THE JOURNAL FOR LIVE EVENT TECHNOLOGY PROFESSIONALS

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LIVE SOUND INTERNATIONAL

25th ANNIVERSARY

INSIDE

THE REAL THING

Success is in the details with Duran Duran on tour.

SPLITTING AUDIO FOR BROADCAST, STREAMING AND MORE

WORKING WITH BOTH ANALOG & DIGITAL CONSOLES

REAL WORLD GEAR: COMPACT LINE ARRAYS

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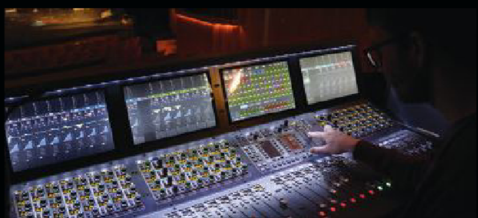
–Charlie Bradley, monitor engineer, Duran Duran, Annie Lennox, Snow Patrol

"After just two weeks on tour, I can say that S6L has the fastest and easiest workflow of any console, with the best sounding preamps and a fantastic control surface."

–Gerard Albo, FOH engineer, a-ha, Patti Smith, Tom Jones

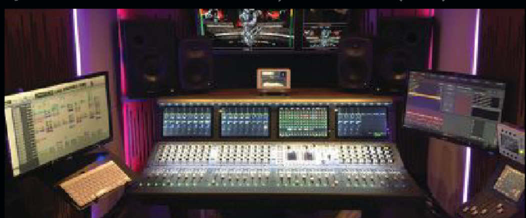


National Theatre of Iceland, Reykjavík

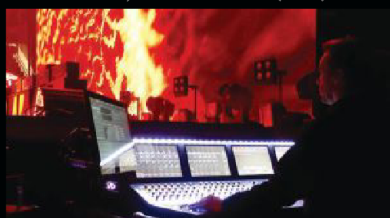
Rich Steeb, Blue Rodeo (FOH)

Rob Allan, Massive Attack (FOH)

Myles Hale, Black Sabbath (monitors)



Transmix Oy OB Truck, Helsinki, Finland



Robert Nevalainen, Bryan Adams (monitors)

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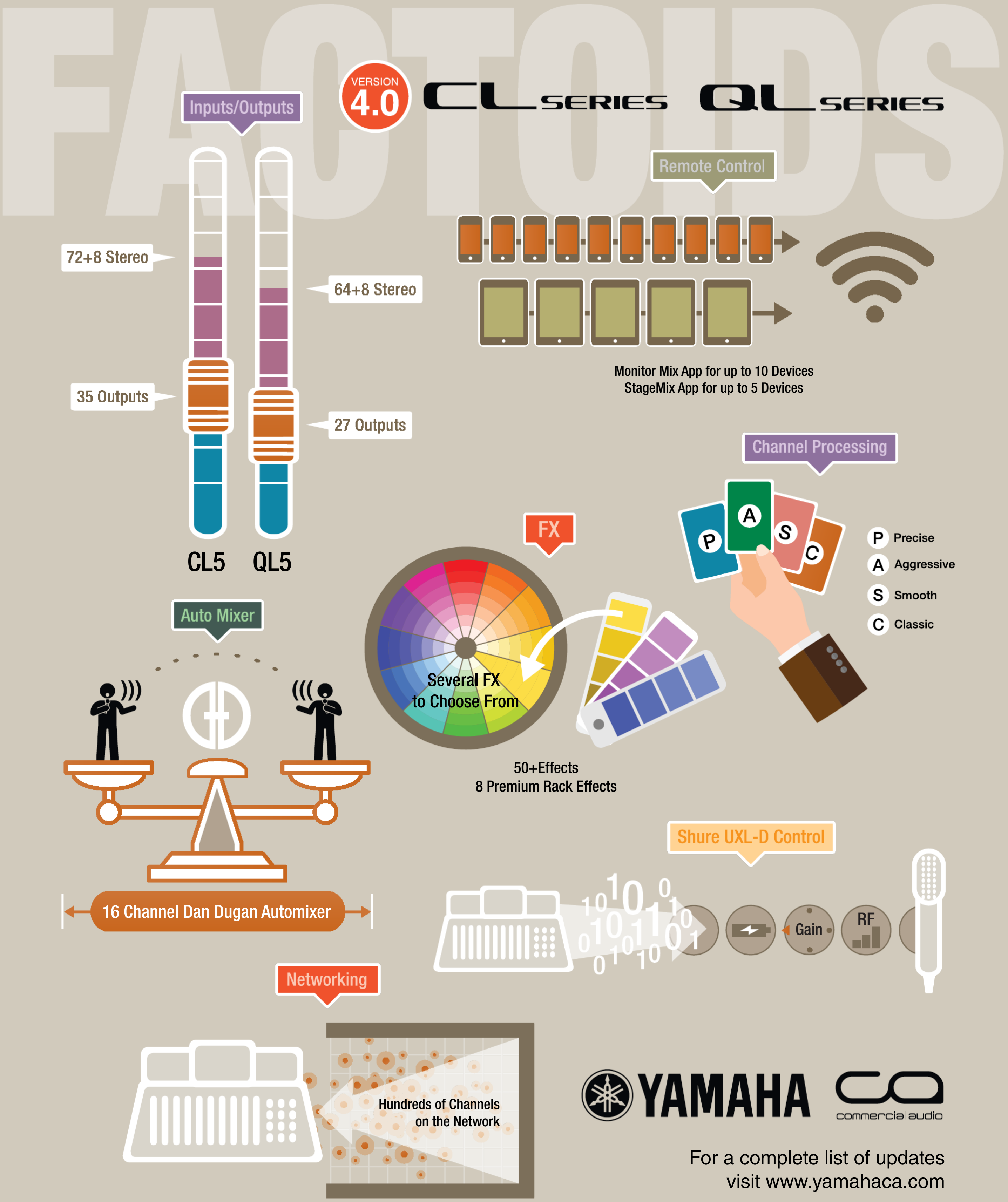
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From the Editor's Desk

THE PROLIFERATION of digital technology in pro audio has undeniably changed many things for the better. But the market is still in a period of transition to digital; it will continue to be ongoing for the foreseeable future and may never really be "complete."

The aspect of splitting audio signals for various feeds (in addition to the main system) appears to be one of the areas where the transition is happening perhaps more slowly than we might assume. In his look at the topic in this issue, Gary Parks talked with individuals at several leading sound companies to get their approaches and views, and what he found may be surprising. It seems that "good old analog" still has a place in that facet of system work.

Elsewhere in the issue, Craig Leerman offers two pieces that both provide a treasure-trove of useful information in working with analog and digital consoles as well as getting the most from a system. In addition, Jonah Altrove contributes part 1 of an in-depth look at the vital issue of impedance, while Kevin Young presents a profile of Karrie Keyes, a top monitor engineer and a whole lot more.

Also don't miss Andy Coules' guidelines for musicians to get the most from their collaborations with sound engineers, as well as a classic column from Sully providing insight on working with the next generation of audio professionals.

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

Keith Clark

Keith Clark
Editor In Chief, Live Sound International/ProSoundWeb
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25th ANNIVERSARY

ON THE COVER: Iconic pop/rock group Duran Duran performing on the current Paper Gods Tour. *(Photo by Todd Kaplan)*

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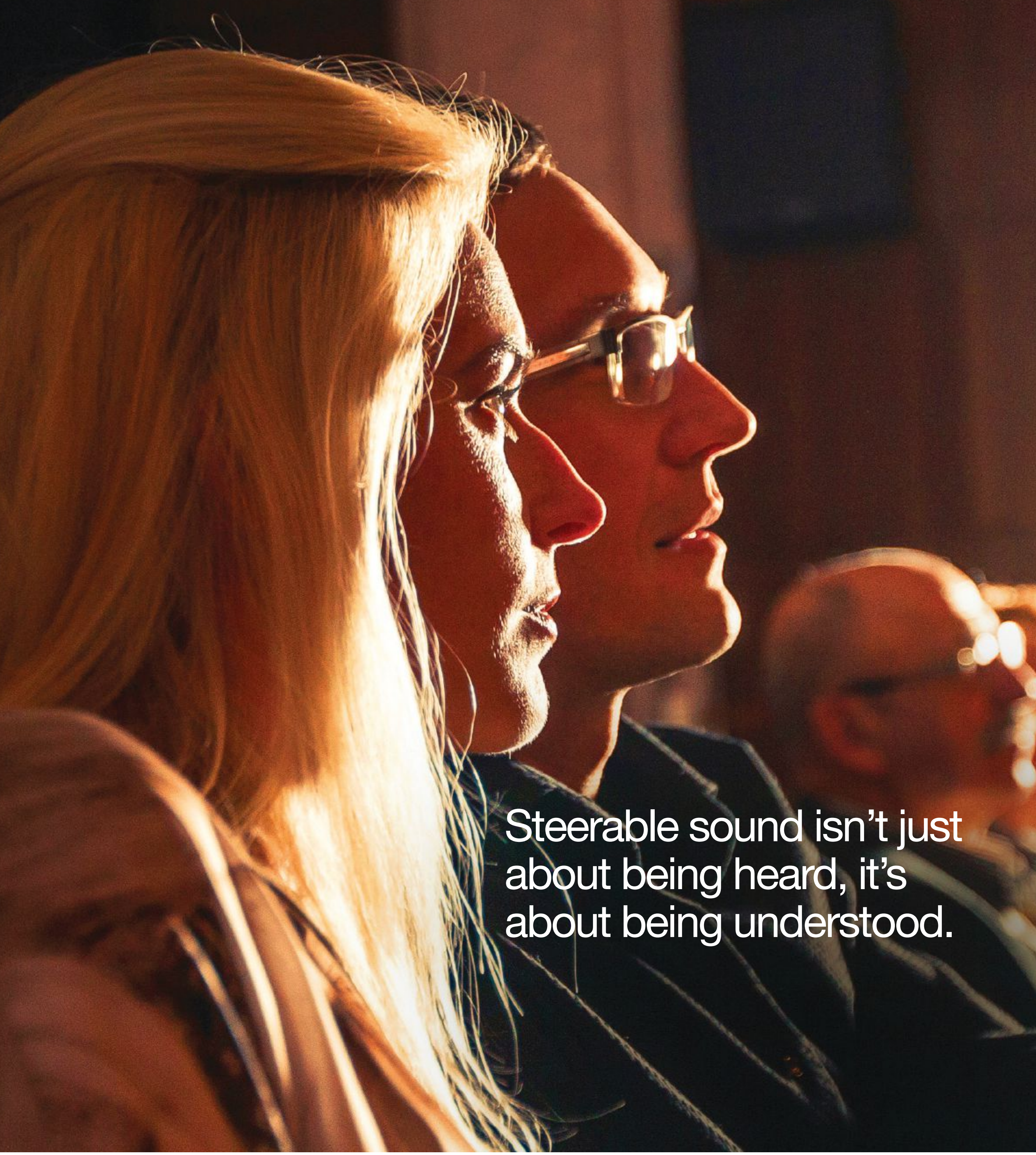

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Audio-Technica ES931

A 6-inch condenser gooseneck microphone available in multiple configurations that include the ES931PM with standard AT8534 wall/ceiling plate power module, the ATND931 Dante model with ATND8734 wall/ceiling plate power module, and the ES931/MIC basic model (mic only). Each iteration is available in one of three polar patterns (cardioid, hypercardioid and MicroLine), and in two color options (black and white). The rugged gooseneck retains virtually no "memory," permitting quick adjustment into the desired shape and position. www.***audio-technica.com***

BASSBOSS LA88

A self-powered compact line array designed to deliver broad coverage and long throw. Horizontal dispersion is stated as 140 degrees, while frequency response is 50 Hz to 19 kHz (+/- 3 dB) in full-range mode. The dual-channel EIAJ Powersoft class D amplifier is rated to supply up to 3,000 watts. Components include dual 8-inch (2-inch voice coil) woofers in a vented LF section with proprietary multi-aperture midrange loading, joined by two 1.7-inch-diaphragm compression drivers on 1-inch-throat isophasic waveguides. The cabinet measures 9.7 x 25.2 x 19.5 inches (h x w x d) and can hang with an overall width of 28.5 inches with rigging pins inserted in the flyware, offering the ability to pass through 30-inch doors while pinned together on an optional transport and ground-support cart. Weight is 55 pounds. Touring versions of the Baltic birch enclosure are finished with a black polyurethane bedliner coating, with indoor install versions available in white or black. A waterproof fiberglass finish is available for permanent outdoor installs. ***www.bassboss.com***

Avid VENUE | DNT-192

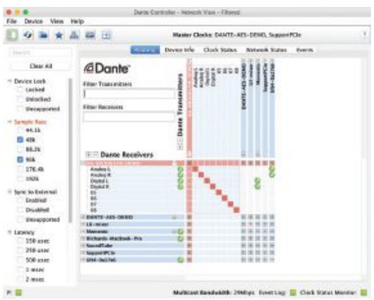
An 16-channel option card for the VENUE | S6L live mix system that integrates Dante-enabled devices and existing Dante networks. Based on the Dante Brooklyn II module, the DNT-192 supports Gigabit Dante-enabled devices that include external audio mixers, amplifiers, preamps, DSP, wireless microphone systems and more. Up to six cards can be installed in a single Stage 64 for a maximum of 64 inputs and 32 outputs per rack. Dante Controller software enables configuration of cards and devices, along with monitoring signal health and status, and routing audio throughout a network. ***www.avid.com***

Yamaha EMX Series

Models EMX2, EMX5 and EMX7 join the powered mixer series, all with onboard class D power amplifiers stated as delivering 500, 630 and 710 watts, respectively. They include Yamaha SPX effects for hall, plate, room and echo reverbs, as well as a new feedback suppressor. The EMX2 and EMX5 also include an updated 1-knob master EQ sound contour control to adjust overall frequency balance, while the EMX7 offers a flex-type graphic equalizer (Flex9GEQ) with selection of up to nine bands out of a total of 31 for fine tuning +/-15 dB. The EMX2 has 10 inputs, four mic preamps and three stereo line inputs. The EMX5 and EMX7 each have 12 input channels. All three are equipped with phantom power, and input 4 has a high-impedance input for acoustic guitar or bass without need for external processing. ***usa.yamaha.com***

Audinate Dante Controller 3.10

The latest version of free software allows users to manage Dante networks from laptops connected via wi-fi. By connecting a wireless access point to a



Dante network, adjustments can be made from any location in a facility. A new Advanced Filter can be used to refine the range of devices displayed to narrow and refine views when managing larger networks. Filter parameters include device and channel names, sample rate, latency setting and lock status. 3.10 also supports new Device Lock to remotely lock supporting Dante hardware and software devices, employing a user-selected 4-digit PIN. ***www.audinate.com***

JBL Professional Intellivox HP-DS170

A column array loudspeaker suited for reverberant environments where improved speech intelligibility is required. It is equipped with proprietary digital directivity synthesis (DDS) technology that utilizes advanced vertical beam shaping to control audio in environments with complex acoustic challenges. The beam shape and other settings can be adjusted via Harman's WinControl software. A single cabinet is stated as being capable of maintaining an even sound pressure level of 106 dB across an audience area of up to 30 meters (100 feet) in length. The unit's custom 6.5-inch cone drivers are powered by an onboard multi-channel class D amplifier working with sophisticated DSP. The HP-DS170 is equipped with RS-485 connectivity and is IP55 rated for indoor/outdoor applications. ***www.jblpro.com***

BSS Soundweb London BLU-USB

An audio interface for integrating PC/Mac computers with corporate soft-phones in local conference audio systems through BLU link digital audio transport, for sending audio to, and receiving audio from, the BLU link bus. The BLU-USB is designed to solve the distance limitations associated with USB transmission. By converting audio to BLU link close to the source, the processor can be located in a different location up to 100 meters (300-plus feet) away. It's compatible with any BSS Soundweb processor or any other device that includes BLU link. The unit can sit on a table as well as be mounted under a table or on a wall. ***www.bssaudio.com***

Earthworks FlexMic Series Expansion

FMLR models have a single-color LED light ring (choice of red or green) at the top of the microphone that can be programmed via an external media control system to indicate on/off status. They also include the company's patented polar technology that allows orators to move as much as 70 degrees off-axis on either side, as well as above or below the mic, while maintaining the same sound quality and intelligibility. FMLR models have a fully flexible gooseneck and are available in 13- and 19-inch versions. Meanwhile, FMRLR models have a rigid center with flex at both ends, offered in 19-, 23- and 27-inch lengths. All FlexMic versions are available with either a cardioid or hypercardioid polar pattern. ***www.earthworksaudio.com***

Ashly Audio nX-Series

An expansion of the series of multi-mode amplifiers includes models rated to deliver 75 or 150 watts per channel in a single rack unit frame that can house either two or four channels. Like all models in the series, there are base models (nX). Meanwhile, nXe models add Ethernet for monitoring and control while nXp models provide networking and built-in Protea matrix processing. Optional Dante or CobraNet cards can be added to nXe or nXp models. The amplifiers can drive low impedance (2, 4, 8 ohms) or 25-, 70-, and 100-volt constant voltage systems via rear-panel dipswitches. ***www.ashly.com***

Fulcrum Acoustic FLS115

A subcardioid subwoofer that's a companion to the company's FL283 subcardioid line array. It utilizes proprietary Passive Cardioid Technology that reduces excessive rear low-frequency radiation without the need for additional drivers, amplifiers, or signal processing. The enclosure and rigging are designed to accommodate up to 20 degrees of splay between adjacent cabinets, providing for sharply curved arrays. ***www.fulcrum-acoustic.com***

EM Acoustics M-C12 & M-C15

A stage monitor series with choice of 1.3-inch HF/12-inch LF (M-C12) and 1.3-inch HF/15-inch LF (M-C15) coaxial drive units. Both incorporate passive crossovers, requiring only one amplifier channel and zero external processing. Recessed 35 mm pole-mount sockets and M8 threaded rigging points are also included. ***www.emacoustics.co.uk***

Peavey P2

A powered full-range column loudspeaker loaded with eight 4-inch custom HF drivers, joined by a 12-inch subwoofer with a 3-channel input. Input 1 utilizes a variable mic/line input with a vocal boost selector, input 2 offers dual 1/4-inch inputs, and input 3 has dual RCA connections. Onboard DSP adds an optional frequency boost/cut. Also included are an XLR-thru output and a fan-cooled power supply. ***www.peavey.com***

ISP Technologies CX110i & CX114i

Self-powered coaxial loudspeakers incorporating a new generation of coaxial drivers combined with precision-matched active crossovers and equalization. The CX110i includes a 10-inch coax woofer with 2.5-inch HF, while the CX114i offers a 14-inch coax woofer with a 3-inch HF. Dispersion for both is stated as 80 degrees conical. Also available are weatherized versions that are built of exterior-grade 18 mm Baltic birch plywood, fully sealed with a polyurea spray-on finish with UV protection, and finished with hot zinc-dipped and powered coated grilles with stainless steel screws and hardware. The onboard Speakon connectors and mounting bracket are also weatherized. ***www.isptechnologies.com***

Funktion-One F132

A horn-loaded subwoofer incorporating a single Powersoft 32-inch M-Force 10 kW linear transducer that's based on a patented moving magnet linear motor structure. It carries a stated frequency response of 24 Hz to 65 Hz. Each F132 is approximately 5 feet deep and 3.7 feet wide. ***www.funktion-one.com***

Galaxy Audio EB4

In-ear monitors utilizing single titanium drivers to achieve fidelity with extended bass response. Frequency response is stated as 20 Hz to 20 kHz (+/- 3 dB). Aluminum alloy construction makes the units light in weight and enhances durability. Three pairs of Silicon sleeves (small, medium, and large) are provided for users to select an optimized fit. EB4 has a standard 3.5 mm stereo jack. ***www.galaxyaudio.com***

Clear-Com FreeSpeak II-Base II

A base station and splitter box (FSII-SPL) to complement recently-released wireless intercom beltpacks and transceiver antennas. The base station provides digital signal transmission over 1.9 and 2.4 GHz bands with a cascade menu and a browser-based Core Configuration Manager (CCM) for system configuration and real-time changes. It supports up to 25 full-duplex wireless beltpacks and covers a large production area with up to 10 distributed-antenna transceivers and two transceiver splitters spanning up to 20,000 meters from the base station over fiber connection. Two built-in optional SFP fiber connectors enable either a native single-mode or multi-mode fiber link between base stations and the FSII-SPL splitter. Multiple 2- and 4-wire ports are also available to connect with analog partyline and 4-wire devices. The splitter offers five RJ45 Ethercon connectors, LED status indicators for each connection, and a DIP switch for setting Cat-5 or fiber operation. It can be mounted into the 1RU splitter rack, which holds up to two splitters, or on a microphone stand or against a wall. ***www.clearcom.com***

TEAM EFFORT

A musician's guide to getting the most from sound engineers. *by Andy Coules*

For the performing musician there are few things worse than bad sound. If the mix is bad in the PA the audience can't hear the music, and if it's bad in the monitors it makes it difficult to play together effectively. Whatever level artists are at in the music industry, they spend valuable time, effort and money preparing for those precious minutes on stage but all of those efforts can quickly be undone by subpar sound engineering.

It's no coincidence that an engineer is the first crew member most bands typically hire; having someone out front who knows your songs and cares about your sound makes all the difference. It's an observable fact that bands who have confidence in the engineer are more relaxed and deliver better performances.

But for those situations where there's no choice but to work with the house engineer, here are a few tips for getting along with – and thus getting the most from – that engineer.

STICK TO THE SCHEDULE

The first thing is to be punctual. Turning up on time for sound check with all band members and equipment present ensures getting the maximum time to get sorted, and it also allows the engineer to comfortably deal with you. Being late automatically increases the amount of work the engineer needs to do while also compressing it into a shorter span of time. It's a simple matter of professional courtesy; turning up on time can so easily avoid starting the day on the wrong foot.

SOUND CHECK IS NOT REHEARSAL

Getting a bunch of musicians together to rehearse on a regular basis can be a total nightmare. Finding that one night when everyone is free is frequently an almost impossible task, so the temptation exists to "run through the new stuff at sound check."

However. it's important to understand that the primary purpose of sound check is to enable everyone to properly set up and to ensure everyone is comfortable on stage. This involves checking that all of your equipment is present (including those pesky power and signal cables), that it all works, that everyone can physically fit on the stage. and that the amps and monitors are distributed appropriately. Then it's just a case of line checking the individual instruments and running through a couple of songs to get the monitor mixes sorted.

The sound check also enables the engineer to get a feel for the music so he/she can start to build the mix. A good engineer will also work with the musicians to ensure the stage levels are suitable such that he/she can add the right mix elements to the ambient sound to produce the desired sound in the house. Therefore musicians should always arrive fully rehearsed and ready to go.

SOUND CHECK SONG SELECTION

It's always a good idea to think carefully about what songs to play in sound check. Be sure to include every configuration of instruments and vocals to ensure that it all works and the levels in the monitors are good for everyone.

This is also a good time to clarify with the engineer who sings lead and backing vocals, and to point out any mix elements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that require particular attention. Bear in mind that some band members aren't able to ask for monitor changes while playing so try to avoid doing long songs in their entirety. If necessary, just do half songs or a quick verse/chorus to get an idea of the monitors, suggest changes, and quickly move on.

Another useful aspect of sound checks is that they help in rehearsing the actual changeover itself, which can greatly help the show run smoothly, so try to ensure everyone is present. Bands that sound check without all members present always take more time in changeover, and this can throw off the entire schedule.

WARMING UP

Vocalists need to understand the importance of warming up; like any muscle, it's a simple fact that it works more optimally when it's warm. Go to any small gig and watch the vocalist and I guarantee that they get noticeably louder by the third song. This is important, not just from the point of view of caring for the voice and getting the most out of it, but it also helps in setting appropriate monitor levels. If the whole band sound checks at 70 percent of its normal performing level, then the entire exercise is pretty much pointless.

I've offended more than one singer by suggesting vocal lessons, but this isn't always because they can't sing. A qualified instructor teaches proper breathing and other tips for getting the most from the voice, and they'll also be able to provide a range of exercises for warming up.

IT'LL SOUND FINE WHEN IT'S FULL

This may sound like an excuse to cover a bad-sounding sound check, but there's much truth in this simple maxim. Every room sounds drastically different when it's empty (i.e., during sound check, before the audience arrives) because human bodies are excellent absorbers that usually (and often drastically) reduce the amount of reflected sound.

I've never come across any guitarists with ears in the back of their knees.

Therefore most experienced engineers aren't trying to build the definitive house mix during sound check, but instead are focused on combining the mix elements in a rough configuration, ensuring that everything is present, correct, and doing it's job. Anticipating how the room will change when the audience is present is a cornerstone of mixing live sound, particularly in small venues with difficult acoustics and less-than-amazing PA systems.

One thing to definitely avoid is stepping out front during the sound check and giving the engineer mix advice (such as turn this up, turn that down, etc.). I fully understand that musicians have a vested interest in how it sounds, but such advice is meaningless and

vaguely annoying. Whenever offered such advice, I simply explain the situation (about the empty room) and ask them to trust me.

This is not to say that musicians aren't allowed to offer mix advice, it's *their* music after all, and most engineers welcome insight into the sound or how they'd like it conveyed. But it's better to have a quick chat with engineers and then leave them to it. One of the best examples of this I've encountered is when a band turned up, gave me a set list and a sheet of rough notes such as "this is the lead vocal, this vocalist just talks between songs, this guitar should be louder than the others, watch the keyboard player for solos," and so on.

GET THE BALANCE RIGHT

With monitor mixes bear in mind that only in smaller venues is the front of house engineer also expected to mix the monitors; at most other (i.e., bigger) venues, these are two completely separate jobs done by two different people. When an engineer is wearing both hats, please be aware that constantly asking for changes in the monitors takes away from his/her time to focus on the house mix.

One commonly uttered phrase that I find completely unhelpful is when a musician asks for "a bit of everything" in the monitors. If I put a bit of everything in, the result is a loud "mush" that not only makes the stage sound messy but also reflects and affects the main mix. My advice to musicians is to listen to what they can hear naturally and request only those mix elements that are needed to be able to play/sing in time/pitch. Most things happening on stage can already be heard, and after the obligatory "more me" request not much more is really needed in the monitors.

KNOW THE EQUIPMENT

This may seem basic and obvious but it's quite important for musicians to be familiar with their own equipment prior to gigging with it. Make sure everything needed is available, along with carrying spares for items such as power leads, signal cables, batteries, strings, etc.

If the backline is being provided, spend a little time in sound check becoming familiar with it, i.e., getting the desired sound from an unfamiliar guitar amplifier. Don't expect engineers to do this; their job starts when sound comes out of the amp, and what happens before that is the domain of the musician.

I've lost count of the times when guitarists have no signal coming out of their amps, so they turn to me, shrug, and expect me to sort it out. All I'll do is logically trouble shoot the signal path, which is another thing musicians should be able to easily do for themselves.

REMEMBER WHERE YOUR EARS ARE

In many years of working in live sound, I've never come across any guitarists with ears in the back of their knees, so be aware of this when positioning instrument amps on stage. Elevating or tilting an amp so that it points at players' heads (and thus their ears) can help a great deal in being able to hear themselves.

Not only does this foster playing to full potential, it also assists in keeping the volume level of the amps lower, giving engineers a fighting chance of getting the vocals on top of the mix. I always ask guitarists to set the level of their amps so they can hear themselves clearly above the drum kit and let me do the rest.

If there's more than one guitar in the band, take a little time to make sure that they're sonically working well together – too much bottom end can make a mix muddy and too much top end can unleash the loudness wars that typically ensue when two (or more) guitarists fight to hear themselves (which can quickly escalate and obliterate the mix in one fell swoop of collateral damage).

In my experience, very few musicians think about how the sound of their instrument fits together sonically with the rest of the band; they just generate the noise and hope the engineer can make sense of it. But a little time and thought during rehearsals can help a great deal in getting the balance right at the source and thus make the engineer's job so much easier – and produce an optimum result, which is the main goal.

Putting on a live show is a team effort, so by treating engineers with respect and courtesy, chances are that they'll treat you likewise and everyone will have a fun, productive and rewarding show. LSI

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

Cover Story

THE REAL THING

The success is in the details with Duran Duran on tour.

by Gregory A. DeTogne, photos by Todd Kaplan

TRUE, THEY MAY HAVE begun as a group of art school, experimental rockers from the UK that quickly rose to superstardom in the 1980s on a wave of polished good looks and MTV-driven hits, but those that initially dismissed Duran Duran as simply another manufactured, throw-away pop group got it dead wrong.

Still going strong today with four of its five original members intact, the band has placed 21 singles on the *Billboard* Hot 100 to date, sold over 70 million records, and continues to burn it up out on the road with this year's Paper Gods Tour, supporting an album of the same name released last September.

As in the past, sound reinforcement this time out was provided by Eighth Day Sound (Cleveland), with veteran front of house engineer Snake Newton showcasing both his own skills and the power of an Adamson Systems E15/E12/E119 rig from behind an Avid VENUE | S6L, a console also being used on the tour by monitor engineer Charlie Bradley. With the tour currently playing an eclectic mixture of sheds and arenas, a typical configuration of the arrays utilizes 10 E15 mains hanging in the mains per side along with two or three of the wider dispersion E12s underneath.

Arrays of six E119 subwoofers additionally go airborne, with another half-dozen relegated to the ground. Coverage is further extended with out-firing arrays, flown next to the subs, made up of E15s over E12s.

"I'm a big fan of 15s, especially when you have big drums and Les Paul sounds like this show does" Newton admits. "After a couple hours of using 12s for this sort of music it starts to sound like they're trying too hard. These boxes do a great job, all while keeping our footprint fairly compact. They aren't a one-trick pony either: The top-end is really sweet, they are very hi-fi. Overall they are quite capable of doing a lot more than just providing big, beefy brawn."

FITTING IT ALL IN

At this point in their career, Duran Duran has such a huge catalog of material that it's next to impossible to play all of their hits within the confines of a two-hour show and still have time to work in new material as well. "They've been through so many incarnations in terms of their music," Newton notes, "and reinvented themselves so many times. That's really a big part of the challenge here. Your drum sounds, for example, have to match the drum sounds of the era of each song. Effects, the gating, tonality of the instruments, it all changes constantly."

Even a casual listen to the set list illustrates Newton's point. On a song like "Ordinary World" from the band's mid-career, it's not very warm sounding at all – the snare drum explodes in a burst of gated reverb, the melodies are raw and edgy. Newer material, conversely, is more disco-punk, with dry, sort of fat sounding drums. Then there are songs like "A View To a Kill" which doesn't sound like any other, and tunes developed with hip-hop producers.

"Our aural palette is indeed quite diverse," Newton (who hails from the UK as well) says with an air of English understatement. "I found long ago that getting each song to sound like the real thing – what the fans want and expect, in other words – requires its own snapshot.

(Above) Duran Duran performing on the current tour, with frontman Simon Le Bon on a Shure wireless system with BETA 58A capsule.

(Left) One side of Adamson Systems E Series arrays in place prior to a recent show.

Each is so unique that if I loaded the one I've created for the song 'A View To a Kill' for another like 'Girls on Film,' it would sound like a bomb going off in the arena."

First developed back when he was on an Avid VENUE Profile, the snapshots evolved into a collection of 75 in total, representing songs that the band had, at one point or another, played live. When the time came to transition to his current VENUE | S6L, Newton could've easily transferred everything he'd done over to the new platform, but chose to start from scratch instead.

"I was running a lot of Waves plugins on the Profile," he explains. "And the S6L doesn't run them. That's one of the main reasons I chose to start anew, and in retrospect, it was a wise decision. My snapshots had become like an old tree with branches rambling and winding in every direction. I realized that I'd be better off cutting it down and starting with a new one. In the end I was pleased with the results. I've found myself not missing Waves as the desk is so good sounding."

Neither Newton nor Bradley were out of inputs on their old Profiles, but the latter had maxed out his outputs and was resorting to all kinds of desperate measures to boost his capacity. When it was announced that the S6L was available, Bradley was right on it, and the first day the band heard the new board in their ear mixes they were sold. Newton followed suit, and spent a couple of weeks at home with his new desk building session files using a multitrack recording and gain settings established by Bradley during an arena tour.

A BIT OF THE OLD

Despite the presence of 300 processing channels on the S6L, Newton still maintains a collection of outboard gear. An avowed fan of Universal Audio plugins, he subscribes to the notion that you can't do a Duran show without RMX16s on drums, and to that end the authentic RMX16 digital emulation developed by AMS Neve and Universal Audio as a plugin found its way into the UA Apollo interfaces he keeps in his rack.

A Lexicon emulation is also kept at hand for lead singer Simon Le Bon's vocals, while an Eventide Harmonizer emulation stands in with effects on bass guitar and certain songs. A DBMax broadcast level maximizer from TC Electronic adds further dimension to Newton's sound, serving as a five-band limiting device additionally loaded with dynamic EQ.

"The thing I've found while working within the digital world for so long," Newton confides, "is that the nicer and cleaner it gets, the more I want to sneak in a bit of old analog sound."

Eclecticism is the rule when it comes to stage inputs, with offerings spread about from Sennheiser, AKG, and Shure. Vocal mics across the board at three positions including center stage are wireless BETA 58As. A BETA 98A captures saxophone,

Front of house engineer Snake Newton at his Avid VENUE | S6L.

while Shure wireless is the choice on guitar and bass. With drums, pretty much everything is miked from within – with snare being one exception. ("This way, you can throw the toms into a ridiculous state of reverb without getting spills from the cymbals," Newton contends).

Starting at the bottom with kick drum, a classic BETA 52/91 combination resides behind the drum head. Sennheiser 604s can be found inside the toms. Snare top is indeed miked outside at the top with a BETA 56, but at the bottom inside you'll find an AKG C418 condenser. AKG C414s ride overhead, and on hi-hat there's a C414 and a Crown CM310 mounted underhead.

The band universally relies upon Sennheiser in-ear monitor systems using JH Audio Roxanne earpieces. The only real loudspeakers onstage are a drum sub and a pair of 10s behind the bassist that are little more than props given this player's propensity for roaming the stage far and wide. The guitar backline includes a Kemper profiling amplifier running direct out the back; a speaker on its upstage left is there for use only when feedback is desired.

NO HIDING PLACE

Kyle Walsh serves as crew chief from Eighth Day Sound. Joining techs on the crew managing guitar, bass, and drums is keyboard tech Ozzie Henderson, a man tasked with keeping keyboardist Nick Rhodes' world in top running form. In a move designed to eliminate the need to travel with the various unreliable vintage keyboards needed to produce Duran Duran's wide-ranging

A perspective of the very clean Duran Duran stage.

Racked-up Sennheiser and Shure wireless systems, joined by Avid stage boxes, adjacent to the monitor position.

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sound, Rhodes took a turn into the virtual world with an Ableton sampler serving as the single platform from which everything would emanate.

Rather than use emulations of the classic synths and keyboards he has used over the years, Rhodes decided that since he still owned his vast collection of Jupiter 8s and every other instrument he'd ever counted upon in the band, he would sample the real thing right from the actual original sources. Henderson accepted the unenviable task of sampling every note on every instrument, and sometimes looping them to obtain a sustained sound. While requiring a tremendous amount of effort on his part, the results are nothing short of authentic.

"Details mean everything to Duran Duran fans," Newton adds, giving clarity to his role and the work of his peers. "They come to these shows wanting to hear every note that every player onstage produces. If a gate doesn't open on a floor tom, it will get noticed. There's no hiding place out here, and there shouldn't be.

"Along with every detail, every nuance, it's my goal to bring excitement to the crowd with my mix. There is a certain amount of theatrics required. This is not a timid mix at all, at times it's even overblown and needs to consume everyone in the room. I approach it all not with big fader moves, but a lot of important, small fader moves. That leaves me with the space to feature the passages everyone wants to hear.

"It may be my mix, but the music in its entirety belongs to the crowd." LSI


Monitor engineer Charlie Bradley also switched to the new VENUE | S6L.

Gregory A. DeTogne *is a writer and editor who has served the pro audio industry for the past 35 years.*

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TAKING THE INITIATIVE

Monitor engineer and mentor/ educator Karrie Keyes

by Kevin Young

KARRIE KEYES ENJOYED a unique view of the 1980s-90s punk rock scene as well as the rise and evolution Pearl Jam, one of the more popular groups of the past 20-plus years. In fact, I recently caught up with the noted monitor engineer during a break on the band's current tour, which finishes up later this month with dates at Fenway Park in Boston and Wrigley Field in Chicago.

Keyes' career also includes live mix work with RHCP, Fugazi, Sonic Youth, solo projects by Pearl Jam lead vocalist Eddie Vedder, and briefly with Neil Young, among others. Although based in California for most of her 30-year career, she also lived in Seattle briefly during the height of the Grunge era. Eventually, though, she says she'd like to work out of New Orleans.

"I'm trying to make that work," the 48-year-old engineer says. "My kids are almost out of the house and I'd like to be doing more, possibly one-off work, when Pearl Jam's not touring."

MAKING A CHANGE

Born and raised in Los Angeles, Keyes fell into audio by chance, she says, although she'd always gravitated towards arts and music as a teenager. "I played flute and clarinet from 4th grade through my two years of high school, but I didn't have the dedication to play professionally," she says. "At that point no other music career options were offered, at least through school."

At the time (the mid-1980s), careers in audio weren't something often presented to people – particularly young women – as an option; something Keyes and front of house engineer Michelle Sabolchick Pettinato have set out to change by founding Soundgirls.org. Primarily aimed at supporting women in professional audio by providing mentorships, forums, content, and development tools to members, the community welcomes men as well. In fact, Keyes estimates that men make up 35 percent of the membership.

Still, she points to a lack of encouragement provided to young women interested in working in audio as the driving force for Soundgirls. Additionally, in part, the organization's website is also meant to compensate for diminishing arts programming in public school curriculums, something that Keyes noticed when her twin girls moved from the Montessori schools they'd attended to public high school.

"I was fairly horrified by the lack of arts funding," she says. "When I was in school, if I didn't have arts and English programs, I would have dropped out. That was a guiding light when we started Soundgirls."

The intention is to provide resources for young people starting out in production, as well as for those currently working in the industry, along with a means of connecting industry veterans and emerging professionals to share advice and provide support.

INSPIRING ENCOUNTER

It's not the kind of resource Keyes had access to when she started her career in audio in 1986. And it was only by chance that she got it, making a serendipitous decision to attend a Black Flag show instead of going with friends to a concert by the band Fear that was being staged closer to home.

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Attending the Black Flag gig meant public transit and potentially being stranded in LA all night, but Keyes, never one to do things the easy way, went anyway. Before the show, she made her way to the front of the audience and started talking with a young mix engineer named Dave Rat, owner of then-fledgling audio company Rat Sound, who was working the gig. It's a meeting she credits with inspiring her to pursue a career in audio.

Keyes explained to Rat that she wanted to learn about what he did. Her first lesson was wrapping microphone cables, she adds. After a subsequent gig in Palo Alto, she returned home certain of her future career.

"That experience highlights what I tell people starting out: seize opportunities. I was in the right place at the right time. I couldn't walk into Rat Sound now and say, 'I want to learn how to do sound.' At that time, however, the company was a couple of people doing punk rock shows who said, 'Sure, come help. We need help,'" she notes, laughing.

Personally and professionally, Rat and Keyes made a good match, and soon the two became a couple, working shows together and living in a warehouse – without hot water – for two years until building up enough work/income to afford to live elsewhere. "Until then every dime went back into the systems, insurance, or gas," Keyes says.

"We were together for 10 years," she continues, "and we raised two amazing daughters – now 20 years old – while working in this crazy industry. We actually decided to split up when I was pregnant, realizing our relationship wouldn't last. And it's impressive, looking back, that we could make that decision calmly and rationally, and we've co-parented the girls ever since."

Keyes working with fledgling young audio professionals at a Live Sound Camp for Girls.

LEARNING THE CRAFT

Doing so while managing conflicting touring schedules wasn't always easy, but, as one of relatively few women in audio, Keyes is no stranger to challenge. Early on at Rat Sound, she did whatever needed doing – loading and unloading trucks, working every show she could, learning the gear, the craft and how to troubleshoot. The pay was minimal, but the experience integral to her career.

Some jobs were a bit daunting, she notes, adding, "But it's all relative. The less experience you have, the scarier the gig. At least they should be. I've seen people who say, 'I've got this, no problem.' And I'm watching them, thinking, 'You don't have this. Slow down, ask for help, and people will help. You should have some apprehension if you don't have experience."

One of Keyes' first regular gigs was with The Untouchables, as system tech for their Southern California gigs. Later, when Rat, then the band's monitor engineer, moved to FOH, Keyes took over on monitors through to 1990.

She continued working with Rat Sound for 20 years, finally leaving in 2005 to focus her efforts exclusively on touring and live sound. "I did the books for years and paid the bills because I was better at that than in the shop with a table saw," she explains. "The last five years I mainly did human resources and management, but that wasn't what I wanted to do."

STAGE MATTERS

Although Keyes has occasionally mixed FOH, her preference is monitors. Beyond being able to mix quickly, learn from your mistakes and consistently improve your chops, it's important, when mixing for the pickiest listeners in the room, she says, to realize that you're providing a service.

One of the most important things for a monitor engineer to understand, Keyes insists, is to know when to do nothing: "With Pearl Jam there are set cues, and once the rig is up and running there shouldn't be a lot of changes. If there are, something's wrong."

She adds that when engineers try to mix monitors as if practicing their FOH mix, it's frustrating for the band. "I see that and I'm horrified and they're like, 'Well you're just sitting there doing nothing,' and I say, 'Yeah, that means everything is going great."

When things do go south, she advises: "You can't take it personally, because then nothing's going to be solved and everybody will be upset. I think sometimes engineers don't take into account how stressed the musicians may be about the gig and interpret it as being difficult.

"When you're starting out, probably the best trait to have is to actually care about the gig and not come off as, 'I just want to move to FOH.' You have to be willing to listen. Even if what the musicians are telling you doesn't make sense you have to figure out what they need and make them comfortable and secure so they can concentrate on performing."

Even when a touring unit has been together as long as Pearl Jam and their crew – essentially becoming family – Keyes says, issues still crop up that require tact, diplomacy and patience in abundance.

That said, the fact that the band's system has remained relatively consistent over time, helps mitigate those issues.

"They've always toured with production and the monitor system doesn't change much tour to tour, so we're not starting from scratch and we're able to focus on what's working and what isn't." That said, "Pearl Jam's always had volume issues," she adds. "So the question is always, 'How are we going make it comfortable for everybody on stage so no one's losing their sound."

DIFFERENT APPROACHES

As a result, the monitor system has evolved over time. Currently, the band utilizes a mix of in-ear monitors and wedges; specifically Rat S Wedges – which only go out with Pearl Jam – for Vedder, and EAW Microwedges for the other band members.

"About 10 years ago we convinced the band that ears were the way to go and everyone was on ears except our drummer, Matt Cameron, but now IEMs seem to be going by the wayside," Keyes says. "I would say at this point only our lead guitar player, Mike McCready, relies on 'ears.' Everyone else has a combination of wedges and uses either one ear, ears in/out, or ears not on, but could be. At this point there's two stereo ear mixes, two single/mono ear mixes and then our singer and drummer are not on ears."

She also cites a recent move to a DiGiCo SD5 from her previous desk as helping to solve roughly 95 percent of the issues they had while also improving workflow and sound quality on stage.

Her approach to monitors with every band she's worked with is similar: "I just talk to the band and figure out what they need, and I'm fortunate that I've been in few situations where I walk into rehearsals without knowing the band." As examples, she cites RHCP, who she worked with from 1990 to 2000, and Fugazi, which hired her because she mixed their local shows regularly and knew their needs.

"But if I were to do a Fugazi gig tomorrow, I wouldn't use a Pearl Jam monitor system. It would be minimal," she notes. "The first tour I did with Fugazi, a few days in they said, 'The monitors are kind of hard right now.' And I said, 'What's going on? They said, 'Well, before we had you the monitors sucked. That made us angry and fueled our performance. Now the monitors are fine and there's nothing making us angry.'

"So I said, 'Sorry. I could mess them up for you.' It's funny what drives bands," she adds, laughing.

Carolina Anton, head of the Soundgirls Mexico chapter (and live sound engineer), with Keyes.

Keyes has enjoyed numerous standout moments during her career. Among them mixing monitors for Neil Young on the Mirror Ball Tour and RHCP at Woodstock 3, but she remembers her time with Fugazi particularly fondly. The no-frills approach the group took to production required a lot of work, but owing to their fiery live show and dedication to performance. "There was a magic to it," she says.

GOING & GROWING

Keyes' main gig remains touring with Pearl Jam, and while her passion for mixing is undiminished, it's matched by her commitment to Soundgirls: "It's a far-reaching, all-consuming passion." She and Pettinato launched the organization in 2013 after having met, for the first time, as participants on an AES Conference panel – "Women of Professional Concert Sound" – the year before in San Francisco. "We had a great time in a room with other women, our peers, and I think that was the first time that ever happened for any of us," she says.

Initially, the founders envisioned Soundgirls "as a place to network and find other women in our jobs," but since, operating under the sponsorship of The Northern California Women's Music Festival, it's grown substantially. For example, one recent initiative is a series of four-day Live Sound Camps for Girls in four different cities in the U.S.

While Keyes believes that women are much more readily accepted in audio and the production industry in general than when she started out, in her view the need for a community like Soundgirls remains important. "I've always worked with an amazing group of men who have always been supportive. I've never felt I was working with sexist jerks or had to deal with sexual harassment."

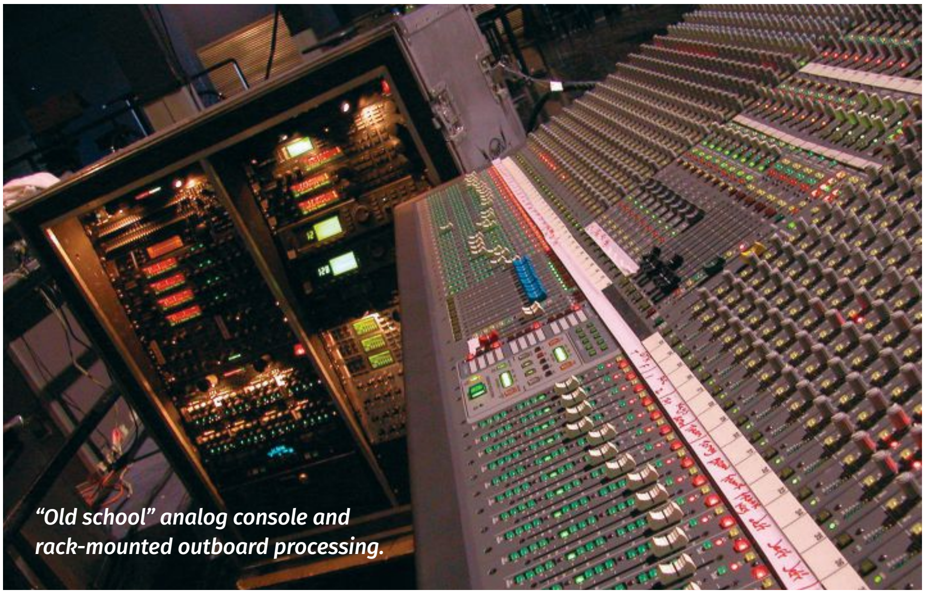
Still, given that women are estimated to make up only about five percent of the workforce in professional audio, the more visibility and resources Soundgirls can provide, the better. "You can't be something that you never see," she concludes, bluntly, "and trade magazines can't feature a women every month but we can – so we interview women each month, publish their stories, and they become role models." LSI

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

ANALOG & DIGITAL

Setting up and working with both types of consoles.

by Craig Leerman



"Old school" analog console and rack-mounted outboard processing.

One day on a freelance gig I walked into the room to discover that the A/V company provided me with an older analog console with two racks of outboard gear. While setting up front of house and patching in all of the effects and processing, I found myself wishing for a digital console.

The very next freelance gig I was presented with a brand-new digital console and no extra gear at FOH. As I waded through menus trying to set up a console that I 'd never used before and struggled to read the manual I'd just downloaded on my phone, I found myself wishing for an analog console and some simple outboard processing.

The death of analog consoles and processing hasn't happened yet, nor will they go away in the foreseeable future. Many sound companies and installations still utilize analog components, and there are "old sound guys" like me who still like to grab knobs instead of sort through menus. For shows with just a few inputs, analog is usually the most cost-effective choice, and for some larger shows they're a proven item that (often) sound great while offering all of the necessary features needed.

If you work in only one venue, you learn the console and system, and have adapted a workflow for that gear. If you freelance a lot like me, there's the need to adapt to whatever console and equipment is provided. Here are some of the approaches I've developed over the years.

BEING READY

I have slightly different setup routines depending if it's a digital console or an analog model with outboard processing. The first thing I do, if at all possible ahead of time, is ask what gear is being provided so I can download the manuals and quick start guides if I don't already have a copy. There are folders on my laptop filled with manuals for consoles, outboard processors and recording units, so answers can be found quickly at a show even if I don't have internet access.

If I find out about the gear in advance then I print out a quick start guide and/or pertinent pages of a manual (like how to change aux sends from pre to post) for quick reference at the gig. It's also a good idea read the manuals on unfamiliar gear before the show.

With analog consoles at FOH, I start by setting up the outboard racks close to the console so the patch cables can reach. Depending on how the equipment is packaged I may place the console on top of the outboard racks to save space or put the outboard racks off to the side at a 90-degree angle.

Next up is figuring out what's needed with respect to inputs, outputs and processing. Digital consoles usually have processing available for every input and output, but analog requires a bit of forethought, including strategies and "workarounds" when processing channels and devices are limited.

PATCHING IT UP

Once the game plan is ready, I kick things off by patching the outputs for the main loudspeakers as well as any delay and fills. This involves running short patch cables between the console outputs and the outboard EQs in the rack, then patching the amplifiers (or powered loudspeakers) from the outboard EQ units. There may be the need to patch in a delay unit as well if running remote loudspeakers. Once the loudspeakers are patched, it's time to can address any output feeds that are

needed, such as an audio send to video world or a feed to lobby loudspeakers.

Next up is patching in effects and channel processing like compressors and gates. Most analog systems that I run across have between two and eight channels of compression available, so it's a good idea to plan ahead with a limited number. With analog consoles, outboard processing is usually "inserted" into a channel via an insert jack. Some consoles have two send and return connectors that are usually 1/4-inch TRS (balanced) for inserting external gear, but more common is a single 1/4-inch TRS insert jack that provides an unbalanced send and return on a single plug.

Digital consoles, particularly more recent models, carry an impressive amount of processing onboard.

A special "insert cable" that consists of a 1/4-inch TRS plug on one end that breaks out into two 1/4-inch TS plugs at the processing end is used to interface the outboard gear. The cable is wired so the Tip of the TRS is the send to one of the breakout legs, the Ring is the return on the other breakout leg, and the Sleeve is wired to the sleeves on both breakout legs. I always carry some insert cables to shows as they seem to be the most forgotten item on the pack list.

Now it's time to plug in the inputs and label the console. I also take the time to label the aux sends as well as the outboard gear so I know where they're patched in.

A BIT EASIER, BUT...

Digital consoles are a bit more simple to set up as most offer the necessary processing onboard. Before doing anything with a digital desk, I like to start from scratch, wiping it back to the factory settings. Some models have a default scene that can be recalled, some require a few clicks in the menu, and others may require a boot-up while holding down a few buttons to reset back to the default.

My reasons for starting with a clean slate are simple. I don't know what the last user did and don't want to get "bit" during my show trying to make an adjustment only to find out that the last user changed a setting that was not readily apparent, like switching all of the auxes to post fader.

Once the console is reset, I still start at the outputs and make sure there's an EQ in-line with the loudspeakers. Many models have an EQ assigned to each output. but a few require assigning any needed processing to a specific output from a limited number of items onboard.

After outputs comes inputs, and I assign each channel the processing it requires. Again, some boards have a limited number of processing and effects, and these must be chosen and patched before they can be used. If the console has scribble strips, label each input and output as you go. I also like to label the console with tape, especially for items that don't have scribble strips like user-assigned buttons. At this point it's also a good idea to save the settings on the console in a scene.

If I'm using a remote app for mixing on a tablet, now is the time to make sure it's working and then walk around the venue to see if it stays connected around the room.

ANALOG TENDENCIES

On most corporate gigs and small festival band gigs, the supplied analog console usually has between 24 and 32 inputs and four or eight subgroups. Larger shows may get a larger frame board with 40-plus inputs and VCAs. Since VCAs aren't available on many shows, I use the subgroups to make my job a little easier by grouping like inputs together.

With four groups available on corporates, I normally place the podium mic into its own group, any presenter wireless into a group, table microphones for panel discussion into a group and Q+A audience mics in another. With four subgroups available on band gigs I split up the groups into drums, guitars, keyboards with horns (if any) and vocals.

It can help to add a bit of compression to a podium and lavalier mics used by presenters. With music it's likely to add some compression to kick drum, bass guitar, lead guitar and vocals.

I work with a lot of headline singers who have some background vocalists behind them. One trick that comes in handy when there's not a lot of outboard compression channels is to run all of the background vocals into a subgroup and compress the group as a whole, but insert a compressor into the lead singer's channel and then just run them straight to the L+R masters. That way there's tailoring available for the lead vocalist and the ability to add compression to the background singers, and without taking up an entire subgroup for one vocalist.

On corporate gigs it's not uncommon for me to get stuck


Inspiring
Every
Moment

with a very small analog console with limited channel EQ but there's a high-profile presenter who needs some drastic EQ help to sound good in the room. If the console has a channel insert jack, I insert an outboard graphic EQ or parametric EQ into the channel.

I carry a stereo 15-band graphic just in case there's not an extra one provided onsite. This EQ has often come in handy for the mains when the A/V company thinks that a small console with built-in five- to seven-band graphic is all you need and they don't provide an outboard EQ.

An insert cable with 1/4-inch TRS plug and two 1/4-inch TS plugs.

Other outboard gear regularly carried to freelance gigs include a stereo leveler, stereo compressor/gate and multi-channel feedback suppression processor. While some audio folks laugh when they see my feedback unit, they soon realize that it does a great job taming wireless lavalier mics because each of the 24 filters can have a bandwidth as narrow as 1/80 of an octave. It works great when inserted on the lavalier subgroup.

Another limitation on some smaller analog consoles is a lack of buses. More than a few times I've been mixing monitors from a smaller front of house board and have run out of aux sends. If the console has a matrix section, it can be used to set up side fill mixes, and in a pinch, some individual performer mixes.

STAYING ORGANIZED

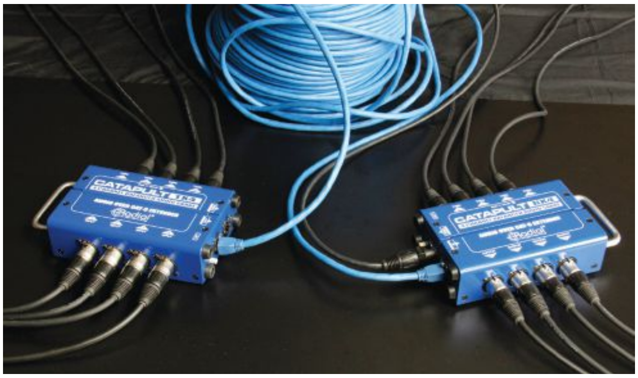
Even the smallest digital consoles usually include comps and gates on every channel but many of them lack VCAs (also called DCAs) or subgroups. An easy way to get subgroup control on a digital board that does not offer groups is to use a post fade aux bus. Simply assign every input channel that you want in the group to the same post fade aux send. Make sure to un-assign those channels from routing to the main L+R outputs. Now assign the output of that aux send to the L+R mains and it acts as a subgroup.

While the aux master may be on a different layer, many consoles have a user-configurable layer that can be tailored to specific needs. For example, I place my "money channels" like lead vocalist, lead guitar, podium, presenter wireless and others on the user layer, along with any effects masters and subgroups or VCA/DCAs. This way, mixing can be done mostly on a single layer.

One of the few drawbacks with digital desks, at least for me, comes when using a digital audio network instead of an analog snake. The problem is that an intercom channel or lighting DMX can't be run on the same cable as can be done if the analog snake has extra channels. Sure, the solution is running a few single cables from FOH to backstage, but finding hundreds of feet of extra XLR cable onsite can be an exercise in futility and carrying around hundreds of feet of "extra" cable as a freelancer is not an option.

My solution is to deploy a snake system that provides four analog audio runs down a single shielded Cat cable. A small reel with 100 meters of cable makes for a compact package I can keep in my truck. Radial Engineering, Whirlwind and others offer nice 4-channel snake boxes for Cat cable. In addition to comms and DMX, they work great for analog mics, line level returns and even AES signals.

Speaking of networks, Dante has become the de facto standard of audio networks, so I carry a few small gigabit switches to shows to help route signals. I also just added a small wi-fi router to my bag. It comes in handy for interfacing an iPad for remote console control and might also come in handy with the new version (3.10) of Dante Controller able to connect to networks over wi-fi. *(For more info see Loading Dock, page 9 in this issue.)*



Catapult, a recently released 4-channel Cat snake from Radial Engineering.

Finally, for almost every gig, I bring a small utility mixer just in case the console has a problem or a submixer is needed (or would come in handy). In the past this was an 8-channel analog unit but recently has switched to a very compact digital mixer. With 16 inputs that have 4-band EQ and onboard multitrack recording, I sometimes just replace the console provided by the A/V company with my own mixer and a small rack.

The key as a freelance audio technician is to be able to adapt to the equipment that is provided no matter if analog or digital, and to be prepared outside of that to make the show happen, and as well as possible. Making the client happy insures I get another call. LSI

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

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MOST IMPORTANTLY, QUALIFIED

A conversion from punk to useful human being. **by Sully**

EDITOR'S NOTE: This fine article was featured in the July 2005 issue. We reprint it here in celebration of our 25th anniversary.

Back when I knew everything, I dragged an obscenely heavy rig up an intolerably tall mountain in a woefully underpowered truck to an impossibly small theater in the backwoods of northern Pennsylvania.

Since this was when I knew everything, I had the somewhat surly local crew stack my compact TMS-3s two wide, four high, with the 15s coupled for maximum bass energy. Because stacking that box four high without a fork was sort of like pushing a dump truck up a hill with the brakes on, some of my help that day were engaging in a spirited debate regarding my experience level and the wisdom of my stacking choice.

Later, I began tuning the PA, carefully twisting up the low-frequency bandpass on the crossover while listening to selected cuts from *Back in Black*. I was shortly satisfied that when my client arrived and listened to my subtle, intuitive tuning, he would turn to me with a grateful smile and request the EQ across left and right be removed since there was obviously no need for it. I practiced my thank-yous in a bathroom mirror backstage to achieve a proper balance of humility and steadfast resourcefulness.

The artist that day was Frankie Valli. About an hour later Jim Sanders (a.k.a. "Redford") strolled in, walked out to front of house, looked at the PA, looked at me, looked at the PA again, then stuck out his hand and said, "I'm Redford... please don't tell me you have the 15s coupled."

This is not in the script. "Um, yeah... Since we don't have additional subs, that's what we do," I uttered with humility and steadfast resourcefulness.

Redford: "Kid, don't your parents have any Frankie Valli records? This is a coverage show, not an AC/DC gig."

Me, summoning all of the forces of good, told him not to fear, I'd just re-stack the PA. He peered at me with kind eyes that spoke the words, "Thank you, you are truly a professional." Then his mouth added something like, "Buddy, that crew is gonna play tether ball with your intestines if you tell 'em to pull that PA down."

Since I knew everything, I was sure that my wizened 22-year-old brain would have a solution soon... if I just gave it a minute or so. Unaware of the delicate cerebral dance transpiring in my head, Redford continued, "Ah, screw it, I'll deal. Otherwise you appear to be about the right size for one of those (expletive) guys to gaff tape a plumbing fixture to your head and stick you in a trash can at load out... Come on, let's get a drink. You like scotch?"

I do now.

These encounters have always reminded me of a late-night talk show with people like Redford as the unexpected surprise guests. You know, you're expecting the guy from the San Diego Zoo to come out with a cockatoo that's destined to crap on someone's shoulder when suddenly, Bono appears to do an acoustic version of the first four songs from *Joshua Tree*.

They're like snow days in July. Situations that are dripping with the hallmarks of misery *by audio*, but miraculously are rescued by something very simple – in this case, a personality. Inside of a short three-minute conversation, Redford had managed to convince me to join his team. Instead of me versus him, with me grudgingly doling out small concessions to his wishes, I began suggesting more work for myself: "Redford, you want I should tie into the house delays? You want the front fills in stereo? You want me to put a bag over Frankie's head so he stops making that noise?"

To be clear, I was not an engineer sycophant when I was a puppy. But I had concluded – based upon my hours of experience – that force of personality and the imitable ability to *hang* typically superseded any true mixing ability.

This mangled Aristotelian logic brought me to some simple conclusions: the louder and friendlier a guy was, the less qualified he was. Quiet and unassuming? Well obviously *still waters run deep*. Time would later prove this and many of my other theories worthless, but I was 22 and knew everything.

Turned out Redford would be the beginning of my conversion from punk embryo to useful human being. He demonstrated to me that in addition to being loud, funny, and sarcastic, he was most importantly *qualified*.

He was batting in the *Show* and I was pretty impressed with myself for hitting off Little League pitchers. Never once, though, did he point that out to me. He so dwarfed me in skill, yet had absolutely no need to mention it. In fact, quite the opposite, during the day he would ask *me* questions and solicit *my* opinions.

At night, he'd spin up a mix that made Frankie sound way better than he had any right to, finish his drink, shake my hand assuring me we'd "fooled them again," and then disappear. I'd spend the rest of load out striking the PA while collecting murmured compliments from the exiting audience. With each one I dutifully disavowed my role in the night's success promising to pass the praise on to the mix engineer. And I always did. Or I mostly always did. Secretly I hoarded small pieces of each note and nod, hoping later to puzzle them together into a map that might point *me* to the *Show* someday.

I lost track of Redford. No matter. I've never lost track of the course he accidentally or purposefully set me on.

Had he spoken *at* me, I would have turned all medieval and disaffected 20-something on him, demonstrating my lethal ability to maim with a single contemptuous roll of my eyes. Instead, he *included* me as an equal part of his adventure, casually pointing out things he had discovered along the way. I was free to participate or not, but I was welcomed along if I wanted to come.

When it came time for the show, he would continue to point out the foibles of the world with one hand, while with the other, he made the PA broadcast his right to speak. I think about Redford sometimes when I'm faced with someone young, eager and already in possession of the secrets of the world. It causes me to pause and consider inflicting patience on them instead of strychnine.

Don't get me wrong, I'm perfectly comfortable vetting young engineers by drowning them under mountains of broken cables and castors to see how badly they really wanna play... But I know in doing that, I'm striking a bargain to pass on information that was passed on to me. The method of delivery will determine how well they absorb those lessons.

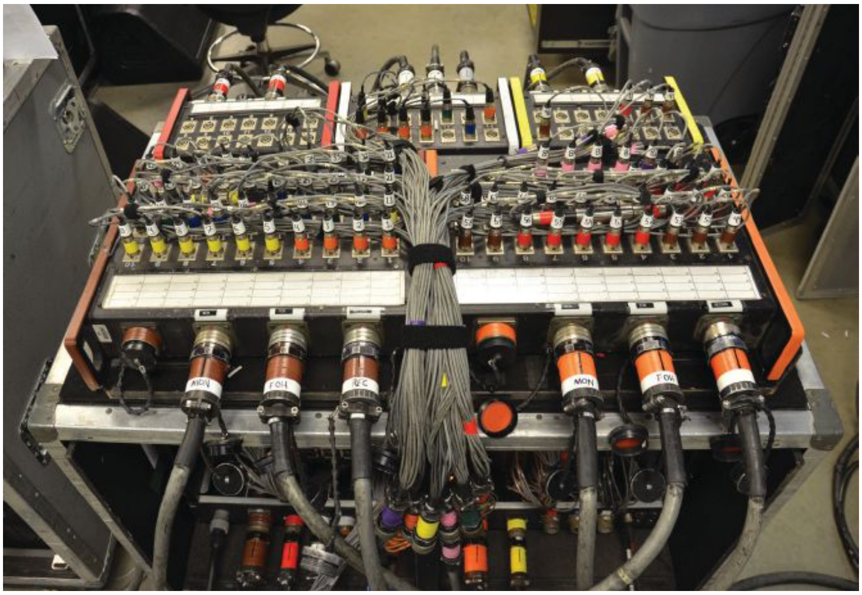
In the end, they're the next round, no matter what we say. My generation will pull the bus door shut after our final load-out, and then it'll be their turn to live out all of the Jackson Browne songs. Might be nice though if we offered them a drink when we see them off to the *Show*. LSI

***Sully** is a long-time live sound professional and mix engineer who's toured with several top artists.*

SPLIT WORLD

Passing live audio to simulcast, stream, broadcast, and record.

by Gary Parks



PHOTOGRAPHY BY COURTESY OF CLAIR GLOBAL

A live concert or event often serves a wider audience than those seated in the arena. In addition to the audio in the house and video at the side of the stage, it might also be simulcast to a separate location, streamed on the web, recorded for archival or other purposes, and/or broadcast via radio or television.

To accomplish this task, audio signals from microphones and instruments must be split beyond the traditional front of house and monitor positions and sent to other locations, be they other control rooms at the venue or inside production trucks. Complicating matters more, these locations will probably be on different legs of the electrical system, potentially adding ground loops and hum to the signal.

With the more widespread adoption of digital consoles and networking in live sound applications, there's much more potential to access the same signals and be tied together in ring or other topologies with the audio remaining in the digital domain. I was curious whether (and how) these newer technologies are being used.

My search began at the Monterey Jazz Festival, where the main stage shows are simulcast to the Jazz Theatre and other locations onsite as well as often shared via radio or webcast. McCune Audio (San Francisco) handles the audio and video for the festival. I then talked with folks at several major sound companies about how they handle the issues of signal splitting, grounding, and interfacing consoles.

MAIN STAGE SIMULCAST

The 6,500-seat arena at the Monterey Jazz Festival, where the main acts perform, is the focal point of the simulcast and broadcast activities. DiGiCo SD10 consoles are located at FOH and monitors, with their D-Racks at the side of the stage near the monitor station. Four video cameras are positioned in the house for full-stage shots as well as on either side of the stage for close-ups.

The video and simulcast control center is set up behind and underneath the main stage. All camera shots are called from this location, audio is embedded with the video, simulcast feeds are processed and distributed, and in 2014, the show was also webcast. Archiving is done from a production truck outside the arena, with an Avid VENUE Profile console building its own mix.

This mix is also sent to radio, as it has been every year with the exception of the 2015 edition of the festival. The raw feeds from the onstage cameras are provided to screens at both FOH and the archiving truck so that the engineers have visual information about what's happening on stage, even during changeovers between acts.

To accommodate the different mix requirements for FOH, monitors, archiving, and simulcast, stage signals are split three ways. Though all mixing consoles are digital, with A/D converters on stage, the split itself is analog via a custom Ramtech STGBX-54 three-way splitter. Each of the 54 channels can be input directly or via four 12-channel and one 6-channel Ramtech CPC onstage sub-snakes.

The SD10 console at FOH is directly connected, and the monitor and archiving consoles receive transformer-isolated feeds. "Any mic that's in the system, even if it's only for a record input, has to connect to FOH for phantom power," adds Nick Malgieri, FOH mixer.

From the splitter, the stage signals go from each of the three multi-pin outputs to the A/D converter boxes. Both FOH and monitors use two 32-input DiGiCo D-Racks, with an optical fiber loop to transmit the signals to the consoles. To feed the Profile console for archival recording, the third split goes to a VENUE Stage Rack using MADI digital protocol. The video control area receives two separate stereo mixes, from FOH and the archival truck, and uses one or the other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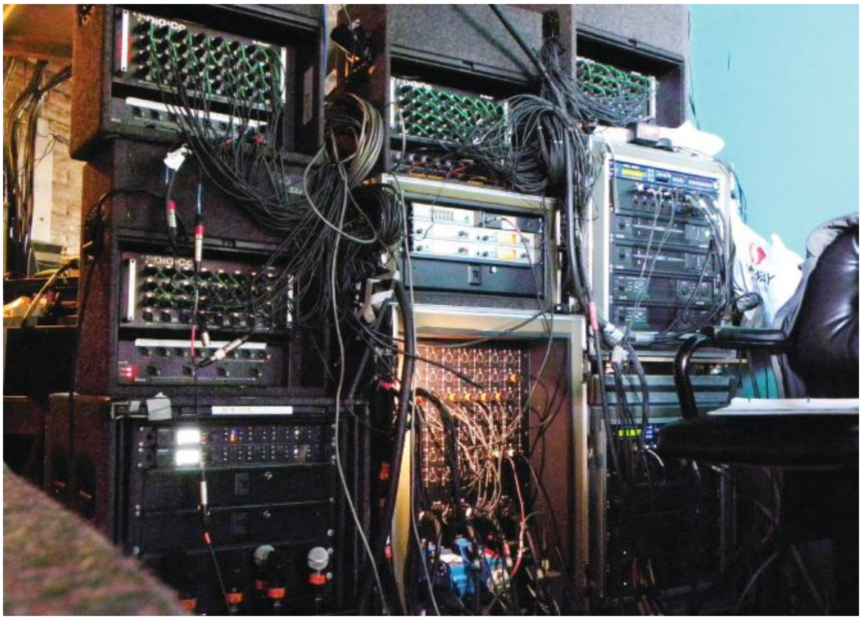
to embed with the live-edited video signal – which goes from there to simulcast, recording, and webcast.

On the decision to use an analog split rather than digital networking to share audio signals, Malgieri explains, "There are ways we could network it all together, but for speed and efficiency we keep it an analog split. That way, no one is tied to anyone else. Sharing preamps together would make us interdependent, which wouldn't be conducive to a festival-style event with fast changes and guest engineers."



PHOTOGRAPHY BY EVA BAGNO

"Split world" adjacent to the monitor position at the Jazz Theater at Monterey, the focal point of simulcast and broadcast activities.

RECORDING & ARCHIVING

Ron Davis has been mixing and producing the Monterey Jazz archival recordings for many years. The mix is independent of FOH, starting with the raw signal "straight off the mic" plus the audience mics above the stage and in the house for crowd response. Having his own multi-channel feed from the stage allows him to "fine-tune the mix for recording purposes," he notes, since his environment is more conducive to critical listening.

The VENUE Profile console interfaces with Pro Tools, and Rob Macky monitors the recording along with other technical details. Other members of the team include an onstage liason, who is in touch continually with a comms person in the truck (also connected with FOH, video, and other positions throughout the venue), and another who archives the recordings to digital media as soon as they're finished.

Davis says that 48 channels are usually more than enough for the acts plus the audience mics, though at times he needs to drop a couple of inputs. In those cases, he may choose one of a stereo pair of mics or just use the DI from the bass rather than adding the mic on the cabinet. His mix is patched to the video area as a potential simulcast feed, and is fed to any radio broadcast trucks airing the show. The archival truck also receives the FOH mix for redundancy.

MONTEREY JAZZ SIMULCAST

Because simulcast is an important feature at the festival, a control area is designated for video and tasked with live video for the side-stage screens, creating the simulcast feed and monitoring the venues where it plays, video archiving and performance MP4 recordings for the artists, and at times, webcasting. The crew includes the camera operators, directed by Jesse Block from the control room, a person controlling the video and audio embedding, another on recording, and a "grounds technician."

Simulcast receives mixes from FOH and the record truck, plus an ambient mic submix. Malgieri states that "Depending on who's ready first, video makes a judgment depending on what's coming down the pipe on which mix they're going to go with. This could change for each act." Also, having both mixes available provides a backup in case of trouble, giving the same content but different mixes.

The simulcast feed goes to the Jazz Theatre, where patrons who have purchased ground passes for the other stages can experience what's happening on the arena stage. The signal is sent about 750 feet via fiber to the theatre, and then decoded into L/R audio to full-range loudspeakers and subwoofers, with video projected on a large screen. The Premier (VIP) Lounge is a smaller venue, closer to the arena, and it also receives the simulcast via HD/SDI.

SIGNAL SPLITTING ON THE ROAD

Beyond Monterey, I checked in with several other touring companies to learn how they handle signal splitting for live events, especially when multiple splits are required for broadcast, recording, or similar applications. In most cases, an "old-school" analog splitter with transformer-isolated outputs is the rule on the road (at least among those I spoke with), rather than sharing a common digital signal among the various applications.

Dave Skaff, senior tour support for Clair Global (Lititz, PA), says that "There seems to be two distinct camps between the live world of traveling music and broadcast. The live mixers are very ingrained with having their own head amp control. To give them that control, a digital split is kind of ruled out." A downside is that each console position needs its own stage racks with an analog split. Continuing, he adds, "In the broadcast world, the idea of using one set of head amps, and having several people follow with digital trim or some kind of gain tracking is a fairly accepted way of doing

things – their comfort level is higher."

On a recent U2 tour, Skaff notes, "We did entertain the idea of having digital splits, with certain people having control of stage racks and others using digital trim for levels." During the planning, he adds that there were incidents where, if the digital loop went down, "you lost a lot of control." The show was especially complex, with six different consoles that would be on the loop – FOH plus a backup, and three separate monitor setups with a backup.

Skaff notes that some of the tour staff's fear of relying on the newer technology came from second-hand conversations they've "heard from others," plus Clair's own observations of small glitches that persuaded them to stay with the tried-and-true methods. The company went back to a custom-designed 6-way analog splitter for the U2 shows so that each console would have control over levels, with proper loading for six mic preamps per channel and transformers that accept a wide range of signal levels without saturation. There were also conventional splitters that facilitate 3- or 4-way passive splits.

For a show that also needs to accommodate broadcast trucks, the engineer might ask for an isolated analog split or a digital split sent to them as AES3, or possibly a MADI split off the stage racks; Skaff has had all of the above requested recently. Many tours will provide an open analog split, available for a recording truck or other production application.

SMALLER OR LARGER FLEXIBILITY

Dave Rat of Rat Sound Systems (Oxnard, CA) discussed with me a "baseline method" of signal splitting, using a custom-built XLR panel with two 56-pair Whirlwind W4 MASS connector outputs. For smaller shows, the choice is usually a single panel for FOH and monitors, while for larger shows or ones that require separate recording or broadcast feeds, the approach is multiple panels with a single input and a pair of ISO outputs – with the direct signal going to FOH.

Rat Sound has also designed multi-connector panels that are fed from stage boxes, and by changing the tails, the switch between opening and headlining acts can be accomplished more quickly and reliably. Occasionally there are bands where both the FOH and monitor engineer are working with the same mixing console, and each will use a common digital split; any additional feeds for a production truck are likely to come from the analog splitter.

He observes that recording seems to fall into two categories. The first is an isolated split recording where the signals go to a recording truck, or in the case of the Red Hot Chili Peppers, to a console located in a remote room in the venue that is fed from a separate A/D rack stage-side and mixed there. This mix might go to broadcast or another application. The second is that the FOH or monitor console sends a recording feed, taking the mic outputs in their raw form to a multitrack recorder.

Greg Snyder of Thunder Audio (Livonia, MI) also confirms that even though many of the latest mixing consoles can share a stage rack and digitally split the signal, his team often opts not to share mic preamps and to use an analog split. He finds that off-the-shelf splitters can be very reliable, and that "with today's digital consoles, we find that passive splitters are very easy to use as go-to packages."

Snyder adds that quite often the FOH engineer will create a mix to be embedded with video, which is transported from the console to the video truck via an analog snake or fiber interface. He notes that mixing for both live and video "requires that the engineer be very conscious of the mix they're providing so that it will be usable for broadcast."

PHOTOGRAPHY BY EVA BAGNO

Ron Davis, mixer and producer of Monterey Jazz archival recordings, at his Avid VENUE Profile console.

HALL OF FAME & MORE

When I caught up with him recently, Mark Dittmar, the live broadcast events engineer at Firehouse Productions (Red Hook, NY), had just returned from the Rock & Roll Hall of Fame show, which he's worked for several years running. This year's event combined a live show for about an audience of about 15,000, plus broadcast, at Brooklyn's Barclay Center arena.

In addition to Firehouse's live audio setup that included infrastructure, splits and comms, All Mobile Video provided the television truck and a Music Mix Mobile truck did the audio mixing. Firehouse handles overall coordination of the show, and then, Ditmar notes, "informs the others how we're handing things off to them."

Both 3-way and 6-way splitters were

deployed to route audio signals to the various stage racks, and then to mixing consoles in the venue and out to the trucks. "We always split everything analog; we don't do any digital sharing," he explains. "That has a major negative impact on the speed and workflow that we're doing. We keep everything analog in the split world, and then it goes digital from that point on out. With how fast changes come at us in this type of show, it's proven to be impractical for any type of preamp sharing."

He also points out that the FOH music and production desks, monitor desk, music mix desk, and broadcast desk are usually different makes/models, and there's typically only one song during rehearsal to set levels and EQ, along with a quick camera check, and then it's on to the next act.

Firehouse utilizes modified Whirlwind splitters, with Dittmar noting that "one of the cornerstones of our company is having absolutely zero split issues." It's not uncommon to see a 192-input show split six ways, so the company uses a very specific grounding scheme and is "militant" about sticking to it. Part of the splitter's design is focused on enforcing proper grounding, and the tech crew also follows a rigid power distribution scheme that also reinforces best practices in grounding. Dittmar concluded our conversation by stressing the basics: "Splitting and grounding is something where you can be 99.9 percent correct, and the 0.1 percent that you're wrong about brings everything down."

GROUNDED IN ANALOG?

While digital networking has matured greatly over the past several years and can effectively distribute audio signals to multiple sources reliably, there are still some areas within the audio chain where old-school analog devices remain a standard. Signal splitting and isolation seems to be one of those areas.

In part, this is a practical decision driven by the nature of shows being set up in different venues every night while accommodating the rapid changes between acts and the desire of engineers to have full control over the inputs into their consoles. There also seems to be some resistance to surrendering that control, based on prior experiences with earlier networking technology and anecdotes from fellow engineers.

The bottom line is that analog splitting is a proven solution for sharing live audio. It will take more time, positive experiences and perhaps technical development before digital splitting becomes more commonplace in live sound. LSI

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

NOTHING LESS

Optimizing existing components and systems.

by Craig Leerman

We've all been there. Working a show or even attending as a spectator, the house system looks more than adequate but it underperforms. There are many reasons it happens, including broken components, lousy musicians, poor acoustics, operator error, and more.

Finding out if something is wrong with a system or individual components usually takes very little test equipment. The first and most important tool is our ears. Listen to an installed system when it's not in use or set up the PA at the shop to check it out. Hook up everything as always then play some known test tracks and really listen.

A friend mixes at a church for the teen praise band. The system is 4-way, portable, purchased used from a local DJ. My friend regularly complained that the mains didn't quite sound right but never took the time to analyze it. One day I visited and helped him set up the system. We plugged it together, fired it up, and realized pretty quickly that all of the 6-inch midrange drivers weren't working.

The problem? The system was wired as a 3-way rig with an amplifier for the subs, another amp for the lows, and what he had assumed was an amp for the mids and highs. Opening up a loudspeaker showed the mids were wired to the other Speakon connector on the back of the cabinet, so there was no amp (or crossover) feeding them. The previous owners had either used a different amp rack/wiring scheme, or never used the mids. Simply taking the time to listen led to the discovery of the problem.

MULTIPLE TRANSGRESSIONS

Back when my company had a repair bench and did warranty work, one customer brought in a bunch of floor monitors and said all of the HF drivers had blown within a few weeks after buying them. After opening up the cabinets it appeared that the light bulb fuses to protect those drivers had been dislodged and/or fallen out of their holding clips. From the battered look of the monitors, it was quite apparent that they'd been tossed around so much in a short time that it had impacted the fuses.

Over the years I've seen cracked plastic, fiberglass and even metal horns, shifted transducer magnets, and all sorts of physical damage to components that were not all visible from the outside. Blown drivers probably top the list. As a result, every component in a system should regularly get a preventative maintenance check to make sure everything is working properly and to look for signs of misuse.

A common but often missed item is an incorrectly wired component or cable. This can wreak havoc where a driver or entire loudspeaker box is out of polarity with the rest of the system. As we all know (right?), if two signals are at the same frequency but one signal is reversed in polarity, those signals can cancel each other out.

Cables are easy to check with a standard cable tester. Drivers in the loudspeaker are a little more tricky unless you open up the box and trace the conductors from the connections to any internal crossovers and then to the drivers. One way to easily check gear is to employ a polarity test unit. I've been using a Cricket CPTS from Galaxy Audio for years to test polarity of loudspeakers, microphones, cables, signal processors and even mixers. The Cricket has a sending unit and a receiving unit and even works down a long snake or through house wiring.

In addition to reversed polarity issues, cable testers can also find miswired and broken cable connections. The CBT-500 multi-unit tester from Hosa Technology and the HCT-BD tester from RapcoHorizon are both staples of my toolkit, but there are many other good models on the market.

SAME OR DIFFERENT

Next on the list is making sure various parts of a system are working correctly with each other, and things like crossover and amp settings are correct. If the units are from the same manufacturer and there's a dedicated system controller with factory settings, there's not much you should need to do.

On the other hand, if utilizing loudspeakers, DSP or crossovers with external amplification from different manufacturers, now's the time to double check that every setting is configured correctly, especially gain structure. The key here is avoiding overdriven or clipped signal that distorts.

I start by turning off all gain controls on every piece of gear in the signal chain. Next I use a test signal (i.e., a sine wave) and set the channel trim, channel fader and master fader so the mixer is outputting a maximum signal without clipping. The clip indicators and metering on the mixer serve as a reference.

Then without changing the mixer's signal output, it's time to adjust the level controls on each following piece of equipment. The goal is for them to run as high as possible without clipping. One other thing: when mixing, never run a console's output into clip so there's some headroom available.

APPLYING THE BASICS

Another thing I've seen a lot is that the provider doesn't bring enough rig for the gig, and/or deploy what they have correctly. When I started in pro audio, there was a formula that most sound companies used stating there should be 10 watts for every person in the audience. Now, we all know (right?) that wattage is not a good measurement of volume because various loudspeakers have different sensitivity ratings, with placement and coverage angle also factoring into the equation. But back then it was a good starting point.

The most powerful amps my company carried in that era were Crown DC300s and Peavey CS 800s (rated at 340 watts RMS into 4 ohms and 400 watts RMS into 4 ohms, respectively), but with our horn-loaded cabinets, we could get a respectable 100 dB about 100 feet away (measured with a Radio Shack level meter, of course). Today I know a little more about audio, and specifically a little thing called the Inverse Square Law, which states that when the distance from a sound source is doubled, the sound pressure level (SPL) will be reduced by 6 decibels (in an area without reflective surfaces).

Now some of you may start thinking that you've heard that line array cabinets only drop off around 3 dB per doubling of distance – and that even might apply if one were using a column of just small cone drivers. But for hybrid waveguide and cone woofer-type boxes that typically make up most modern line arrays, the 6 dB rule still applies.

Let's look at an example of a smaller PA for an outdoor festival with a single full-range and sub cabinet per side. With a seated audience that begins about 10 feet from the loudspeakers, you decide that a level of 95 dB is a pretty good volume for those closest to the stage. At 50 feet, the SPL drops to 81 dB and at 100 feet, the crowd is only getting about 75 dB, barely above the background noise.

So if the audience spans deeper than 100 feet, the folks in the back will have a hard time hearing. How do we solve this problem? One solution is deploying additional loudspeakers (delays) further out in the coverage area. The delays are positioned forward of the mains, electronically delayed so that their output arrives at the audience at the same time as the output of the mains. For reference in this regard, the speed of sound is approximately 775 miles per hour (1136.6 feet per second) at sea level in dry air at 75 degrees (Fahrenheit). Most delay units provide adjustments in feet and milliseconds, but if only millisecond are available, a good rule of thumb is to add one millisecond of delay for every foot the loudspeakers are forward of the main PA.

UP IN THE AIR

Poor audience coverage is another ongoing problem. There's a park in Las Vegas where a contractor installed loudspeakers on lighting poles for background music and announcements. They're about 14 feet above the ground and are aimed 90 degrees to the side. The announcements are unintelligible and very low in volume. Simply pointing the cabinets down toward the audience would vastly improve the situation.

Same with many portable PA systems – pointing a single 90-degree horizontal dispersion cabinet straight ahead (per side) is not going to provide quality coverage to audience members located off to the sides. Additional cabinets should be deployed to cover that portion of the crowd.

Another way to address coverage is height. Get the cabinets higher up in the air and point them down at the audience. This can be as simple as using a "tilter" device on the loudspeaker stand.

When using standard trapezoid-type top cabinets on larger outdoor shows, I sometimes place them on a section of

scaffolding (that is staked and braced) above the heads of the crowd. Add some AADs (acoustic aiming devices, a.k.a., black-painted pieces of wood) to tilt the cabinets downward, and voila – solid coverage. (Be sure that after getting the angles set to strap the cabinets securely to the scaffolding so they don't fall.)

DISTRIBUTED BOOM

Subwoofer deployment is another aspect to consider. The goal is for them to blend in with the output of the tops. Subs usually operate in a general frequency range of 30 to 110 Hz, which means wavelengths of about 10 to about 35 feet long. Anything within one-quarter (1/4) of the wavelength in distance can affect the output, including floors and walls (a.k.a., boundaries), as well as additional subs stacked next to each other.

If we suspend a sub in the air, the output emanates in all directions, and if it's more than 8.75 feet away from any surface (a one-quarter wavelength of 30 Hz or 35 feet), it does not get any gain in output from a boundary. Place the sub on the floor (called half-space loading) and there's an additional (theoretical) 3 dB of output because the energy that would have traveled down is now reflected up from the floor.

By placing a sub on the floor next to a wall (quarter-space loading) we add 6 dB more output, and moving it to a corner (eighth-space loading) adds 9 dB. (While this looks great on paper, in the real world the numbers won't be that high because of interference from the boundary and the acoustic space.)

`Now, place the sub one-quarter wavelength from a rigid wall, and it's output will bounce off the wall and return to the sub, making it about one-half wavelength, with the phase about 180 degrees from the original signal. This results in destructive cancellation. Depending on distance from a boundary and the pitch of the signal, frequency response is affected.

Most subs radiate energy in an omnidirectional pattern, and this pattern can be modified by other subs operating nearby. A common approach is placing subs at the left and right sides of the stage, which can result in a "power alley" where the bass frequencies from each side combine and add power. The audience seated in the middle hears the output from both sides equally and gets a lot of low end. However, as you move off center, phase and time delay issues cause cancellations, and for those off to the sides or in some of the null zones, the low end can be weak.

A center location with one or more subs can prove ideal for achieving a smoother coverage pattern, but this method may result in putting too much bass back onto the stage unless cardioid cabinets or configurations are implemented. Center flown sub arrays are popular for installs, especially in theaters and houses of worship, but not so much for live events, largely due to logistical considerations and related issues.

Cardioid and end fire sub arrays can be made from multiple cabinets offering directionality and pattern control while producing less sound behind the subs. These work pretty well in keeping bass off the stage and out of backstage areas.

A popular cardioid approach involves stacking the subs on top of each other, with one cabinet (usually the middle one in a stack of three) reversed to point to the rear. Pattern control is attained by using delay so that the signal at the rear is 180 degrees out of phase with the front-facing cabinets, canceling much of the sound. Several manufacturers offer cardioid subs with either rear-facing drivers or vents to help cancel out sound behind the cabinet.

End fire arrays consist of multiple cabinets that are aligned in a row, arranged one in front of the other with all pointing forward. All of the subs are delayed in relation to the rear-most cabinet, making it possible to project powerful, directional bass over long distances.

FINISHING IT OFF

One more key is optimizing a system for the room or location. This is generally called system tuning, and it involves both our ears and a measurement rig utilizing quality, purpose-designed software such as Smaart from Rational Acoustics. It shows us the input signal, compares it to the output signal, and even reveals how a room is affecting loudspeaker output. It's also easy to see things like frequency response, phase response, delay

and more, and it's all information that can help us "dial in" a system.

The final step is back where we started: listening. Once a system is set up and the coverage is on target, play known tracks and listen to the tone. (Also double check the coverage.) At this point, I make as few changes as possible, then set up a mic on a stand at stage center, pointing straight up. In this process, first verify that the mic channel EQ is flat. Next, bring up the input of the mic until a bit of feedback occurs, and then use the main EQ (graphic or parametric if available) to pull down the frequency that's ringing down until it stops.

Push the fader up again until another frequency rears its head and remove it. Do this until multiple frequencies start to ring together, or there's plenty more headroom in the mic than needed.

Play tracks again and listen to see if the music still sounds good. If I've hacked the EQ to death trying to get rid of feedback I start again, but this time using the channel parametric to get rid of a few frequencies.

While all of this seems like a lot of work, it's essential to the craft and our role of audio professionals. We owe performers, audiences and our colleagues nothing less. LSI

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

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SEAMLESS IN SEATTLE

The SeafoodFest stage hosts popular local bands, with EAW Redline Series loudspeakers serving as stage monitors.

Combining two recent developments to serve a popular annual event.

by Sam McLean

FOR MORE THAN FOUR decades, SeafoodFest has celebrated the Seattle neighborhood of Ballard's historic connection to the fishing industry as well as its strong Nordic roots. Now based on NW Street downtown adjacent to Salmon Bay, among the attractions and revelry is a live music stage located at the heart of the two-day event that presented mostly local talent, including Helo Sequence and Saturday night headliner Mudhoney.

Last year marked the initiation of support of the festival by sister companies Eastern Acoustic Works (EAW) and Mackie (the latter based in nearby Woodinville, WA), providing a sound reinforcement package capable of covering an audience area of more than 100 feet wide and stretching several hundred feet down the street. This time out, the companies teamed up to supply their latest technologies in the form of the public debut of EAW's just-unveiled RADIUS line of loudspeakers working with Mackie's new AXIS digital mixing system.

"SeafoodFest takes place right in the middle of a bustling urban setting," explains Mike Stewart, executive director of the event. "Setup is on Friday and with everything going on in the neighborhood, it can be quite a challenge. RADIUS and AXIS made setup a breeze – very impressive."

EAW Radius arrays and subwoofers at the live music stage.

MEETING REQUIREMENTS

The live music program began each day at 1 pm, with 12 bands in total taking the stage through the festival. Each group performed for about an hour, with 30 minutes allotted between sets. The new system components helped meet the typical festival requirement of quick and seamless transitions during changeovers.

Specifically, the self-powered RADIUS main PA consisted of left-right hangs of six RSX208L line array modules joined by 12 RSX18 subwoofers stacked six per side on the ground. Front fill was managed by two RSX89 loudspeakers, one at each front corner of the stage and firing inward.

The 3-way RSX208L is loaded with two 8-inch cones (LF and LF/MF), and high frequencies are handled by dual 1-inch compression drivers. Each module is stated by the company to produce up to 125 dB whole space SPL, plenty of output to meet the application. The RSX18 incorporates an 18-inch cone (3-inch voice coil) loaded in a vented enclosure, with both the full-range and sub boxes carrying class D amplification and integral DSP limiting.

Chris Mael, Mackie's director of procurement, planning and analysis, who's also been mixing shows and involved in live sound since his teens, held down the fort at front of house. "Flying the arrays was very easy," he notes. "The flyware is self-explanatory. We had two guys and got it flying really fast. We didn't have to tackle the crazy puzzle of trying to figure out how the pieces went together."

MIX CAPABILITIES

The other significant facet of the new technology package was the AXIS mixing system, which debuted earlier this year and is primarily targeted to small- to mid-sized production companies. It combines a 32-channel DL32R rack-mount digital mixer and DC16 control surface, and includes 32 remote-controllable Onyx+ mic preamps as well as 18 outputs with built-in DSP that can be allocated in a variety of ways.

In addition, the FOH configuration at the festival included SmartBridge, which can house up to three iPad devices in integrating with AXIS. It has "smart sensing" to know when an iPad is in place and also facilitates customization of each iPad view for enhanced workflow flexibility.

For this application, the sound team opted to place EQ and compression for a specific channel on one iPad, with a second iPad dedicated to providing a master view of all inputs and outputs. The third iPad (and its SmartBridge station) were reserved for guest tech/engineer use.

"The AXIS interface allows me to get anywhere on any particular channel really quickly and shows a lot of information up front, which makes it easy to navigate," Mael explains. "When you go wireless with the iPad, the Master Fader app delivers the same intuitive experience. It gives you complete control over the system, so you always feel at home. Many guest engineers are not fans of mixing on a console they've not used before, but in this case all it took was a few minutes of playing around on it and they were sold."

OUT OF THE BOX

Meanwhile, RADIUS includes automated functions like array self-detection and optimization designed to cut down on setup and tuning time. At the same time, the new EAWmosaic app provided convenient main system optimization from anywhere in the area via a tablet or laptop.

"With as many years as I've mixed FOH and monitors, I've gotten pretty particular about the PAs that I like to mix on," Mael explains. "You can only do so much with EQ before it starts sounding unnatural. If I can get my hands on a rig that sounds good right out of the box? That's really good. Everyone who listened to the system was extremely complimentary about the sound quality – even when it was loud – which says a lot about the system."

Finally, (Audinate) Dante networking connectivity, built into both major components, further enhanced both setup and integration. Despite the festival being the first public use of both, and the usual hectic nature of the festival scene, Mael and the sound team claim a virtually flawless event from a system point of view.

(Above) Chris Mael setting up the Mackie AXIS digital mixing system at front of house. (Below) A look at the coverage area, with the audience spilling well down the street.

"A lot of engineers walk into a festival apprehensive about the console and PA they have to use," Mael adds. "I was happy to be able to tell them that RADIUS is clean, clear and had plenty of throw and a ton of headroom. And with AXIS, you have a console that is predictable, sounds good and is easy to use. Needless to say, everyone here was quite happy."

It's a viewpoint reinforced by SeafoodFest executive director Stewart: "For Mackie and EAW to come in and produce sound in a close urban environment and get it set up and ready to go with extremely fast changeovers is fantastic. In particular, this year everyone noticed the sound quality, which was amazing." LSI

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

LOAD IT UP

Impedance, resistance and why they matter, part 1.

by Jonah Altrove

It may not seem so at first blush, but there are a few concepts as pervasive in live audio as the concept of impedance. It's relevant to microphones, direct boxes (DIs), preamps, processors, amplifiers, loudspeakers, equalizers, crossover filters, and even the cables that connect them all together. In this article we'll explore the principles of impedance, and examine how these concepts apply directly to our work in the field.

OHM SWEET OHM

In order to understand impedance, we need to start by looking at the three fundamental electrical principles of voltage (E), current (I), and resistance (R). This would be an entire article on its own, so we'll just have a quick review here. Traditionally, these concepts are introduced using the venerable, if limited, metaphor of water flowing through a pipe. But why talk about water when you can talk about cake? Here's a more engaging analogy:

Imagine the fabric bags of frosting that bakers use to decorate cakes. Squeezing the bag forces the frosting out a nozzle in the bag's corner. The flow of the frosting would be analogous to current (I) which is the flow of electrons in a circuit. The frosting flows because of the pressure applied to the bag, just as current flow is caused by the application of voltage (E), which is, more or less, "electrical pressure." The nozzle resists the flow of frosting, just as resistance (R) opposes current flow in a circuit.

If we wish to increase the frosting flow (increase current), we can neither squeeze the bag harder (increase voltage) or use a nozzle with a larger opening (decrease resistance). These relationships are expressed in the formula E=IR, known as Ohm's Law. Voltage is measured in volts, current in amperes or amps for short and resistance in ohms, which is represented by an Omega (Ω).

P =	E =	I =	R =
EI	IR	P/E	E/I
I^2R	P/I	√P/R	P/I^2
E^2/R	√PR	E/R	E^2/P

Figure 1 – Calculating all four circuit parameters.

Add to the mix that you can multiply voltage times current to get power (P) in watts, and with a little algebra we can calculate all four circuit parameters – P, I, E and R – if we know any two (**Figure 1**). You probably don't need to worry too much about committing all that to memory – if you use the formulas regularly, they'll become second nature. It's really the same idea stated a bunch of different ways. The important thing to realize is that a change in any of the variable affects all the others.

In North America, residential electricity runs at 120 volts. We can calculate that a 100-watt light bulb draws a current of .83 amp and has a resistance of 144 ohms. Now let's swap out the 100-watt bulb for a 60-watt bulb. The power is less, but the voltage is still 120 volts, which means the 60-watt bulb has a higher resistance – the voltage is unable to push as much current. Ohm's Law predicts a resistance of 240 ohms, and a current of .5 amp. For a given voltage, increasing resistance causes a decrease in current.

IT'S GETTING HOT IN HERE

But there's a problem – if you use an ohmmeter on a 60-watt light bulb, you'll get a value for less than the expected 240 ohms. We're missing two critical puzzle pieces. The first is that incandescent bulbs get hot. Very hot. In fact, over 90 percent of their power is dissipated as heat, not light. What matters to us is that resistance increases with temperature. This is why the giant superconducting magnets in particle colliders are cooled with liquid helium – to keep the resistance as low as possible.

It's also why the response of a sound system changes as it plays louder. Warm voice coils have higher resistance, and pass less current from the amplifier. This is part of the reason we have system limiters – a loudspeaker can only convert ("transduce") so much power into sound. Past a point, it doesn't get louder, it gets hotter. The heated coils in the loudspeakers lose efficiency and headroom, an effect called power compression. In the extreme, this heat buildup can be the kiss of death to a system – there's a reason we refer to failed components as "burned out."

It takes a pretty robust rib to remain linear all the way to its limits, fighting the changing resistance as the program materials heats the voice coils, but the last few years have seen more attention paid to this issue by manufacturers. For example, this is the issue that Meyer Sound is tackling with its LEO family of loudspeakers.

So that's one piece of the puzzle: a cold light bulb will show a lower resistance than a bulb heated to operating temperature.

AC/DC

The other reason that the light bulb doesn't meter the resistance predicted by Ohm's Law is that the it's an AC device. Here's where we need to make the distinction between resistance and impedance: "resistance" is the total opposition to current flow in a DC circuit, such as a flashlight.

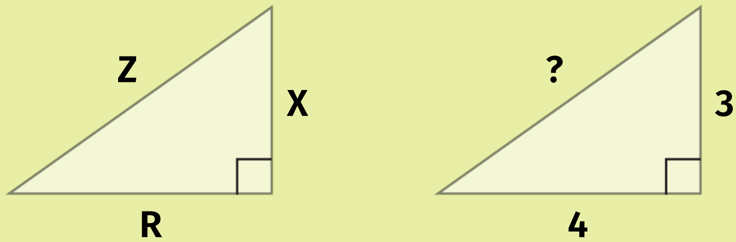
However, with AC, there's more to it. AC circuits will exhibit the same DC resistance, but there is additional opposition to current flow created by circuit components such as capacitors and inductors. This additional opposition is due to the way these components react to the alternating current, and is called reactance (X). The AC circuits opposition to current flow consists of both resistance and reactance, so now we need a new term to describe the total opposition. All things considered, how much does the circuit impede current flow? This is its impedance, represented by a "Z." (Also keep in mind AC resistance, which is a frequency at which the voltage and current are in-phase (phase angle = 0), such as the resonant frequency of a loudspeaker.)

As with resistance, reactance is also measured in ohms, but we can't simply sum the two. A circuit having 4 ohms of resistance and 3 ohms of reactance does not have an impedance of 7 ohms. Conceptually, resistance and reactance are considered as two sides of a right triangle, with a 90-degree angle between them. The impedance is represented by the length of the hypotenuse, so we can use the Pythagorean Theorem (c2 = a2 + b2) to find it. You can see that 4 ohms of resistance (R = 4 ohms) and 3 ohms of reactance (X = 3 ohms) produce an impedance of Z = 5 ohms (**Figure 2**).

There are two types of reactance: capacitors cause capacitance, and inductors cause inductance. These act opposite each other, meaning they subtract, not add. If a circuit exhibits 8 ohms of inductive reactance (XL) and 6 ohms of capacitive reactance (XC) then the total reactance is 8 – 6 = 2 ohms, which is the value we would use in the impedance triangle. This is like walking 8 meters in one direction, then turning around and walking 6 meters back – you're only 2 meters from where you started. Since the light bulb's filament is inductive, this accounts for the rest of the discrepancy.

power factor, and many modern amplifier employ power factor correction to improve their efficiency.)

When it comes to audio signals, the situation is much more complex: our frequencies vary from 20 Hz to 20 kHz, three orders of magnitude. The huge frequency range means that our circuits will react far differently at 20 Hz than they will at 20 kHz. I won't impede you with my formulas (ha!) but the calculation of X_L and X_C incorporate a frequency variable, f, without which it's impossible to do the calculation. It's therefore impossible to discuss the behavior of a reactive circuit unless we consider frequency.



$$Z^2 = R^2 + X^2$$

$$Z^2 = 4^2 + 3^2$$
$$Z^2 = 16 + 9$$
$$Z^2 = 25$$
$$Z^2 = 5\Omega$$

Figure 2 – The impedance triangle.

R VS. X

The reason we need to draw a distinction between resistance and reactance is that reactance changes based on the frequency applied to the circuit. This makes sense: Reactance is the circuit's response to alternations in the current direction, so it follows that changing the speed of these alternations would affect the reactance.

In terms of electricity, such as in the case of our light bulb, this is a moot point since North American power is always distributed at 60 Hz, so reactance is a one-and-done calculation. This is how electricians use the blanket term of "resistance" though they're technically describing impedance. For them, frequency is a constant and therefore so it reactance. (Note, though, that even at 60 Hz there are still resistive and reactive properties that must be considered. A load can be purely resistive, purely reactive, or a combination. This is accounted for by the

The formula for capacitive reactance (X_C) is a fraction with the f in the denominator. As frequency increases, X_C become smaller – less and less opposition. Inserting a 79 µf (microfarad) capacitor into an AC circuit creates 100 ohms of reactance at 20 Hz, 2 ohms at 1 kHz, and .1 ohms at 20 kHz. Increasingly high frequencies pass more easily, and increasingly low frequencies are impeded, so capacitance creates a simple high-pass filter.

The formula for inductive reactance (X_L) has the f in the numerator, so XL increases with frequency. A 318 µH (microHenry) inductor causes .04 ohm of reactance at 20 Hz, 2 ohms at 1 kHz and 40 ohms at 20 kHz. Increasingly low

frequencies pass more easily, and highs are increasingly blocked, so inductance creates a simple low-pass filter. (If you design filters, you know that 318 MμH is a strange value for an inductor, but the resulting figures clearly illustrate the change in response over frequency).

It's easy to see that impedance varies with frequency. This isn't what most people think of when they hear the term "impedance" but it's at the heart of how equalizers and crossover networks function, so it's vital to understand.

Let's return again to our light bulb example, and this time let's run the AC at a frequency of 120 Hz. Since the light bulb filament is inductive, it would present a higher impedance to their higher frequency, and the bulb would be dimmer.

Through a pure inductor, doubling *f* also doubles X_L and thus passes half the voltage. "Half the voltage" is -6 dB, which is why a simple single-pole crossover filter has a slope of 6 dB/octave. (Remember, an octave is doubling of frequency). To create steeper slopes, we cascade reactive elements. A 24 dB/octave filter circuit has four reactive elements that contribute 6 dB/octave each.

If impedance varies with frequency, why so do we have loudspeakers labeled "Impedance: 8 ohms"? The truth is that the Z rating on most loudspeakers is a worst-case scenario. Many loudspeakers have impedance curves that look like roller coasters. We usually want to know the worst-case rating – the lowest point on the curve – because this tells us the most load that the loudspeaker could place on the amp.

Remember, lower Zs translate into higher loads on the amp since there's less to oppose the flow of current. So an 8-ohm-rated loudspeaker exhibits a much higher impedance throughout most of its frequency range. (The rated impedance is always higher than the lowest point on the curve. The IEC standard says that the rated impedance cannot be higher than the low point * 1.2. So, an 8-ohm loudspeaker could have a Zmin closer to 5 ohms.)

Very low impedances are undesirable because the amp is approaching a short-circuit (0 ohms). In the sub-2-ohm range we have to deal with a whole slew of unpleasantries, including decreased amplifier lifespan, increased distortion, higher current draw, overheating, smoke, fire, and loss of employment.

In the next article, I'll further these concepts and offer additional practical examples. LSI

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

SHURE PSM 300

Evaluating a recently released wireless personal monitoring system. *by Nicholas Radina*

A major player in the in-ear monitoring game is Shure with the well-designed and dependable PSM family of wireless stereo personal monitoring systems, which includes the entry-level PSM 200 as well as the touring friendly, full-featured PSM 900 and PSM 1000. The newest member of the family is the PSM 300, which utilizes much of the technical heritage of the PSM 900/1000, yet at a more accessible price point for both entry-level users as well as for rental houses, monitor engineers and musicians needing more advanced options.

The PSM 300 offers two options with regard to body pack receivers, both working with the half-rack P3T transmitter. The lower-priced P3R receiver, utilized for this review, is a slim, plastic unit with simple-to-use volume and mix controls, powered by two AA batteries (up to six hours of continuous use). It ships with Shure SE112 earphones.

Shure PSM 300 personal monitoring system with P3R receiver option.

Meanwhile, the P3RA receiver is housed in a heavier-duty metal case with additional features for RF and audio adjustments. It sports an LCD display with menu-based navigation, as well as 2-band shelving EQ with high/low boost and a volume limiter to set maximum gain level range. Supplied with Shure SE215 earphones, it too can be powered with two AA batteries or (optionally) Shure's SB900 lithium-ion rechargable batteries that extend runtime.

Aside from the transmitter, receiver, 1/4-wave antenna and external power supply, PSM 300 systems also come with a rack-mount kit. The earphones for this review (SE112) were accompanied by different sized inserts as well as a nice carrying pouch. A well-written quick start guide and a proper manual complete the package. Shure provides everything needed to be up and running quickly, including a fresh set of AA batteries!

Back panel I/O facilities on the P3T transmitter.

GENERAL OVERVIEW

Both PSM 300 system versions send two channels of 24-bit digital audio to performers, with a choice of creating a mix on each body pack with adjustable stereo balance or proprietary MixMode 2-channel mono blend. Also onboard are one-touch IR frequency scan and sync to find and assign clean channels. Signal-to-noise ratio is stated as up to 90 dB, with operating range of 300 feet (90 meters).

There are up to 15 compatible frequencies per band, with a stated 24 MHz tuning bandwidth (regionally dependent). Available frequency bands include G20 (488.150 – 511.850 MHz), H20 (518.200 – 541.800 MHz) and J13 (566.175 – 589.850 MHz).

And, the PSM 300 incorporates the company's patented Audio Reference Companding to foster sound quality by allowing the system to avoid companding until it's necessary. This eliminates low-level artifacts, reduces distortion, and improves transient response.

The half-rack P3T transmitter, housed in a metal and plastic casing, has a clean, professional look. The uncluttered front panel includes an input level knob to set incoming audio level; IR sync window for aligning a body pack to a receiver for syncing; sync button to be pressed once the wireless scan has been completed; group buttons that select the group/channel if setting manually; and a clear LCD that displays, at a glance, audio level, group, channel and TV channel (nice!), as well as the mode the unit is operating in (mono or stereo/MixMode).

The back panel offers four 1/4-inch balanced TRS jacks — note there are no XLR connectors. These are labeled and function as follows:

- Audio-In Left/Mono for connecting to a mixer output or source "left" when monitoring in stereo or MixMode. It's also used for mono, just be sure to select "Mono" mode on the accompanying switch.
- Audio-In Right for connecting to a mixer output or source "right" when monitoring in stereo or MixMode.
- Loop Output Left & Right for passing signal to another PSM transmitter or other devices.

There are also two switches. One provides selection of the input level from the audio source/mixer, either +4 dBu line level or -10 dBv. Unless you're using a consumer device such as a phone, tablet or laptop, I advise sticking with the +4 dBu setting. The other switch sets the transmitter to operate in either mono or stereo/MixMode.

FUNCTIONAL MATTERS

The P3R receiver pack in my review system is plastic but feels solid and is quite light weight and slim – nice build quality. I was impressed with the rigidity of the antenna. Over time, less rigid antennas tend to droop, which can advisedly affect RF reception.

A single knob on the top of the pack adjusts volume and turns the unit on/off. It's joined by an 1/8-inch stereo jack. There are also two small LEDs that wrap around from the top to the front side of the unit. One LED is blue, illuminating when a RF signal is present. The other LED indicates battery strength by via colors (green, amber, red).

The right side of the pack has a nice low-profile pan knob with a solid center detent. You can also think of this as a level knob when using the MixMode option.

On the front of the unit, there's a sleek door that, while gently squeezing both sides, opens to reveal the battery compartment, group, channel and scan buttons, as well as the sync IR window and a 2-character, 7-segment type display.

BRIEF OBSERVATIONS

Here are some general impressions I formed while checking out the system. First, I was quite impressed with the scanning speed/procedure of the receiver. Remember the body pack is used for scanning – not the transmitter. Quick tip: run the scan from the location where you'll be using the pack. The scanning procedure is simple and outlined quite well in the manual.

MixMode is unique to Shure and offers interesting options. If operating in MixMode, the side pan knob of the body pack can be utilized to blend between two sources such as a general band mix and your vocal. Simply use the left and right inputs on the transmitter as inputs for each source or mix. Do keep in mind that these are two mono sources, of course.

Shure states six hours of receiver operating time on fresh name-brand alkaline batteries. In my work with the system, I never operated it for more than four hours, so this just wasn't a concern. However, the design makes it easy to confirm battery level at a glance by using the top-mounted battery level LED – green is good for approximately 5-7 hours, amber for 1-3 hours, and flashing red means 30 minutes or less. Obviously, your mileage may vary due to battery brand and usage conditions.

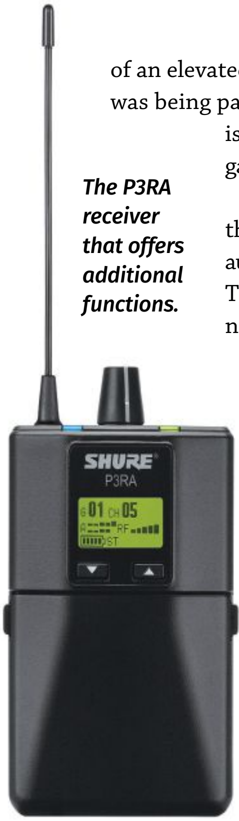
A nice feature to conserve battery life is an intelligent "auto-off" feature if the receiver doesn't receive audio after a period of time. Very handy.

IN THE FIELD

I deployed the PSM 300 on two very different types of gigs. The first was on a salsa band show. My "non-sound" life finds me smacking cowbells and playing timbales with a few Cincinnati salsa bands. Setting up the unit and scanning for the best clear frequency was a breeze. Although the lack of XLR inputs tripped me up, a handy adapter saved the day.

I opted to use the standard stereo mode, which I recommend due to the clarity gained from panning as well as the reduction in volume needed from the pack (a sneaky brain trick). Once the monitor mix was adjusted to my liking, off I went. Even though I have molded earpieces, I opted to use the supplied SE112 buds and was pleasantly surprised with their fit. Obviously there's not going to be the isolation offered by molded units, but these buds are a good start.

The bass response of the SE112s was a bit heavy, but in my experience, unless you're using molded buds with the isolation benefits they offer, bass response can suffer. Shure's choice to emphasize the bass is a nice compromise. I noticed a bit of an elevated noise-floor when no audio was being passed along, which I assume is the result of an intelligent gate and headphone amplifier.



The P3RA receiver that offers additional functions.

My other field application of the PSM 300 was to transmit audio to another sound system. The gig was a wedding and I needed background music to play concurrently in two different rooms. I could have simply used a long XLR, yet due to doorways and foot traffic (as well as the time to gaff it all down), that wasn't an attractive option.

So I simply set up the transmitter next to the mixer and placed the receiver in the other room with a 1/8-inch-to-XLR adapter plugged into the headphone output (while paying attention to proper gain structure). After finding the limit to the system's range (less than 300 feet, but hey, I was transmitting through a few walls and a kitchen), it worked great.

One additional field test came when I loaned the system to a fellow musician to try out at a small club gig. He reported back that the scanning and setup was quick and intuitive. He also commented on the generous headroom in comparison to his current system as well as the "not worrying about it" battery life.

The benefits of using personal monitoring are wonderful, but I've been frustrated with the performance and quality of some lower-priced system options. Particularly in this light, I think Shure has a hit on their hands at a great price point with the PSM 300.

U.S. MSRP: starts at $699 with P3R receiver; starts at $799 with P3RA receiver. LSI

Nicholas Radina *is a long-time audio engineer and musician based in Cincinnati who also tours as the monitor engineer with the band O.A.R. He invites your input via his website at NicholasRadina.com.*

TURBOSOUND INSPIRE IP2000

Putting a column loudspeaker/sub combo through its paces. *by Craig Leerman*

The Turbosound iNSPIRE iP2000 is a loudspeaker system comprised of modular, full-range column arrays and a companion 12-inch subwoofer. Each column array is comprised of 16 cone drivers plus a horn-loaded super-tweeter that can be easily mounted on the sub, locking to it and thus eliminating loudspeaker stands and cables.

The iP2000 is outfitted with proprietary Klark Teknik SST (Spatial Sound Technology) for helping to create enhanced 3D acoustic environments as well as DSP presets for application type and loudspeaker positioning. The DSP is more than just selective EQ enhancement. A 24-bit processor analyzes the incoming signal and intelligently applies specific filters, making subtle real-time adjustments to the crossover, compressor/limiter, three discrete class D amplifier channels and more.

The Turbo Control remote control app is available for iPhones and iPads, linked via Bluetooth stereo audio streaming for wireless connection. The multi-channel amplification and onboard 3-channel digital mixer makes for a true "plug and play" package.

The iP2000 is the largest model in the iNSPIRE series that also includes two modular column and subwoofer models, the iP1000 and iP500, two subwoofer models with stereo satellite output, the iP12B and iP15B, and a passive satellite loudspeaker, the iP82, all utilizing the same onboard mixing, processing, and class D amplification.

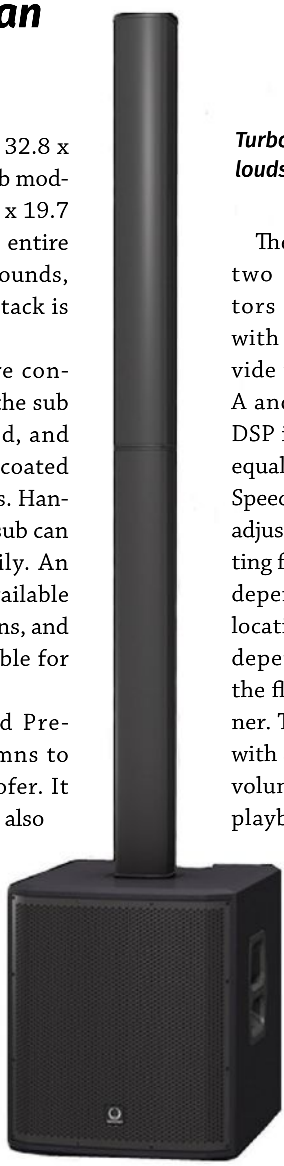
PHYSICAL CHARACTERISTICS

The iP2000 disassembles into three pieces for easy transport. The two column modules measure just 32.8 x 4.0 x 3.5 inches while the sub module checks in at 19.1 x 16.5 x 19.7 inches (both h x w x d). The entire assembly weighs in at 68 pounds, and when assembled, the stack is 81 inches high.

The column modules are constructed of aluminum and the sub cabinet is made of plywood, and both have durable powder-coated perforated steel mesh grilles. Handles on each side mean the sub can be moved/positioned easily. An optional shoulder bag is available to carry and store the columns, and a protective cover is available for the subwoofer.

A mating system called Precision-lock joins the columns to each other and the subwoofer. It mates each section together, also making all electrical connections via a multi-pin connector. No tools are required as the sections simply push together by hand.

The iP2000 has a stated frequency response of 45 Hz – 20 kHz (+/- 3 dB), max SPL of 123 dB, and nominal dispersion of 120 degrees horizontal. All controls and connections are located on the sub. There's a nicely sized LCD screen that displays parameter settings and a single encoder knob that can be pressed as well as rotated to facilitate changes. There are also four buttons to further define parameters, marked Process, Setup, Exit and Enter.



Turbosound iNSPIRE iP2000 loudspeaker system.

The back side of the sub hosts two combo TRS/XLR connectors for inputs A and B, along with two XLR outputs that provide un-processed copies of the A and B inputs for linking. The DSP includes four factory preset equalizations labeled Music, Live, Speech and Club, and these can be adjusted to taste. The position setting facilitates DSP compensation depending on the loudspeaker's location, providing three presets depending if it's positioned on the floor, near a wall, or in a corner. The digital mixer is outfitted with 3-band EQ, channel volume, volume for a Bluetooth attached playback device like a phone or tablet, sub level and master volume.

GETTING RIGHT TO IT

Usually with product evaluations, I try things out on my test bench first and become familiar with operation before any field applications, but this time was different. The day I received the iP2000 was also the day of a party in my neighborhood to celebrate a child's birthday and some other milestones.

What had been conceived as a gathering for an expected 10 or so guests had quickly grown to a full-fledged event attracting several dozen attendees joined by a bounce house, snow cone machine and a feast that could feed a small village. Of course every party needs music so I decided to give the iP2000 a "trial by fire" in covering what had grown to be a quite large and noisy area — and the system didn't disappoint.

Turbosound provided a single iP2000 stack consisting of two columns that attach to the sub. First I unboxed the top modules and was surprised at how small and light in weight they are. Each module has a multi-pin connector that mates with the unit below it, and this connector is surrounded by four large pins that stick out a few inches. The pins hold each module in place, and the rear pins are slightly wider so there's no mistaking which way to fit the modules together.

The sub also has a receiver unit with multi-pin, so the top modules simply slide into place on the sub. It's a slick, simple and intuitive build/interconnect system. And remember, the multi-pin handles all electrical connections so no cables are required.

Next, I was quite pleased to find that the control panel is top-mounted on the sub and not on the back. I could just look down to change parameters and not have to get on my hands and knees and bend over to see the screen to make adjustments. It also means that changes are easier to make with the unit located next to a wall.

I placed the sub in proximity of an AC outlet and then attached the top modules. The menu was intuitive and there wasn't even need to consult the manual. I plugged an iPad into the A and B inputs and quickly had some music playing. Sound quality was really quite good out of the box, and the sub had a lot of thump for its size.

CONNECTIVITY & COVERAGE

Eager to check out the app, I downloaded it on my iPhone. The phone quickly discovered the loudspeakers via Bluetooth, and then in using it, I found that the app provides a lot of control. I could adjust the volume of both A and B channels, the volume of a third channel for Bluetooth streaming audio, the master volume, sub volume, and EQ, as well as switch between the four preset EQ modes. Walking to the far corner of the coverage area, I found solid connectivity at more than 100 feet away.

During the event, numerous guests commented on how great the system sounds, with some also pointing out how nice it looks. The 120-degree horizontal dispersion is solid, even off to the sides, just out of the pattern. I had no trouble filling the entire outdoor space with a single stack. At 81 inches tall, the tweeter and some of the mid-range cones are above the heads of a standing crowd, so when folks would stand between the loudspeaker and myself, it was still clear rather than muffled.

The I/O facilities on the iP2000 12-inch-loaded sub. The column tower simply plugs and locks in at the top.

After the event I checked out the Bluetooth streaming capabilities. The iP2000 allows streaming audio to one or two loudspeaker stacks, with selection of which stack is the master. The app also automatically detects stereo mode and adapts the GUI accordingly. I also evaluated the app and connectivity on an android phone and had the same success. Fast, solid, and no problems at all.

Satisfied that the Bluetooth worked as advertised, I plugged a microphone into the system to listen to a live voice through the loudspeaker. The iP2000 has a speech mode setting so I tried it first. It's a little thin-sounding to my ear but can be warmed up with the EQ. And both the live and music modes sound great with a voice.

THE BOTTOM LINE

As previously noted, I'm a big fan of column-format loudspeakers for a variety of reasons. One is a preference for the look of the slim profile as opposed to a more clunky loudspeaker on a tripod stand. I also like a wider horizontal dispersion than the typical 60 to 90 degrees. My clients like how columns can be placed in tight spaces and still perform to expectations while not blocking sightlines.

Anyone in the market for a column/sub combo should definitely put the iP2000 on the list. It delivers high-quality sound in a compact footprint that's easy to transport and flexible in placement. It's a perfect system for DJs, and also makes for a more-than-suitable rental PA as well as being cable of meeting myriad applications in schools and churches where inexperienced techs and volunteers will have no problem getting up, running and optimized without a glitch.

U.S. MSRP: $799 LSI

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

COMPACT LINE ARRAYS

Good things can come in smaller packages.

by Live Sound Staff

Compact line arrays are those we've defined as having low-frequency drivers measuring 8 inches or smaller. They present a scalable solution – a single array will provide main coverage to a relatively small space, and the addition of more loudspeakers easily expands coverage capabilities.

Compact arrays offer a relatively small footprint in terms of size and weight, yet are capable of delivering significant output. They're conveniently portable due to the size/weight factor, and are suitable for a range of fixed installations as well. For dynamic full-range music presentation, arrays are usually accompanied by at least one subwoofer, some of them designed to fit seamlessly within a flown array structure.

Many manufacturers also offer line array modules of various sizes that work together. Most commonly, we see compact modules at the bottom of larger arrays to supply front fill reinforcement at concerts, festivals and other large-scale events, and they can also be placed on the stage lip to provide low-profile front and near fill. Another application is side fill in tandem with larger main arrays.

Flexibility with rigging continues to evolve, providing added array structure options. Compact arrays also present an increasingly popular approach of groundstacking as mains capable of satisfying the requirements of smaller applications without the need for rigging infrastructure.

The majority of compact arrays are available with dedicated amplification and sophisticated DSP packages, either onboard or separately rack-mounted. Others are designed to work with a range of amplification and processing devices, and some companies offer a choice of either approach. A number of models are now equipped with proprietary and/or Dante networking capability.

Progress on the compact line array development front continues, with the following listings of the latest models intended to inform as to what's available as well as provide a means of differentiation, at least from a specification standpoint. Enjoy this look at a wide range of options from around the industry. LSI

d&b audiotechnik T10
www.dbaudio.com

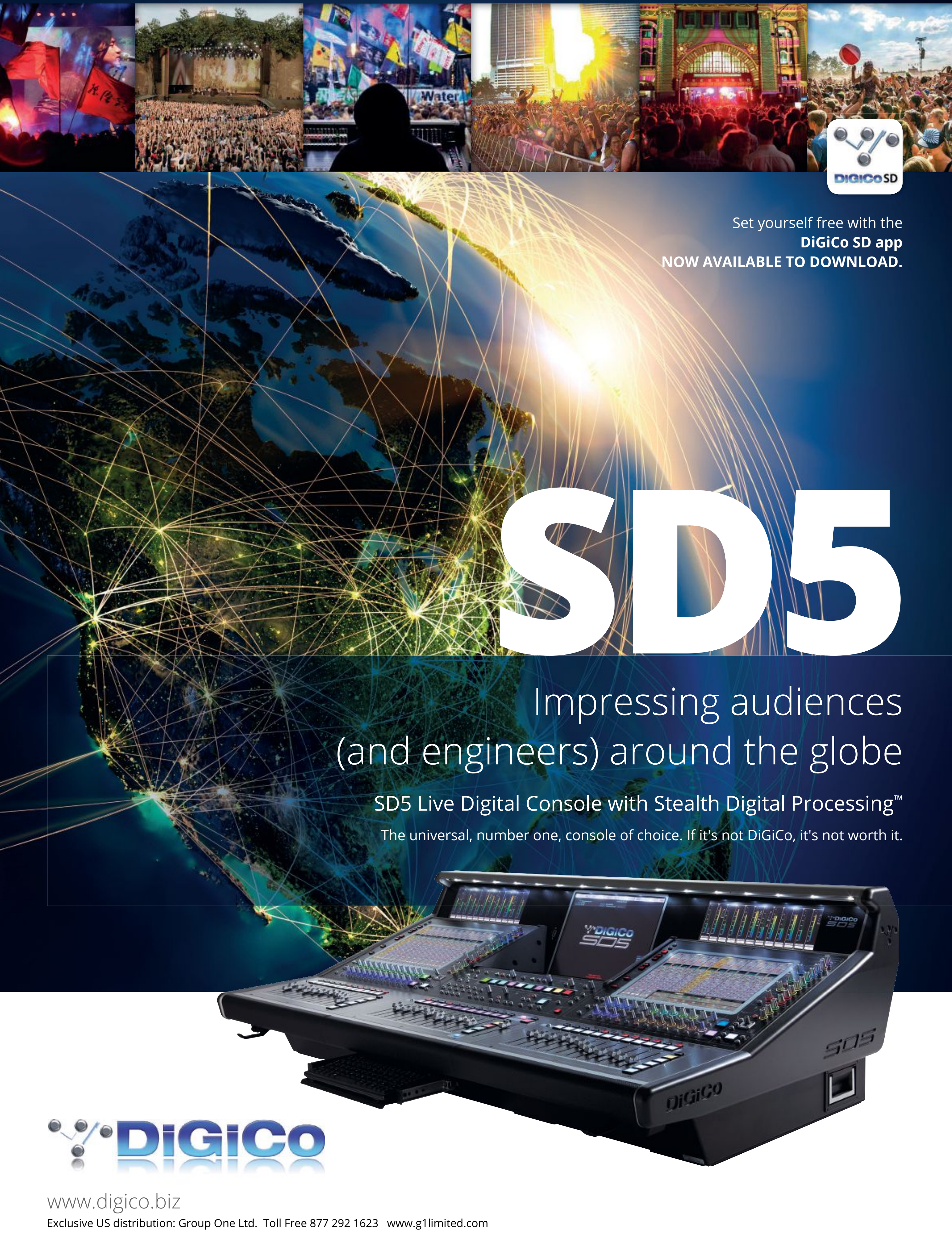
Configuration: 2-way
LF: 2 x 6.5-inch cones neodymium cone drivers
HF: 1 x 1.4-inch-exit compression driver on rotatable waveguide
Frequency Response: 68 Hz - 18 kHz
Dispersion: 105 (h) x 15 (v) degrees or 90 x 35 degrees single point
Crossover: Internal passive
Rigging Angles: 0 to 15 degrees in 1-degree increments
Power: d&b amplification (D6, D12, D20, D80, 10D, 30D)
Weight: 24 pounds
Size: 7.7 (h) x 18.5 (w) x 11.8 (d) inches

Renkus-Heinz IC²
www.renkus-heinz.com

Configuration: 2-way
LF: 4 x 8-inch neodymium cone drivers
HF: 4 x 1-inch compression drivers
Frequency Response: 60 Hz - 20 kHz
Dispersion: 120 or 90 (h) degrees; adjustable vertical from 10 to 80 degrees per beam
Rigging Angles: Digitally steerable; adjustable from -30 to +30 degrees
Power: Self-powered (class D, 8 channels), multi-channel DSP
Weight: 75 pounds
Size: 18.5 (h) x 28.5 (w) x 11.5 (d) inches

Meyer Sound M'elodie
www.meyersound.com

Configuration: 3-way
LF/MF: 2 x 8-inch cones (neodymium)
HF: 1 x 1.2-inch-exit driver with REM (neodymium)
Frequency Response: 76 Hz - 16 kHz (± 4 dB)
Dispersion: 100 (h) x 11 (v) degrees
Crossover: 320 Hz and 1.1 kHz
Rigging Angles: 0 to 11 degrees, 1-degree increments
Power: Self-powered, 1,275 watts tri-amped; DSP
Weight: 62 pounds
Size: 9.2 (h) x 28.5 (w) x 12.8 (d) inches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RWG Spotlight Listing

NEXO GEO M6 Series
www.yamahaca.com

A very versatile, arrayable cabinet design, the GEO M620 represents renowned NEXO product design, flexibility, high SPL output, and exceptional sonic clarity for speech and music sound reinforcement. Joined in the GEO M6 Series by the GEO M6B bass extension, it extends beyond NEXO's tradition of live music applications toward the corporate AV sector and a wide range of fixed installation opportunities where speech reinforcement is the primary requirement.

GEO M620 loudspeakers can be used as single or in pairs, in curved arrays of three, and in line arrays up to 12 cabinets long, depending on application and enabled by a range of simple accessories that allow flying, wall-mounting, pole-mounting and ground-stacking, pairing with any of NEXO's wide range of sub-bass cabinets to create a highly portable SR solution.

The GEO M620 is a full-range unit for stand-alone, curved array or line array applications. Extremely compact in size (and light in weight at less than 22 pounds), it uses a NEXO-designed long-excursion, high-efficiency 6.5-inch LF driver and 1 x 1-inch throat compression driver on a BEA/FEA optimized HR Wavesource to deliver superior sonic performance.

TECHNOLOGY FOCUS: *Cost-effective amplification is via NEXO's NXAMP4x1 TDcontroller with three cabinets per channel; i.e., a 12-cabinet system can be powered by just one amplifier and controlled over a Dante or EtherSound network.*

OF NOTE: *Using NEXO's HRW patented waveguide for optimum HF coupling, the M620 performs in a variety of configurations, facilitated by a 3-point rigging system. GEO M6B bass extensions can be deployed inside a M620 line array, allowing the user to increase line array length for improved directivity control and output in the low-mid frequency range.*

KEY SPECIFICATIONS:
Configuration: 2-way
LF/MF: 1 x 6.5-inch cone
HF: 1 x 1-inch-throat driver on optimized HR Wavesource
Frequency Response: 80 Hz - 19 kHz
Dispersion: 80 or 120 (h) x 20 (v) degrees
Rigging: Proprietary NEXOSkeleton rigging system; can also be groundstacked or pole-mounted on subs
Power: NEXO NXAMP amplifier/controller recommended (1 unit can drive up to 12 cabinets)
Weight: 21.4 pounds
Size: 7.4 (h) x 14.6 (w) x 10.2 (d) inches

Martin Audio MLA Mini
www.martin-audio.com

Configuration: 2-way (cellular drive)
LF: 2 x 6.5-inch neodymium cone drivers
HF: 3 x 1.4-inch neodymium compression drivers on constant-directivity waveguide
Frequency Response: 76 Hz - 18 kHz
Dispersion: 100 (h at - 6 dB) x 10 (v) degrees
Rigging Angles: Vertical coverage can be fine-tuned electronically
Power: Self-powered (class D, 9 channels), multi-channel DSP
Weight: 30.4 pounds
Size: 8.3 (h) x 19.7 (w) x 14.8 (d) inches

K-array KH15
www.k-array.com

Configuration: 2-way
LF: 2 x 8-inch neodymium cone drivers
HF: 2 x 1-inch neodymium planar wave drivers
Frequency Response: 70 Hz - 20 kHz
Dispersion: 120 (h) x 15 (v) degrees
Rigging Angles: Adjustable
Power: Self-powered (class D), 2 x 750 watts; DSP onboard
Weight: 34 pounds
Size: 6.3 (h) x 24 (w) x 10 (d) inches

QSC WideLine-8 WL3082
www.qsc.com

Configuration: 3-way
LF/MF: Dual 8-inch neodymium cone drivers
HF: 1.4-inch-exit neodymium compression driver
Frequency Response: 68 Hz - 18 kHz
Dispersion: 140 x 10 degrees
Rigging Angles: Adjustable in 1-degree increments from 0 - 10 degrees
Power: Tri-amped; LF - 250 watts, MF - 250 watts, HF - 85 watts
Weight: 38 pounds
Size: 9 (h) x 20 (w) x 15 (d) inches

TURBOSOUND Flashline TFS-550H
www.turbosound.com

The Flashline TFS-550H is a compact yet powerful 3-way bi-amped line array designed for a wide range of medium-scale touring and installation applications. Each module utilizes dual high-excursion 6.5-inch neodymium cone drivers with compression loading, along with a high-efficiency 4-inch neodymium Polyhorn MF driver and a proprietary 1-inch neodymium HF compression driver on a Dendritic horn.

The cabinet is symmetrically loaded to create coplanar symmetry with smooth and consistent horizontal and vertical coverage, the cylindrical domain equivalent of a coaxial arrangement for individual spherical sources. In addition, the precise positioning of the neodymium drive units saves space within the enclosure and enables a very compact footprint.

Recommended amplification is provided in the form of advanced, powerful 10000DP 4-channel amplifiers incorporating both Lake Processing and Dante networking.

TECHNOLOGY FOCUS: *Utilizing TURBOSOUND technologies deployed in the larger Flashline and Flex Array systems, the HF Dendritic waveguide has multiple hornlets with identical path legends for a flat constant-phase wavefront with a horizontal dispersion pattern of 100 degrees. When coupled to the horn, the waveguide shaped the spherical wave of the compression driver into the cylindrical wavefront – a key component of the line source array behavior.*

OF NOTE: *The touring flybar provides multiple pick-up points, and allows arrays of mid/high enclosures, bass cabinets, or a combination of both types to be flown in a visually unobtrusive array.*

KEY SPECIFICATIONS:

Configuration: 3-way
LF: 2 x 6.5-inch neodymium cone drivers
MF: 1 x 4-inch neodymium cone driver
HF: 1 x 1-inch compression driver
Frequency Response: 80 Hz – 20 Hz
Dispersion: 110 (h) x 8 (v) degrees
Rigging Angles: 0 to 12 degrees in 1-degree increments
Power: 10000DP amplification recommended (DSP, networking); LF: 400 watts peak, MF/HF: 240 watts peak
Weight: 29.5 pounds
Size: 7.6 (h) x 21.7 (w) x 15.6 (d) inches

Electro-Voice XLD281
www.electrovoice.com

Configuration: 3-way
LF: 1 x 8-inch neodymium cone (EV DVN2080)
MF: 1 x 8-inch neodymium cone (EV DVN2080)
HF: 2 x 2-inch voice coil neodymium drivers (EV ND2S-8)
Frequency Response: 60 Hz - 20 kHz
Dispersion: 120 x 10 degrees (90 degrees horizontal available – XLD291)
Rigging Angles: 1-degree increments
Power: Biamp and triamp modes (200/200/80 watts)
Weight: 48 pounds
Size: 9.9 (h) x 28.6 (w) x 14.5 (d) inches

VUE Audiotechnik al-4
www.vueaudio.com

Configuration: 2-way
LF: 2 x 4-inch Kevlar neodymium cone drivers
HF: 1 x 1-inch beryllium diaphragm compression driver
Frequency Response: 100 Hz - 18 kHz
Dispersion: 90 (h) x 10 (v) degrees
Rigging Angles: 0, 2.5, 5 and 7.5 degrees between array elements
Power: V4 Systems Engine (power, DSP, networking)
Weight: 19 pounds
Size: 10.3 (h) x 19.9 (w) x 5.5 (d) inches

FBT Mitus 206LA
www.fbt.it

Configuration: 2-way
LF: 2 x 6.5-inch neodymium cone drivers
HF: 1 x 1.4-inch neodymium compression driver on a waveguide
Frequency Response: 68 Hz - 20 kHz
Dispersion: 100 (h) x 10 (v) degrees
Rigging Angles: Adjustable in 2-degree increments from 0 to 10 degrees
Power: Self-powered (class D), LF - 600 watts, HF - 300 watts; DSP
Weight: 30.8 pounds
Size: 7.5 (h) x 22.6 (w) x 15 (d) inches

RWG Spotlight Listing

L-Acoustics Kiva II | *www.l-acoustics.com*

Kiva II is a new ultra-compact modular line source delivering an impressive peak SPL of 138 dB (6 dB of max SPL versus its predecessor), maximized amplifier density with 16 ohm impedance, and a new rugged enclosure material.

Weighing only 31 pounds (14 kg), the product's elegant lines and flush-fitted rigging allow it to melt into any architecture, making it a natural fit for installations in performing arts centers and special events demanding minimum visual obtrusion, and particularly in L-ISA multi-channel configuration installations.

Beside the whopping SPL, Kiva II features a reinforced weather resistance, a rigging visual safety check, and new rigging accessories to expand the range of applications.

KEY SPECIFICATIONS:
Configuration: 2-way
LF: 2 x 6.5-inch neodymium cone drivers
HF: 1 x 1.75-inch compression driver
Frequency Response: 70 Hz - 20 kHz
Dispersion: 100 (h) x 15 (v) degrees maximum
Rigging Angles: 0 to 15 degrees, 1-degree increments from 0 to 5 degrees and 2.5-degree increments beyond 5 degrees
Power: L-Acoustics LA4/LA8/LA12X amplified controller
Weight: 31 pounds
Size: 6.9 (h) x 20.5 (w) x 14.1 (d) inches

TECHNOLOGY FOCUS: *Despite its ultra-compact format, the Kiva II features L-Acoustics' Wavefront Sculpture Technology, giving it a long throw capability and delivering even SPL from the front row to the back of the audience. Its coplanar transducer arrangement and new K-shaped coplanar transducer configuration generates a symmetric horizontal coverage of 100 degrees without secondary lobes over the entire frequency range.*

OF NOTE: *The internal passive crossover network uses custom filters. L-Acoustics amplified controllers ensure the linearization and the protection of the transducers (L-Drive).*

RCF HDL10-A
http://rcf-usa.com

Configuration: 2-way
LF: 2 x 8-inch neodymium cone drivers
HF: 1 x 2.5-inch voice coil titanium compression driver on custom waveguide
Frequency Response: 65 Hz - 20 kHz
Dispersion: 100 (h) x 15 (v) degrees
Rigging Angles: Adjustable in 2-degree increments
Power: Self-powered, digital amplifier, split 500 watts for LF, 200 watts for HF (1400 watts peak); DSP
Weight: 45 pounds
Size: 11.5 (h) x 22.4 (w) x 17 (d) inches

Adamson Systems Metrix
www.adamsonsystems.com

Configuration: 3-way
LF: 1 x 8-inch Kevlar neodymium cone driver
HF: 1 x 1.4-inch-exit compression driver
Frequency Response: 95 Hz - 18 kHz
Dispersion: 120 (h) x 5 (v) degrees
Rigging Angles: Adjustable at 6 positions of splay
Power: LF - 250 watts AES, 1000 watts peak; HF: 110 watts AES, 440 watts peak
Weight: 53.5 pounds
Size: 8.6 (h) x 21.2 (w) x 16.5 (d) inches

EAW RADIUS RSX208L
www.eaw.com

Configuration: 3-way
LF: 1 x 8-inch cone driver
LF/MF: 1 x 8-inch cone driver
HF: 2 x 1.4-inch voice coil compression drivers (horn-loaded)
Frequency Response: 65 Hz - 18 kHz
Dispersion: 120 (h) x 12 (v) degrees
Rigging Angles: Adjustable in 3-degree increments, 0 to 12 degrees
Power: Self-powered, triamplified; modified class D, max output of 3 x 500 watts; DSP
Weight: 40.5 pounds
Size: 9.8 (h) x 27 (w) x 13 (d) inches

WorxAudio By PreSonus XL1
www.worxaudio.com

The new WorxAudio XL1 is a 2-way, high-efficiency line array loudspeaker designed to be an ultra-compact high performance system for the reproduction of speech and music program material. Available in both touring and install versions – both powered and passive – the XL1 Is an exceptional choice for a wide range of sound reinforcement applications.

The XL1 incorporates a large-format 1.4-inch-exit compression driver, coupled to a stabilized, proprietary Flatwave Former, to deliver clear but penetrating highs over a predictable and controlled coverage area. Dual 8-inch cone transducers coupled to A.I.M. minimize cone filtering throughout the operating range.

The sturdy multi-ply enclosure and powder coated steel grille withstands the most demanding portable applications. Custom flyware enables easy arraying to a precise coverage area. And with integrated Dante audio networking capability, powered XL1 versions offer a no-hassle, self-configuring, true plug and play digital audio networking experience.

OF NOTE: *AIM (Acoustic Intergrading Module); large format compression driver; controlled symmetrical pattern control; flat phase and power response; FlatWave Former (wave-shaping device); TrueAim Rigging offering 1 degree increments*

COMPANION PRODUCTS: *Powered versions of the WorxAudio XL1 incorporate the new PDA-1000R power amplifier with integrated DSP, Dante networking capability, and WorxControl (a loudspeaker management system). The PDA-1000R is a 2-channel, class D power amp that delivers 500 watts per channel and includes XLR input and XLR pass-thru.*

KEY SPECIFICATIONS:

Configuration: 2-way
LF: 2 x 8-inch neodymium cone drivers
HF: 1 x 1.4-inch-exit compression driver on proprietary Flatwave Former
Frequency Response: 65 Hz - 18 kHz
Dispersion: 160 (h) x 10 (v) degrees
Rigging Angles: TrueAim Tour rigging adjustable in 1-degree increments
Power: Choice of self-powered and passive; WorxAudio PDA-1000R amplifier recommended; DSP
Weight: 59 pounds
Size: 10.2 (h) x 24.5 (w) x 18 (d) inches

JBL VERTEC VT4887ADP-DA
www.jblpro.com

Configuration: 3-way
LF: 2 x 8-inch Differential Drive neodymium cone drivers
MF: 4 x 4-inch neodymium cone drivers
HF: 2 x 1-inch neodymium compression drivers
Frequency Response: 55 Hz - 22 kHz
Dispersion: 100 degrees horizontal
Rigging Angles: Selectable splay increments between 0 to 15
Power: Self-powered (DSP/networking) Total power is 1100 watts continues, 2200 watts peak
Weight: 87.5 pounds
Size: 11 (h) x 31 (w) x 22.1 (d) inches

Alcons Audio LR7
www.alconsaudio.com

Configuration: 2-way
LF: 1 x 6.5-inch neodymium cone
HF: 1 x RBN401 4-inch pro-ribbon driver on a proprietary "Morpher" lens
Frequency Response: 74 Hz - 20 kHz
Dispersion: 120 or 90 (h) x 15 (v) degrees
Rigging Angles: 0 to 15 degrees in 1-degree steps
Power: Alcons amplifier/controller recommended
Weight: 17.6 pounds
Size: 6.9 (h) x 14.1 (w) x 10.7 (d) inches

dBTechnologies DVA M2P
http://dbtechnologies.com

Configuration: 2-way
LF: 2 x 6.5-inch neodymium cone drivers
HF: 2 x 1-inch neodymium compression drivers
Frequency Response: 68 Hz - 20 kHz
Dispersion: 90 (h) x 15 (v) degrees
Rigging Angles: Adjustable, several angles from 0 - 15 degrees
Power: 150 watts RMS, 300 watts peak
Weight: 15.7 pounds
Size: 7.5 (h) x 18.1 (w) x 13.6 (d) inches

Clair Brothers i208
www.clairbrothers.com

Configuration: 3-way
LF: 1 x 8-inch cone driver
LMF: 1 x 8-inch cone driver
HF: 1 x 1.4-inch compression driver
Frequency Response: 60 Hz - 20 kHz
Dispersion: 120 (h) x 10 (v) degrees
Rigging Angles: Adjustable at 0, 2.5, 5, 7.5 or 10-degree increments
Power: LF & LMF (each): 400 watts program, 800 watts peak; HF: 220 watts program, 440 watts peak
Weight: 62 pounds
Size: 9.2 (h) x 28.9 (w) x 23.9 (d) inches

Fulcrum Acoustic FL283
www.fulcrum-acoustic.com

Configuration: 2-way
LF: 2 x 8-inch ceramic cone drivers, proprietary cardioid deployment
HF: 3 x 1.4-inch neodymium drivers
Frequency Response: 54 Hz - 18.6 kHz
Dispersion: 90 (h), vertical is array dependant
Rigging: 4 to 20 degrees in 2-degree increments; 0- and 2-degree splays with optional rear link bar
Power: Single-amplified w/DSP; 500 to 1000 watts at 16 ohms recommended
Weight: 57 pounds
Size: 12.8 (h) x 21.3 (w) x 19.3 (d) inches

Outline MINI-COM.P.A.S.S. iMODE
www.outlinearray.com

Configuration: 2-way
LF: Quad 5-inch, Double Parabolic Reflective Wave Guide loaded
HF: Dual 1.75-inch, Double Parabolic Reflective Wave Guide loaded
Frequency Response: 100 Hz - 20 kHz
Dispersion: Asymmetrical horizontal directivity - steps of 15 degrees can be set between 60 to 150 degrees manually; vertical depends on array height and curvature
Rigging Angles: 0 to 7.5 degrees, .5-degree increments
Power: Self-powered, 2 x 500 watts; DSP
Weight: 52.9 pounds
Size: 13.8 (h) x 21.7 (w) x 16.4 (d) inches

D.A.S. Audio AERO 8A
www.dasaudio.com

Configuration: 2-way
LF: 1 x D.A.S. 8MN 8-inch neodymium cone driver
HF: 1 x D.A.S. M-60N 1.75-inch neodymium driver
Frequency Response: 95 Hz - 20 kHz
Dispersion: 90 degrees nominal
Rigging Angles: Adjustable in 1-degree increments from 0 - 10 degrees
Power: Self-powered; LF - 250 watts, HF - 100 watts; DSP
Weight: 39.6 pounds
Size: 9.6 (h) x 20.7 (w) x 14 (d) inches

BASSBOSS LA88
www.bassboss.com

Configuration: 2-way
LF: 2 x 8-inch cone drivers with multi-aperture midrange loading
HF: 2 x 1.7-inch diaphragm compression drivers on isophasic waveguides
Frequency Response: 50 Hz - 19 kHz
Dispersion: 140 (h) x 8 (v) degrees
Rigging Angles: Adjustable in 1-degree increments, with fixed front pivot point
Power: Self-powered, class D Powersoft amplifier (3,000 watts peak); DSP
Weight: 55 pounds
Size: 9.5 (h) x 24 (w) x 19 (d) inches

ISP Technologies HDL2208
www.ispproaudio.com

Configuration: 2-way
LF Drivers: 2 x 8 inch neodymium cone drivers
HF Drivers: 1 x 2.6 inch neodymium compression driver on multiple-aperture waveguide
Frequency Response: 68 Hz - 16 KHz
Dispersion: 100 (h) x 10 (v)
Rigging Angles: 1 to 10 degrees with 1-degree increments
Power: Self-powered, 3-channel proprietary DCAT amplifier (850 watts, RMS)
Weight: 62 pounds
Size: 9.1 (h) x 24 (w) x 19 (d)

Alto Professional TOURMAX SXA28P
www.altoproaudio.com

Configuration: 2-way
LF: 2 x 8-inch cone drivers
HF: 2 x 1.4-inch neodymium compression drivers
Frequency Response: 77 Hz - 18 kHz
Dispersion: 100 x 7.5 degrees
Rigging Angles: Adjustable at -20, 0, +20 degrees
Power: 800 watts program, 400 watts continuous
Weight: 48.1 pounds
Size: 10.7 (h) x 24.4 (w) x 16.7 (d) inches

SLS Audio LS8800
www.slsaudio.com

Configuration: 2-way
LF: 2 x 8-inch cone drivers
HF: 1 x PRD1000 ribbon driver
Frequency Response: 72 Hz to 20 kHz
Dispersion: 110-degree horizontal
Rigging Angles: 1 to 10 degrees
Power: Biamp; LF - 500 watts; HF - 60 watts
Weight: 60 pounds
Size: 9.6 (h) x 28.3 (w) x 13 (d) inches

Biema CAVA
www.biema.us

Configuration: 2-way
LF: 2 x 8-in cone drivers
HF: 2 x 1-inch compression drivers
Frequency Response: 100 Hz - 18 kHz
Dispersion: 90 (h) x 8 (v) degrees
Rigging Angles: 0 to 8 degrees, adjustable in 1-degree increments
Power: Self-powered (class H), 600 watts; DSP
Weight: 96 pounds
Size: 10.4 (h) x 31 (w) x 24.8 (d) inches

Renkus-Heinz has appointed **Alberto Mantovani** to the position of engineering manager. A senior engineer with more than 30 years of expertise in the technology sector, Mantovani began his career in Italy in the 1980s, developing complex embedded systems for factory automation. He also held high-level sales and technical marketing positions with Rockwell Semiconductors Systems in Europe before moving to California in 1997, where he co-founded two high-tech enterprises.

"Alberto comes to Renkus-Heinz with a unique and diverse range of skills and expertise, from product development and product management to team leadership and business development," states Renkus-Heinz CTO **Ralph Heinz**. "He has quickly become an important member of the Renkus-Heinz team, and we look forward to a long and mutually successful relationship."

Big Mix Capabilities For A Montreal Showcase

Yamaha digital consoles were deployed at 17 stages (12 stages at both front of house and monitors) for the recent 37th Montreal Jazz Festival at the Place des Festivals, which attracted more than two million attendees and presented over 500 concerts. The consoles were supplied by sound provider Solotech (Montreal), and the line-up included two new RIVAGE PM10s at the TD Stage, the largest of the outdoor stages.

Dany Legendre, front of house engineer at the TD Stage (pictured here), notes, "What I like about the Yamaha PM10 is the preamp. The system, without using any of the cool plugins, sounds great right from the start. But, if you want to get into the plugins, it's amazing what you can do with this console. The signal flow is really straightforward. If you're used to Yamaha consoles, it's really simple; it only takes about five minutes to be up and working on the console."

Gaétan Bouffard, monitor mixer at the stage, adds, "The learning curve of the PM10 was really easy. I became very confident about using it at the TD Stage as well as being able to show it to visiting engineers. The board sounds fantastic."

Dave Caulwell has joined **Riedel Communications** to oversee business development for the eastern U.S., bringing more than 15 years of experience in all aspects of event technology and production to the position. Most recently he served as vice president of production for TechniCom Audio & Visual, where he was responsible for the strategic direction and guidance of the production team as well as securing new business.

"Having served the production and live-event industries in key technical, operations, and management roles, Dave has a strong understanding of our potential customers' technical and business requirements," says **Joyce Bente**, president and CEO of Riedel North America. "In combination with his proven ability to build relationships and guide strategic growth, these strengths make Dave a valuable addition to our growing U.S. team."

Audio-Technica has recognized Riverdale, NJ-based **On the Road Marketing** with the Samurai Award for the 2015/2016 fiscal year, honoring the company's service to the upstate New York territory. The award was presented to On the Road team members **Tim Chamberlain** (senior account manager for upstate NY) and **Mark Meding** (marketing principal) by A-T president/CEO **Philip Cajka** and **Andrew Pernetti**, territory manager, professional markets.

"We're proud of Tim Chamberlain's success over the past year and to honor him with the Samurai Award," states Pernetti. "He has continually distinguished himself with a high level of service and support. We appreciate the relationships he has developed with his customers and his extensive knowledge of the A-T product line." LSI

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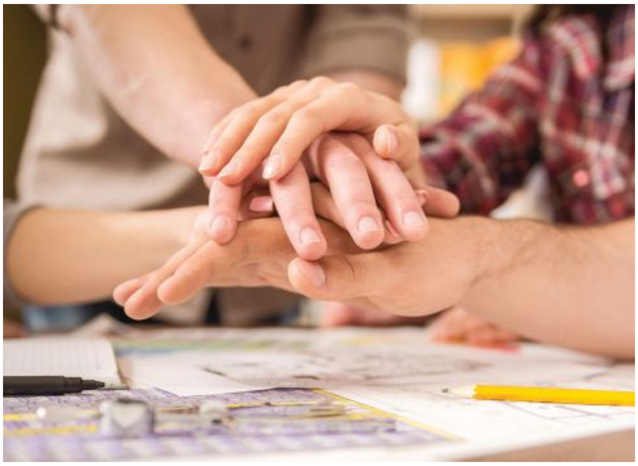
THE RIGHT STUFF

Insights on serving the actual needs of clients.

by Jonah Altrove



A few years ago while working as an A1 for a local production company, I took a call from a young married couple who held an annual jam band festival on their property. In previous years the rig had been a potluck of cobbled-together "prosumer" gear, but they were seeking to bring in professional production to deal with bigger acts and a larger draw. For an expected attendance of 6,000, they requested side-by-side dual stages (plus the accompanying dual roof systems and lighting rigs), as well as large LED screens, and of course, the requisite audio package.

I had a deep sense of foreboding. Being familiar with the region, my sense was pretty well-tuned about what sort of turnout could be expected, and I knew 6,000 was a pipe dream. I needed to tell them that their eyes were bigger than their budget. This can be tricky, because as production professionals, we must be sincere and receptive to the wishes of our clients and to try to meet their needs in a responsible manner. But notice the use of "needs" rather than "wants" in that previous sentence. In my view, an overkill production spec that the client will never pay off is just as impossible as a bare-minimum spec to meet requirements.

So I sat down with them to discuss the situation, stating, "Let me start by saying that we will rent you whatever you want. If this package is what you really want, we'll provide it. However, you've hired us not only to provide the equipment, but also because of our knowledge of these types of events. This is what I do for a living, and I need you to understand that I have serious concerns about the costs of a production this large. I know you don't like hearing this, but I'd rather you be a little upset now than very upset later when you owe a lot of money to a lot of people."

Although I happened to be the one doing the talking, this wasn't just my view – it was our company philosophy. Our first question with this and every other project: "What products and services are appropriate for the given application?"

The couple and I worked through every aspect of their production budget. We went out to the parking lot and paced off the requested stage dimensions. Actually seeing it helped them realize it was far larger than necessary. We completely cut the video package and also moved to a single-stage configuration and a slimmed-down lighting rig. I spec'd small-format arrays that would provide more than adequate audio coverage while reducing weight and power requirements. All of this allowed us to minimize necessary roof capacity and generator size, and it all added up to a price quote that was reduced by an order of magnitude.

If you've ever handled smaller/independent events, you can probably guess the outcome: 174 tickets were sold. After the show, the husband approached me and said, "You know, I'm going to have to sell my motorcycle to pay you guys, but if you hadn't talked me down, I'd have to sell my house. So thank you for not just giving me what I asked for."

While I later left the company to pursue freelance work, I continue to apply the same approach. As an independent engineer, I don't have a warehouse full of gear to rent to my clients; my job is to offer solutions and expertise, not equipment. I talk at length with them about their needs and how to best meet them. If I truly feel that another engineer or company will be a better fit, I tell them. By referring the gig to others who in turn do a great job, I preserve and strengthen my relationship with the client. This isn't the case if I'm poorly equipped and do a lousy job.

A referral also doesn't necessarily mean giving up the gig. There's nothing wrong with saying, "I'm committed to your event so I'd like to bring in this other entity as well to make sure we get it right." Communication is vital, because all of the gear in the world doesn't matter if our clients don't trust us to provide what's most appropriate.

There's an old customer service adage that conveys this all quite well: "Remember that customers don't want 1/4-inch drill bits. They want 1/4-inch holes." LSI

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