronics at an early age. "His mother had been a concert pianist and he had an encyclopedic knowledge of classical music, but he also loved jazz and had a huge collection of records and CDs," Brutsche says. "His family was in the insurance business and his father expected him to go into that, but he loved audio and wanted to go into recording, which was kind of a weird thing to do at the time, particularly just to go do it yourself and figure it out."

John (Jack) DeGolyer Maxson

Maxson was serving as the president of the Delta Recording Company and Spot Productions before starting up Showco in his garage in 1970. Those early days of modern concert sound were a time when many "just fell into the industry," Brutsche explains. "Most of us were musicians, but Jack was unique."

When promotion company Concerts West received word of the system built by Maxson and Brutsche, it asked the pair to provide sound reinforcement at a show for Spirit in Los Angeles, and soon after that successful inauguration, began booking the system on a regular basis.

"Three Dog Night loved it and wanted to use the system every night," Brutsche recalls. "So we rented a truck, Jack got in it and started mixing their shows while I began building another system in Jack's garage. Then (Led) Zeppelin came through town and I went down, did Zeppelin, and they also loved the system. So I rented a truck and went on the road with them. That's how it all started."

THE ARCHITECT

Showco quickly grew to become one of the premier sound reinforcement providers in the world, with a client list that included The Rolling Stones, David Bowie, Paul McCartney and Wings, and James Taylor (all of whom Maxson mixed personally), helping pioneer the post-Woodstock concert industry by providing state-of-the-art, road-ready audio for stadium, arena and festival rock 'n' roll.

"We had to invent things as we went; build equipment, design speaker cabinets, figure out how to make it all work and go up and down quickly, and Jack was really good at that sort of thing," Brutsche says.

He credits Maxson's knowledge as a driving force in the company: "Jack was a really deep thinker and clever about coming up with ideas for how systems should be designed; millions of ideas that made the equipment work. He was an expert in audio, but his knowledge of music was also really deep. Music was just part of him."

That understanding of music combined with technical prowess and a knack for finding solutions others might miss played heavily into Showco's technology, Brutsche adds, referencing the company's first mixing console, the Superboard – a 30-channel, 8-bus board created in 1974, the first touring mixer with parametric EQ. "Jack was the architect of it – as far as the features and what it should do – and it was a huge success for us," he adds.

A NEW DIRECTION

In 1981, Maxson's focus turned to another production element, when he co-founded Vari*Lite. The company pioneered a new automated stage lighting concept that, like Showco, helped change the game, and again, Maxson's unique ability to see through challenges was hugely instrumental, Brutsche explains.

"We were sitting in a Dallas BBQ joint called Sallih's thinking about how we were going to make this product, and out of

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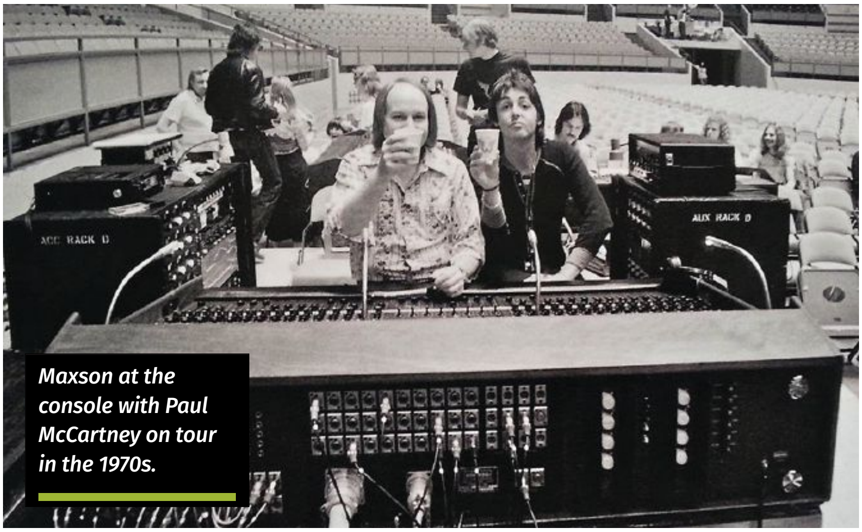
nowhere Jack just says, 'You know, if we have two more motors we can make this thing move.' The concept for the whole thing came together in that instant." Just 12 weeks later they had a working prototype.

Like Showco, Vari*Lite became a groundbreaking company, even garnering Prime Time Emmy Awards for Outstanding Achievement in Engineering in 1991, 1994, and 2001. Still, Maxson's passion for audio, production technology and music was matched by an enduring desire to serve his community. "His dream was to be a philanthropist," Brutsche notes. "He always told me that if he could do anything he wanted, that's what he wanted."



Maxson at the console with Paul McCartney on tour in the 1970s.

MANY PASSIONS

Maxson pursued that role with his typical drive, generous with both time and money in support of multiple organizations, including The Dallas Arboretum, Dallas Symphony Orchestra and the Museum of the American Railroad in Frisco. About the latter, Brutsche says, "He loved heavy machinery – trucks and trains – and read countless books on the railroad. Not history, technical books. So when a train went by he could tell you what kind of locomotive it was, what engine it had and who made it."

Above all, Maxson was dedicated to his family, particularly to his wife, Sally, as well as his daughter, Peg, from a previous marriage. Professionally and personally, he responded to adversity by rising to the occasion, such as when Sally was struck with Alzheimer's disease.

Brutsche concludes about his friend, "He took care of her himself for as long as he possibly could. He really was impressive; a very strong person, quiet, unassuming, kind of reserved, but a deep thinker with broad interests." LSI

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

LSI Loudspeaker Demo Coming To USITT 2017

The LSI loudspeaker demo will be presented at USITT 2017 in St. Louis next March. Held at the Ferrara Theatre at show site America's Center, the demo will provide the opportunity to check out at least a dozen larger and smaller format loudspeakers from several leading manufacturers.

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

Go to livesound-demo.com to find out more about the demo at USITT in St. Louis and to register to attend.

Gary Zywiecki has joined the Vancouver, BC-based team at **Radial Engineering** as manager of engineering. He is heading up all R&D activities from product concept to launch, leading a multi-disciplinary team of engineers, advancing products that span from vintage vacuum tube technology to digital and wireless technologies.

Zywiecki brings more than 25 years of expertise in product development to his new role. His career path has included work in the wireless, GPS, telecommunication and control software industries, and he also has extensive experience in the rapid deployment of disruptive technology.

EAW has named **Jim Bobel** to the position of North U.S. sales engineer, with the announcement coming from **Jim Newhouse**, the company's North American director of sales. Bobel is responsible for managing all EAW sales in the northern states, ranging from the Dakotas to Kansas and across to the East Coast.

"Having watched Jim in his previous positions, I've always respected his relationship building and customer focus," states Newhouse. "His ability to grow new business and lead has set him apart from his peers. We are proud to have him join our team."

Martin Audio has appointed **Dan Orton** to the newly created position of product group manager, reporting to **James King**, director of marketing. Orton's duties include management of both the product portfolio and product support team, and he's also tasked with delivering improved territory application support and monitoring.

With extensive pro audio experience in both the live sound and installation markets, Orton has worked in a variety of engineering development and application engineer roles with both Turbosound and L-Acoustics. "Dan is the perfect choice for this new role and I am excited by his technical expertise and insight that will prove invaluable as this business continues to grow," King states.

L-Acoustics' holding company, **L-Group**, has announced the acquisition of **CAMCO**, which has more than 25 years of experience developing amplified controller technology and holds several patents in the trade. Based in Wenden, Germany, CAMCO counts 50 employees and distributes its products in 60 countries, in addition to supplying components to professional sound system manufacturers.

"For over a decade, CAMCO has been a key supply partner of L-Acoustics. Their expertise in electronics for the professional sound industry has contributed to the success of the L-Acoustics amplified controllers," explains **Hervé Guillaume**, managing director of the holding company. "Welcoming them into the L-Group family of companies will allow both L-Acoustics and CAMCO to continue growing our research and development expertise. This merger will give us the tools to develop future products to better serve our clients and explore new markets." LSI

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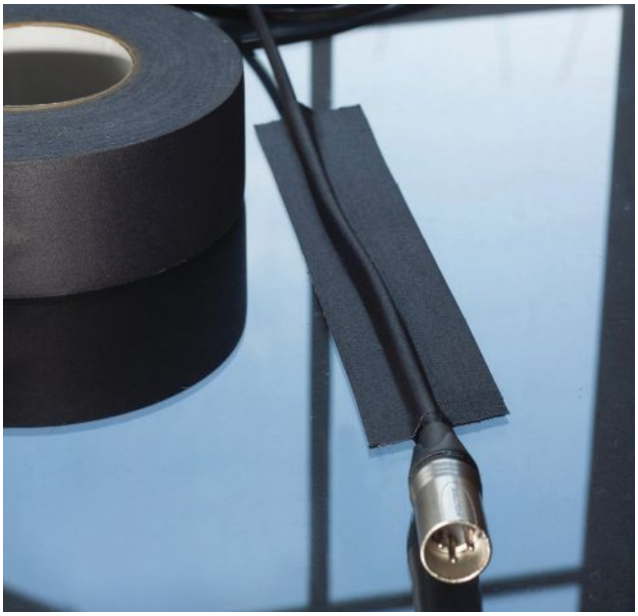
THE LITTLE THINGS

A top-flight system doesn't matter if...

by Craig Leerman



Sure, you have a great microphone collection, the latest digital console, and fantastic loudspeakers – that's everything required of a sound company, right? Well, not exactly.

The truth is that it's the little things that can make the most significant difference between a "good" or "great" gig. Here's a list of some (not all) of the "must haves" at any show, no matter how big or small.

Gaffer ("gaff") tape — not "duct" (or "duck") tape, or masking tape, or five-rolls-for-a-buck tape from the local swap meet. Gaff tape is the choice because it is purpose-designed for our applications while leaving little to no residue on cables, floors, carpets and most other surfaces. (The presentation is professional, there's far less time needed for "de-gunking" cables in the shop, and clients appreciate the cleanliness.)

Proper extension cords — not the cheap orange or yellow cords from the local big box store. They don't have the proper thickness of jacket and insulation for use in areas where people gather, like clubs and theatres.

The right cables — the National Electric Code (NEC) calls for Extra Hard Usage (SO) cables in areas where they may be subjected to physical damage, such as where people walk or carts roll over them. For locations where they're not subject to physical damage, Hard Usage (SJO) cables should be the choice. In addition, utilize 10- or 12-gauge cables for longer runs because voltage drop occurs with distance. This effect is more pronounced the longer the cable and the smaller the wire gauge.

Spare cables — including mic, instrument, loudspeaker and power varieties. Also carry different lengths in case a run is longer or shorter than expected. Also don't forget barrel joiners for speakON and 1/4-inch terminated cables in case two need to be connected to extend a run.

Adapters — imagine the possibilities and then further understand that we can't always imagine them all. For example (and "unexpectedly") there may be a need to adapt a cable to interface the PA with a house system to feed sound to a lobby and/or underbalcony area. Also don't forget turnaround adapters; these "gender benders" come in handy for switching the gender (male or female) of existing cables for intercoms, returning signals through snakes, and more.

1:1 transformer isolation — a.k.a., "humbuckers" or ISO units, they use a transformer to break the physical connection between two pieces of gear that may be plugged into different outlets, which can cause noises like hum or buzz through a system. Think about a mixing console plugged in at front of house and powered loudspeakers plugged in at the stage area, a very common scenario.

The right booms — stock a decent assortment of shorter booms and/or adjustable length booms, as well as "shorty" mic stands. Long booms tend to stick out into the performance area, particularly when used on drums and instruments, and can get in the way of performers. Also stock a few round base stands, which come in handy for tight placement areas and for performers who prefer that style.

Extra mic clips — Mr. Murphy dictates that clips tend to break right before the show starts. Swapping them out is the easiest thing in the world – if you have spares on hand.

Music — clients don't usually expect us to be DJs (unless that's expressly why we were hired), but they do expect us to provide walk-in/out music and/or to play something light in the background during a dinner. Create and carry playlists of versatile tracks, particularly of the smooth jazz/soft rock/instrumental variety.

Utility mixer — if the mixing console dies things can get dire really quick. Always carry a small backup (with at least 6 to 8 channels) as a potentially show-saving backup. It may not be big enough to handle every input, but that's where our skills in routing come in, right? LSI

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

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