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

LIVE SOUND INTERNATIONAL

INSIDE

MARKING A MILESTONE

LSI Celebrates 25 Great Years

EXPLORING SYSTEM TIME & PHASE ALIGNMENT

SOUND IN THE ROUND FOR MUSE ON TOUR

THE VAULT: WATCHING THE SIGNAL FLOW

25th ANNIVERSARY

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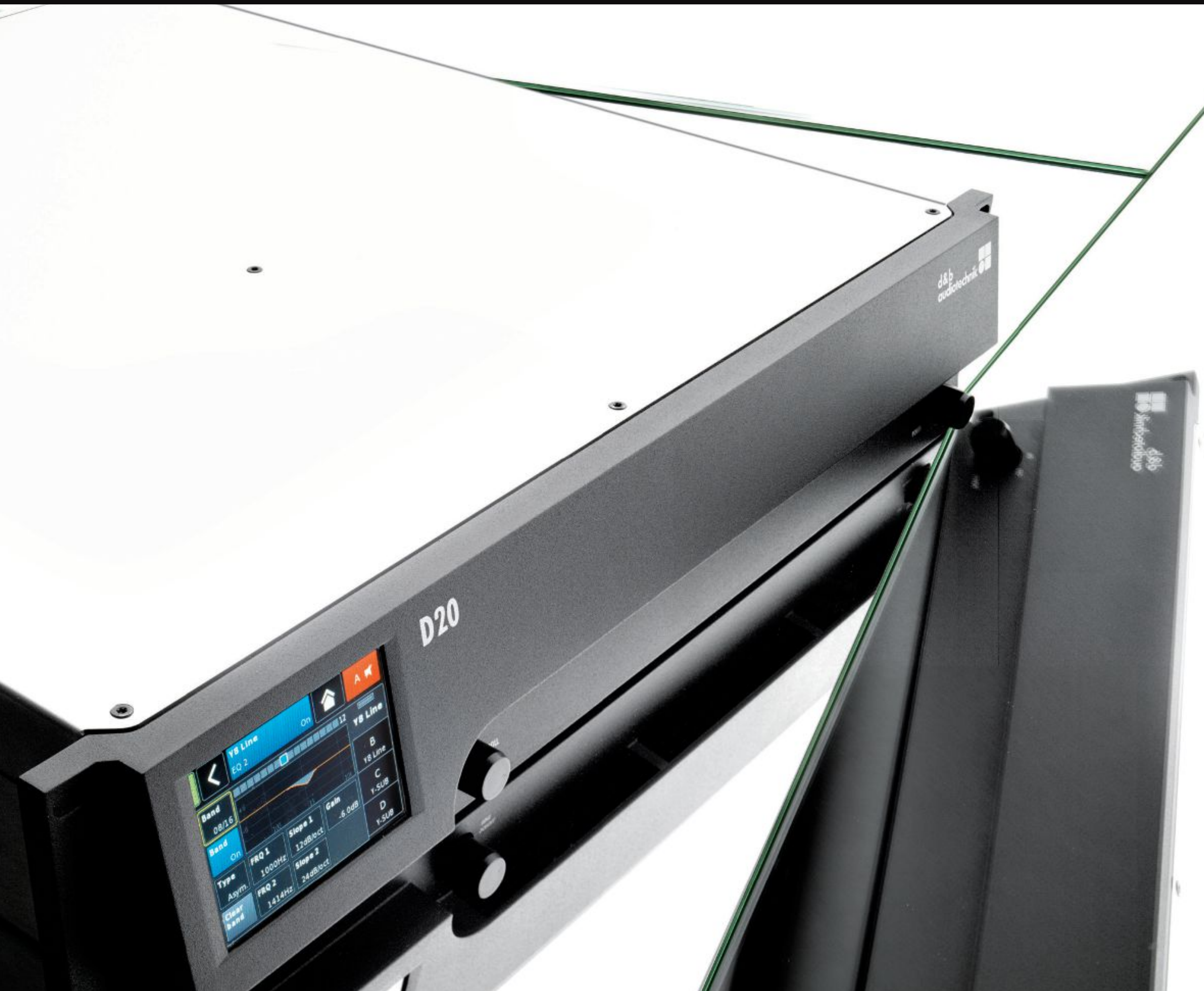

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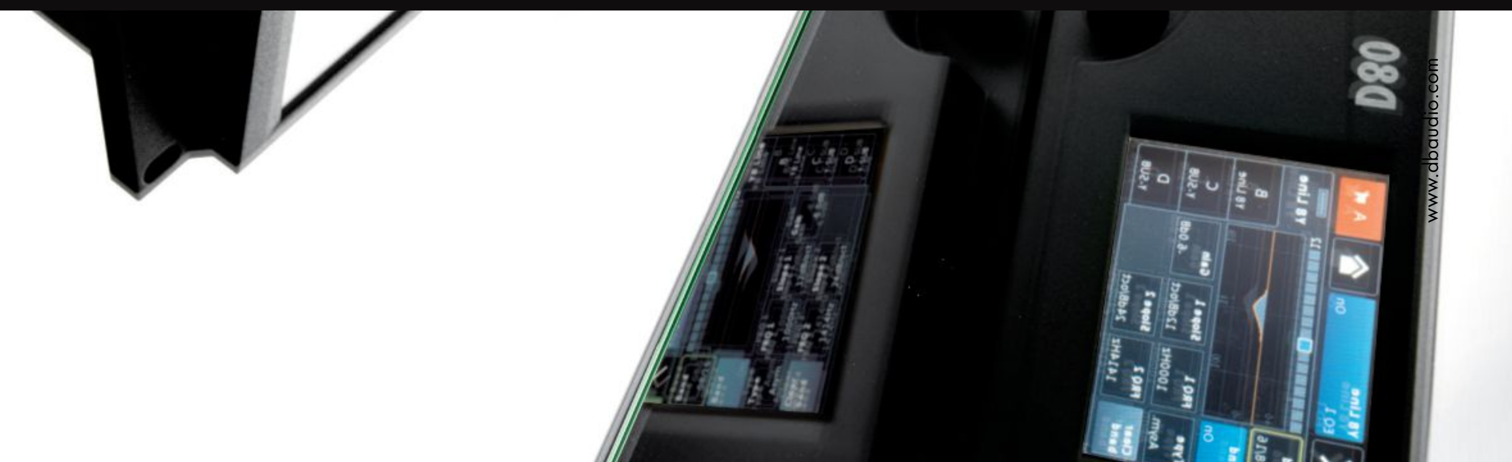


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LIVE SOUND INTERNATIONAL

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From the Editor's Desk

ORIGINALLY TITLED *Live Sound! and Touring Technology*, this publication began as a bi-monthly supplement to *RE/P* (Recording Engineer/Producer) in January 1992. As we mark the 25th anniversary this year of what we now call *Live Sound International*, it's interesting to note that despite the considerable impact of the "digital revolution" on the ways we receive and consume information, traditional media continues to play a useful, beneficial role to millions worldwide.

The print publications that survive and thrive are built on strong foundations. The inaugural issue of *LSI*, under the direction of editor Mike Joseph, serves as an excellent example and set the precedent for where we are today. The vast majority of contributors to that first effort were experienced audio professionals sharing their knowledge and understanding of important aspects of live sound reinforcement. Sound familiar? It's a great heritage that has obviously served us well and continues to this day.

In this issue, we pay tribute to that legacy by featuring an in-depth article by Barry McKinnon that appeared on pages 16 to 22 in the first edition. We're also treated to an interesting re-work of a previous piece focusing on working with true singers by Mark Frink, an original contributor who continues delivering consistently relevant editorial virtually every month. Meanwhile Bob McCarthy supplies valuable insight in clarifying the concepts of system phase and time alignment.

And as always, there's much more, including a revitalized graphic format that retains the original essence while enhancing the way the information is presented. Besides, it's nice to spruce up the old place every once in a while. Enjoy the issue.

Keith Clark

Keith Clark
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ON THE COVER:
A collection of *Live Sound International* covers over the past 25 years.

LIVESOUND INTERNATIONAL

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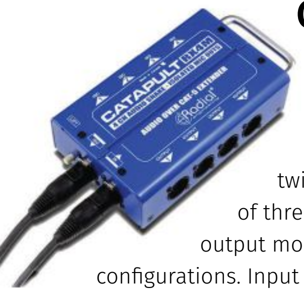
Allen & Heath ZEDi-10 & ZEDi-10FX

Both models combine a compact mixer with a 24-bit/96 kHz 4x4 USB interface to enable multitrack recording and playback to a Mac or PC, or to an iOS device. They're bundled with Cubase LE music production software and the Cubasis LE mobile music production app. Both offer four mono channels with separate XLR and TRS jack sockets, one stereo channel, and a second stereo input for reverb returns or playback. The mono channels include DI high-impedance circuitry for the jack sockets, allowing guitars to be plugged directly into the mixer. Also included are 3-band EQ, 60 mm faders, FX/aux/monitor and headphone outputs, channel monitoring, 48-volt phantom power, and an internal universal voltage power supply. The ZEDi-10FX also provides multi-FX models such as reverbs, delays, doublers, chorusing, and others. ***www.allen-heath.com, www.americanmusicandsound.com***

Radial Engineering Catapult



A four-channel audio snake that sends analog and AES digital audio signals over standard cat-5 shielded twisted pair cable. A choice of three input modules and three output modules provide a total of six configurations. Input modules include XLR inputs along with two Neutrik Ethercon outputs and four additional XLR outputs. Output modules offer Ethercon input, thru-put, and two sets of XLR outputs. These can be ordered without transformers for sub snakes or to transmit AES audio signals; with mic isolation transformers for signal splitting between front of house, monitors or the recording system; or with line level isolation transformers to eliminate system noise. A ground lift switch is available at each end for the Ethernet cable connection. ***www.radialeng.com***

K-array KU44 & KU26

The KU44 is an arrayable bass element that complements mid-high loudspeaker models with the company's proprietary Slim Array Technology. Stated frequency range is 50 Hz to 150 Hz. It incorporates dual 4-inch neodymium transducers joined by dual 4-inch passive radiators. There's a choice of two different values of impedance (8 ohms or 32 ohms). The enclosure is made of steel, and a variety of hardware accessories are available. The KU26 is a compact subwoofer designed as a companion to the company's Lyzard, Vyper, and Tornado loudspeakers. It operates in a stated frequency range of 45 Hz to 300 Hz, and is electronically protected. The KU26 includes a 6-inch neodymium transducer that's outfitted with a double voice coil (16 ohms + 16 ohms) for selectable impedance settings. as well as a 6-inch passive radiator. ***www.k-array.com***

Mackie Master Fader v4.0

The latest version of the control app for the company's DL32R, DL1608 and DL806 digital mixers integrates iPad, iPhone and iPod touch support within a single app, and also adds support for the iPad Pro. It now provides the ability to view a real-time RTA beneath any output's PEQ or GEQ. A built-in oscillator has been added for the DL32R, and it's signal (pink noise, white noise or sine wave with selectable frequency) can be routed to any channel or physical output. The user interface is now completely scalable, including the ability to use the new Split View in iOS 9. A quick assign function simplifies assigning channel ID. Master Fader v4.0 is available for download at the Apple App Store. ***www.mackie.com***

Lectrosonics IFB-VHF

A wireless microphone system that includes the IFBT4-VHF frequency agile compact transmitter and the IFBR1a-VHF receiver, designed to operate in the less-trafficked VHF broadcast band in offering 239 frequencies between 174 to 216 MHz. The transmitter has 50 mW power output for long-range use and incorporates a multi-use XLR input jack and DIP switches with settings for dynamic mic, line level, Clear-Com and RTS intercom sources. It can also operate in Digital Hybrid Wireless mode for additional compatibility with future Lectrosonics products. A pilot tone signal controls the audio squelch on the receiver to eliminate noise when the transmitter is turned off, and it also prevents the receiver from locking on to false signals. ***www.lectrosonics.com***

RCF M18

A digital mixer incorporating an onboard Wi-Fi access point with integrated antenna that enables all functions to be controlled wirelessly. It operates at 2.4 GHz and 5 GHz. All 18 input channels include gate, dynamic compressor, and 4-band parametric EQ that provides three different modes (standard, vintage and smooth). Eight input channels also have discrete mic preamps offering 60 dB of gain range, controllable via the MixRemote app (available for iOS). And, two (out of 10) line inputs can be switched to Hi-Z mode. Also onboard is a suite of plug-in algorithms and effects. The back panel is populated with eight mic inputs (with six XLR and two combos), 10 line inputs, six aux outs, headphones out, and two XLR main outs. Three separate aux buses are also available to feed the internal effects. A footswitch socket is supplied for internal FX mute or program change, as well as MIDI in/out. A USB port enables use of external storage for the internal two-track player and recorder, while an Ethernet port facilitates computer connection. ***www.rcf-usa.com***

AKG K182

Foldable, closed-back headphones with 50 mm transducers with a rated frequency range of 10 Hz to 28 kHz. A 3D-axis professional folding mechanism means the K182 can pack into compact spaces. The detachable cable has a 1/8-inch-to-1/4-inch screw-on adapter. ***www.akg.com***

Shure MOTIV Version 1.1

Upgrades to the mobile recording iOS app designed for use with the company's MOTIV digital condenser microphones include five-band EQ and gain adjustment controls for MV51 and MV5 mics as well as the MVi digital audio interface. There's also now Bluetooth support for instant playback over loudspeaker systems, and transport controls in the control center to simplify the process of pausing, playing, fast-forwarding, and rewinding a recording. A dark mode setting has also been added for low-light environments where a bright screen might be a distraction (i.e., clubs and other live event venues). Version 1.1 is available for download at the Apple App Store. ***www.shure.com***

WorxAudio Technologies TrueLine XQ10

Incorporating four direct-radiating, 10-inch cone drivers with 2.5-inch voice coils in a tuned enclosure, the XQ10 is designed to be used for low-mid frequency sections in multi-way loudspeaker systems or as an extended bass section. It can be arrayed and is available in both touring and install configurations, as well as powered and biamped (install only) versions. The powered versions include WorxAudio's PDA-2000 amplifier, which is available with Dante and AVB audio networking options, onboard DSP, and the company's AllControl software integration. Two parallel-wired NL4 Neutrik speakon connectors are provided, along with Neutrik powercon connectors for AC power, plus XLR M/F (in/out). The enclosure, which measures 22.5 x 24.5 x 28 (h x w x d), is made of 15-ply Baltic birch. The company's TrueAim Tour side-plate rigging is available, along with optional array frame accessories. ***www.worxaudio.com***

Bose Professional Panaray Series IV

Updated versions of the Panaray 802 and Panaray 402 loudspeakers that offer enhanced installation options. The 802, which supplies 120- x 100-degree (h x v) dispersion, adds new side-threaded inserts and an optional accessory U-Bracket. It comes in a black finish, measures 13.3 x 20.5 x 13.2 inches (h x w x d), and weighs 30 pounds. The 402, with 60- x 120-degree (h x v) dispersion, has new rear threaded inserts with industry-standard mounting to accommodate optional pan-and-tilt brackets. It's available in black and white, measures 23.3 x 8.1 x 8 inches (h x w x d), and weighs 16 pounds. ***http://pro.bose.com***

BEST OF BOTH WORLDS?

The collaborative powers of the studio and live reinforcement.

by Karl Winkler

When first starting down the audio path, I eagerly read every scrap of information I could get my hands on about how the "pros" approached their craft. What kind of gear were they using? What techniques? What was the "secret sauce"?

In the late 1980s, I was much more interested in recording and had not yet seen the light about live sound reinforcement. And in those days, the Lord-Alge brothers were a big deal in the recording world. While *Home Recording* magazine was more my speed back then, I'd drool over the pictures of studios and equipment in *Mix*. Then, in the early 90s, *EQ* magazine was created to bridge the gap. One of the things I remember reading in an article of the time was Tom Lord-Alge stating that he "learned the craft of pulling a mix together quickly by doing live sound."

I recall this because it was surprising. I saw live sound as a rather crude affair, at least from what I'd witnessed in my limited experience. I mean, those gorgeous studios in *Mix* were using Neumann microphones, Pultec EQs, Universal Audio LA2A compressors and Neve consoles – the cream of the crop. These tools dwarfed what was being used on the live side. And this was years before I heard touring sound pioneer Albert Leccese espouse his three rules for live sound: 1) Make noise. 2) Keep making noise. 3) Make it sound good if you can.

DIFFERENT BUT THE SAME

But fate intervened in the mid-90s when I learned of a sound tech position available with the Air Force band out of Washington, DC. I decided to go for it, to follow Tom's observation about "getting some live chops" that could be applied in my future studio career. Except it didn't turn out that way.

I caught the live sound "bug," coming to enjoy the thrill of mixing shows in front of an audience. More importantly, I came to understand and respect the world of live sound due to the myriad challenges and rewards woven into its very fabric. Different gear, yes, but no less advanced. Different techniques, of course, but no less sophisticated.

A funny thing happened along the way – I noticed that a lot of long-time live sound people looked down on the studio world, similar to the way "jazzers" look down on classical musicians (and vice versa). What was going on? Well, just as I had thought as a studio person that live reinforcement was a bit déclassé, the live folks tended to think that the recording world was rarefied and "out of touch with the real world."

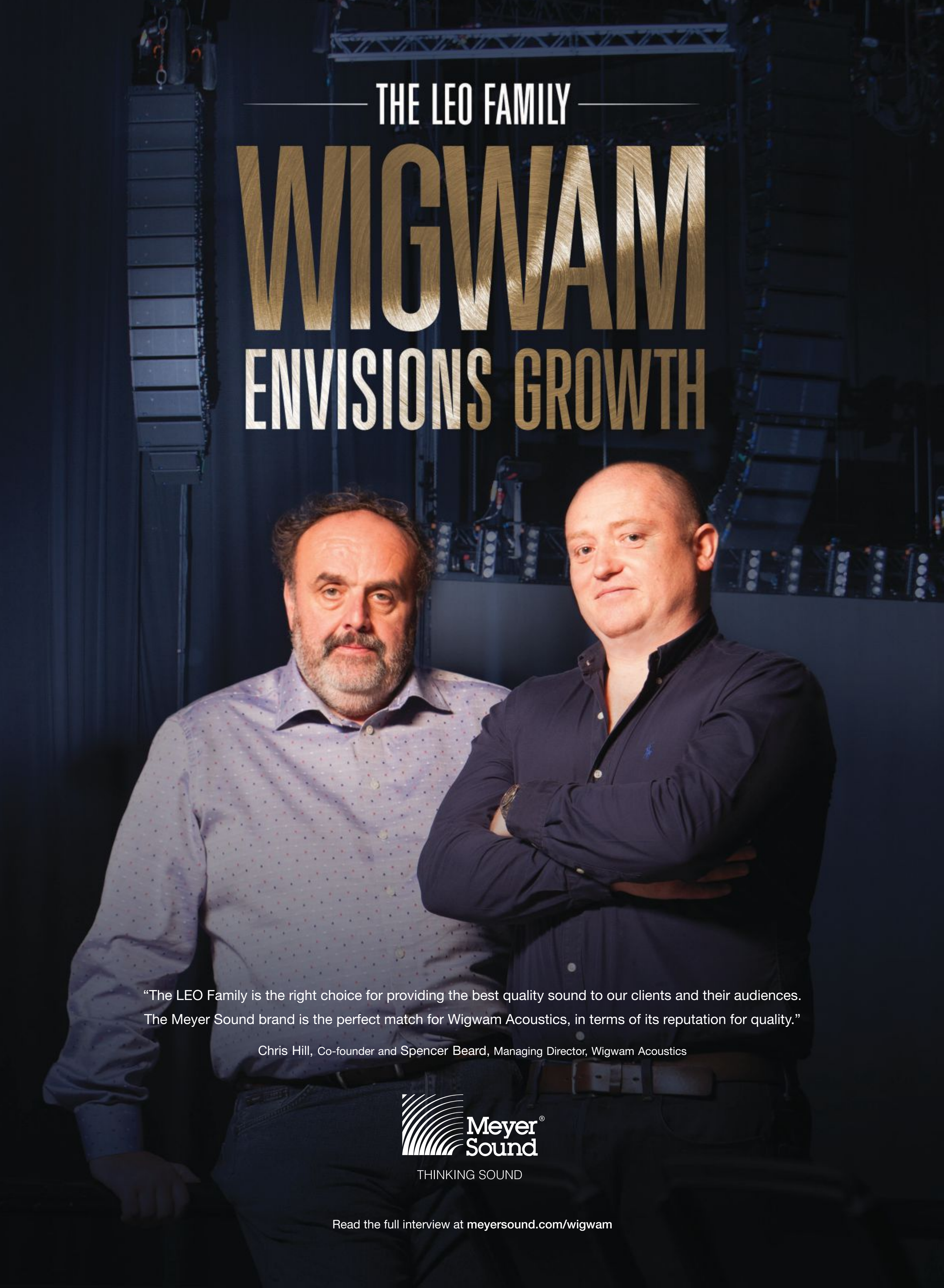
Fast forward to now. I've come to believe that while there's an element of truth to both belief sets, there's also been a healthy exchange of ideas, best practices, and encouragement rather than divisiveness. One side delivers recorded sound to millions, the other side delivers live sound to millions. So both sides form a whole that is critical to the success of the artists involved and the enjoyment of consumers worldwide.

Both make the most of available techniques while seeking to do even better, both make the most of available technology while driving for something better, and both work closely with manufacturers to help achieve what is today impossible but tomorrow a reality.

COMMON FACTORS

With that in mind, here are some of the best things I've learned from each of these two facets of pro audio, beyond the initial truth about learning to pull a mix together quickly from the live world.

First, microphones matter. Simply changing the mic can make or break a


THE LEO FAMILY
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“The LEO Family is the right choice for providing the best quality sound to our clients and their audiences.
The Meyer Sound brand is the perfect match for Wigwam Acoustics, in terms of its reputation for quality.”
Chris Hill, Co-founder and Spencer Beard, Managing Director, Wigwam Acoustics
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singer's voice, and in the studio, a lot of time was often spent finding the right one. And, as it turned out, the 1990s saw a revolution in mics for live sound. It started with the incorporation of studio models on the road, followed by the development of purpose-made studio-quality mics like the Neumann KMS105 and Shure KSM9, to name perhaps the two that are most well-known today in live sound circles.

Next, if the artists aren't happy with their cue mix, they aren't happy, period. This applies both in the studio when tracking and on the stage when performing. The advent of wireless in-ear monitoring systems brought about another revolution in live sound, making it more like the studio experience. It's resulted in happier artists, which certainly benefits their live performances. Sure, it's made monitor mixing more challenging in some ways, but that's the price of progress. And it's been worth paying.

It doesn't take long for live sound people to learn that the show will start on time, no matter what.

From the studio: careful choice of outboard gear can really enhance the mix. Outboard gear has long been used for live sound, of course. Perhaps the difference today is that with the incorporation of studio-quality stage mics, premium mic preamps, in-ear monitoring and superior loudspeaker systems, the sometimes subtle effects of great outboard gear can be heard much more clearly.

Speedy troubleshooting is critical. It doesn't take long for live sound people to learn that the show will start on time, no matter what. Thus any problems *must* be solved, quickly and regardless of circumstances. On the other hand, studio people know that hours can't be billed if the console won't turn on.

High bit and sample rates make a difference – to a point. Sure, we definitely want to use the highest sample rate possible (within reason), and the largest bit depth. But the reality is that high-end playback equipment is rare among consumers. Most recordings are listened to in the MP3 format or in cars. This is not to say that recordings shouldn't be a high-quality product! But it's the reality, and therefore is a tree worthy of attention and consideration in the forest of sample/bit rate obsession that some exhibit.

In live sound, there's a different challenge: cumulative latency, particularly noticeable in an IEM signal path. Any time analog audio signal is converted to digital, there's latency. It depends on the sample rate, but is generally in the range of 1.5 to 3.5 milliseconds, round trip. If this were all, it would be fine for most artists in most situations.

But heaven forbid if there's re-sampling going on, which adds at least another couple of milliseconds. Re-sampling happens when interfacing equipment running on different sample rates. So, in effect, what this means is that the system designer needs to run everything on the lowest common denominator, even it is "only" 44.1 kHz (and be aware of the master clock). This way, it's possible to make the A/D and D/A round trip only once to avoid adding any further latency beyond the basic amount.

WIN-WIN

The point is that both sides face issues, even if they manifest themselves in different ways. And we can both learn something (actually many things) from each other, as has been proven.

We all share the same "industry" that's called pro audio, so it's best to try to stick together to address challenges in furthering both the science and the art. While there may be six degrees of separation between any two people in the world, there are probably only two degrees between folks plying the pro audio trade. Let's treat each other like true colleagues, and we will *all* be the better for it. LSI

Karl Winkler *serves as vice president of sales/service at Lectrosonics and has worked in professional audio for more than 25 years. Reach him at karl1@karlwinkler.com.*

DAVE SHADOAN

A journey that's had a lot of interesting stops along the way. *by Kevin Young*

TO DAVE SHADOAN, what sets Sound Image apart are its people, partnerships built over the long haul, and most importantly, unfailing service to clients. Clearly it's paying off and then some, with the Escondido, CA-based company serving thousands of shows annually, with numerous touring systems out simultaneously in addition to one-offs and seasonal leases.

"I can't think of a single incident where we've missed a truck, a load-in, or a show. If we say we're going to do something, we do it," states Shadoan, the company's CEO/president. "In the long run it comes down to trust. People hire Sound Image because they trust our staff. Our job is to make sure artists are able to do the best they can, whether it's for an audience of 20,000, 2,000, or 200 people. And we're only as good as our last show."

In addition to the headquarters in Escondido (just outside San Diego), the company also has a busy operation in Nashville while continuing to expand internationally, most recently with United Audio Companies, a venture developed with SSE Audio Group in Europe, announced in August 2015. Sound Image was also early to system integration, entering the market more than two decades ago and now with full-service staff and facilities in both Arizona and California.

Further, the company also holds 10 patents, a tribute to its focus on innovation, with one (of many) notable endeavors being a pioneer in the use of composite materials to construct loudspeaker cabinets.

READING THE SIGNS

"Life is a journey" goes the popular quote, and it's personified by Shadoan. Born in 1956 in the west-central Ohio town of Sidney, by grade school he'd received an early indicator of his future. "We took a test – I didn't even know what it was for – then a few weeks later some of us were brought down to the lunchroom/basketball courts where there were instruments laid out on tables. 'You're here because you've shown an aptitude for music,' I was told. Where that came from, I don't know. But my mother loved music and loved to sing. She'd put on Elvis Presley and Conway Twitty records and get me to sing along. Maybe when you're 5 or 6 years old you start paying attention to stuff like that."

He chose the saxophone, both because it was the "coolest looking" instrument and one of his mother's favorite songs was "Yakety Sax" by Boots Randolph. A gift of perfect pitch, combined with a love of mathematics and the ongoing development of an encyclopedic knowledge of songwriters and production credits, fueled his efforts to learn popular

music and made him a hit with his band teacher: "By 7th grade, as homework, she'd say, 'Dave, you're going to write the parts for this instrument'."

Music, however, wasn't the main focus. In fact, his primary motivation after finishing high school was putting serious distance between himself and Sidney. So at age 18, he and a friend headed out, carrying two motorcycles in a beater panel truck that he used to carry gear and to get to shows to watch up and coming bands. In his pocket was money he'd earned at various janitorial jobs and from the sale of his '69 Chevelle. (Even then, Shadoan, who freely admits to a "car problem," had multiple vehicles.)

The two friends traveled up the Eastern Seaboard to New York City, through the Appalachians, and finally pivoted to California for the winter. San Diego wasn't the final destination they had in mind, however. "Our plan was to go to Oregon, Washington, or Alaska, and get jobs in the forestry service, sit up in a big tower and look for smoke like Smokey Bear," he laughs.

But reality intervened; funds had become scarce. "I did what everybody does when they've got no money: found a way to feed myself." Owing to random meetings with people on the beach and skateboarding on the boardwalk, he discovered ways of doing so. "For example, I met a guy nicknamed Abalone Bob, who sold seafood to local restaurants. He taught me to dive, and therefore we ate quite well."

Along the way he also imported Persian rugs and sold 'gray market' Mercedes-Benz autos. "And I never should have stopped," he says, laughing. "Selling those cars, I'd literally double my money in 30 days."

Sound Image and SSE Audio Group sealing the deal on the United Audio Companies venture. Left to right: Mike Sprague (Sound Image), John Penn (SSE), Dave Shadoan, Jesse Adamson (Sound Image), Yan Stile (SSE) and Dan Bennett (SSE).

MAKING CONNECTIONS

Getting into professional audio was largely happenstance, Shadoan notes: "Basically I'd meet people and one thing led to another. A guy introduced me to another guy who introduced me to Ross Ritto."

Ritto had formed Silverfish Audio Associates with partner Joel Silverman in 1971 in Rochester, NY, and they'd worked with Ashly Audio on the design of mixing consoles in addition to a primary focus on touring sound reinforcement. In 1975, Silverfish was chosen as the system provider for Jimmy Buffett, a significant development, and by 1978 the company moved to San Marcos in northern San Diego County.

Upon meeting, Ritto and Shadoan became fast friends. Shadoan began to invest money with Ritto into what was to become the new company. "But I was still hanging out with my friend from Sidney," he notes. "We hadn't made up our minds yet that we weren't eventually going to go be Smokey Bear. But the first time I touched a console was because of Ross. It looked like it wasn't that difficult, and at that time, it wasn't – there were only so many channels. I was also infatuated with getting on a bus and traveling."

In the early 1980s, the new partners renamed the company Southern California Sound Image. "We were total opposites," Shadoan says of Ritto. "If you want to find a good business partner, find someone who doesn't have any of your behavioral traits and you'll be fine. Ross was very organized. We each did some things well and others poorly. My best trait was understanding we had to grow the company at all times, at any cost, to be technologically cutting edge. That's still part of our mantra today."

Shadoan mixed monitors almost exclusively. "I can count on my fingers the number of times I mixed front of house. Monitors allowed me to get close to the artists and build long-term relationships. House guys came and went, but I, and Sound Image, got asked back." He learned about gear from the ground up, building loudspeakers, amplifier racks, cables, power distros, feeders, and more. "As a matter of fact, I was 'paint boy' too," he adds.

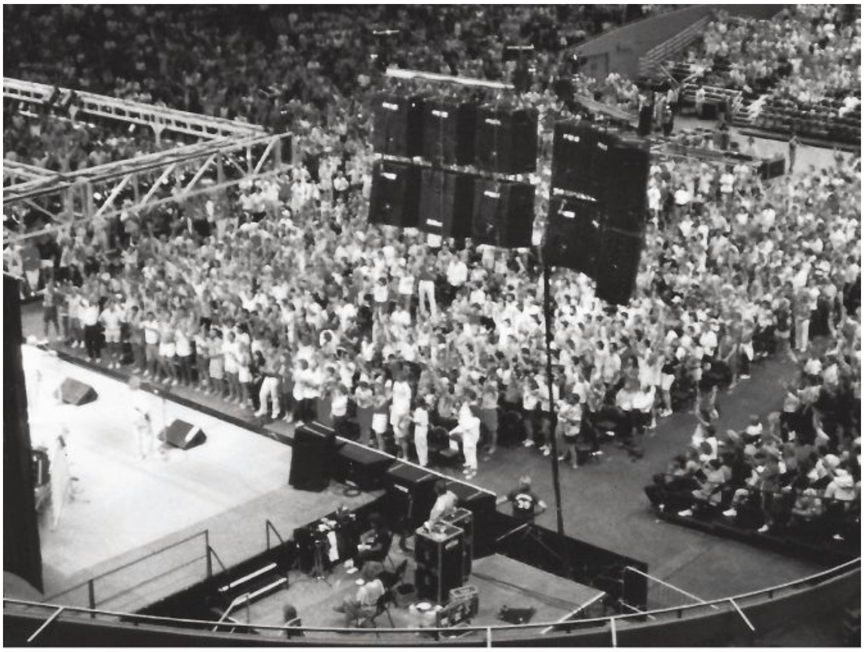
Ritto had already been developing relationships with artists like Buffet and others, with Shadoan's outgoing personality and dedication helping bring more clients to the table, including The Robert Cray Band. It began when Cray needed a last-minute monitor engineer for a gig at the Universal Amphitheatre, with Shadoan dropping everything and driving like a bat out of hell to the venue.

"I literally walked in the door as they were counting off the first song and ran over to the console," he says. "Nobody paid me much mind, but when they went into the break, Robert and Richard, the two main guys in the band, walked over and said, 'That was incredible.' And I replied, 'Yeah, you guys are amazing.' And they said, 'No, the sound'." To this day the band maintains an association with Sound Image.

INNOVATION & GROWTH

In those days, when the equipment that sound companies needed to assemble systems didn't exist, they built it. If they did buy something, they inevitably had to hot-rod it. It was a practice by all touring companies of the era, and Sound Image was no exception. "In the 80s, the guys who were getting ahead were the guys who were going at it. You had to have some idea how to build this stuff, because nobody built it for you." Today, he adds, "With the right price tag, a purchase order and a computer, I could build a system, soup to nuts, without modifying anything and never touching a piece of (existing) gear."

In 1984, loudspeaker design engineer Mike Adams, who Shadoan credits as key to the company's success (he's been with the company 42 years and now serves as senior VP of engineering), unveiled the PhaseLoc Series of custom Sound Image loudspeaker enclosures, which led to



Above, a look at one side of the PA deployed by Sound Image for Jimmy Buffet in the '80s, and below, one side of the rig (JBL VTX Series) for Linkin Park on tour in 2014.

PHOTO CREDIT: VIC WAGNER

the creation of the flagship G5 system. Phase-Loc employed mechanical time alignment, dual enclosure splayed baffles, and double-wide cabinets deploying an L-Track rigging system, reducing set-up time by half.

It was a turning point, Shadoan states. "In those days, most systems were two-way, three-way or possibly four-way systems. Nowadays everything is accurate, precise. But when we started doing this, people asked, 'How do you want the speakers pointed?' And we'd reply, 'At the people'."

Sound Image expanded to Nashville in 1989, a move that was particularly fortuitous with an explosion in the popularity of country music. Barbara Mandrell, whose 1989 tour was one of the largest of the time, was integral to the company's work, even requesting the opening of the Nashville office. "She was everything to us; a really good business person and a good friend," Shadoan says. More country acts joined the fold, including Clint Black and Brooks & Dunn, and to this day, the genre contributes to a fair percentage of the company's touring revenue.

The contracting/integration division was created in 1991, followed a year later by a new regional and corporate division. Then in 1994, Sound Image became the first touring company to deploy a fiber optic networking system with power amplifiers, which became the basis for the landmark QSControl system from QSC. Next came the 1995 move to the current headquarters in Escondido.

From the early '90s (left to right): Jeff Grogg (former VP of intergration), Ross Ritto, Mike Adams and Dave Shadoan.

MAKING TRANSITIONS

Also around this time, Adams and Ritto began the development of a new waveguide technology for loudspeakers using composite, graphite carbon fiber materials. (Shadoan credits design engineer Mark Engebretson, who worked with Sound Image on the project, for coining the term "waveguide technologies.") The effort eventually resulted in the establishment of the company's Audio Composite Engineering (ACE) Department in the mid 1990s.

Ritto's passion for boating and friendship with multiple America's Cup winner Dennis Conner directly inspired the work with composites. "If you look at a boat without a deck, it looks like the back of a speaker box, and Ross literally came in one day and said, 'We're going to make speakers out of this stuff'," Shadoan says.

Work with equipment manufacturers, which had begun during the hot-rodding days, continued to bear fruit into the next millennium. In 2000, Sound Image aided in the development of JBL VerTec line arrays, and in 2001, signed a distribution agreement with QSC. Shadoan also stopped mixing around the same time: "I went from a console to a calculator exclusively in 2001. The last show I mixed was Boz Scaggs, who's like my brother."

The organization – and Shadoan in particular – took a major hit in 2009 with the passing of Ritto, who'd been previously diagnosed with cancer. "It's still difficult," Shadoan says about the loss of his friend, the person whose passion directly inspired his own love of audio and the best business partner he could have asked for. "Ross and I were so compatible. Even though I knew him 30-plus years, in all that time we had maybe five knock-down, drag-out 'screaming-and-breaking-stuff' fights."

While the loss of Ritto was staggering, it also serves to this day to highlight the strength of what he helped build: a team that continues to look to the future and move forward, headed by strong veteran leadership supported by a steady stream of new talent.

So what's next? "I don't know," Shadoan says, laughing. "If you find out, let me know. If someone had told me I'd be where I am now when I was that 18-year-old with the van who was looking to be Smokey Bear's sidekick, I never would have believed it.

"I tell my daughter, and all kids, to find something that they love and stay there. Just do it. Some day all of that hard work, passion, loyalty and everything you've put into it will reward you – not always financially, but usually – and besides, everything in life ain't always about money anyway." LSI

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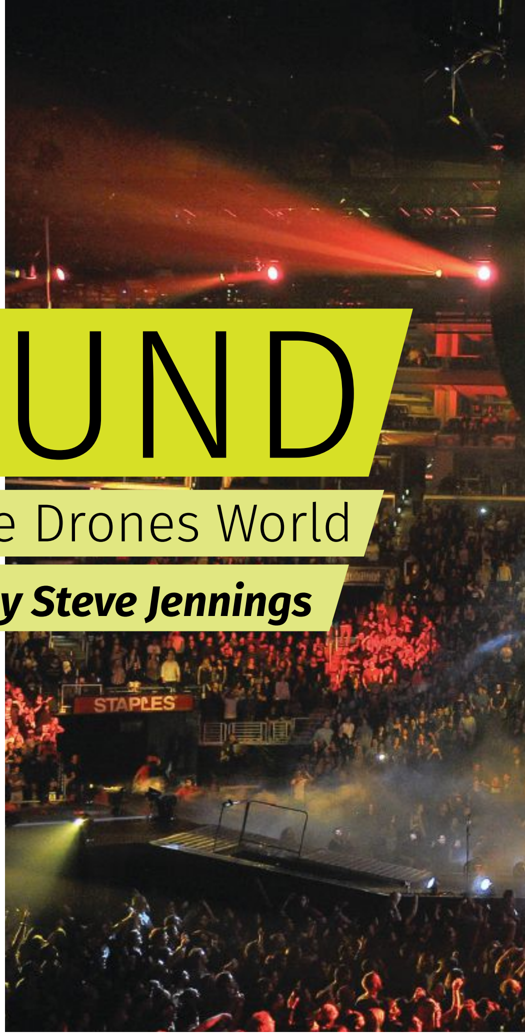
IN THE ROUND

Muse takes a 360-degree turn on the Drones World Tour. *by Gregory A. DeTogne, photography by Steve Jennings*



AMONG THE PANTHEON of arena rock legends, Muse just may be having a truly transcendent moment with its Drones World Tour, which kicked off in late November in Mexico City prior to settling in for the holidays with a pair of sold-out shows at the Staples Center in LA. With early 2016 dates in Las Vegas, San Diego, Chicago, Montreal, and well beyond, the English rockers are bringing their brand of aural and visual spectacle to fans in the round from a 360-degree stage. Reinforced with a formidable audio cache assembled by Skan PA Hire Ltd., the show travels in 24 trucks and uses 32 hang points for the PA alone.

Ambitious by any measure, the show, according to Muse frontman Matt Bellamy, follows in the footsteps of Pink Floyd's The Wall Tour. "It's our version of 'The Wall,' basically," he told BBC Radio 2. "There's a whole swarm of drones, and a stage like a double-headed arrow."

The drones of which Bellamy speaks are, for insurance purposes, referred to as Helium Flown Objects, or HFOs, by the production crew. Making their swarming presence felt from above at specific times during the show, they're controlled by a networked tracking system. Narrow and low-slung, the stage creates an aura of intimacy in the arena spaces it was designed for.

GETTING IT COORDINATED

To create the show's sound reinforcement blueprint, Skan PA Hire's Matt Vickers drew from an inventory of d&b audiotechnik components housed at Skan PA's Newbury-based headquarters in the UK. Flown within four separate zones, each of the PA quadrants comprises three independent hangs of its own employing d&b J-Series J8 cabinets, J-SUBs, and V-Series V8s. In total, there are 72 J8 enclosures and 72 V8 boxes that are flown, and 32 J-SUBs, some of which are flown and some of which are deployed under the stage along with eight J-INFRA Subs and four V-Series subs per side.

Power for the loudspeaker rig is provided in force via 84 d&b D80 amplifiers. System processing falls under the guidance of Lake controllers, which, along with the amps, are all housed come show time in a "space station" flown high above the crowd. Not merely a repository for gear, the space station is manned by crew members and engineers from the audio, video, rigging, and lighting teams, as well as HFO wranglers.

For audio crew chief Liam Tucker, the biggest challenge among many when assembling the PA in the round is that the stage can't go in until everything flown is in place. "Because this show is in the round, the cables can only go in one way," he says the day after the sold-out LA dates have successfully concluded. "So therefore if the lighting crew or anyone else runs into a problem above, the rig can't go up, and then the stage can't go in. With no stage we have no power to monitorworld or to start creating the PA network. No stage also means we have nothing to put the subs under, so we don't have a full system to configure. The audio team is generally the last to be ready because we are so dependent upon everyone else being set to go."

Joining Tucker on the audio crew are system tech Joachim Dewulf, front of house tech Eddie O'Brien, space station wingman/denizen Rob "Snowy" Wilkins, PA tech and support band coordinator Scott Maxwell, and wingman Matt Besford-Foster. Muse veterans Marc Carolan and Adam Taylor take charge of moving faders at front of house and monitors, respectively.

DEVELOPING A PROCESS

An Irishman amidst a crew packing plenty of English presence, Carolan offers the chance for this American writer to further his understanding of English dialects as well as how to wring the most out of a Midas XL4 analog console. "I've been using an XL4 on-and-off ever since I began mixing Muse 14 years ago," Carolan tells me. "I love the preamps, plus it's an ergonomic thing for me. I ran into a challenge this time because I can

Above, Muse performing in the round on an early tour date. At right, Marc Carolan at his front of house station prior to showtime.

only address 127 scenes via MIDI with the XL4. I used those all up on the last tour, so this time I had to come up with a system that allowed me to automate and expand, but not rely on the XL4's computer as much."

His solution was to make good use of a Midas PRO2C sidecar, which is not really a sidecar in his scheme of things. But better to let him explain it himself: "The one thing Midas has done with all its digital kit is made the gain structure freakishly identical to their analog stuff. Therefore, in the past when I've used a PRO2 as a sidecar with my XL4, it felt just like an extension of the main board in terms of how I set up the gain structure. The interface and busing were just seamless.

"Now I'm using half of a PRO2C's resources to basically serve as automation for the XL4, and the net result of my efforts is that I've created an XL4 that's as automated as a digital board. I can keep my hands on the faders and eyes on the band during the show, and all the automation simply happens in the background. I don't ever have to think about it. If the purpose of any of these techno-complications we give ourselves is to simplify the mixing of the show, I think I've succeeded nicely."

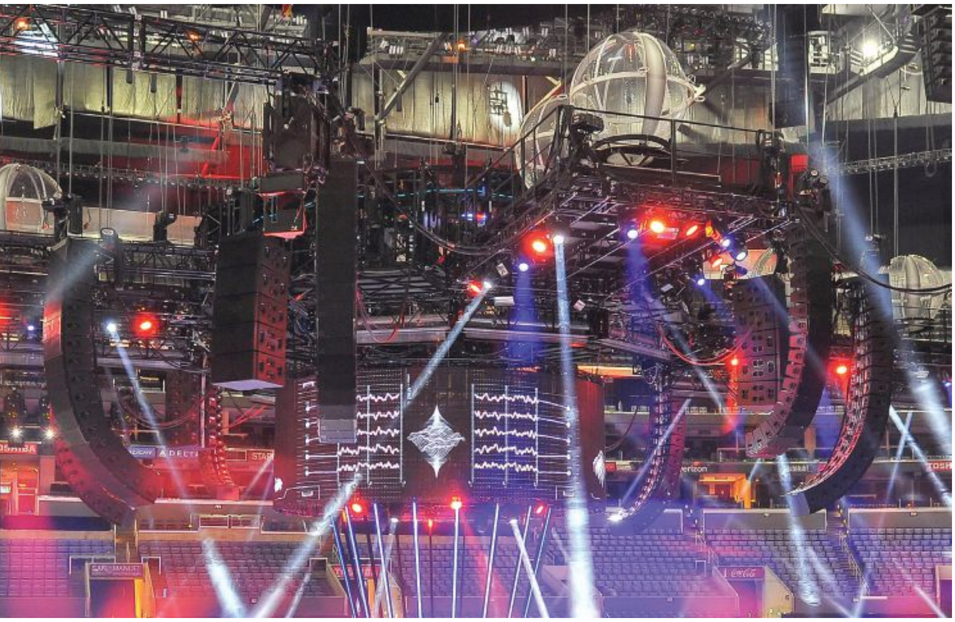
THE RIGHT BALANCE

Still standing with both feet firmly grounded in the analog world on this tour, Carolan naturally draws from an eclectic palette of outboard processing. "A lot of this stuff is, well… it's just classic," he says, first noting his Bricasti M7 reverbs, which see use on vocals and toms. Other recognized gold standards among his collection include a Line 6 Echo Pro delay modeler and Eventide H3000 harmonizer he calls upon for vocal effects, Yamaha SPX2000 added to snare and other percussion elements, and a dbx 120A subharmonic processor that gets sparingly splashed across the toms.

"I have a good palette to draw from," Carolan continues. "But I like to think I use it tastefully, especially given the added amount of reflections and higher RT times we create playing in the round. Given our 360-degree configuration, the room comes into play a lot more in terms of trying to get even coverage. I've had to become very aware of managing the relationship between reflective and direct sound. The way I approach it is to get the system to do the work of balancing things. That way my mix can remain a mix, not a tool to reach sonic engineering goals."

Part of the approach relies upon d&b audiotechnik's ArrayProcessing functions, which are found within the ArrayCalc simulation software. Incorporating filter algorithms that calculate and optimize the performance of the d&b line arrays over the entire 360-degree area of coverage, it's an ace in the hole in terms of achieving tonal balance.

"With ArrayProcessing," he says, "I gain the peace of mind that the tonal



Top to bottom: The production's ring of flying d&b audiotechnik J- and V-Series arrays. The Midas PRO2C at front of house providing automation for the XL4, flanked by a collection of outboard gear. Adam Taylor in his space beneath the stage that accommodates his stage monitor kit.

and dynamic content of my mix is really spreading evenly across everyone's ears. It makes a huge difference in this environment, where direct versus reflective sound is so important. I learned from the engineers at d&b that even the smallest tweaks can have maximum impact. Just a single dB often greatly alters whether what you perceive is in your face or heard outside of its spatial context. I can bias this system toward my needs of being linear in terms of power over distance, or linear in the sense of frequency response over distance. I have precise control over the amounts of energy present in the room, at all times and in all places."

HEARD, NOT SEEN

For his part, monitor engineer Adam Taylor deals with his mixes with an equal eye on every nuance. "Just like out front, being in the round makes my job difficult, as everyone is essentially in front of the PA all of the time," Taylor points out. "There's no hiding place ever."

Just trying to find Taylor at a show is a dilemma in itself. Located not far from Carolan's house position, he is nonetheless virtually invisible inside his bunker-like environment beneath the hammerhead of one of the thrusts, with little more than a letterbox to peer out of to physically see the stage. He mixes on a Midas PRO9, pushing the board hard with 79 inputs coming from the stage.

Keeping his stage volume down is imperative, so everyone is on wideband Sennheiser 2000 Series in-ear monitoring systems feeding either custom UE 11 buds from Ultimate Ears or generic Westone UM Pro 30s. Other than a lone bass amp, there isn't a single traditional cabinet onstage, with guitars all using Kemper profiling systems instead.

Beyond the three core members of the band (frontman/multi-instrumentalist Matt Bellamy, bassist/backing vocalist Chris Wolstenholme, and drummer Dominic Howard), keyboardist/utility player Morgan Nicholls is on this tour. For the size and scope of this show, microphone inputs onstage are surprisingly spartan, with wireless Sennheiser transmitters sporting Neumann KMS 105 capsules being the choice for Bellamy and BETA 57A mics from Shure standing in for Wolstenholme's backing vocals. At drums, a beyerdynamic M88/ Shure BETA 91 combination captures kick; there's also a Shure SM57 on snare top, while hi-hat and ride cymbal get an AKG 451 and the overhead crash cymbals are individually miked with diminutive Neumann KM 184s.

"We're struggling more than ever with spill," Taylor notes. "Even though we have this massive stage, the band has actually wound up closer to the drums compared to gigs past, and then of course there's the matter of playing in the round. All of our mic choices are calculated to help combat these problems while still providing the fidelity and performance the band requires."

A partial view of the very clean stage that's also in the round.

MAKING IT SO

Taylor's mixes for the band are generally a bit of everything, with prominence given to each individual as needed by that person. While he is keen on the sound of the Midas PRO9 and its onboard dynamic EQ, he still uses some outboard gear to reach his mix goals. To that end a TC2000 and a Yamaha SPX1000 are MIDI-controlled and have a scene per song. With the TC unit managing longer delays and "bits of fancy stuff," the SPX1000 makes itself heard on snare.

When Taylor calls up his scenes, he has four stereo mixes for each member of the band. From his outputs, signals travel through Aphex Dominator limiters along with a small amount of output EQ from the board for three of the mixes. For Bellamy's mix, he uses a four-band GML Model 8200 parametric equalizer from George Massenburg Labs. Apart from that, the only other piece of outboard gear within his inventory is a Summit DCL-200 dual-channel tube compressor and limiter.

Sonically, the Drones World Tour finds the band and its production in top form and over the top, with an abundance of technology at its disposal that despite its multimedia madness, never descends into gimmickry. "We've succeeded in turning something extremely difficult, expansive, and inert into something that is impressively musical and in motion," crew chief Tucker says on a closing note. "There's a mammoth amount of kit at large here, a mammoth amount of tech. To pull this show off every time is a big ask, but I guess we've proved it's possible, wouldn't you say?" LSI

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

TIME & PHASE ALIGNMENT

And when and where to choose them for system optimization. *by Bob McCarthy*

The setting of delay times in signal processors is one of the principal techniques of system optimization. In most cases the timing is set to "align" two (or more) signal sources so as to create the most transparent transition between them. The process of selecting that time value can be driven by time or phase, hence the relevant terms are "time alignment" and "phase alignment."

These are related but different concepts and have specific applications. It's important to know which form to use to get your answers for a given application.

Time alignment connotes a synchronicity of sources, e.g., they both arrive at 14 milliseconds (ms). Phase alignment connotes an agreement on the position in the phase cycle, e.g., they each arrive with a phase vector value of 90 degrees. Time alignment is most applicable when the sources are matched and have the same operating frequency range, e.g., a full-range main loudspeaker and the same model used as a side fill. Phase alignment is called upon when sources cover different frequency ranges, e.g., mains and subs.

Both time and phase alignment together may be required when unmatched sources cover matched frequency ranges, e.g., Papa Bear mains, Mama Bear side fills and Baby Bear delays. These are the broad strokes. Now let's dig deeper.

AND THEY'RE OFF ...

How can we explore a complex subject such as phase and time without resorting to "click-to-the-next-article math"? We will use both analogy and pictures of the real stuff in action.

The first analogy is a relay race. The first runners are aligned to a single starting point. The race begins with the starter pistol, the moment of time alignment between all sources. If the runners travel at the same speed, they are both phase aligned (their radial position on the track) and time aligned (the elapsed time puts them the same distance from the start). If one runner goes faster than another, then both phase and time fall out of alignment. If the difference reaches a complete lap, then the phase is aligned (again) but the time is not.

It's a relay race, which means the first runner for each team must hand off the baton to the second. The critical element here is that the two runners on our team must be phase aligned to make the handoff. The second runner intersects the first at the designated radial meeting point (the phase) regardless of the time (one team may be ahead of another but the handoff occurs at the same place). Each handoff is a "crossover" of the baton to

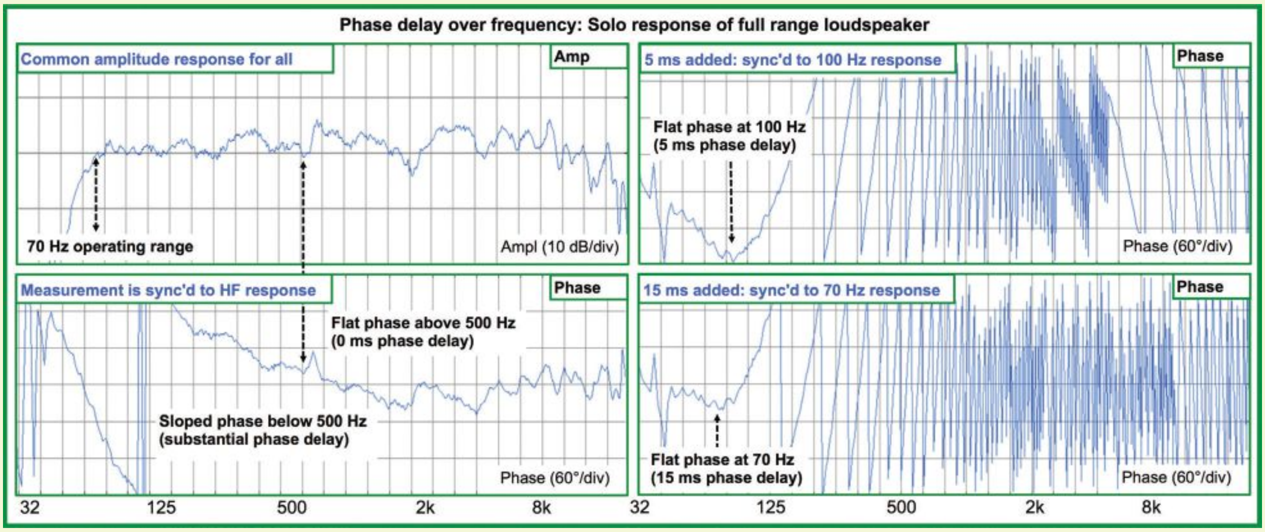


Figure 1 – Phase delay example for full-range loudspeaker.

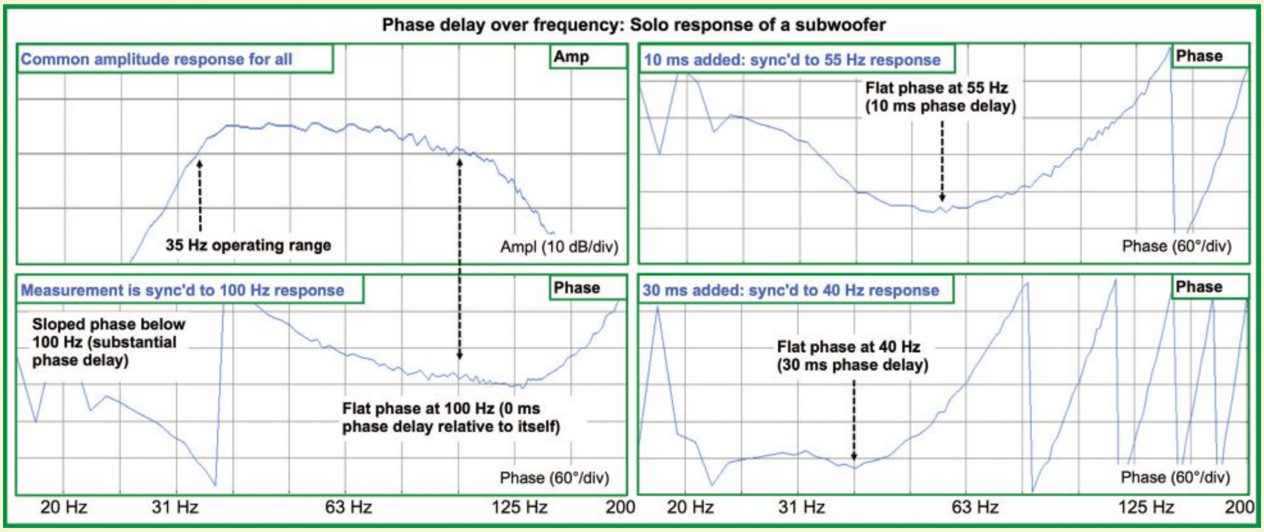


Figure 2 - Phase delay example for subwoofer.

another member of the team.

Our 3-way sound system is like a relay race, with tweeter, midrange and subwoofer running the segments. They must be phase aligned at each crossover to keep from dropping the sound to the ground and blowing the race.

TERMINOLOGY

Lets go through a bit of phase terminology to standardize the discussion, specifically: phase shift, phase delay, phase offset and phase alignment. Phase shift is frequency-dependent delay quantified in degrees, phase delay is the same thing quantified in ms, and phase alignment is the process of phase matching at a particular frequency and location.

The term "group delay" is used by some for phase delay but the distinction is not relevant here. Let's illustrate by example: A filter attenuates the amplitude and causes the phase response to bend 90 degrees at 1 kHz: phase shift of 90 degrees or phase delay of 0.25 ms. The same would go for a loudspeaker that has low frequencies lagging behind the highs (as in 99.9 percent of loudspeakers). One loudspeaker has 90 degrees of phase shift at 1 kHz and the other does not: phase offset of 90 degrees or 1/4 wavelength (the difference).

Time offset terminology is easier because it is frequency independent. Time offset causes phase offset, however, which is frequency dependent. A time offset of 1 ms causes 3,600 degrees of phase offset at 10 kHz, 360 degrees at 1 kHz, and 36 degrees at 100 Hz.

APPLICATIONS

The easiest way to visualize the need for time alignment is mismatched latency between devices in a common path. Latency is frequency independent, so the difference is a fixed time offset. The solution is time alignment by delaying the earlier signal.

The propagation time of a sound source through air is effectively "acoustic latency." Two matched loudspeakers arriving from different acoustical path lengths have a latency offset that can be compensated by time alignment.

Phase alignment comes into play when devices have unmatched phase responses over frequency. This should be a minor issue in analog electrical signals unless they're unmatched in terms of their upper and lower limits. Differences in AC coupling filters at the bottom end and TIM filters at the top end can cause phase offsets around the extremes. One solution could be exotic phase alignment filters, but the simplest would be matching the amplitude responses first, which may reduce or eliminate the phase differences.

Time alignment is straightforward. We can use the impulse response of a modern analyzer and read the time offset directly. This is true any time the ranges of the devices being aligned are matched over the large majority of their ranges. They don't have to be exact.

For example, a typical underbalcony loudspeaker has more very high frequency response and less low frequency range than the mains, yet time alignment should work fine because they have 6-plus octaves of overlap. Subwoofers ranging from 30 Hz up to 100 Hz can be merged with mains covering down to 60 Hz. There's less than an octave of overlap, which means time alignment is a poor choice (phase alignment is used).

MOVING TARGETS

Now let's get to the tough reality about loudspeakers. They have lots of phase shift and phase delay. A good quality active loudspeaker can be engineered to keep the range from 500 Hz on up within ±60 degrees of phase shift (< 1 ms of phase delay). By 100 Hz we can expect 5 ms of phase delay, rising rapidly below that. This is important because in the LF range, the

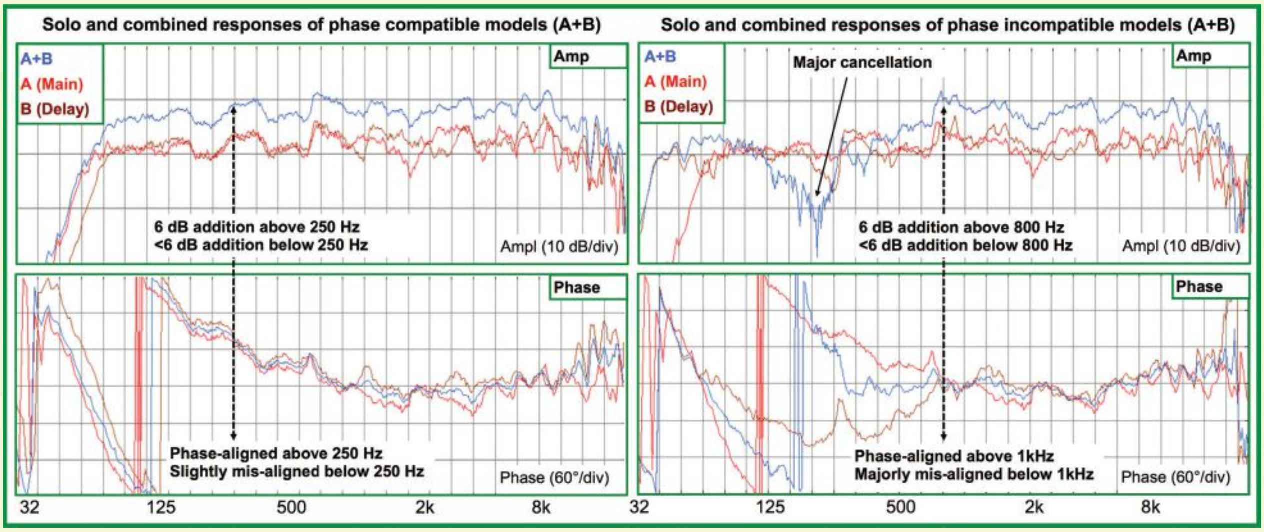


Figure 3 - Examples of phase compatible and incompatible loudspeakers.

loudspeaker can't be characterized as having a single arrival time. It has arrival times that span a very large range. For example, a 2-way system that reaches down to 70 Hz might have 10 to 15 ms of phase delay at the bottom (**Figure 1**).

A subwoofer covering from 30 Hz to 100 Hz typically shows more than 30 ms of phase delay between its upper and lower range, with a continuous range of values in between (**Figure 2**). We kid ourselves when we say the sub energy arrived at the mix position at 100 ms because it actually arrives spread over a 30 ms range around that.

How can we time align a thing that has 30 ms of slop factor over a course of two octaves? If you ever wonder why your analyzer has a hard time finding an impulse on subwoofers, think about the fact that the energy is spread over time – a lot of time. The ambiguous impulse response reading of the analyzer is the result of the temporal ambiguity of the system being measured, not the analysis method (even if you stretch out the measurement window and restrict the measured bandwidth).

The mains, by contrast, have widespread agreement about arrival time, with their upper six octaves all within 0.5 ms (at least the good ones do). This is why you see that beautiful impulse spike.

The causes of LF phase delay accumulation are a complex mix of electroacoustic behaviors and filters (maybe another article, another day), but the trend is observable on virtually any loudspeaker. Therefore our main array has phase shift (phase delay) that increases in the low end.

Matched models have matched phase delay, which means they have no phase offset (and hence no time offset). Such a matched pairing is inherently phase compatible. Any phase offset between them would have to be the result of time offset (such as a longer path) and would be remedied (if desired) by time alignment (**Figure 3**).

A different loudspeaker model may accumulate phase shift at a different rate over frequency, and therefore we will find phase offset begin to accrue between the pair. This is like the relay race where one team is faster than the other. If the amount of phase offset is small, we would classify the pairing as "phase compatible," which means time alignment is still an applicable tool. If the offsets are large, then we will need to use both time alignment (to synchronize the compatible areas) and phase alignment (to reduce phase offset in the incompatible areas).

A marriage between 2-way and 3-way systems is a classic phase compatibility challenge. All-pass filters are the most typical tools for phase alignment of incompatible models.

MAKING CONNECTIONS

Now on to our second analogy: freight trains. When did the train cross Main Street? The first car crossed at 12:00 but the last one did not cross until 12:05. Our loudspeakers cross the mix position like that. First the highs, and then later, the lows. Let's give each octave it's own car and see how it works (**Figure 4**). If we had a loudspeaker with 4 ms of phase delay per octave, we would see evenly sized cars that are a total of 36 ms long.

That would be a terrible loudspeaker, but stick with me here. We could have another train running next to it on a parallel track. Each car would line up to its counterpart, i.e., the trains have lots of phase delay but zero phase offset. They are time aligned.

Now let's move this closer to audio reality. The first six cars (16 kHz to 500 Hz) are so small they only occupy 0.5 ms of track, but after that the cars get bigger and bigger until they stretch all the way to 36 ms (like the previous one). If we run this train next to the first one, they will match at the first and last cars – but the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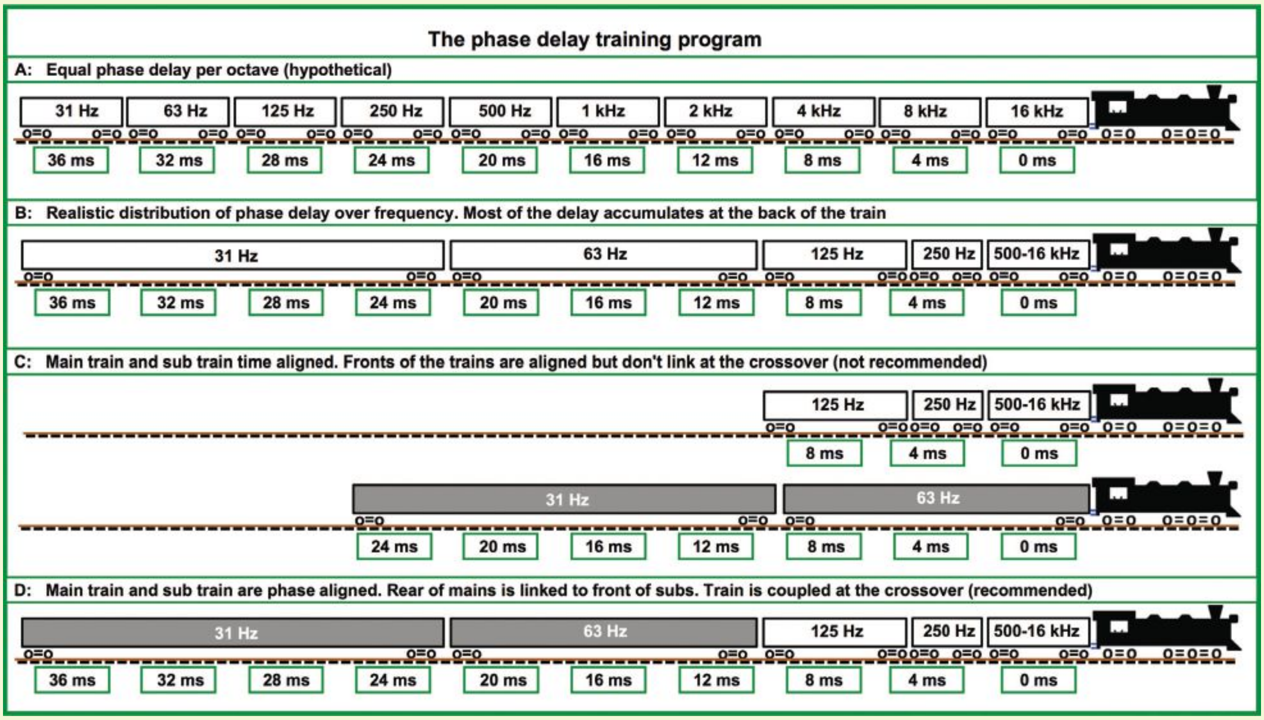


Figure 4 - Phase delay train analogy.

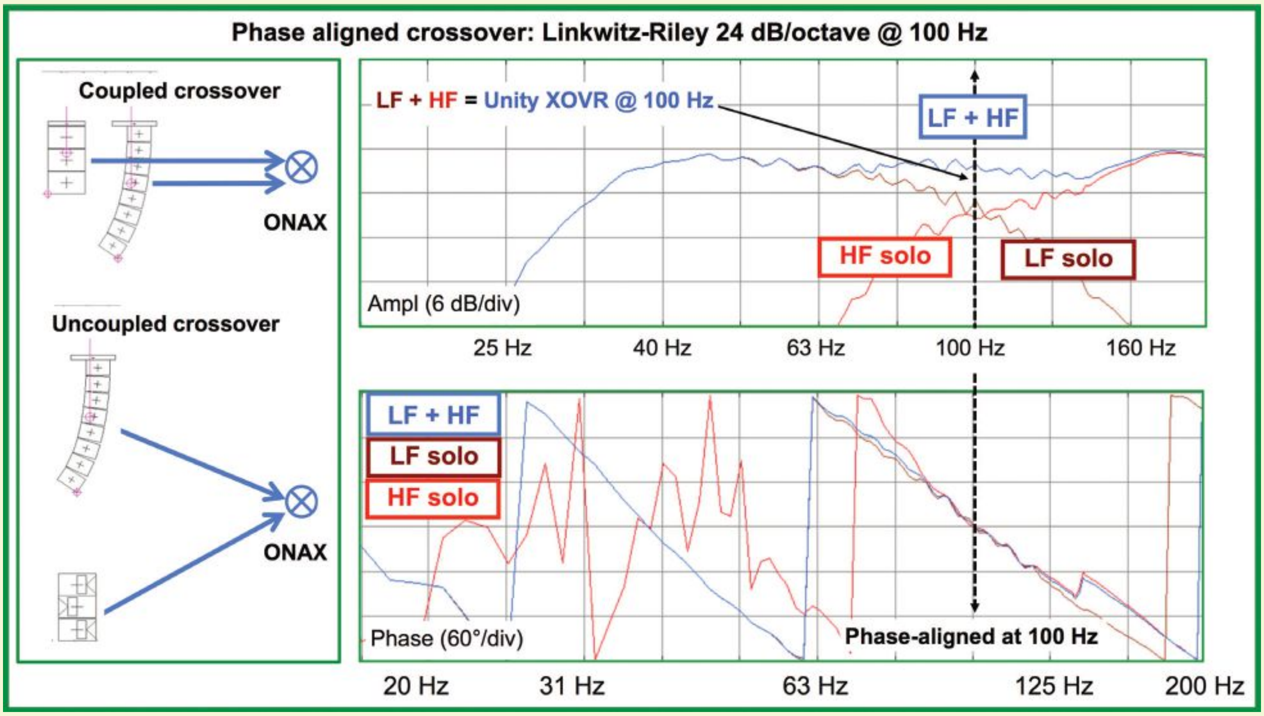


Figure 5 - Phase alignment example for main and subwoofer.

middle cars don't match. The front and rear cars are time aligned but the middle section is full of phase offset. The systems are not "phase compatible."

How do we connect a main train to a subwoofer train? This is a series connection (unlike the time alignment on the parallel track). Connect the rear of the main train to the front of the subwoofer train. The cars are phase aligned at the coupler between the two sections.

THE PROCESS

We can time align a full-range loudspeaker to a full-range loudspeaker or a subwoofer to a subwoofer since each pairing covers the same range. It's when we want to marry mains to subs that time alignment goes out the window in favor of phase alignment. The same principal applies for a crossover between an HF driver and a LF driver in a 2-way box, but in the modern world, this is handled by the manufacturers, right?

So how do you do it? That's the easy part. Time alignment is a 3-step process. Use the impulse response because it will give readings directly in the applicable unit: time. Observe the impulse response of A (solo), and then B (solo), and delay whoever got there first. Then take a look at the frequency responses and verify that maximum addition has occurred. A little trick here is to observe the highest frequency range bump the delay up or down by the smallest allowable increment (e.g., 0.01 ms) and see what gives the best addition.

Phase alignment follows the same steps, but we are going to use the phase response instead of the impulse. Look at A solo, then B solo, and add delay to the early one until the phase responses match (**Figure 5**). Polarity inversion can be considered in cases where this allows for a smaller amount of delay to be used.

For example, a 100 Hz crossover might be phase aligned by 7 ms of delay on the subs or 2 ms and a polarity reversal. The later approach minimizes the overall time spread over the spectrum. Another approach to minimize overall time stretching could be all-pass filters in the affected area.

There are many ways to approach the alignment of sources that mix together. All of them affect phase and time. It's very helpful to know which of the alignment paths is most suitable to your particular application. LSI

__Bob McCarthy__ has been designing and tuning sound systems for over 30 years. His book Sound Systems: Design and Optimization is available at Focal Press (www.focalpress.com). He lives in NYC and is the director of system optimization for Meyer Sound. Bob also offers special thanks to Merlijn Van Veen for his help with this article.

SPECIAL CONSIDERATIONS

The engineer's guide to working with true singers.

by Mark Frink

Welcome to mixing monitor sound for a singer. There are many kinds of professional entertainers, songwriters and celebrities, but when working for that breed of performing artist who sings exceptionally well, a refined audio approach helps them do their best on stage.

When she's in her zone, a palpable connection to every member of the audience produces goose bumps and wet eyes. She's easily identified because she holds the microphone in her hand.

THE MICROPHONE

For singers, the microphone is the handle for the entire sound system, pro audio's "mother of creation." Young singers may not prefer a particular mic, but eventually become attached to a specific model. All other equipment is easily substituted, but that same mic, in the palm of her hand, provides daily consistency and confidence to sing her best.

With young singers, it's possible to try different vocal mics, but after using one model for years, its sound, balance and weight feel familiar and it's simply a comfortable pair of shoes. Changing mics for older singers is a huge challenge that should never be tried on a day with the pressure of a show later.

While hard-wired mics provide the best performance and value, today's singer often requires wireless freedom of movement. Improved companding helps wireless sound more like wired, and digital wireless provides noise-free audio without companding at a cost of three milliseconds of latency.

As a result, choosing a vocal mic often includes choosing a wireless system. Adding to Shure's wide assortment of industry-standard vocal capsules, many mic makers have offered capsules that fit Shure handheld transmitters. Lectrosonics in turn released a handheld transmitter that fits Shure-compatible capsules, Sennheiser and Neumann capsules using an adapter, as well as its own HHC condenser.

Lectrosonics HH transmitter that accommodates Shure (and many other) capsules.

Dynamic mics have advantages on loud stages, are more rugged and don't need phantom power, but true singers favor condenser mics for their transparent, accurate sound. Condenser mics are detailed and crisp, but tend to pick up nearby sounds, so some distance between singer and band helps on loud stages.

Mics with tighter supercardioid or even hypercardioid polar patterns better isolate the singer's voice on louder stages. Hard-wired condenser mics must be used with their high-pass and pad engaged, as its proximity effect is too boomy and her voice is too loud for the capsule on big notes.

Safely store the show mic until she arrives on stage. Use the spare when you need to talk into her channel so you're not putting your germs in her mic. If her channel seems a bit quiet when you speak into it, that's because she produces more level than most. And I probably don't need to tell you that there's no smoking anywhere near the stage until the singer has left the venue.

While frequency response can be tailored with EQ, polar response and handling noise can't be adjusted electronically, making them important features, the best reason for choosing the right vocal mic is that it can help their singing.

MONEY CHANNEL EVOLUTION

Vocal compression is a trademark of pop vocals, and yesterday's analog front of house engineers always inserted a vocal compressor, ranging from dbx and Drawmer standbys to vintage or boutique tube and optical compressors.

Singers hate compression, but engineers use it to create the "studio sound" that exemplifies pop music by reducing the natural dynamic range to something that can sit comfortably in a mix while a singer goes from a whisper to a scream and the band moves from acoustic ballads to up-tempo rockers.

Without compression, even Tony Bennett would startle the blue hair in the front row.

Besides a compressor, inserting a graphic EQ into the lead vocal channel is an old trick that allows quick EQ choices beyond a console's 4-band EQ. Enterprising engineers would even insert one channel of a stereo GEQ and use the other channel before the compressor's side-chain or "detector" input, causing a strident part of the singer's upper register to be compressed early by pushing that frequency in the detector.

However, emphasizing a frequency in a compressor's side-chain still causes the entire signal to be compressed. The move towards brighter-sounding condenser vocal mics and a compressor's difficulty reducing transient sibilance encouraged the use of dedicated de-essers to keep live vocals sounding natural.

Outboard de-essers provide a single tunable band of dynamic EQ focused to reduce very high frequencies in the 4 kHz to 10 kHz sibilance region. Soon the idea of dynamic EQ for other frequencies became popular and the BSS 4-band DPR-901 became the hallmark FOH outboard vocal processor in the 1990s.

BSS DPR-901 parametric dynamic EQ.

Touring FOH engineers have also used premium multi-function studio "channel strip" processors that combine a mic preamp, compressor, de-esser and/or EQ. There are also a variety of channel strip plugins and digital live sound environments for building custom strips from a wide variety of plugin emulations.

Many engineers call the Waves C6 their one indispensable plugin, reducing and even replacing vocal channel EQ. The C6 is a six-band compressor with a "paragraphic" user interface. It combines four channels of multiband compression with two additional floating bands that can be used for de-essing sibilance and de-popping plosives. Multi-band compression is also standard built-in processing on many premium digital consoles.

Screenshot of the Waves C6.

MONITORS

A chanteuse will often have repertoire with a large vocal range and phrases ending in big notes. As singers move from their lower to upper register, their voice responds differently and tailoring the sound system's response to match produces smoother transitions. Sustained notes to which vibrato is added produce rich overtones. Singers benefit from frequency response tilted towards the highs in their monitors to help them control register change and vibrato.

While FOH engineers can endlessly tweak, snip and polish a singer's voice in the house mix, monitor engineers have fewer options. Vocal compression is never used in the singer's monitors, as it makes it harder to sing. So when a singer hits a big note, the stage monitors faithfully reproduce it, while the voice gets knocked back in the mains by 3 to 10 dB by the FOH compressor.

The result is that the singer hears the sound jump out of the stage monitors on big notes, while it collapses in the mains. The singer hears softer notes resonate in the venue, while big notes dip and move onto the stage where the uncompressed sound is dryer, other than its reverb, but with fewer acoustical reflections.

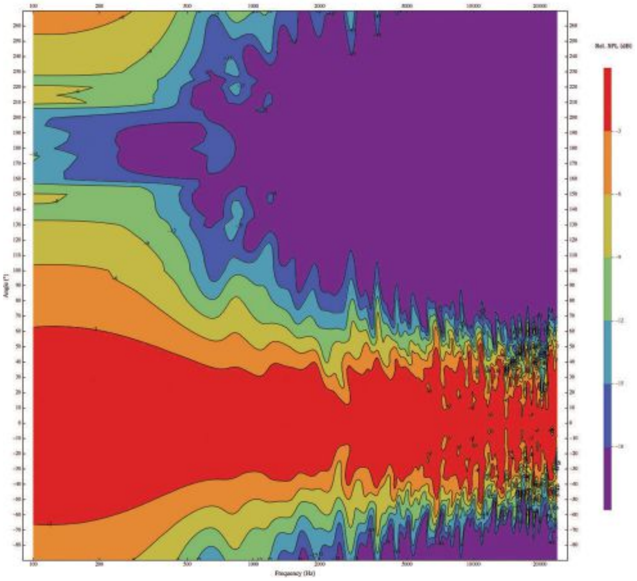
Good singers put up with it. Great singers learn to cheat the compressor with mic technique. They know which big notes FOH compression will suck up and momentarily pull the mic away from their mouth, using the inverse square law to reduce the power of their voice by increasing the distance with their arm.

When the mic is pulled away on big notes, the singer avoids the compressor and the vocal collapses far less from her perspective. Younger singers have difficulty understanding this; older ones do it instinctively.

The vocal sound reflecting back from the venue is a vital part of what singers using wedge- or side-fill-based monitoring hear on stage, making it important that FOH engineers keep the vocal and its reverb well on top of the band in the main mix. Not only does sound bounce back on stage from venue walls and ceiling, it comes back on stage from the sides and back of the mains.

Point source arrays lose directivity in the lowest octaves and often have a midrange side-lobe, so the main vocal is already loud on stage at those frequencies. Reducing those frequencies in the vocal for the singer's wedges and side fills keeps it sounding natural on stage. Delaying the singer's wedges and side fills in relation to the mains can help them sound more like a single system.

Line arrays have better pattern control than point source arrays at lower frequencies, due to improved LF coupling, but only with arrays of sufficient length to control their lowest frequencies. While women don't sing much lower than 250 Hz (B3), tenors' lowest notes are an octave below. That means a male singer in a venue with 6-box line arrays will feel significant amounts of their mic's proximity effect on stage from the mains (and his acoustic guitar wants to feed back at 160 Hz).



Horizontal contour plot of a 3-way loudspeaker.

COURTESY OF JBL PROFESSIONAL

IN-EAR MONITORS

Most real singers generally dislike in-ear monitors (IEMs). Putting something in their ears blocks sound in the venue from reaching their ears directly, but also destroys the binaural effect of sound laterally cross-feeding. It's fatiguing to wear IEMs for long periods of time because they sound unnatural to a brain that's built to hear binaurally.

Vocalists also have a requirement to interact with their audiences, responding to calls from the fans: "We love you!" IEM engineers attempt to add this to the mix with hard-panned downstage audience mics that are fadered-up between songs to help singers hear and localize fans accurately, but it's a poor substitute.

The Sensaphonics Active Ambient system gives performers back the natural sound of their ears by embedding binaural microphones in custom molds to add natural sound back into their ears when desired with the flick of a switch on their bodypack.

Sensaphonics Active Ambient system.

IEMs force performers to hear through a monitor engineer's mix, placing a higher requirement on getting everything to sound natural through miniature transducers squeezed into ear canals. Besides individual instruments and voices being heard directly at the ear instead of in an acoustic space, reverb itself must replace a venue's acoustics, requiring a powerful, natural sounding reverb.

REVERB

Reverb plays an important role in a singer's monitors. Algorithms with dense Early Reflections (ER), as well as a 4-way crossover are preferred. Turning down the reverb and listening to the ER for a natural sound helps audition reverb presets.

The 4-way crossover allows the reverb to be tailored to complement a venue's natural acoustics. Shortening lower frequencies that already dominate the stage helps. Leaving high mids the longest helps brighten the room for the singer.

Next, EQ the reverb to sound natural, cutting each region that sounds metallic or artificial. Finally, pre-delay is the critical adjustment; it must vary from 20 to 30 milliseconds, depending on the room. Like any special sauce, a little bit goes a long way – don't overdo it.

WEDGE LAYOUT

There are many designs and approaches to wedge-based monitoring for different types of performers, but for singers, it's about their mic and their reverb. But first, carpeting the stage is recommended. Inexpensive indoor-outdoor carpet comes in 6-foot rolls and, not only makes the stage sound better, it feels good. Even one piece across the downstage edge can help.

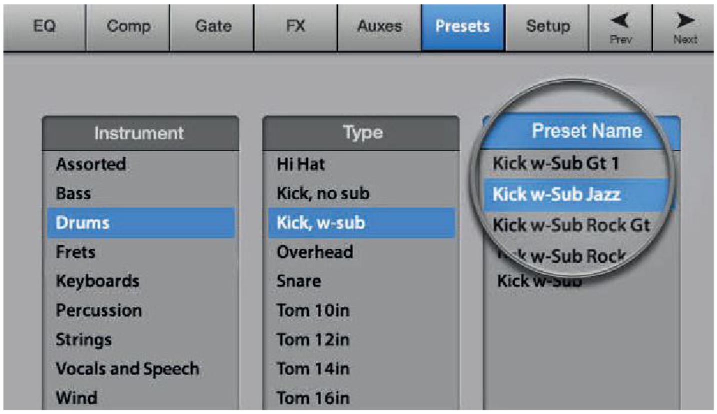
Stereo wedges should be a dozen feet apart and facing each other so the sound comes from each side, maximizing the reverb's stereo effect. With the back of the wedges propped up with a two-by-four, their horns are on-axis when standing downstage center.

She travels to the sides of the stage so a second wedge, similarly angled, is needed about 12 feet past the first. This second pair can be a mono mix, as she'll only hear one at a time. This mix can be high-passed more than the others, as lots of low-frequency energy is coming off the mains at side-stage. The so-called upstage "butt-fill" is a fourth mix that helps when she steps towards the band.

The main thing that goes in the singers' mix is her vocal and its reverb. In reverberant halls, a small amount of piano for pitch and maybe kick or hi-hat for time may be needed. See you in catering. LSI

Mark Frink *is an independent engineer who has mixed monitors for a few singers and is available this coming summer.*

POINT-AND-SHOOT **MIXING**



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

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

IN, OUT, AND ABOUT

Interconnect capabilities of recent digital consoles/mixers. *by Craig Leerman*

Digital consoles typically aren't limited in terms of interconnect to what's physically included within the control surface, expandable with a variety of stage boxes that provide a wealth of inputs and outputs as well as routing and more in both the digital and analog realms. Many of these boxes also carry the console's "mix engine" and remote controllable microphone preamps, along with facilitating networking with other system components and handy for all sorts of outboard recording and playback, broadcast feeds, and interfacing with additional consoles.

The most utilized pro audio networking protocol is Dante from Audinate. AES, MADI, CobraNet and EtherSound are among other widely used protocols, while several manufacturers have also developed protocols for interfacing their own audio devices, and sometimes others. The biggest difficulty with the variety of network protocols is that they're not easily compatible with each other in a "plug and play" way; however, in many cases this can be solved with "bridge" devices.

A nifty new bridge introduced last year is the Orange Box from DiGiCo, providing a pair of DMI (DiGiCo Multichannel Interface) card slots that allow users to choose from a wide range of protocols in creating an audio path, including Dante, Hydra 2, MADI (on coax or cat cable), Aviom, Optocore, ADC,

Allen & Heath DM48, one of three dLive DM MixRacks.

DiGiCo Orange Box

AES, DAC and Soundgrid. Further, the developing AVB (Audio Video Bridging) set of standards is focused on creating an interoperable ecosystem of low-latency, time-synchronized, highly reliable synchronized networked devices using open standards through certification by the AVnu Alliance.

Against this backdrop, let's take a look at the connectivity offered by a variety of recent digital consoles and their stage boxes, along with some increasingly popular rack-mount mixers that have hit the market within the past year or so.

The new **Allen & Heath dLive Series** is available with three surfaces (20, 28 and 36 faders) and three DM MixRacks that also contain the mix engines. All of the surfaces are equipped with 8 XLR mic/line inputs, 8 XLR line outputs, 2 digital stereo AES3 ins, 3 digital stereo AES3 outs, and a USB stereo recording and playback. Each surface also provides a dual redundant GigaACE gigabit link for the accompanying MixRack, a redundant DX link for I/O expansion, 2 I/O ports (128 channels, 96 kHz each) and 2 network ports.

DM MixRacks are available in 32 mic/line ins x 16 outs, 48 mic/line ins x 24 outs, and 64 mic/line ins x 32 outs. All can work with up to 128 input channels and 64 mix outputs by adding DX expanders or patching digital sources. There's also dual redundant GigaACE gigabit links to the control surface, 2 redundant DX links for I/O expansion, 3 I/O ports, a 48 kHz port for the ME-1 personal mixing system, and 2 network ports.

The **VENUE S6L** from **Avid**, now shipping, is also available with a choice of three control surfaces in sizes from 24 to 32 faders (plus 2). All surfaces have 8 onboard mic/line ins and 8 outs as well as 4 stereo AES ins and 4 stereo AES outs.

Two available mix engines are differentiated primarily by channel and plugin counts. The E6l-144 supports up to 144 channels and 64 mix buses (plus L/C/R) and 125 plugin slots, while the larger E6L-192 delivers 192 input channels, 96 mix buses (plus LCR) and 200 plugin slots. Each engine also

Left: Avid Stage 64 for the VENUE S6L.
Above: Front and back panel of the Mackie DL32R.

includes 5 USB ports and an AVB-192 Ethernet AVB network card with two ports, selectable as etherCON (copper) or SFP (fiber), with redundant ring topology. (Support for up to two AVB cards is coming soon.) And, option cards are also available for recording and playing back up to 64 audio tracks via AVB. MADI and Thunderbolt.

VENUE Stage 64 racks provide the main I/O for the S6L. Each rack can be fitted with up to 64 ins and 32 outs, accompanied by a selection of a variety of analog and digital option cards, including AES, ADAT, and Dante.

The **DiGiCo S21** offers a lot of processing and routing options in a compact package. It includes 21 faders and 2 touch screens, and can process 40 flexi (stereo or mono) input channels and 46 output buses, including 16 flexi aux/submaster buses. It carries 24 mic/line ins and 12 analog outs plus 2 AES I/O (mono). Dual DMI option card slots accept up to 64 additional I/O per slot, and 2 Ethernet ports facilitate networking. The USB2 I/O interface accommodates recording and playback of up to 48 channels.

Stage box choices abound, with DiGiCo offering 8 different racks that can be configured with analog and digital ins and outs including AES/EBU, Dante, AES-42, ADAT, HD-SDI, and Aviom. Some of the larger stage boxes also accept DiGiCo D-TuBe inputs, which are tube preamps that can replace the last 8 inputs.

The **Mackie DL32R** is a rack-mounted mixer that looks like a stage box. The compact (3 RU) unit is controlled wirelessly via an iPad running the Mackie Master Fader app. Mixing can be accomplished using iPads, iPhones or iPod Touch, with up to 20 control devices able to access the mixer at one time. The DL32R comes with 36 input channels and 28 output buses, while the onboard USB 2.0 interface provides 32 x 32 recording and playback for PC and Mac, as well as directly to a 2.0 USB hard drive.

Additional onboard connections include 24 XLR mic/line ins, 8 XLR/TRS combo mic/line ins, 14 XLR outs and an AES out. An optional Dante card is available that allows the DL32R to network with larger systems.

A recent entry from **Midas** is the **M32R**, a 16 (plus 1) fader, rack mountable version of the M32 console. While compact, it supplies up to 40 simultaneous input channels and a 32 x 32 USB 2.0 audio interface for recording and playback. Onboard connections include 2 AES50 networking ports that allow up to 96 ins and 96 outs, 16 XLR mic/line ins, 8 XLR outs, quad 1/4-inch I/O and dual 1/4-inch or RCA I/O, and ULTRANET protocol port.

Stage boxes for the M32R include the DL16, which has 16 XLR mic/line ins and 8 XLR outs, and the DL32 with 32 XLR mic/line ins and 16 XLR outs. Both are also equipped with MIDI I/O for bidirectional communication between M32 consoles and on-stage MIDI devices, along with 2 ADAT outputs providing 16-channel digital output on two optical TOSLINK connectors and 2 AES50 SuperMAC ports for cascading additional DL stage boxes.

The latest development from **PreSonus** is the **CS18AI**, an AVB control surface for the RM Series of mixers that allows them also act as mix engines/stage boxes. The CS18AI control surface has a built-in 4 x 2 AVB network for sending and receiving audio sources with RM mixers, fostering the use of talkback microphones and playback devices at front of house. The two returns are shared with the monitor line outputs and headphone output, and can be sourced from any output of the mixers, including the solo bus, main mix, or any aux mixes. The CS18AI, which sports 18 moving faders, can also be used as a controller for the Studio One DAW.

The key to the system for live shows are the StudioLive RM32AI and RM16AI mixers, which are rack-mount units taking the form of a stage box that are controlled via the PreSonus UC Surface touch control software for Mac, Windows, and iPad (or used with the CS18AI). The RM16AI is equipped with16 XLR mic/line ins, a pair of RCA ins, and 8 XLR outs. The larger RM32AI has 32 XLR mic/line ins, a pair of RCA ins, and 16 XLR outs. Both have XLR L/R and mono outs, as well as DB25 jacks that mirror the line-level mix outputs, MIDI I/O, and an integrated, continuously bidirectional 52-in/34-out, FireWire 800 recording interface. An optional Dante card is also available.

The more compact **Roland Pro AV M-5000C** joined the flagship OHRCA M-5000 console a few months ago, with the recent version 1.2 software update adding a host of features for both consoles, including remote control software, customizable and resizable window sets, offline console setup and management, real-time analyzer, de-esser and much more. The M-5000C is outfitted with the same features as its big brother, just in a smaller form factor with 16 + 4 faders instead of 24 + 4. The internal mix architecture is not fixed, and any of the 128 audio pathways can be assigned as an input, output or mix bus.

Onboard connections on the M-5000C include 16 XLR mic/line ins, 8 XLR outs, 2 AES I/O, MIDI I/O and 3 REAC network ports. These are joined by 2 expansion slots for optional Dante, Soundgrid, MADI or REAC cards. Stage boxes ranging from 8 to 40 channels can accompany either console, with the largest, the S-4000S, offering modular chassis that is typically configured with 32 ins and 8 outs. Additional stage boxes can be linked, and the S-4000M Merge device allows up to four digital snake heads to be merged into a REAC stream, where the master device recognizes it as single REAC slave unit.

The latest console from **Soundcraft** is the **Si Impact**, a 26-fader model providing 40 input channels and 31 output buses. Like other Si Series consoles, it offers one-knob function control and FaderGlow that illuminates the fader track in different colors to provide at-a-glance status information. The Impact adds a new option in the form of its 32-in/32-out USB interface for recording and playback. Onboard, the Impact has 24 XLR mic/line ins, 8 XLR/TRS combo mic/line ins, 16 XLR outs, AES out, and USB port. Expansion slots facilitate networking with MADI as well as Dante.

Two versions of Mini Stageboxes provide remote location of inputs and outputs. The smaller Stagebox 16 is equipped with16 XLR mix/line ins and 8 XLR outs, while the larger Stagebox 32 has 32 XLR mic/line ins and 16 XLR out. Both are rack mountable.

Solid State Logic recently released new V3 software for the new (and more compact) **L300** and larger L500 consoles. It delivers more than 40 new features. including an expanded feature set for the original Query function. There's also an optional Dante interface card, remote control software, console expander mode, user interface changes, new effects, and more. The L500 (now the L500 Plus) offers 256 mix paths, while L300 goes from 128 to 192 (144 fully processed, 48 dry) mix paths.

While smaller than the L500, the L300 still provides 26 faders and onboard connections that include 14 XLR mic/line ins, 12 XLR outs, 4 pairs of AES, MADI (in both coaxial and optical), and optional Dante and Blacklight II cards. The accompanying ML 32.32 stage box supplies 32 XLR mic/line ins, 32 XLR outs, and 16 pairs of AES I/O, with an option an additional rear-mounted 32 mic outs.

Recently making its debut, the **Yamaha TF Series** is available in 3 frame sizes, each with the TouchFlow interface optimized for touch panel control, and practical presets for newer and experienced operators. All models include 34 x 34 recording and playback capability via USB to computer, as well as 2-track recording and playback via USB to a drive.

Yamaha RPio622 for the RIVAGE PM10.

The rack-mountable TF1 (17 faders) offers 40 input channels (32 mono plus 2 stereo plus 2 return) and 16 XLR/TRS combo mic/line ins. The mid-sized TF3 (25 faders) has 48 input mixing channels (40 mono plus 2 stereo plus 2 return) and 24 XLR/TRS combo mic/line ins. The larger console TF5 (33 faders) carries 48 input channels (40 mono plus 2 stereo plus 2 return) and 32 XLR/TRS combo mic/line ins.

All models also have 2 RCA ins, 16 XLR outs, as well as an expansion slot for an NY64-D Dante card for networking. A new stage box slated for release in 2016 is the Tio1608-D, offering 16 XLR mic/line ins and 8 XLR outs. Up to three Tio boxes can be daisy-chained together without the need for network switches.

The flagship **RIVAGE PM10** from **Yamaha**, which is now seeing field applications, consists of the CS-R10 control surface, DSP-R10 DSP engine and RPio622 interface box. The system offers 144 input channels, 72 mix buses, 36 matrix buses, plus stereo A and B or mono buses.

The control surface (36 faders) has 8 XLR mic/line ins, 8 XLR outs, 4 AES I/O, and a pair of MY expansion slots. The engine provides 2 additional MY card slots and 4 HY card slots for Dante or Yamaha's TwinLane audio network. Inputs and outputs are handled by the interface box, which can accept up to 6 RY cards that are available in 16 XLR mic/line ins, 16 XLR outs or 8 AES I/O versions. The interface box also offers pair of HY card slots. LSI

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

OTHER THINGS

Managing wireless systems is about more than coordinating frequencies.

by Ike Zimbel

When deploying wireless, whether it's a single system or several, and whether it's for a tour, a one-off, a broadcast, at a church or wherever, it's important to keep track of a whole range of parameters besides the most obvious one of frequency selection.

Why? The two main reasons are consistency and preparation. In other words, making sure that anything that needs to be replaced, from a single piece to the entire system, is checked and vetted as well as outfitted with the same settings. The goal is to avoid disrupting or delaying the show. Here's a checklist of key parameters, followed by some specifics:

MIC RECEIVERS

- Output level
- Squelch settings
- Sync settings
- Rear panel switch settings such as mic/line and ground lift
- Battery type, for monitoring purposes
- IP address, also for monitoring purposes
- Sample rate and other information when using digital output
- Firmware revision number

Sennheiser ew300 IEM receivers on tour last year with the Rascals.

MIC TRANSMITTERS (HANDHELD AND BELT PACK)

- Input sensitivity/gain
- RF Tx power
- EQ and/or high-pass filter (HPF) settings
- Capsule type/model (with handheld)

IEM TRANSMITTERS

- Input sensitivity
- Tx power
- Mono/stereo
- Sync settings
- IP address

IEM RECEIVERS

- Squelch settings
- Limiter settings
- EQ settings
- Pilot tone
- Stereo/mono

ANTENNAS

- Type
- Placement
- Cable

SPELLING IT OUT

Wireless mic receiver outputs and IEM transmitter inputs are usually (and should be) worked out in consultation with the mix engineers, with the choice usually being between some version of mic or line level. Whatever the choice, make sure that all units are set the same way. It's important to get on this one early because once line check starts, it's far too easy for the engineers to push a pad switch or grab a gain knob to compensate for the one channel that's different from all the rest. This same idea applies to any ground-lift switches on the rear of the receivers; set them all to the "ground" position as a starting point.

With receiver squelch settings, keep in mind that if you've done a reset, the receiver may have defaulted to the lowest squelch setting. In today's noisy RF environments, wide open squelch settings are unlikely to do the job. I recommend starting with a low-to-moderate setting (i.e., 9 dB on a Sennheiser system) and see where things go from there. Sometimes more is needed.

It's important to stay on top of what information is being transferred during the sync process, and it's also important to stay on top of changes as they occur. For example, during rehearsals the sensitivity setting on an instrument belt pack may change a few times as the engineers and backline techs work out the best settings for each player/instrument.

Once this has been nailed down, I recommend entering that setting into the receiver and then adding "sensitivity" to the list of parameters that get synced. That way, you won't end up inadvertently changing the sensitivity during the course of normal frequency changes – and – in the event that a pack fails, the replacement can be synched and ready to go instantly. The same thing applies to various other parameters that can be synced, such as power level, name, cable emulation, etc.

Be aware that many newer systems have different power consumption curves for alkaline and rechargeable batteries, so that needs to be set correctly in order to get accurate battery level display information.

Also with many of the new systems, *all* parameters and adjustments are done in firmware, so performance can change from revision to revision (especially since most revisions are to fix something anyway). On a recent tour, I had two identical receivers in a backline rack. With the squelch set the same on both units, one would open up with the occasional "hit" (with transmitters off) while the other would stay closed and quiet. Upon investigation, I found that the unit that was opening up had much older firmware. I didn't get a chance to update this unit, but with everything else being the same (positioning, antennas and so on), I'm pretty sure that the firmware was the culprit.

Scout the best antenna locations and make sure mounts are on hand.

MODERN COMPLICATIONS

The single most common fault I've seen with remote monitoring systems is "losing" a unit due to an IP address issue. This usually shows up as the unit being present in the display but "grayed out" and unresponsive to control inputs (while happily working away in the actual rack). So if there are static IP addresses for each unit, make sure they're written down or otherwise saved (I use a spreadsheet). And if you are using "automatic," make a note of that too.

The same goes for sample rate and other information when using digital output. This is the first thing to check if a replacement unit won't "talk" to whatever the original unit was connected to (like an AES input in the stage rack).

With wireless mic transmitters, all of the above applies, plus note that once RF power level has been set, it can often be stored in the sync settings on the receiver. In addition, make note of any EQ, HPF or pad settings that are in the hardware of the handheld/belt pack (in other words, not part of the sync process). Finally, make sure mic capsules are suitable for handheld transmitters.

With IEM transmitters, be aware of the input sensitivity, although this is usually up to the monitor engineer. Also check transmit power levels, mono/stereo settings and associated pilot tone settings where applicable.

With antennas, you can never go wrong with tracking the make and model of each one ensuring that they're compatible with your systems. Other things to focus on include gain/attenuation settings (if the antennas include them), frequency range if antennas are narrow band (i.e., filtered to a specific range), and the length and type of cable for the antennas, along with couplers and/or adapters. This information is absolutely necessary for cable loss calculations. Finally, scout locations and mounts for the antennas ahead of time.

OTHER, OTHER THINGS

In live audio, it's been my experience that RF gear tends to move around a lot more (in rental inventory) than say, an amplifier that might live in the same amp rack for years. As a result, there's a very good probability that any given rack of wireless gear will a) have been put together for your show, and b) probably have a variety of different settings from unit-to-unit left over from the previous show(s).

Focus on what the systems are telling you.

So for starters, it's a good idea to perform a re-set on all units to get them back to the same start point. In addition, if you have input as to how the rig gets built (or if you're building it yourself), try to ensure that the units get racked with the lowest bands corresponding to your lowest unit numbers. For example, if rack-mounting Sennheiser 2050 Series IEM systems, start with all of the "A" band units (516-558 MHz), followed by any "G" band (558-626 MHz) units, and then any "B" band (626-698 MHz) units. (The same applies to all systems, regardless of manufacturer).

Why is this important? Because frequency coordination programs like IAS, as well as coordination/monitoring programs like Shure Wireless Workbench, Sennheiser Wireless Systems Manager, and Lectrosonics Wireless Designer all organize frequencies from lowest to highest. Thus if you want to have the numbers and names in your coordination (and monitoring) correspond to the units in the rack, then the systems need to be racked as described here. (For example, "IEM Mix-1" is the top-left channel in the rack, joined by "IEM Mix-2" at top-right, and the coordination looks like 516.125 – IEM Mix-1, 518.525 – IEM Mix-2, and so on.)

Next time, I'll detail practices for keeping track of hardware (handhelds, belt packs and more) on busy shows, along with several other additional facets of wireless that go beyond frequency coordination. LSI

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

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Chris Young and his band performing on the current concert tour.

THE RIGHT BLEND

Two schools of audio technology for Chris Young on tour. *by Sam McLean*

THE WAVE OF YOUNGER country artists capable of consistently topping the charts while packing sheds and theatres continues to roll along, exemplified by singer/songwriter Chris Young. Having just turned 30 this year, though, he's actually been in the public consciousness for almost a decade after winning a Nashville Star singing competition on USA Network in 2006, followed by a nomination as top new male vocalist in 2007 by the Academy of Country Music.

In support of his fifth studio album, entitled I'm Comin' Over, Young is currently in the midst of a North American and European tour, where his full, big-bodied voice that's quite well-suited to the live realm is backed by a band made up of top players. Supporting their efforts are front of house engineer Gary Lewis and system tech Andrew Dowling driving a main system supplied by Sound Image Nashville that's led by Anya arrays from Eastern Acoustic Works (EAW).

Lewis, who's worked with Young over the past five years, notes, "I like to be on the edge of new things. The energy the Anya PA generates fits the show perfectly. The clarity of the vocals and instruments is great, and the coverage is unbelievable."

GOING FOR IT

On the return trip from the European leg of Young's tour earlier this year, Lewis was approached about utilizing Anya for the remaining tour dates in late 2015. With only 48 hours to make a decision, he found himself on a plane to EAW's facilities in Massachusetts for a demo.

"The company has an Anya rig installed at a local performing arts theatre so I was able to load in a virtual mix and get familiar with it," he notes. "At first, the visuals of a straight array kind of throw you off, but the sound is so impressive you quickly forget about that. Ultimately it was a decision between going out with what I knew or taking out something new that could be better. I decided to go for it."

It didn't hurt that Dowling would be onboard to work with the new rig. "He knows the ins and outs of the system like no one else. It's amazing what he can get it to do," Lewis states.

In narrower venues, Dowling has generally deployed single columns of six Anya modules flown left and right, while in arenas, he's opted for two longer columns per side to deliver additional coverage when needed. "It's a constantly changing game," he observes. "Some places don't have the space for two motors, so we use less, while more boxes means even higher resolution. Either way, the coverage is exceptional."

Dowling continues, "The other advantage of Anya modules is that when hung in a column, they're really a 70-degree box, which has worked out quite well. I can take it out a little wider, turn it on a little bit while staying off the walls better than usual, which really makes a difference. As far as headroom is concerned, we could be 'in your face' and then some and still not hurt the system."

A dozen EAW SB2001 subwoofers accompany both configurations, although Lewis states that the mains can generate so much low end that he's barely using the subs. "The only thing I have in the subs is one of my kick drum channels and one of my bass guitar channels, along with one more channel for an effect on some of the heavier songs to drive it in and then pull it back," he details.

RETURN TO FORM

Lewis has returned to an analog console for the tour, a Midas XL4: "I cut my teeth on analog and made the switch to digital along with almost everyone else, but when given the chance, I chose analog. It just sounds so warm and reproduces exactly what I'm getting from stage. I love it."

In addition to the XL4, he carries two racks of outboard gear that's utilized judiciously. Drawmer gates are applied for kick, snare, and toms, joined by a smidgen of Anthony DeMaria Labs (ADL) compression for tightening. Yamaha SPX digital effects units are also applied to toms (SPX990) and snare (SPX2000) to provide a bit of distinction.

A Manley compressor is dedicated to steel guitar, with several tube compressors on hand to do spot duty where needed. "I like to do a very light compression on the guitars – 2:1 or 3:1 – just to get the tubes working," he says. "Most of the warmth comes from the preamps on the console, so it's more like 'color' with a little warmth."

Dynamics are generated largely by the musicians, who Lewis characterizes as "exceptional." There's some additional compression on acoustic instruments and keyboards, used more for boosting certain channels -- putting them out front --rather than compressing. Subtle vocal tailoring is done with a Manley VoxBox.

ADDITIONAL ASPECTS

An Avid VENUE SC48 digital console is positioned to the right of the XL4 at front of house for every show. Outfitted with redundant snakes, it hosts Lewis' live recording platform as well as serving as the back-up to the house desk. "It's nice because if something did happen with the XL4, I could jump right over to the SC48," he notes. "My show is right there."

Dowling is disciplined when it comes analog to digital conversion in the signal chain, something he's even more focused on with this tour due to the analog console. "My rule with A/D conversion points is simple: as few as possible. So with Gary on the XL4, as soon as signal leaves his desk, we make the conversion to digital and it stays digital through the chain," he says. The main system's Lake LM44 processor does the conversion to Dante digital audio, with signal maintaining that status through to the loudspeakers.

Racks of Shure wireless microphone systems travel with the tour, outfitted with SM58 capsules that are favored by both Young and his FOH engineer. "The gain-before-feedback of the 58 is really exceptional," Lewis offers. "Chris likes to move around, and he spends a good amount of time in front of the PA. It also sounds really good with his vocal characteristics and handles his technique really well."

WORKING WITH MORE

Monitor mixes are dispatched from a VENUE Profile digital console to Shure PSM 1000 personal monitoring systems. Additional stage fill for monitoring is delivered by four-box arrays flown left and right.

The last tour leg of 2015 went a quick six weeks, just enough time to try out the new PA before the holidays. Additional dates in the Midwest kicked off early in the new year, moving along to the East Coast in late February.

"When the opportunity came up to take this rig out, I was intrigued. I'd heard a lot about it and thought it would be a great opportunity to see how it performs with a country artist. None of us have been disappointed," Lewis concludes. "I feel like an analog board provides me with more to work with, and I've found that the same is true with this new PA. The tour is a perfect mix of old school and new school technology blended together for the very best audience experience." LSI

Above, system tech Andrew Dowling (left) and front of house engineer Gary Lewis at the Midas XL4 analog console during a show. Below, a look at one of the six-module EAW Anya arrays deployed at a theatre.

***Sam McLean** is a long-time writer working in pro audio, based in the U.S.*

HARMONIC PURSUITS

A variety of approaches for treating vocals in the mix.

by Chris Huff

No place is an engineer's talent more tested than with vocals. A singer's range, frequency characteristics, and vocal style all contribute to the specific sound and how it's treated in a mix.

Song arrangement is another influence. Anything's possible, whether it's a lead with backing vocals, multiple leads taking turns, guest vocalists, duets, and many other combinations, and this interplay is another factor in mix decisions. With so many variables and possibilities, here are a handful of strategies that can help in tackling whatever happens on stage.

OUT OF RANGE

Tom Jones once said in an interview that he was offered songs no one else could sing because of his range. He explained that most pop artists didn't have the range to sing those particular songs, and when presented to operatic singers, they made the songs sound, well, "operatic."

From personal experience, we know that vocalists will sing out of their range, but it's still up to us to make them sound good. After all, they – and the audience – expect it. A key is recognizing EQ work done, when they sing in-range, is nullified the moment they sing out of range as their vocal qualities change. That bump at 1.5 kHz might need to become a tight 6 dB cut. In other words, it's time for a remix.

View these mixes as an entirely different singer, where only the gain stays the same. Reset the high-pass filter (HPF) and then go after the harsh frequencies. Singing outside the range tends to emphasize frequencies from 500 Hz to 2000 Hz (in general), depending if the particular vocalist is above or below his/her range. Also, re-evaluate presence (1.5 kHz to 8 kHz) and clarity (2 kHz to 9 kHz). The higher the pitch, the more these frequencies require tuning.

HIGH-FIVE THE HIGH-PASS

The aforementioned HPF, a staple tool in mixing vocals, should be considered with respect to the size of the band. For larger groups, let the low-end instruments own the low end, and roll off the lows from the vocals. This eliminates unhelpful vocal lows as well as stage noise that seeps into the microphone.

With smaller groups, it can help to turn off the HPF. For example, with an act made up of a guitarist and a singer, this will provide the "space" to allow the lows to be used in the mix for a "warmer," more natural sound.

Fixed-point HPFs in consoles tend to be focused on the 80 Hz to 100 Hz mark, but variable HPFs can run much higher – and they should be if the situation requires it. I typically run my vocal filters in the 140 Hz to 160 Hz range, and go as high as 180 Hz depending on the singer. Also note that the frequency properties of microphones can attenuate lower frequencies, i.e., the Shure SM58 rolls off the lows below the 120 Hz region.

BACK ME UP

James Taylor has had some great backing singers over the years, including long-timers Arnold McCuller and David Lasley. Both have distinct voices that could easily be lead vocals. However, through honing their craft and harmonizing together, they create a beautiful combined sound, which makes the engineer's work a bit easier in keeping Taylor's own distinct vocal on top.

Mixing backing vocals starts with considering them as support players, differentiated from the lead (or leads). First there needs to be some all-purpose clean-up, eliminating the bad in general, and usually, rolling off the lows. Next, blend them together with an overall goal of a unified sound. The focus is reducing the uniqueness of each voice. While the lead vocal needs to be clearly understood, backing vocals don't often require that clarity because they're in a support role.

Mixing backing vocals starts with considering them as support players.

The song arrangement defines their volume in relationship to the lead. They can be alongside the lead during a chorus or playing in the pocket, just like a supporting instrument. Effects can also be used to add distinction. Applying more reverb to backing vocals can push them farther back in the mix, which at the same time can give the lead vocal more distinction.

DUELING DUETS

As mix engineers, we aren't always blessed with harmonious singers. The problem can (again) be in the song arrangement and/or when the voices share defining characteristics in the same frequency range. When faced with this, I've had good success in emphasizing the voice that's slightly more central to the arrangement to add definition while also lightly softening the other voice.

When the vocalists share a song evenly, use the distinctness of the better sounding vocalist to define the song, again while cutting a few dB in the same range for the other so they aren't battling for the same space. Optionally, both can be smoothed out and softened – it depends on factors unique to the situation.

CAPTAIN FANTASTIC

In his book *Mixing Secrets for the Small Studio*, Mike Senior explains how the "fantastic-ness" of a song needs to grow as the song moves along, verse to chorus, and so on. The first verse has energy and then the chorus kicks in with a bit more. Next, the mix drops back a little until the next verse all the while building in intensity.

Fantastic-ness can grow during the song, but it can also ebb and flow throughout. For example, "What Do You Mean," a recent song by Justin Bieber (hey, I've got a teenage daughter), exhibits ebb and flow quite well. Instruments are layered in over the first 30 seconds, and then at the 46-second mark, the mix drops back to the way it started. This song has three patterns regarding instrument arrangement thst alter slightly throughout the song, eventually propelling the song to the 2:18 mark where it's more upbeat than ever.

Mixing live, one hopes for a band that understands good arrangements and changing song dynamics. When that doesn't happen, there are options to create movement and fantastic-ness, and vocal mixing plays a large part in the process. Song movement can be manipulated, on a basic level, with vocal channels, mix groups, and the master fader. The groups, assigned for things such as guitars, keyboards, drums, percussion, and vocals, allow easy intensity changes in the song.

The vocal channels enable changing up how the lead and backing vocals interact. The second verse can have a softer lead which propels into a stronger chorus. For the final chorus, the backing vocals might be as loud as the lead. It's for this very reason my right hand usually stays over the vocal channels.

The master volume enables the overall song volume to build – a great way to end a set. Try this by starting that last song at the same volume spot as the other songs but bump it a little at the second chorus and alter it throughout the song. By the end of the song, subtle master fader boosts can result in a more energized audience.

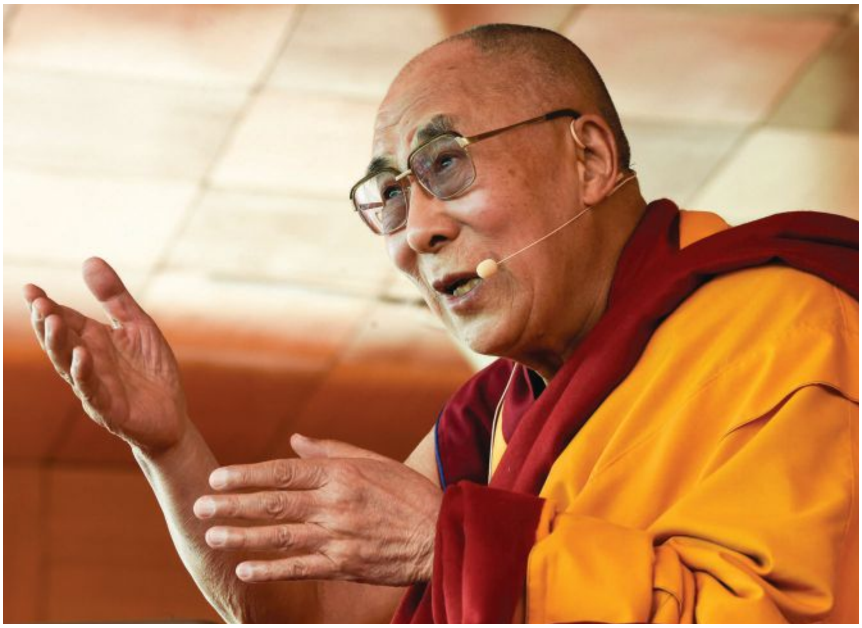
Vocals make or break a song. Use these strategies to tackle the variety of ways vocals are used in music. LSI

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

MINIATURE ADVANTAGES

The design and application of headworn and lavalier microphones.

by Bruce Bartlett



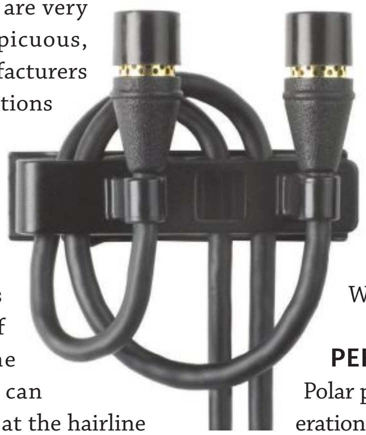
The Dalai Lama hosting a prayer meeting, outfitted with a DPA d:fine cardioid headset microphone.

Headworn and bodyworn microphones serve many valuable roles in live sound reinforcement. The two most common are headset/earset and lavalier designs, both allowing performers/presenters to roam freely around the stage/platform, hands free, with impressive sound quality and good gain-before-feedback. Nearly all are condenser designs; some are wired and some are wireless.

The headset/earset category is defined by light weight and comfort. Earsets have a mount that wraps around one ear, while headsets further stability by using an adjustable band that wraps around the back of the user's head that's attached to mounts that wrap around both ears. Both types extend a slender boom that places the mic element at the side or in front of the mouth for exceptional gain-before feedback. The booms are very thin and inconspicuous, with many manufacturers also providing options to match the skin tone of users.

Lavaliers usually clip to the shirt, tie or lapel about 8 inches under the chin of the user, and the smallest models can also be concealed at the hairline (common in theatrical productions). They're a good choice for users who are uncomfortable wearing a headworn mic – just keep in mind that gain-before-feedback and isolation will both suffer a bit. For critical applications, a dual lavalier approach provides redundancy. Some are supplied with special clips that hold two mics, or it's available as an option.

A singer/guitarist once told me that he worked on a very small stage with no space for a mic stand. He tried an earset mic, and it sounded natural and provided good isolation from his guitar. But the feel of the mount around his ear was too disturbing for him, so he switched to a lavalier. Although it picked up a fair amount of his guitar, he was satisfied with the sound. Whatever works!



Left: Dual Shure MX150 lavs on a single clip.
Right: Audio-Technica AT831b.

PERFORMANCE PARAMETERS

Polar pattern is an important consideration, particularly with lavalier mics. Omnidirectional patterns allow greater head movement without the user getting off-mic, and are less susceptible to mechanical and breath noise. Directional (cardioid and supercardioid) patterns have higher gain-before-feedback and pick up less background sound (better isolation). However, note that a directional lav on the chest often has an intermittently colored tone quality, while an omni tends to sound more natural.

All types have a rising high-frequency response, which compensates for the mic being off-axis to the mouth. Since highs radiate straight out of the mouth, but not so much to the side or below, the sound picked up can be dark or muffled unless the mic boosts the highs to compensate.

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At the same time, that high-end peak can cause high-frequency feedback with floor monitors – something to watch out for.

EXAMPLES

Let's have a look at specific models to get an idea of what's available. The **Shure MX150** is a subminiature lavalier that's available with a choice of cardioid or omni capsule, as well as XLR or TQG (TA4F) connectors. The XLR version includes an in-line preamp that's actually an impedance converter; it converts the unbalanced medium-Z capsule signal to a low-Z balanced output on an XLR connector. A multi-position tie-clip reduces cable noise by routing the cable in loops, and can be mounted horizontally or vertically. The same clip can accommodate single or dual mics

The **Audio-Technica AT831b** is a directional lavalier marked by reduced feedback and suppression of background noise. It's designed for voice and also works well as an instrument mic, especially for pickup of acoustic guitar with included guitar adapter. An integral 80 Hz high-pass filter provides easy switching from a flat frequency response to a low-end roll-off.

The **Countryman E6 Earset** puts the mic element at the side of the mouth. It's one of the company's most popular models, offering selectable omni or directional response. The capsule and boom, available in four colors, are so tiny that they're nearly invisible. Other features include a detachable cable and adjustable frequency response by changing protective caps.

The **Point Source Audio CO-8WS** is an omni earset that includes the

Countryman E6 Earset.

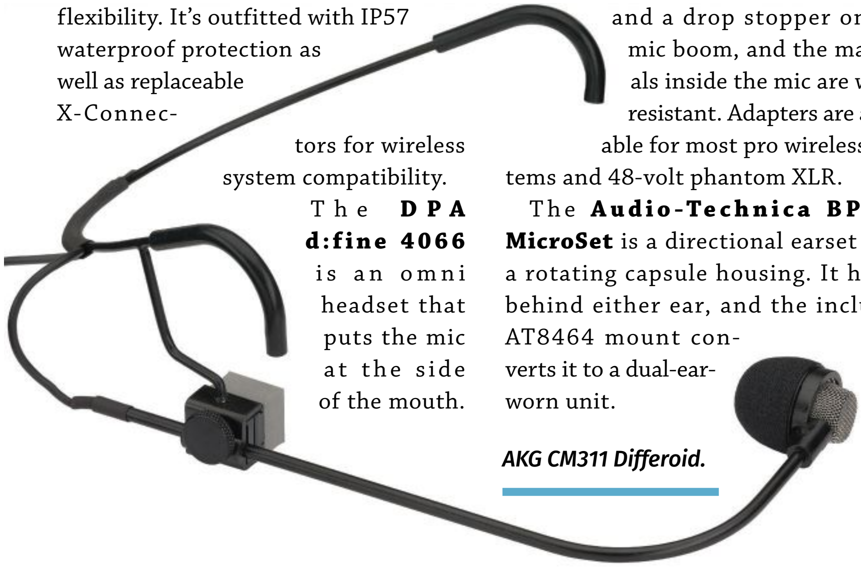
Point Source Audio CO-8WS.

DPA d:fine 4066.

company's "Unbreakable Boom" that is bendable to 360 degrees, making it highly flexible in terms of placement. It can be worn over either the left or right ear, with a dual earset option enhancing flexibility. It's outfitted with IP57 waterproof protection as well as replaceable X-Connectors for wireless system compatibility.

The **DPA d:fine 4066** is an omni headset that puts the mic at the side of the mouth. (A cardioid version is also available.) It has a detachable boom that can be positioned at either left or right. Available in three sizes and colors, it also has a double-vent protection system and a drop stopper on the mic boom, and the materials inside the mic are water resistant. Adapters are available for most pro wireless systems and 48-volt phantom XLR.

The **Audio-Technica BP894 MicroSet** is a directional earset with a rotating capsule housing. It hooks behind either ear, and the included AT8464 mount converts it to a dual-ear-worn unit.



AKG CM311 Differoid.

The **AKG CM311** is a directional headset designed for touring applications. The proprietary Differoid (differential cardioid) design cancels sound from floor monitors, resulting in high gain-before-feedback and solid isolation. It's available in different versions, such as the CM311 L with a 3-pin mini XLR connecter that fits all AKG wireless belt pack transmitters.

And this is just the tip of the iceberg when it comes to choices in miniature headworn and lavalier microphones. More than a dozen manufacturers offer a variety of types, and at a wide range of price points. LSI

***Bruce Bartlett** is a microphone engineer (www.bartlettaudio.com), a live sound and recording engineer, and an audio journalist. His latest books are Practical Recording Techniques (6th Edition) and Recording Music On Location 2nd edition.*

INTERESTING TIMES

On working for a support band on tour. *by Chez Stock*

While continuing the journey to find my niche in this industry, I'm being exposed to many new things. One of which is booking a front of house gig with a band that is supporting a well-known band doing underplay shows around the U.S. Working with a support band is new to me – previously I've only worked with the headliner, mixing or as a system tech. I took this gig with a whole slew of assumptions about how we, as a support band, would be treated, and you know what they say about assuming things…

Initially, I was super excited. The prospect of mixing FOH with another, more seasoned (A-list if you will) engineer on the headliner's crew filled my head with ideas of chatting about compression ratios and multiband compression and FX processing. I was to share the beautiful DiGiCo SD10, a console that I've toured with in the past as a monitor tech for Queens of the Stone Age. I was excited to get some time on a great console, with the consistency of at least traveling with our own mics and the same board at every show.

HELLO MR. MURPHY

What could possibly go wrong? Well, seemingly everything, and immediately so. The hardest part for me was not being able to tech my own gear. For the very first show, the headliner's FOH had graciously created a file for me, with inputs labeled and FX built in. Amazing. I was so stoked!

Unfortunately, as things tend to go as a support band, our allotted sound check time came, and we weren't allowed on deck. Of course, that meant the console was not mine to fiddle with either. So we waited and waited, and then waited some more. I've played this game before, except on the other side. The headliner needs more time to nail down the IEM mixes or to add a song, requiring extra rehearsal time, etc.

But hey, it's cool. I knew the PA (a beautiful d&b audiotechnik J-Series rig), and I knew the console. Even a half-hearted "throw and go" could sound great with those two components on the back end.

Finally, with about a half hour to spare before doors opened, I was given the keys to the rig. I quickly tuned it, which was just like old times with the d&b system, and the stage was pinned and ready to go. I called for the kick drum over talkback, and this was when the fun began.

"Kick drum please." Silence. Open up the preamp, throw on headphones, nothing. Check phantom power, nothing. I turn to the headliner's engineer, who is also the FOH tech, and ask him if the multi has been switched over because I'm not seeing/hearing signal. He calls up to his tech on stage, who confirms that we are in fact in the appropriate stage rack and are fully patched. Odd. The headliner's engineer steps over and starts making some patching changes.

We get the first eight channels sorted; I have almost all of my drum kit coming up. Time is slipping by, and I'm now in line check mode. Confirm that the correct audio is popping up on the right channel and move on, since we barely have any time left before doors.

Then we move onto channel nine. Nope, no signal. WTF is happening? Stay calm. I look to the headliner's engineer and ask if he can have the stage tech please confirm the patch is correct because my overhead is definitely not coming up on channel nine. After many minutes and many, many soft patches, it becomes apparent that the stage rack is patched completely wrong and in addition to the mispatches, there's also a bad card! Yikes.

POWER OF PERSEVERANCE

Somehow, we managed to get all of my inputs to show up on the console where they were supposed to, and I had about 30 seconds of a song before doors opened. Did I mention that this was my very first show with this band? I'd sat in on rehearsals, so I knew the sources well but had never mixed them before tonight. Needless to say, this was not my favorite way to start a tour.

Throw and goes can be fun, but not when it's a matter of gross mispatches causing my sound check to be delayed. But I stayed at FOH after doors opened and did some very basic console setup. Turned on compressors, set ratios with the thresholds high. Put high-pass filters on channels where it was appropriate and made sure the EQ was at least turned on. I let management know, very politely, that I would prefer them to give me at least half of the first song before they started giving me "mix notes."

A few minutes later, our show starts, and I'm flying around the console, tightening gates and lowering compression thresholds. I felt pretty good about the result considering the, uh, *chaotic* nature of the sound check. My band records every show and gives me mix notes on how to fine tune things for future shows, and thankfully, everyone was very happy with the sound of our first show. Management only had a few suggestions throughout the show, a little boost of this, a little cut of that. All in all, success!

I let management know, very politely, that I would prefer them to give me at least half of the first song before they started giving me "mix notes."

NOW YOU TELL ME...

As it turns out, I learned at the next show that the second stage rack for our patch had been flipped wheels to the sky on top of the primary stage rack. So the headliner's tech had patched my inputs as if the rack was right side up, meaning not only were my inputs coming up backward in banks of eight, they were also coming up in reverse. So my 1-8 inputs were actually patched into 48-40.

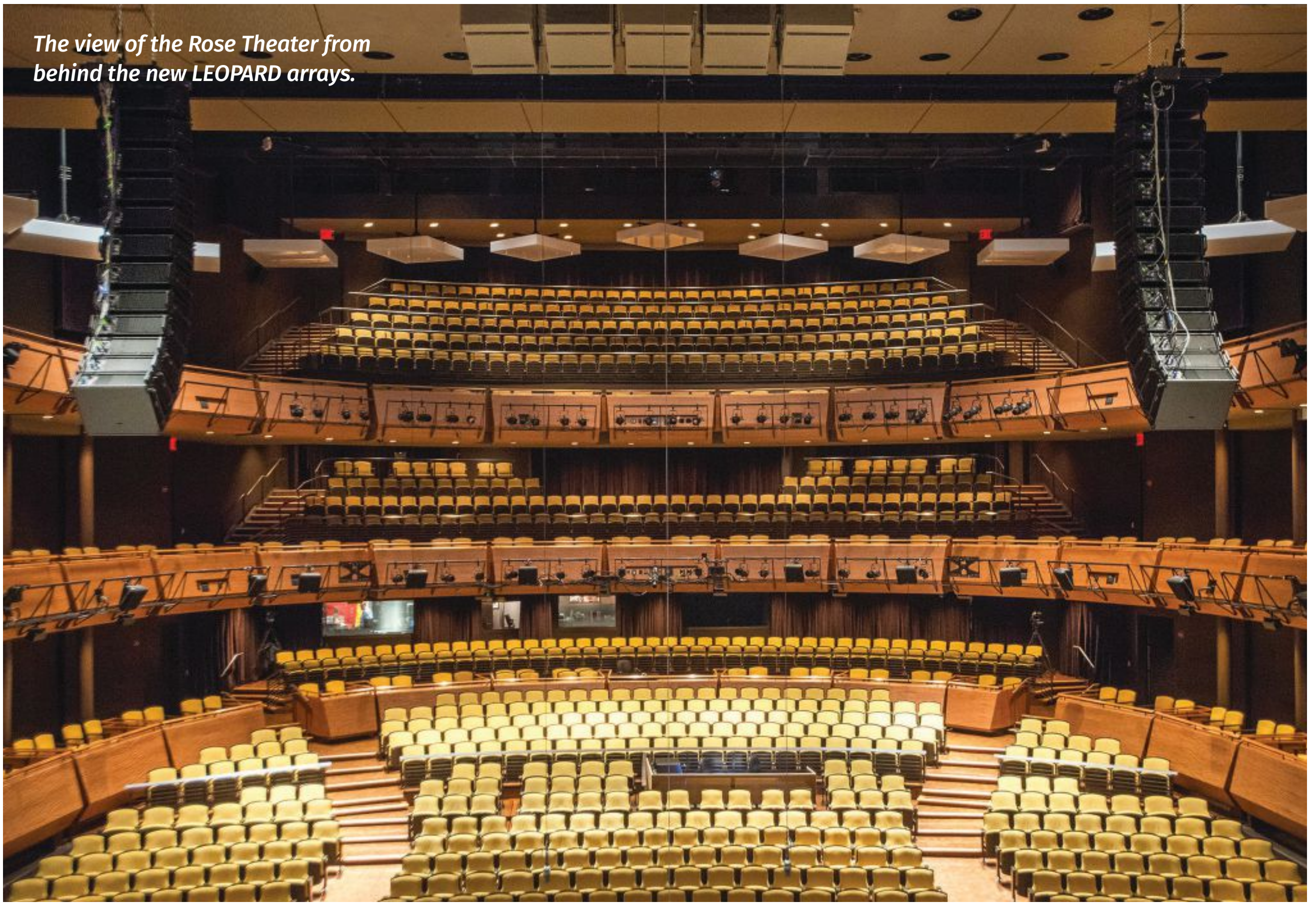
If I'd been allowed to access the stage rack or had time to go on stage myself, I think I probably would've realized this issue immediately when the inputs presented themselves so oddly at FOH. But since I was just the support engineer on a shared console, it wasn't my place to demand access to "their" gear.

Lesson learned. LSI

***Chez Stock** is FOH and tour manager for several independent artists, including Yuna, Dorothy, and Empress Of. Read more from her at SoundGirls.org.*

ARTFUL AUDIO

Loudspeakers in a range of recent live/performing arts venue installations. ***by Live Sound Staff***



The view of the Rose Theater from behind the new LEOPARD arrays.

PHOTO CREDIT: STEPHANIE BERGER

TRANSPARENCY FOR JAZZ AT LINCOLN CENTER IN NYC

The just-renovated Rose Theater at Jazz at Lincoln Center (JALC) in Manhattan has a new Meyer Sound LEOPARD linear system, part of a comprehensive upgrade that sees Meyer Sound loudspeakers implemented in venues throughout the JALC complex. The extensive sound system portion of the project was coordinated by Sound Associates of New York.

The refurbished system for the 1,233-capacity theater is designed to provide controlled, even coverage throughout the irregularly shaped room while meeting the needs of the Jazz at Lincoln Center Orchestra. It incorporates dual arrays of 12-each LEOPARD loudspeakers and five 900-LFC low-frequency control elements in a cardioid cluster over the stage. Other components include UPQ-1P, UPM-1P, UPJunior VariO, UP-4XP loudspeakers, M3D-Sub directional subwoofers, and Galileo Callisto processing.

"This is a big band with trumpets, trombones and saxophones, and that creates a considerable amount of unamplified acoustic energy coming off the stage," explains Doug Hosney, chief officer at Frederick P. Rose Hall. "What we needed to be able to do – as transparently as possible – was to lift instruments like bass and piano into that acoustic energy for balance, but not have it sound like those instruments were artificially amplified.

"The line arrays we had before were considerably larger and not particularly linear in response," he continues. "You had to drive them pretty hard before they would open up, so we wanted to replace them with arrays that operated cleanly at lower levels as well. That's where LEOPARD stepped up to the table and just blew away the competition."

DELIVERING A PROVEN L/C/R APPROACH AT CANNES

The Claude Debussy Theatre at the Palais des Festivals in Cannes, France (home of the Cannes Film Festival) has a new system utilizing recently introduced Adamson Systems S Series line arrays to serve a busy, diverse schedule of live events. The 1,068-capacity theatre, the second largest of five auditoriums in the venue, has been upgraded with a left-center-right loudspeaker configuration devised by system designer DV2 (Adamson's distributor in France) and installed by rental/integration company Dushow.

Left and right arrays each consist of 10 S10 enclosures with two S119 subwoofers positioned at the top of each hang, while the center cluster is made up of six S10 modules. Two additional E219 subwoofers are ground stacked. Four Adamson Point 12 compact 2-way loudspeakers supply front fill and out fill coverage. The approach is similar to another system deployed at the venue by DV2, which enjoys a long affiliation with the Palais des Festivals.

"This design has delivered exceptionally well for 15 years in the Auditorium Louis Lumière," explains Didier Dal Fitto, DV2 technical director. "As a result, the Palais sound crew decided to use the same concept for Debussy. Dushow installed two of the new S119 subwoofers at the top of the left and right arrays, which gives enough subwoofer pressure to cover conferences applications. A complement of two E219 stacked subwoofers and Point 12s are added for more solid music programs."

The Minnesota Orchestra performing beneath new L-Acoustics Kiva arrays.

Seven Lab.gruppen D120:4L amplifiers provide power to the main PA and subwoofers. A Dante digital network is the backbone of the system.

MODERNIZING ORCHESTRA HALL IN MINNEAPOLIS

A $52-million renovation of the 2,000-seat downtown Minneapolis home of the Grammy Award-winning Minnesota Orchestra is highlighted by a new sound reinforcement system headed by L-Acoustics loudspeakers supplied and installed by Allied Audio Services of Grove Heights, MN.

"The acoustics are fantastic for the orchestra, which is the primary focus," explains Joel Mooney, technical director of the orchestra. "But you also have to meet all the needs of modern music and productions. Technology has changed since the last time we remodeled the hall, so we took this opportunity to upgrade the sound system."

The loudspeaker contingent comprises a left-center-right configuration of Kiva array elements together with SB15m and SB18 subwoofers, all of which are powered by LA4X amplified controllers. "We have 14 Kivas in the center hang, which is primarily for speech and general classical reinforcement, and then we have 13 Kivas on the left and 13 on the right," Mooney explains.

The left and right hangs are both topped by a pair of SB15m subwoofers to provide flown low frequency extension. In addition, a pair of SB18 single 18-inch subs are positioned behind a fascia created during the remodel below the stage lip.

"We tried all the big names; the reason we chose L-Acoustics was because we felt it matched everybody's desires, both the people in the audience and the technicians who come through our hall. The brand meets all riders," Mooney adds.

New Adamson Systems S Series arrays and more at the Claude Debussy Theatre in Cannes.

Renkus-Heinz CFX-series arrays flanking the stage in ballroom 2 at the Grand Lagoon Ballroom.

INDIVIDUAL & UNITED SOUND IN MALAYSIA

Located just outside of the Malaysian capital of Kuala Lumpur, the Sunway Resort Hotel & Spa is part of a huge complex that includes the Grand Lagoon Ballroom, recently outfitted with a flexible new system utilizing Renkus-Heinz loudspeakers. The ballroom, which can accommodate up to 2,000 for a range of events, can be divided into three separate halls, with the new system needing to serve each sub-space individually and all three when united.

"The ceiling is at least 7 meters (23 feet)," explains Keith Tan of Mercoms Systems, which served as the system designer. "The walls are mostly made of wood frames with some soft cloth-wrapped padding in the middle and glass mirrors at the top, and the floor is carpeted. So although the ballroom has some glass surfaces, it also has absorptive surfaces, and there are no pillars. But the depth of the ballroom was a challenge. It was vital to be able to throw clear sound all the way to the rear while achieving relatively even coverage."

All three spaces are served by left-right pairs of arrays, each with four CFX-101LA modular point-source arrays. For ballroom 2 (the middle hall, which also has an extended stage), Mercoms Systems also chose two CFX15S (passive, 15-inch) subwoofers, while the other two halls each have a single CFX15S sub. All loudspeakers, including subs, were flown by installation firm Mega Teknik Jaya. A Biamp Tesira server provides loudspeaker management, including crossover settings and EQ, in addition to combining the systems when the partitions are open.

"With CFX-101LA and CFX-15S in all three sub-ballrooms, the system sounds consistently great with the partitions in place and when opened up into a single space," Tan notes. "It perfectly complements the layout of the ballroom."

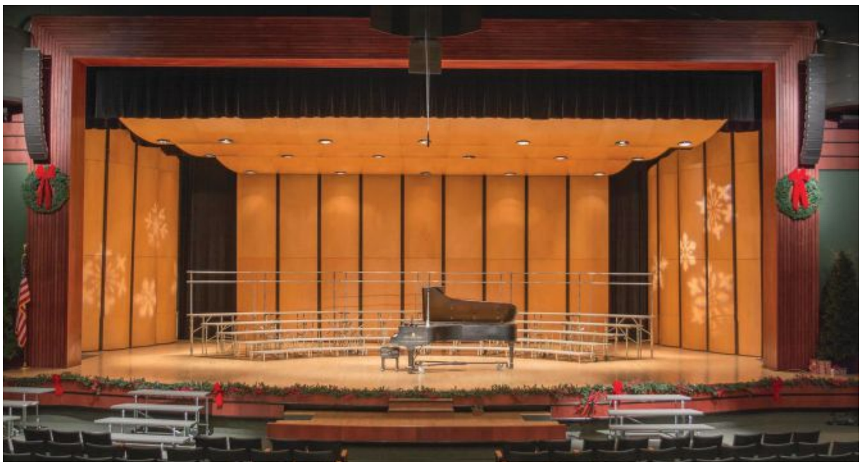
LOW-PROFILE FIDELITY FOR THE ARTS IN IDAHO

The 1,500-seat Swayne Auditorium at Northwest Nazarene University (NNU) in Nampa, ID, which hosts multiple chapels each week as well as university performing arts concerts, theatre department productions, and more, has a new system led by NEXO GEO M6 loudspeakers in a design formulated by Production Services International (PSI International), which has offices in Boise, Salt Lake, and Denver.

Noah Bard, operations manager for PSI International, notes that the old system had run its course and needed to be replaced, adding, "Aesthetically, NNU wanted the sound system to disappear as much as possible. The old center hung system was very much a part of the room. The change had been a part of the initial master plan for upgrades."

The new main loudspeaker system contingent consists of a stereo pair of GEO M6 arrays, 12 per side, each with nine M620 full-range modules joined by three M6B low/mid-frequency modules. Two NEXO LS18 subwoofers are adjacent to each array, concealed by scrims. Audio power is delivered by two NEXO NXAMP 4X1s for the mains and a NEXO NXAmp 4X4 for the subs, with all three amplifier-controllers outfitted with EtherSound networking cards.

PSI International also added a Lake processing card to the previously installed Yamaha M7CL-ES Console and an outboard 5045 Neve processor. "We use the Lake processor for additional system processing along with the base system processing provided by the NXAmps," Bard explains. "The 5045 processor was primarily installed for use during chapels for headset and podium use."



Discreet arrays made up of NEXO GEO M6 modules at left and right in the Swayne Auditorium.

The Hall supported by appropriately scaled Martin Audio MLA Mini arrays.

QUALITY AUDIO, HAPPY NEIGHBORS IN BROOKLYN

The Hall in Brooklyn, a live performance venue located next to MP Taverna that's one of several upscale Greek restaurants from chef Michael Psilakis in New York City, has a new Martin Audio MLA Mini system to both provide quality sound reinforcement and control noise overspill to neighboring residences. Canal Sound & Light (NYC) deployed the new system, which serves a narrow room that's less than 50 feet wide by 110 feet deep and topped by a ceiling that reaches just 11 feet.

"The challenge was to find a compact line array system that could produce superior audio and cut the sound off at a certain point so the client wouldn't get noise complaints from restaurant patrons and neighboring condos and homes," explains Jeffrey Kwan of Canal. "Because it provides that kind of control in a small format, the MLA Mini system perfectly matched the criteria for this project. We installed two Mini 4-enclosure arrays, one per side, with an MSX sub mounted underneath each side of the stage."

To cover the onstage needs of performers, an eclectic lineup of funk, rap, acoustic, neo-soul and rock bands, Canal added a monitoring system with two Martin Audio LE1200 and three LE1500 wedges driven by MA3.0 and MA2.0 power amps.

"One of the owners had an engineer working there as a consultant who actually requested a Martin Audio system, which worked out great for us," Kwan adds. "The system sounds absolutely amazing. We first did a demo in the room during construction and even then, just playing back regular tunes, they were blown away by the way it sounds."

UPGRADING A CLASSIC IN LOS ANGELES

The AV Installation division of 3G Productions (Los Angeles, Las Vegas) recently implemented a new system utilizing d&b audiotechnik loudspeakers at the legendary El Rey Theatre, a live venue located on LA's Miracle Mile that hosts concerts by a variety of artists. Built in 1936 as a single-screen movie theater, the El Rey was converted into a live music venue in 1994, offering a seating capacity of 770 seats with a small VIP balcony.

3G was commissioned to replace the venue's original sound system, which according to David Myers, director of AV Installations, was "over 20 years old, broke down a lot and had too many problems. They needed something that sounded better and was reliable, which we delivered."

The main system utilizes five d&b audiotechnik Q1 loudspeakers per side with a Q10 elements over center stage for front fill. Four d&b B2 subwoofers are ground-stacked in bunkers underneath the stage; all loudspeakers powered by d&b D12 power amplifiers. The system was installed in just three days, all the venue could afford to spare given its crowded schedule, and was used the day after completion.

El Rey house engineer Mark Rea states, "I've been a huge fan of d&b for years and it's great that we now have such a high quality system in the theater. Dave and his team at 3G designed a sound system that fits the acoustics of the room perfectly and is easy to operate for the different guest engineers that come through." LSI

New mains for an old classic, with d&b audiotechnik Q-series in place to support live performances at the El Rey Theatre.

WATCHING THE SIGNAL FLOW

25th ANNIVERSARY

In sound reinforcement systems, distribution is everything.

by Barry McKinnon

EDITOR'S NOTE: This fine article was featured in the very first issue of Live Sound International, published in January 1992. While understandably a bit dated, it still provides a wealth of valuable technical information on an important topic in sound reinforcement.

Signal distribution can get unwieldy at times. One of the worst situations is the slow evolution of a sound reinforcement system. The growing demands on the system require the addition of components until some critical point is reached where it just seems to get more and more awkward to hook everything up and make it work. Well take a look at the often talked about G'Zintas and G'Zouttas of a sound system (as in: it G'Zinta here and G'Zoutta that thing there).

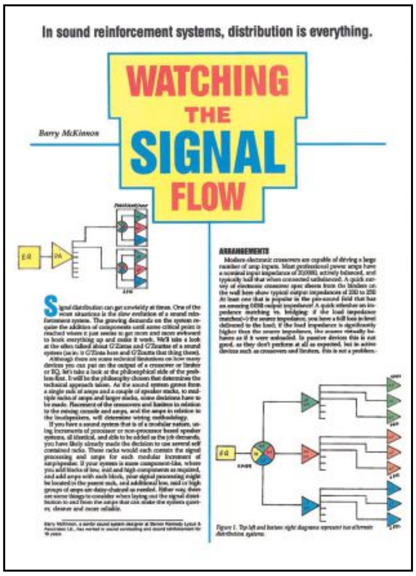
Although there are some technical limitations on how many devices you can put on the output of a crossover or limiter or EQ, let's take a look at the philosophical side of the problem first. It will be the philosophy chosen that determines the technical approach taken. As the sound system grows from a single rack of amplifiers and a couple of loudspeaker stacks to multiple racks of amps and larger stacks, some decisions have to be made. Placement of the crossovers and limiters in relation to the mixing console and amps, and the amps in relation to the loudspeakers, will determine wiring methodology.

If you have a sound system that is of a modular nature, using increments of processor or non-processor based loudspeaker systems, all identical, and able to be added as the job demands, you have likely already made the decision to use several self-contained racks. These racks would each contain the signal processing and amps for each modular increment of amp/speaker. If your system is more component-like, where you add blocks of low, mid and high components as required, and add amps with each block, your signal processing might be located in the parent rack, and additional low, mid or high groups of amps are daisy-chained as needed. Either way, there are some things to consider when laying out the signal distribution to and from the amps that can make the system quieter, cleaner and more reliable.

ARRANGEMENTS

Modern electronic crossovers are capable of driving a large number of amp inputs. Most professional power amps have a nominal input impedance of 20,000 ohms, actively balanced, and typically half that when connected unbalanced. A quick survey of electronic crossover spec sheets from the binders on the wall here show typical output impedances of 20 ohms to 25 ohms. At least one that is popular in the pro-sound field has an amazing 0.05-ohm output impedance! A quick refresher on impedance matching vs. bridging: if the load impedance matches (=) the source impedance, there is a 6 dB loss in level delivered to the load; if the load impedance is significantly higher than the source impedance, the source virtually behaves as if it were unloaded. In passive devices this is not good, as they don't perform at all as expected, but in active devices such as crossovers and limiters, this is not a problem.

The original logo.


In sound reinforcement systems, distribution is everything.

WATCHING THE SIGNAL FLOW

Barry McKinnon

How page 1 of this article looked in the first issue.

The electronic crossovers mentioned are rated to meet their performance specs driving 600-ohm loads, and with the output impedances shown, should be quite happy with that load. Let's just grab a calculator and see what that means. Let's see, an amp is 20,000 ohms, divide that by 600 ohms for the minimum load on the crossover. That's approximately 33 amp inputs in parallel to get down to the minimum rated load of 600 ohms. Or, looking at it another way, a single crossover output can drive more amps than you can afford. Does that mean you really want to drive all those amps from a single crossover output? Well no, not likely, and there are a number of reasons.

In either of the philosophical situations discussed earlier, it is necessary to send the same signal to a number of different places. In one case, a full range signal is being distributed to a number of identical amp racks to be divided up by the crossover/processor; in the other case, crossover outputs are being sent to additional amp blocks, as required. In either case, paralleling a large number of inputs on a single output increases the chances of noise pickup, and decreases the reliability of the signal source. In the case of a parallel network, any noise picked up on any branch of the network is connected at the source to all other branches. Similarly, each branch of the network is a weak link for the whole network. A short circuit on one cable will pull the whole network down, as the source busily drives all the signal into the short.

Note that it doesn't have to be a dead short to be a problem. If the short is 1,000 ohms, that is still adequately low, in comparison to all those 20,000-ohm inputs the source is driving, to send the signal where it does no good. If the impedance of the short is approaching the output impedance of the source, you will start to see current limiting and distortion, just in case you hadn't already noticed there was a problem. Daisy-chaining is a parallel network as well, unless the daisy-chain in/out is buffered with a stage of unity gain amplification. There is an additional headache with daisy-chained inputs: The whole chain can be jeopardized by a problem in a cable early in the chain. Try to sort that one out in a hurry! No fun at all.

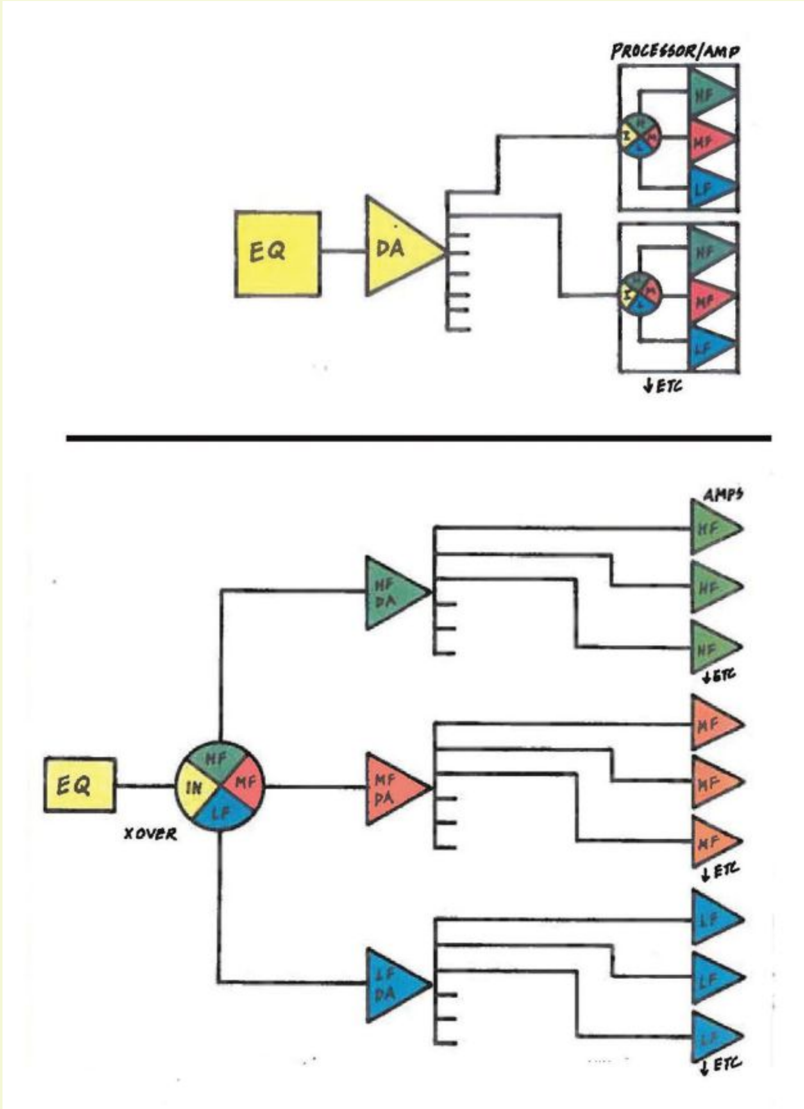


Figure 1 – Two diagrams that represent two alternate distribution systems.

The magic remedy that will lead to cleaner, quieter, more reliable systems is the distribution amp. By using a distribution amp (DA), you can reduce problems in distributing signals. It is easy to see how you can improve reliability. A DA has a number of individually buffered balanced amp outputs that are isolated from each other, preventing a short circuit or an induced noise (as opposed to a grounding problem related noise, which can still show up everywhere) from affecting the other sends. A problem developed in one signal line will not affect the whole system. This reduces the opportunity for problems and speeds troubleshooting as well. See the two diagrams of possible distribution schemes in **Figure 1**.

GAIN STAGING

Another opportunity to improve signal quality presents itself when using a DA. Because many DAs have high level capabilities, it is possible to elevate the levels used to distribute signals. Signal-to-noise ratio is often limited by the noise floor present in the signal distribution of the system, as opposed to the amplification or gain stages. The signal may have been fairly quiet when it left the console, but cables passing to and from amps might drape over an AC cable, sit too close to power transformers, run by a lighting cable, whatever… your noise floor is often limited at the site. I have seen some amps that are their own worst enemy for generating noise in the system, in that the power transformers have stray fields that induce noise in other amps and low level electronics placed above and below them. The way to claim some signal-to-noise back is to raise the signal level in the distribution chain.

Raising the signal level you send through the noise jungle by 6 dB to 10 dB provides a 6 dB to 10 dB S/N (signal-to-noise) ratio improvement, assuming the noise floor is limited by conditions after the DA. This will require choosing a DA that has the kind of output level capabilities needed, and installing input pads on the receiving devices to drop the level back down to a level that can be dealt with. Some pro amps have a switch to change the gain, allowing an input signal of over 2 volts (V) to be used to reach full power, instead of 0.775V. Many amps already have input sensitivities of 1.75V to 1.95V. This is a handy feature for building a gain structure that reduces noise.

If you are sending an elevated signal of +10 dB more than required to achieve full output of the system, it may require a DA with +30 dB output capability to leave some headroom at operating level. It will also require a 10 dB pad built for the input of each receiving device. If the receiving device has an input level control as the first thing in the chain, you could conceivably just "turn it down," but that will likely leave the control set at a tick

If nothing else, a DA makes it easier to hook up all of the signal leads you require to distribute the signals.

above off, reducing the control range available. If the first stage is active, an input attenuator is an absolute necessity.

If the system is intended to work in theatrical venues or locations with low ambient noise (less than NC20, or noise criteria curve of 20), you may find it absolutely necessary to use this kind of edge to optimize gain structure. A system that sounds quiet in the back of a shop with an ambient noise rating of NC50 may still sound like a noise generator in a room with an NC15 rating, such as a concert hall.

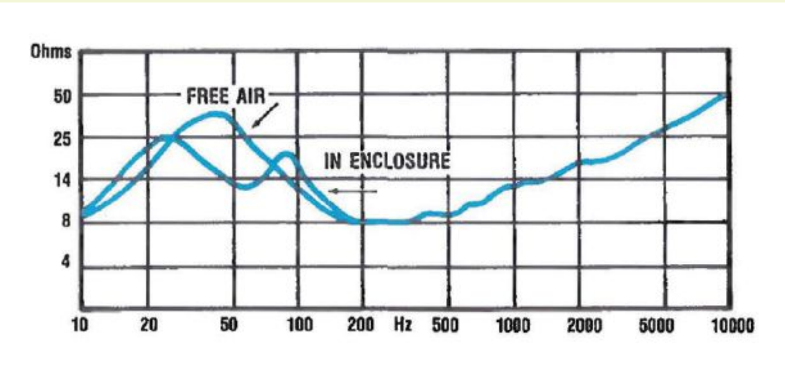


Figure 2 – Impedance curves of a 15-inch loudspeaker in free air and mounted in a tuned enclosure.

A DA also gives you the flexibility to leave the amp volume controls wide open and adjust the drive level with the DA outputs. This is handy from a goof-proofing point of view, but without some passive pads at the amp input or some gain selection capability, this rarely results in good quiet gain structure. If nothing else, a DA makes it easier to hook up all of the signal leads you require to distribute the signals. Good quality DAs with lots of drive level are not inexpensive, but the headaches they eliminate make them quite cost effective.

DOWNSTREAM

We've covered some G'Zinta concerns, now let's look at the G'Zouttas. There has been no shortage of material written on loudspeaker cables in the past few years. It would be nice to use liquid hydrogen-filled copper pipe, or some room-temperature super-conducting cable to connect amps to loudspeakers, but the current reality is that cable has resistance. The important thing to do is optimize the equipment cost/performance ratio in a sound system, where using loudspeaker cable that costs a couple of hundred dollars per foot is not a viable engineering option, no matter how good it could sound.

Let's take a quick look at loudspeaker impedance, because that is the load we will be dealing with. Refer to **Figure 2**, which shows the impedance of a 15-inch low-frequency device, in free air and in a ported enclosure. The minimum impedance point in this case is 8 ohms, with the impedance rising on both sides. The rise with increasing frequency is reactive, that is, resulting from the inductance of the voice coil. The rise with decreasing frequency is again reactive, this time from the action of the cone mass, suspension and air in the box, which is why it changes with the type of enclosure it is operating in. The minimum impedance point is typically where the resistive component of the voice coil is the most important player.

If you measure a loudspeaker with a dc VOM (you know, an ohmmeter) you will get a reading that is close to this minimum impedance point. This also happens to be the point where the biggest rise in impedance happens when you run your loudspeakers hard for a time, and they heat up, (a.k.a., power compression), as it is the resistive component that changes the most in warm wire (but that's another article). It is important to know what load the amp will really see when the system is operating, and this is the load. The amp's minimum safe load impedance is a concern. When paralleling loudspeakers, you should know what the net resulting impedance will really be.

The maximum phase angle the amp can tolerate can be important, too. The phase angle is an indication of the lag time between the current and voltage when putting energy into the loudspeaker system, and also an indication of the amount of energy coming back from the loudspeaker system. The reactive part of loudspeaker impedance is an indication of energy storage in the loudspeaker. Luckily, loudspeakers aren't terribly good at it, rather like the batteries in my laptop, and the energy comes back almost right away. You probably know that there is resistance to moving a cone when connected to an amp, and you may have noticed that the voltmeter will swing back and forth when you push on the cone while measuring impedance. No surprise here, as a loudspeaker is just another form of motor or generator – they all involve a magnet assembly and a coil of wire, and as such create a voltage when the coil and magnet are moved relative to each other.

The amp has to absorb this energy-return created by the loudspeaker cone returning to its rest position, while trying to put new energy into the loudspeaker. Some amps take exception to this, and although most professional amp designs are by now quite well behaved, some amps from years gone by will generate spectacular fireworks if presented with reactive loads.

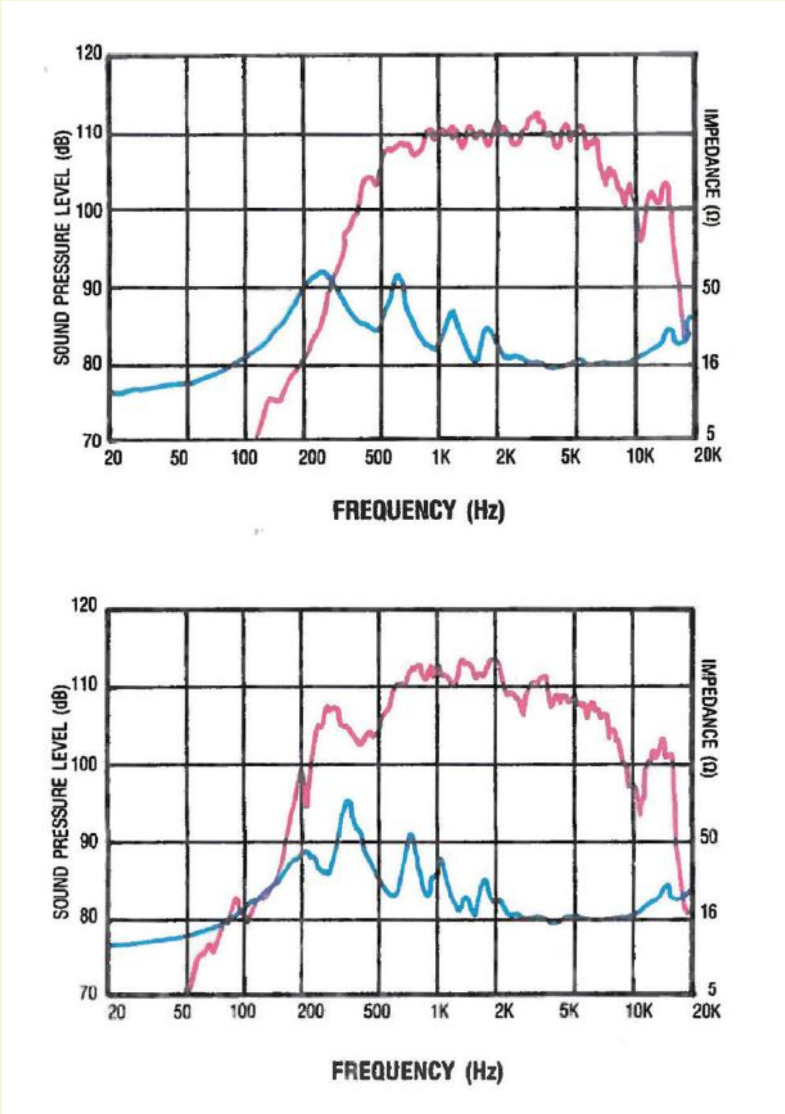


Figure 3 – Impedance curves and frequency responses of the same compression driver of two different horns.

High-frequency devices are sensitive to the horns they are mounted on, and the impedance reflects this. For the most part, the differences are less dramatic than LF devices. See **Figure 3** for the impedance curves and frequency responses of the same driver mounted on two different horns. Note the behavior out of band in the 200 Hz region. This is important when designing passively crossed-over loudspeaker systems, but thankfully much less so in active systems.

The point of this speedy review of loudspeaker impedance is that impedance is a dynamic thing and definitely frequency variable. It is important that you know what your amp will be seeing, and to this end it is important that you measure the real net load that the amp will see, preferably at the amp end of the cable, to take everything into account. Such devices as the Techron TEF or MLSSA analyzer can show the phase response of the impedance curves, and display it neatly on the Nyquist diagram, allowing a quick assessment of the reactive character of the load. Measure well beyond the expected operating bandwidth. Strange impedance behavior out of band can still be a problem if there are any spurious signals that happen to infiltrate your signal chain. Keep in mind, too, that crossovers are not "off" out of band; they are attenuated 18-, 24- or 48 dB/octave, but large signals out of band can still put amplifiers and loudspeakers at risk in nasty situations.

ADDITIONAL EFFECTS

There are other concerns as well. Cable capacitance is not generally a big problem, but that doesn't mean it can be ignored. I saw one installation that used cables provided by the original manufacturer of processor-type sound reinforcement loudspeaker systems. The cables were nice, flexible, small diameter,

It is important to keep cable runs short, not just to avoid wasting power, but to connect some of the amp's damping factor to the loudspeakers.

4-conductor designs that would put the amps into protection without cause. If the cables were long enough, it wouldn't even need a signal to create problems – just turning up the amp volume control was enough to do it. In this case it turned out the specific amp model had an inherent design flaw in grounding that caused greater instability, and the manufacturer was happy to repair or update the units under warranty, but problems such as this were initially caused by high capacitance cable. Excessive capacitive reactance can be seen when doing impedance measurements.

We've already mentioned low-resistance cables, and understand the desire not to waste amp power in heating loudspeaker cables. There are two problems that work against us when dealing with cable impedance. The power lost in the cable is one factor, but another often overlooked factor is the reduced output from the amp seeing a higher impedance load.

Example 1
Assume a 1,200-watt (W) amp channel feeding an 8-ohm load using 12AWG cable that's 50 feet long. Because 2-conductor 12AWG cable represents approximately 0.32 ohms/100 feet, the total load at the amp is 8.16 ohms. Amp output at 8.16 ohms is approximately 1,175W, and the power loss per foot is 0.48W. The total power loss over the 50-foot cable run is 24W, or 0.01 dB. Total power to the load is approximately 1,151W; 49W went away.

Example 2
Assume a 300W amp feeding a 4-ohm load using 14AWG cable that's 50 feet long. Two conductor 14AWG cable represents approximately 0.52 ohms/100 feet. The total load at the amp is 4.26 ohms, meaning the amp output is approximately 282W, with a power loss per foot of 0.34W. The total power loss over the 50-foot cable is 17W, or 0.27 dB. Total power to the load is approximately 265W; 35W disappeared.

Altec Lansing has a really slick program for calculating these numbers called AMPTOOLS, and it greatly speeds up generating similar data. It also has a dynamite attenuator calculator for designing DA pads.

The power loss, even in respectably large cable, can be significant. In the case of daisy-chained parallel loudspeaker hookups, there can be some noticeable differences between the power delivered to the first and last loudspeaker in a chain connected by cables with significant resistance.

DISTRO DETAILS

Weak links in loudspeaker distribution should be avoided, and there are so many places to build them in, such as the connection to the terminals, the distribution bus and the connector contacts. Measurements of load impedance done at the addition of each stage of the distribution chain can show any negative effects. The important thing to remember in building a loudspeaker distribution setup is the use of low resistance terminations and connections. Using electrical hardware designed for high current ac hookup is not out of the question when dealing with high power audio circuitry.

It is important to keep cable runs short, not just to avoid wasting power, but to connect some of the amp's damping factor to the loudspeakers. Damping factor is the ratio of the load impedance divided by the amp's actual output impedance, not to be confused by minimum load impedance. The low output impedance of an amp is what stops (damps) the loudspeaker from moving when the signal is finished. (Try moving a disconnected loudspeaker cone. Then put a piece of wire across the loudspeaker's terminals. That's the nature of damping factor.)

Large amps often have specs that rate damping factors at 1,000, which indicates that the output impedance of the amp is in the range of 0.004 ohms. Tight and clean low frequency reproduction requires a damping factor of 50 or more at the loudspeaker terminals. To achieve that with a 4-ohm loudspeaker requires a total cable resistance in between the amp and the loudspeaker of less than 0.08 ohms. By using the largest wire practical to connect amp outputs to loudspeaker distribution panels, and

CABLE RESISTANCE

2-cond. 14AWG cable approx. 0.52Ω/100 feet
2-cond. 12AWG cable approx. 0.32Ω/100 feet
2-cond. 10AWG cable approx. 0.2Ω/100 feet
2-cond. 8AWG cable approx. 0.1Ω/100 feet

using really large buses where applicable, you can reduce the overall distribution impedance.

Remember that the resistance of the total distribution cable is always added in series with loads. Once in the circuit, it cannot be ignored. It gets in the way of outbound energy and the ability of the amp to put on the brakes in each loudspeaker. With amps routinely able to drive 2-ohm loads, and some that can drive 1-ohm loads, distribution impedances of even 0.1 ohm are becoming a large percentage of the network impedance.

SAFETY COUNTS

The current involved in high-level loudspeaker systems can be substantial. An amp spec sheet off the shelf here tells me that in bridged mono this amp is capable of delivering 2,800W into 4-ohm loads. That's an audio current of 26.5 amps. In a sound system with a good bit of compression and something like thrash metal for program content, it can probably deliver that at near RMS levels. Anyone who has been "buzzed" off a high-power loudspeaker line can directly appreciate the magnitude of audio output.

The matter of safety has to come into play. With the equivalent of almost 4 horsepower (1 hp = approximately 750W) dangling at the end of a loudspeaker cable, it pays to consider it to be a 4 hp lawnmower motor, complete with blade, and treat it with as much respect. Loudspeaker connectors should be capable of dealing with the currents required of them. A phone plug was fine for hooking up your old powered mixers, but serious connectors, such as the Neutrik Speakon and Cannon EP series, are called for in high-power situations. Current handling and short circuit prevention are key elements in connector choice.

Another factor to remember is cable maintenance. As conductors break, often near the connector, the cable resistance will rise. As we have already seen in systems using low load impedances, this will affect system performance. Corrosion of contacts will also play havoc with resistance. A wiping-type contact is a useful feature in a loudspeaker connector.

With a bit of planning and care, signal distribution can be dealt with effectively, providing improvements in signal-to-noise, distortion and reliability. LSI

Barry R. McKinnon *is a principal and a senior consultant at McSquared System Design Group in North Vancouver, BC, and has worked in sound contracting and sound reinforcement since 1976.*

THOROUGHLY MODERN

Power amplifiers for larger-scale applications.

by Live Sound Staff

Today's power amplifiers are far lighter and smaller than their predecessors, and they're now able to generate tremendous audio power from that reduced physical footprint. The primary advancement is efficiency. Greater output efficiency means less heat, and therefore, less weight (reduction and/or elimination of heavy heat sinks), as well as more audio power making it to the loudspeakers.

The most common amplifier topology, by far, is class D (and variations), which uses an on-off switching method for its transistors called Pulse Width Modulation. Because its output devices are either on or off, the efficiency of the amplifier is greatly increased, and this is done without jeopardizing the integrity of the audio waveform by switching positive and negative output transistors on and off many times per waveform cycle.

The method is analog but similar in theory to digital sampling where a 44.1 kHz sampling rate is used to accurately capture a 20 kHz signal. This rapid switching creates a square wave that is then low-pass filtered to recreate the audio waveform.

The addition of DSP into the package is a further enhancement, offering convenience, space and cost savings, operating efficiencies, performance advantages, and more. Many modern amplifiers also benefit from networked control protocols, making them easier than ever to configure and monitor.

These developments have been particularly beneficial to large-scale tours, permitting extreme consistency around the world. Touring engineers can specify a particular loudspeaker configuration that will have predictable results on every continent. Further, "universal racks" can be deployed with different models of loudspeaker inventory by simply recalling the correct preset.

Finally, amplifiers with multiple channels (generally defined as more than two) are now quite common. Again, credit the advent of switching power supplies in class D output topologies, with the weight, size, and dollar-per-watt ratio of multichannel amps dropping dramatically. Further savings can result from less rack space used, thus less racks needed, and for mobile use, less truck space occupied as well as faster load-ins, and load-outs.

Enjoy this Real World Gear look at the latest in modern power amplifiers for larger-scale sound reinforcement applications. LSI

Lab.gruppen PLM 20K44
www.labgruppen.com

Type: Class TD
Channels: 4
Output Power (per channel): 4400 watts @ 2 ohms; 4400 watts @ 4 ohms; 2300 watts @ 8 ohms
Outputs: Speakon, binding posts
Size: 2RU
Weight: 37 pounds
Models In Series: 2 (4-channel PLM 12K44)
DSP: Yes
Networking: Yes

QSC Audio PLD4.5
www.qsc.com

Type: Class D
Channels: 4
Output Power (per channel): 1600 watts @ 2 ohms; 1250 watts @ 4 ohms; 1150 watts @ 8 ohms
Bridge Mono:
Outputs: Speakon (NL4)
Size: 2RU
Weight: 22 pounds
Models In Series: 3 (install CXD Series also available)
DSP: Yes
Networking: Yes

Crown Audio I-Tech 4x3500HD
www.crownaudio.com

Type: Class I
Channels: 4
Output Power (per channel): 2100 watts (3500 watts burst) @ 2 ohms; 2400 watts @ 4 ohms, 1900 watts @ 8 ohms
Bridge Mono: 4200 watts @ 4 ohms
Outputs: Speakon (4-pole), binding posts
Size: 2RU
Weight: 29 pounds
Models In Series: 4 (2-channel I-Tech 5000HD, 9000HD & 12000HD)
DSP: Yes
Networking: Yes

Ashly Audio nX | *www.ashly.com*

Ashly nX multi-mode power amplifiers combine a lightweight, energy-efficient class-D design with an independent switch-mode power supply. They're available in 2- or 4-channel configurations, with 400, 800, 1500 and 3000 watts per channel.

Selectable outputs allow setting low impedance (2, 4, and 8 ohm) or constant voltage (70- or 100-volt) via rear panel DIP switches. Proprietary Ashly EMS (Energy Management System) draws less than 1 watt when in sleep mode (defeatable).

Models include nX (base), nXe (with Ethernet), and nXp (with Protea DSP). Both nXe and nXp models offer optional factory-installed Dante and CobraNet network connectivity. FIR filtering is also now available on nXp. All nX amplifiers are hand-built in the USA and backed by a 5-year warranty.

Model: nXp 3.04
Type: Class D
Channels: 4
Output Power (per channel): 3000 watts @ 2 ohms; 2000 watts @ 4 ohms; 1250 watts @ 8 ohms
Bridge Mono: 6000 watts @ 4 ohms
Outputs: Speakon, balanced Euroblock (aux)
Size: 2RU
Weight: 28 pounds
Models In Series: 4 (2- and 4-channel versions)
DSP: Yes
Networking: Yes

TECHNOLOGY FOCUS:

- *nXe models offer Ethernet control, serial data control, aux preamp outputs, instant standby mode, preset recall, fault condition logic outputs, and optional network audio and digital audio capability.*
- *nXp models add 32-bit SHARC DSP processing (with 48 or 96 kHz sampling) for comprehensive audio processing, with built-in signal generator for test tones and noise-masking, as well as precision load impedance monitoring on each amplifier channel output.*
- *Optional input cards available include AES3, CobraNet and Dante, with AVB in development.*
- *Multiple independent power supplies further bolster performance and reliability.*

Carvin Audio DCM3800Lx
www.carvinaudio.com

Type: Class A/B
Channels: 2
Output Power (per channel): 1800 watts @ 2 ohms; 1150 watts @ 4 ohms; 700 watts @ 8 ohms
Bridge Mono: 3800 watts @ 4 ohms
Outputs: 4-pin twist-lock, binding posts
Size: 2RU
Weight: 15 pounds
Models In Series: 2 (class D, 4-channel DCM2004Lx)
DSP: Yes
Networking: Yes

Powersoft X8
www.powersoft-audio.com

Type: Class D
Channels: 8
Output Power (per channel): 5200 watts @ 2 ohms; 3000 watts @ 4 ohms; 1600 @ 8 ohms
Bridge Mono: 10400 watts @ 4 ohms
Outputs: Speakon (NL4MD)
Size: 2RU
Weight: 52.9 pounds
Models In Series: 2 (4-channel X4)
DSP: Yes
Networking: Yes

Yamaha TX6n
www.yamahaca.com

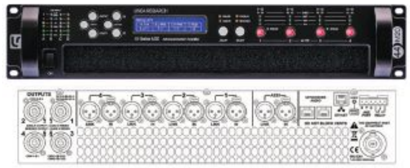
Type: Class D
Channels: 2
Output Power (per channel): 4120 watts (burst) @ 2 ohms; 3000 watts @ 4 ohms; 1800 watts @ 8 ohms
Bridge Mono: 5500 watts @ 4 ohms
Outputs: Speakon (NL4)
Size: 2RU
Weight: 35.3 pounds
Models In Series: 3 (all 2 channels)
DSP: Yes
Networking: Yes (with addition of card)

Electro-Voice TG7
www.electrovoice.com

Type: Class H
Channels: 2
Output Power (per channel): 3500 watts @ 2 ohms; 2500 watts @ 4 ohms; 1500 watts @ 8 ohms
Bridge Mono: 7000 watts @ 4 ohms
Outputs: 4-pole twist lock
Size: 2RU
Weight: 32 pounds
Models In Series: 2
DSP: Yes (variety of modules available)
Networking: Yes

dBTechnologies HPA 3100L
www.dbtechnologies.com, www.americanmusicandsound.com

Type: Class HD
Channels: 2
Output Power (per channel): 1700 watts @ 2 ohms; 1200 watts @ 4 ohms; 800 watts @ 8 ohms
Bridge Mono: 2400 watts @ 4 ohms
Outputs: Speakon
Size: 2RU
Weight: 30.1 pounds
Models In Series: 4 (all 2 channel)
DSP: Yes
Networking: No



Linea Research 44 Series M20
www.linea-research.co.uk

Type: Class D
Channels: 4
Output Power (per channel): 5000 watts @ 2 ohms; 3000 @ 4 ohms; 1500 watts @ 8 ohms
Bridge Mono: 10000 watts @ 4 ohms
Outputs: Speakon NL4
Size: 2RU
Weight: 27.5 pounds
Models In Series: 3 (all 4 channels)
DSP: Yes
Networking: Yes

Crest Audio Pro-LITE 7.5 DSP
www.crestaudio.com

Type: Class D
Channels: 2
Output Power (per channel): 3790 @ 2 ohms; 2450 watts @ 4 ohms; 1425 watts @ 8 ohms
Bridge Mono: 2800 watts @ 4 ohms
Outputs: Speakon
Size: 2RU
Weight: 14.6 pounds
Models In Series: 4 (non-DSP versions also available)
DSP: Yes
Networking: No

MC2 E100
www.mc2-audio.co.uk, www.g1limited.com

Type: Class D
Channels: 4
Output Power (per channel): 3700 watts @ 2 ohms; 2800 watts @ 4 ohms; 1400 watts @ 8 ohms
Bridge Mono: 7400 watts @ 4 ohms
Outputs: 4-pole Speakon
Size: 2RU
Weight: 26 pounds
Models In Series: 7 (2- and 4-channel models)
DSP: No
Networking: No

Peavey IPR2 7500 DSP
www.peavey.com

Type: Class D
Channels: 2
Output Power (per channel): 3750 @ 2 ohms; 2020 watts @ 4 ohms; 1250 watts @ 8 ohms
Bridge Mono: 2400 watts @ 4 ohms
Outputs: 1/4-inch or 1/4-inch 4-pole twist lock
Size: 2RU
Weight: 14.6 pounds
Models In Series: 4
DSP: Yes
Networking: No

NewsBytes THE LATEST NEWS FROM PROSOUNDWEB.COM

PEOPLE

Kelly Fair has joined **Lectrosonics** as western regional sales manager, and **Rebecca Isaacs** has also come aboard as marketing communications specialist.

Fair will be based in California, supporting customers in that state as well as Oregon, Washington, Idaho, Nevada, Arizona, Alaska and Hawaii. He brings more than 20 years of sales, product management and business development experience in pro audio to the position, including roles with Riedel Communications and Sennheiser.

Isaacs is based at the company's New Mexico headquarters, focused on creating and updating marketing materials such as manuals and product data sheets, as well as on written content for trade shows, events and the media. She brings a variety of experience in B-to-B marketing to the position, having held a number of positions that require a creative marketing approach as well as the ability to multitask.

John Krivit is the new president of the **Audio Engineering Society (AES)**, taking over the leadership role from **Andres Mayo**, who will remain on the AES executive committee as past president for the coming year.

Krivit was voted president-elect in 2014, and he also served on the executive committee of the board of governors the past year. Previously, he was the chair of the AES education committee. He's also currently an associate professor teaching at Emerson College and Bay State College in Boston.

Riedel Communications has named **Ryhaan Williams** as its new vice president of sales, Eastern U.S. Based in New Jersey, she is charged with strengthening existing relationships and creating new strategic alliances in the region, in addition to expanding sports markets nationally.

Before joining Riedel, Williams was director of strategic accounts at Chicago-based Joseph Electronics, where she developed new market verticals and product training, and she also provided project management services in addition to fostering relationships with key accounts and consultants.

Italian Speaker Imports, the U.S. distributor of **FBT** loudspeakers, has announced the addition of **Tracy Brefka** as its new special programs manager. Bringing 25 years of experience to the role, previously he spent the bulk of his career with Audio-Technica, starting in inside sales (both professional and consumer) before moving up to assistant sales management and then management.

The supervisory board of **beyerdynamic** has appointed **Robert Winterhoff** as managing director, with responsibility for global sales and marketing. As a part of the company's "Fit for the Future" strategy, the newly created position will strengthen the management team and goes hand-in-hand with other recent appointments within the organization's global structure.

Winterhoff previously worked with Harman International for almost 20 years, holding positions in sales and marketing, business development, program management, quality, and operational excellence.

COMPANIES

Eighteen Sound Srl (Reggio Emilia, Italy) has announced its purchase of the **CIARE** brand (Senigallia, Italy). The purchase takes effect immediately and broadens the scope of the Eighteen Sound product line offering, as well as expanding its access to new global markets.

"This is a direct response to our assessment of our product line offerings and will allow us to offer a broader pallet of products," states **Giacomo Previ**, director of global sales. "Eighteen Sound will revitalize and strengthen the brand, while maintaining its focus on the professional audio marketplace. The CIARE brand will be maintained autonomously and continue its storied 70-year history while catering to its loyal client following and maintaining its solid performance standards."

QSC has been named a 2015 Top Workplace by *The Orange County Register* newspaper. The company, headquartered in Costa Mesa, CA, has earned this distinction in five of the last six years and ranks at the top of mid-sized manufacturers in the region. Top workplaces were elected based on an anonymous employee survey corresponding of 22 statements regarding their workplace experiences and company leadership.

"The talent and passion at QSC amazes me every day," states **Joe Pham**, QSC president and CEO. "It's exciting to see how committed people are to their work, their teams and the whole organization. I feel very fortunate to have the opportunity to work for such a great company."

CAD Audio is celebrating its 85th Anniversary in 2016. The company originally took shape as Astatic Corporation, founded in 1931 by **C.M. Chorpening** and **F.H. Woodworth**, two ham radio operators with a passion for clear audio transmission that led them to develop the D-104 microphone, noted for its ability to perform without static.

The "Astatic" characteristic of that mic became the company's namesake and led

The READERS' CHOICE AWARDS 2016

ProSoundWeb & Live Sound International

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CONSOLES & MIXERS: LARGE FORMAT
Allen & Heath: dLive
Allen & Heath: GLD Chrome
Music Group: Midas PRO X
PreSonus: StudioLive AI Mix System

CONSOLES & MIXERS: SMALL FORMAT
Allen & Heath: Qu-32
Avid: Pro Tools | S3
DiGiCo: S21 Digital Mixer
HARMAN: Soundcraft Ui Series (Ui12, Ui16)
Mackie: DL32R Wireless Digital Mixer
Music Group: Midas M32R
QSC: TouchMix-8 Compact Digital Mixer
Yamaha: CL/QL V3.1 Digital Mixing Console
Yamaha: StageMix 5
Yamaha: TF5 Digital Audio Console

HEADPHONES
Audio-Technica: ATH-R70x Professional Open-Back Reference Headphones
HARMAN: AKG K553 PRO
Shure: SRH940 Professional Reference Headphones

IN-EAR MONITOR: SYSTEMS
Audio-Technica: M3 Wireless In-Ear Monitor System
Shure: PSM300 Stereo Personal Monitor System

INTERCOM SYSTEMS
Clear-Com: FreeSpeak II
Radio Active Designs: UV-1G Wireless Intercom System

INTERCONNECT PRODUCTS
Radial: Catapult 4 Channel Audio Snake
SFM: D6-QUAD-S Series Cat6 SFTP Cables
Whirlwind: Medusa Elite

LINE ARRAYS: LARGE FORMAT (12"+ WOOFER)
Electro-Voice: X-Line Advance
HARMAN: JBL Professional VTX V25-II
PK Sound: Trinity

LINE ARRAYS: MEDIUM FORMAT (8-10" WOOFER)
Adamson Systems Engineering: S10 Line Array
Meyer Sound: LEOPARD Linear Sound Reinforcement System
RCF USA: HDL10-A Line Array

LINE ARRAYS: SMALL FORMAT (LESS THAN 8" WOOFER)
D.A.S. Audio: AERO 8A
dB Technologies: DVA MINI
L'Acoustcs: Kiva
Music Group: Turbosound Flashline Mini TFS550H

LIVE RECORDING HARDWARE & SOFTWARE
Avid: VENUE – S6L
Focusrite Novation Inc: RedNet MP8R
Music Group: Klark Teknik DN9650 Network Bridge
PreSonus: Studio 192 26x32 USB 3.0 Audio Interface
PreSonus: Studio One 3 Professional Digital Audio Workstation
RCF USA: M18 Digital Mixer
RME Audio: Babyface Pro

LOUDSPEAKERS: ACTIVE
VUE Audiotechnik: h-8 High Definition Full Range Loudspeakers

LOUDSPEAKERS: FULL RANGE
HARMAN: JBL SRX800 Series
L'Acoustics: X Series
Music Group: Turbosound Athens TCS152/96-AN
Yamaha: DBR Active Loudspeaker

LOUDSPEAKERS: COLUMN/LINE SOURCE
Meyer Sound: CAL Column Array Loudspeaker
Music Group: Tannoy QFlex

LOUDSPEAKERS: DRIVERS & TRANSDUCERS
Faital Pro: 12RS1066
Faital Pro: HF148
Faital Pro: 18HW1070
Faital Pro: HF144
Music Group: Tannoy Dual Concentric Omnimagnet Driver
Powersoft S.p.A.: M-System

LOUDSPEAKERS: MODELING SOFTWARE
Adamson Systems Engineering: Blueprint AV
HARMAN: JBL HiQnet Performance Manager
L'Acoustics: Soundvision 3.0

LOUDSPEAKERS: PASSIVE
Music Group: Tannoy AMS Series

LOUDSPEAKERS: STAGE MONITORS
Adamson: M212
L'Acoustics: X15 HiQ

LOUDSPEAKERS: SUBWOOFERS
Adamson Systems Engineering: S119 Subwoofer
D.A.S. Audio: UX-221A Subwoofer
HARMAN: JBL EON618S
Meyer Sound: 900-LFC Low-Frequency Control Element
Yamaha: DXS18 Subwoofer

MICROPHONES: CONDENSER TYPE, PERFORMANCE
DPA Microphones: d:screet SC4098 Podium Microphone
HARMAN: AKG C314
Shure: BETA181 Ultra-Compact Side-Address Microphone
Shure: KSM9HS Handheld Vocal Microphone

MICROPHONES: DYNAMIC TYPE, PERFORMANCE
Audio-Technica: AE4100
HARMAN: AKG D5 C
Shure: SM58 Vocal Microphone

MICROPHONES: WIRELESS SYSTEMS
Audio-Technica: System 10 PRO Digital Wireless System
HARMAN: AKG DMS800
Shure: Axient Wireless Management Network
Shure: ULX-D Digital Wireless Systems

NETWORKING TECHNOLOGIES
Audinate: Dante Via Software
Audinate: Dante Virtual Soundcard
Shure: ShurePlus™ Channels Mobile App for iOS
Shure: Wireless Workbench 6.11

OUTBOARD GEAR
Nord: Stage 2 EX 88
PreSonus: StudioLive CS18AI Ethernet/AVB Control Surface
Yamaha: RSio64-D

PORTABLE PA SYSTEMS
Bose: F1 Model 812 Flexible Array Loudspeaker System
Electro-Voice: EKX Powered Loudspeakers
Mackie: FreePlay
Mackie: Reach
RCF USA: EVOX 8
Yamaha: STAGEPAS 600i

POWER AMPLIFIERS
Extron Electronics: NetPA 502 AT and NetPA 1001-70V AT
HARMAN: Crown XLS DriveCore 2
Music Group: Lab.gruppen PLM+ Series Amplifier
Powersoft S.p.A.: Ottocanali DSP+D Power Amplifier
QSC: GXD Series Processing Amplifiers
QSC: PLD 4.5 Processing Amplifier

SIGNAL PROCESSORS
HARMAN: dbx VENU360
Music Group: Lake LM Series
Yamaha: MRX7-D open architecture processor

STAGE MONITORS
Meyer Sound: MJF-210 Stage Monitor
Music Group: Turbosound TMW-121 Wedges

SYSTEM ENGINEER/TECH TOOLS
Music Group: Lab.gruppen Lake Controller
Rational Acoustics: Smaart v.7
Shure: Axient Spectrum Manager

Voting will be open through January 13 at 12:59pm EST.
Winners will be announced on Jan 21, 2016 on Prosoundweb.com.

PROSOUNDWEB.COM/RCA/VOTE

to numerous innovative products, including phonograph pickups and cartridges, while also laying foundations for the production of recording heads. CAD Audio now offers a range of products serving all segments of the pro audio industry.

The **RCF Audio Academy** is being offered again this year. The course is organized by the Department of Science and Methods for Engineering of the University of Modena and Reggio Emilia (Italy), in cooperation with RCF SpA. The primary goal is to provide engineers with specific knowledge of technologies related to sound, transducers, and loudspeaker systems, as well as their application in the professional market.

The course, which starts March 1, 2016, is offered in e-learning mode and requires one week at RCF (also in Reggio Emilia) from May 16 through May 20. Go to www.rcf.it/en_US/education for more information.

Powersoft recently held four in-person listening sessions entitled "Xperience The Power" in Los Angeles and San Francisco, attended by system designers and integrators as well as touring and rental operators, to demonstrate the capability (and highlight the availability) of the next generation of its class D amplifiers and moving magnet linear technology, which is called the M-System.

The M-System utilizes the company's M-Force motor transducer, M-Drive switching mode amplifier module, and Differential Pressure Control (DPC) technology to be able to produce a stated 150 dB SPL from a single cabinet. Powersoft holds five patents on M-System, including two on M-Force, one on M-Drive, and two on DPC. The company states that it is planning future demonstration dates for 2016.

Fits & Starts Productions HOW-TO Sound Workshops, which has been providing training for houses of worship and schools/colleges around the U.S. for 15 years, is utilizing a **DiGiCo** SD9 digital console as a primary teaching tool. Aside from carrying the main PA for the instructor, the SD9 is also utilized to run music and signal so that attendees can see and hear how the console operates.

Visiting approximately 36 cities each year, chief instructor **Mike Sokol** and the F&S team provide a comprehensive, full-day, hands-on opportunity to learn the essentials of setting up and running a sound system. Find out more at www.howtosound.com. LSI

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FOR LOVE OF THE GIG

Drilling down to find answers as to what's most important. *by Jonah Altrove*

In the grand scheme of things, audio engineering is a pretty mysterious field, at least for most people. When asked with I do for a living, "I'm an audio engineer" is often met with puzzlement, so I quickly amend, "You know, I'm a sound person. I run sound for concerts and events." For many folks, this is when the light comes on – one woman said, "Oh, you're the guy, like at my church, in the back who turns all of the knobs." She was pensive for a second, then asked, "So, you know what all those knobs do?"

Sometimes, though, even that explanation proves insufficient. Recently I was the recipient of "Oh, cool man! So, like, what clubs do you DJ at?" Of course, this wasn't intended to be insulting. It's just that there's not that many of us out there doing this.

Then I thought that, maybe, working in relative obscurity is a good thing. When everyone leaving a concert has a comment about the sound, it's usually "The band sounded so good!" If they have something to say about the person operating the sound, it's usually not positive. So we are, at our best, invisible. If our presence is unnoticed, it's considered a successful gig. I subscribe to the Dave Natale school of mixing philosophy. To paraphrase: "People don't come to the show to hear me mix. They come to hear the artist. I just make it louder." We're on a constant search for the most pristine preamp or most recent PA system. So we "live people," as my studio buddy calls us, are a relatively discreet bunch.

People who ask me about life on the road are usually appalled by my description of the long hours, tough conditions, and often less-than-princely pay. I frequently hear, "Why would you want to do that?" And that's the question to which I have no rational answer. My answer is from a simpler place: I love it. I live A/D converters, eat FFTs, and breathe chain hoists. I dream of flawless festival changeovers. I actually have a recurring nightmare in which I'm juggling acts on an analog desk and keep running out of channels (scary stuff!).

For me, it's exhilarating to be able to play a role in a successful production, a memory that audience members will carry with them for life. It's a way to couple my passion for music with my love of science and technology. I like to think of live sound as "elegant precision."

And I'll get involved in any way I can. Watching, learning and studying others for years has contributed greatly to my appreciation for every aspect of production, and broadened the horizons of my knowledge so I can help with whatever needs doing. FOH? Mons? RF tech? System tech? Need me to calculate the force vectors in that rigging bridle, or battle for the optimal placement of the arrays? I'm on it.

On my first "big" show, I was the youngest/smallest and thus the unopposed nominee for the always-glamorous job of pulling 100 feet of feeder through the dirt underneath the SL260. The cables weighed more than me, and by the time the task was complete I was a muddy mess and had hit my head on the screw jacks three times. But I emerged from under the deck with a huge smile. Over a decade later, I still feel the same rush when powering up my console or pinning boxes into an array.

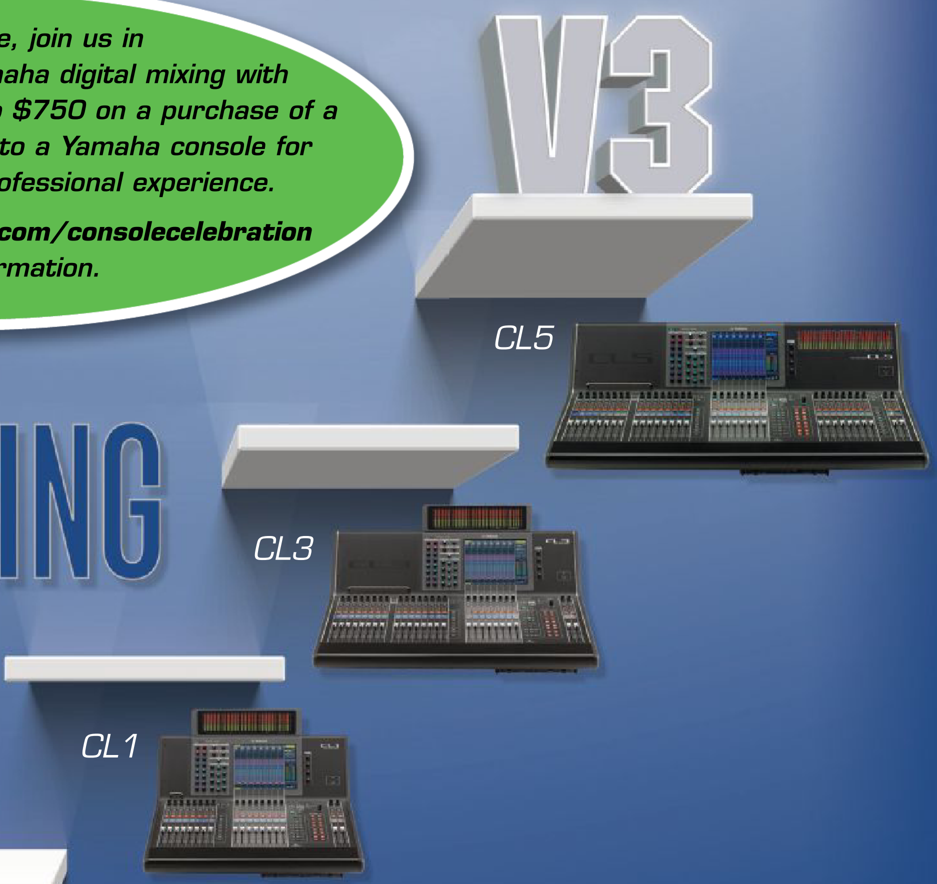
Rational or not, that's my reason, my response to "Why would you want to do that?" Because I love it. It's one of the great things about our particular field – no one gets into it to get rich. It's a labor of love. I love being surrounded by people who are stoked about putting on a killer show. There's probably not a mansion in my future, but as long as I can support myself and get up every morning and look forward to what I do, isn't that more important? LSI

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

For a limited time, join us in celebrating 28 years of Yamaha digital mixing with a special promotion worth up to $750 on a purchase of a CL, QL or TF console. Step up to a Yamaha console for ultimate reliability and a professional experience.

*Please go to **www.yamahaca.com/consolecelebration** for more information.*

STEPPING IT UP



The standards stay, but innovation never ends. Going strong, V3 software for the popular CL and QL Series features a variety of upgrades making our family of digital mixing consoles more efficient than ever. This next step up includes an 8-band Parametric EQ Option with Notch Filters, Real Time Analyzer Function, 4 User Defined Key Banks, Surround Sound Panning and Monitoring and so much more.

Perhaps most notable is the addition of the Dan Dugan Automatic Mic Mixer for the CL Series. Already included in the QL Series, gain distribution for up to 16 speech microphone channels is automatically optimized in real time, achieving smooth, natural level control. What could be smarter than that?

For a complete list of features and to download the software free-of-charge, visit www.yamahaca.com.

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