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LIVE SOUND INTERNATIONAL

25th ANNIVERSARY

INSIDE

YOUTH MOVEMENT

A "young gun" building a production enterprise

THE EVOLUTION OF SYSTEM OPTIMIZATION

ATTAINING QUALITY DRUM KIT SOUND

IDENTIFYING & FIXING MISTAKES IN THE MIX

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From the Editor's Desk

I recently had the pleasure of spending some time with Scott Ciungan, subject of our cover story in this issue, at his company's facility in suburban Detroit and came away heartened by the experience. There's a tendency for (at least) some of us "older" folks to get a bit down on the younger generation, which is natural and "ever thus." (I remember some of the looks

I got from more seasoned people as I was making my way.) But reality is a different story.

I'm fortunate in my work to have encountered many younger audio professionals of various stripes over the past several years, and can't recall a single negative experience. It seems, at least to me, that the future of the industry is in pretty good hands, as evidenced by my (admittedly limited) sampling and more specifically by bright, motivated and disciplined up-and-comers like Scott.

Elsewhere in the issue, Bob McCarthy shares his overview of one facet of the evolution of sound system optimization in the kickoff of a multi-part series, while Craig Leerman is back with another of his patented (well, they should be) primers on essential tools for the gig.

Joe Shambro imparts his experiences with dialing in drum kit sound, focusing on the miking aspect and then bringing in related factors to address. Ike Zimbel offers a nifty take on a very efficient approach to setting up a festival patch, and Andy Coules has some fun while illustrating common mix mistakes he encounters and what to do about them.

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

Keith Clark

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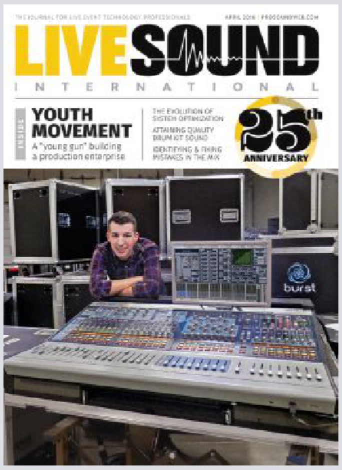

ON THE COVER: Scott Ciungan in the Burst Sound and Lighting shop with an Avid VENUE Series console.

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BPA WORLDWIDE

Audio-Technica ATM230

A hypercardioid dynamic instrument microphone designed for use with toms, snare and other percussion instruments. Joining the Artist Series, the mic's proprietary capsule is designed for high SPL applications, delivering well-rounded audio with solid low-end. Frequency response is stated as 20 Hz to 12 kHz. The hypercardioid polar pattern reduces pickup of sounds from the sides and rear. A low-profile design permits versatile placement around drum kits. Accessories include the AT8665 drum mount and a soft protective pouch. ***www.audio-technica.com***

Alcons Audio LR18

A 3-way line array incorporating the company's multiple-patented pro-ribbon technology for mid and high frequencies. The RBN702rs 7-inch pro-ribbon transducer and an acoustically and electronically symmetrical component configuration help enhance pattern control in both vertical and horizontal planes. Stated power handling of the RBN702rs is 1,500 watts with RMS-to-peak ratio of 1:15. The MF section offers a 6.5-inch midrange transducer with neodymium motor structure that is coaxially mounted behind the RBN702rs. The LF section consists of two reflex-loaded 8-inch neodymium woofers with 3-inch voice coils. The LR18 is driven by two channels of a Sentinel amplified controller equipped with drive processing for each array configuration, including presets for phase-matched LF extensions. Also available is the LB18, incorporating two of the same LF transducers. Adding LB18 modules enlarges array length, extending projection down to lower frequencies while also increasing the throw of low-mid frequencies over distance. The rigging enables angle-setting on the cabinets without lifting the array, with a WLL of 24 cabinets under 10:1 safety. ***www.alconsaudio.com***

Radial Engineering MR5

A MIDI signal translator to control the JX44 guitar signal manager, JX42 guitar/amp switcher, and other Radial products that use the JR5 footswitch. The unit's MIDI input and thru allow it to be daisy-chained with other MIDI devices. Dedicated "guitars" and "amps" XLR outputs connect to the JR5 inputs on the Radial products. A side access DIP switch enables the user to dedicate the MR5 to a single MIDI channel or select Omni mode. MIDI equipment connects with standard 5-pin DIN cables. Because the host Radial product automatically powers the MR5, no external power supply is required. ***www.radialeng.com***

Lectrosonics Wireless Designer Software

An upgrade to the package developed to enhance setup and operation of the company's wireless receiver systems, including Venue, Venue 2 and DR. Along with minor enhancements and workflow streamlining, the latest version now accommodates the importing of frequency lists and also the creation of custom frequencies. This allows for the software to include other wireless systems in the overall frequency coordination. HTML files from IAS (Intermod Analysis System) software from Professional Wireless Systems, with detailed frequency coordination information, can be directly imported.

The software also provides an overall view of Lectrosonics' wireless systems, including all receiver mainframes connected, with a summary of each channel displayed with real-time indications for essential levels and other settings. ***www.lectrosonics.com***

Waves Audio BSS DPR-402

A plugin that can be used as a straightforward compressor, de-esser or limiter. In addition, the processes can be combined. Waves has also added an MS matrix to separately process mid and sides, a mix control to balance processed and unprocessed signal, a noise control to add or remove the modeled inherent noise of the original unit, a gain reduction control, and an option for separate L/R metering. ***www.waves.com***

DiGiCo Stealth Core 2 For SD10, SD8

An upgrade option for both digital console models. With the SD10, the upgrade includes the following increases: from 96 channel strips to 132; 48 output buses to 56 + master + 24 x 24 matrix; 16 to 24 units of digital FX; and 16 to 214 each for DiGiTuBes, multiband dynamic options, and dynamic EQs. With the SD8, the upgrade the following increases: from 60 channel strips to 120; 24 output buses to 48 + master + 16 x 16 matrix, 12 to 16 units of digital FX; and 12 to 186 each for DiGiTuBes, multiband dynamic options and dynamic EQs. The Stealth Core 2 upgrade is due to be released this coming summer. ***www.digico.biz***

Rational Acoustics Smaart v8

A new version of the measurement and analysis software platform that offers greatly expanded control of the software environment while simultaneously providing easier and more streamlined measurement configuration, control and data handling. New facets include a more intuitive measurement configuration, new tab-based interface, multi-window capability, hide/show interface control bars, touch screen-friendly user-configurable command buttons, broadband metering for all devices, integrated control for the Smaart I-O, Smaart-to-Smaart API remote control, and more. *(Editor's note: For more about Smaart v8, see Factory Direct beginning on page 50 of this issue.)* ***www.rationalacoustics.com***

JBL HiQnet Performance Manager 2.0

The latest version of configuration/control software has a redesigned interface, accelerated workflows, and time-saving functions. New features include a drag-and-drop loudspeaker preset selection. Any loudspeaker can be dragged and dropped to any amplifier channel, and the program will create the custom preset configuration in the background. Additionally, JBL Line Array Calculator (LAC) integration has been improved and now offers the ability to import an array created in LAC with just one step. All circuiting information, level, and EQ adjustments developed in LAC transfer to Performance Manager and then to the amplifiers without having to manually move the parameters over from one software program to the other. Version 2.0 also supports new VTX M Series monitors and Crown Audio I-Tech HD firmware. ***www.jblpro.com***

TELEFUNKEN Stage Series

Three new cable lengths add further options to the microphone cable line, including 1, 2 and 5 meter options. The low-capacitance cables are made of 100-percent copper wire, with copper braid shielding for maximum RMI and EMI noise rejection. Non-standard lengths are also available, and models SGMC-10 and SGMC-10R (offered with a right angle XLR) both measure 10 meters (33.3 feet). Cable tie wraps are included. ***www.t-funk.com***

DPA Microphones MMC4018VL

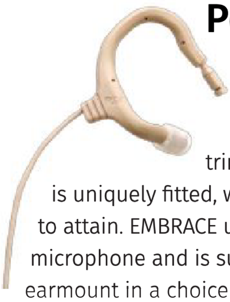
A new capsule for the d:facto handheld microphone that offers a more linear frequency response compared to the original model, which has a 3 dB soft boost at 12 kHz. The capsule has an isolation-optimized supercardioid polar pattern that helps to focus on the sound source with minimum bleed, also enhancing separation, and with SPL handling of up to 160 dB. The capsule can be removed and replaced with any other d:facto capsule to suit different performance requirements. Also, using DPA's adapter system, the mic can be transformed from a wired version to a handheld wireless microphone capable of integration systems from Sennheiser, Shure, Sony, Wisycom, Lectrosonics and Line 6. ***www.dpamicrophones.com***

Lab.gruppen Lake Controller v6.4.3

Provides users of PLM+, D Series and LM Series amplifiers with improved handling of unused modules in the controller while also eliminating several significant "bugs."

A new page called #Unused prevents modules from appearing on the "All Page." Moving any module to an #Unused page will prevent All Page visibility but will not change settings within the module. In addition, All Page is now sorted alphabetically by frame label, which is the amplifier model name by default. And, Lake LoadLibrary v3.8 is also included, offering new TW Audio loudspeaker presets. Lake Controller v6.4.3 is available for download from the company website. ***labgruppen.com***

Point Source Audio EMBRACE



A miniature microphone with pliable earmounts that can be trimmed and shaped so each mic is uniquely fitted, with refitting also quick and easy to attain. EMBRACE utilizes the company's SERIES8 microphone and is supplied with a left and right earmount in a choice of beige, brown or black color for matching hair and skin tone. For extreme camouflaging, the mics also accept theatrical color markers of almost any color. ***www.point-sourceaudio.com***

Turbosound iNSPIRE Series

A line of self-powered modular column loudspeakers currently comprised of iP1000 and iP2000 models. Both incorporate a 1,000-watt power amplifier. The iP1000 has eight 2.75-inch neodymium drivers, a horn-loaded tweeter and a dual 8-inch subwoofer, while the larger iP2000 offers sixteen 2.75-inch neodymium drivers, a horn-loaded tweeter and a 12-inch subwoofer. Also included with both models is a two-channel digital mixer with remote control via dedicated iPhone/iPad app, as well as local control provided via an LCD-based user interface. Wireless Bluetooth stereo audio streaming is also included. ***www.music-group.com/brand/turbosound***

SOMETHING IN THE AIR

Accounting for environmental changes between sound check and show time.

by Dave Rat

Ideas are magnetic and when a useful concept gains popularity, minds will refine and push the limits of innovation, often to the point of absurdity and well past the point of diminishing returns. Meanwhile as a technological feeding frenzy progresses, there are often gaps between productive practices and comprehensive solutions. These gaps are what I find especially interesting to explore and solve.

Take a moment and picture the scenario of overseeing a nightly posh dinner banquet. Each day setting up a table layout for a grand multicourse dinner with perfectly folded napkins, crystal champagne flutes, fine china and pristine cutlery placement. After attending to every detail, confident with your presentation and watching the guests arrive, you notice a rippling motion. The once stable table begins to move as if the tablecloth now lies upon an undulating water mattress.

Your exquisite meal wobbles upon the liquid-filled membrane while the guests obliviously lean and push more and more waves into rolling motion. Glasses are toppling, food sloshing, and a culinary chaos ensues. With a flurry of panic, you and the staff are reaching in all directions to regain control, sop up spills, and replenish the losses.

Just when it seems the entire meal is hopelessly lost to calamity, the intensity begins to ebb, all that will spill or break has already done so. The messes become familiar; you've replaced tall champagne flutes with wide bottom cups on stable saucers. All excess frivolity has been carted away. A balance has been reached and tensions relax with a new and different result than the original intent as the wonderful food and smiles ramp up into a meal to remember, albeit one with a rough start.

The following day once again the task begins, setting a table while carefully attending to each detail, and once again the arriving guests are subjected to a meal that starts as a chaotic mess – some nights more, some less, and sometimes it barely happens at all. You never know when or why and or even if it will happen at all until one day; oh, wait! The table is always made out of ice, and the ice sometimes melts to slush or water between preparation and meal time. It's not a mystery after all.

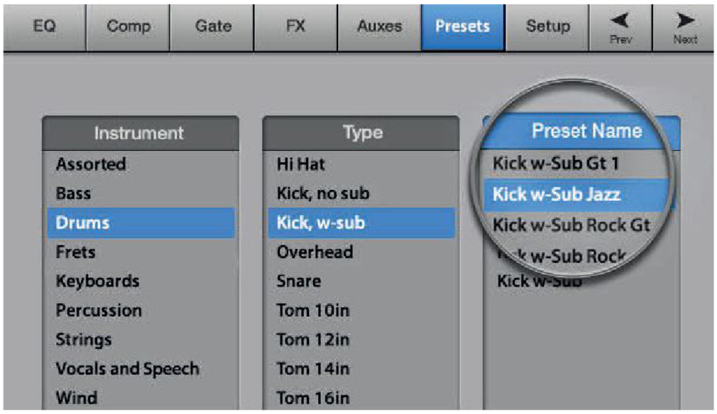
EVERYTHING'S PERFECT

In this equation, we as sound engineers are overseeing the meal, with the system techs as the waiters setting the table and serving the meal. The band is the chef in the kitchen, and the dinner guests are the audience. Not unlike the slosh-water-table parties, we too face the challenges of a changing and interactive workspace. Though the venues we work in have the same physical dimensions throughout the show, the acoustical properties of any given space are constantly changing.

It's common for us to spend a surprising amount of time fine-tuning every single detail of tonal balance, phase interactions, and loudspeaker positioning, resulting in complex EQ curves and a plethora of precisely calculated parameters. Meanwhile, we consistently find that the tonal balance and coverage of the system changes significantly from sound check to show time, and it continues to change throughout the performance.

Experienced engineers and technicians are well aware of these issues and have devised various methods to attempt to compensate. I've seen some make predictive adjustments based on experiences at similar venues. Last-minute line checks are not uncommon where an engineer actually listens to the instruments through the sound system during set change, re-EQing them to correct for any venue tonal shift. I know quite a few touring engineers, including myself, that avoid re-EQing during sound check and

POINT-AND-SHOOT **MIXING**



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

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

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

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

QSC™

instead rely upon the EQ settings from a previous show.

BRIDGING THE GAP

Why is it that the status quo of system tuning is so relentlessly shortsighted? Is the change in sound over the course of a show just some odd mystery that no one truly understands? Is calculating how the sound will change such an unsolvable complexity that we must make last-minute approximations after spending hours tuning?

"It will sound better once the room is full." The words have been repeated so many times that it's actually become humorous to the band as that tired trope of an explanation is offered. We have the sound *now* and then we will have a different sound *later*, and yet it seems we have no way of knowing what that other sound will be except different.

Can we bridge the gap? Is it possible to tackle the complex endeavor of predicting the way the audience and environmental factors will alter the sound? Is predicting the environmental effects of temperature and the influx of humans as complex as predicting weather patterns or launching a satellite, or perhaps is there a much simpler equation that for some reason effective tools have not yet been created?

Since we can know approximately where the audience will be and how many humans will attend, we can calculate the average skin temperature, body size, and about how much clothing humans tend to wear. So it can't be that complex to calculate the approximate sound absorption, sonic diffusion, and thermal effects that clusters of people will have. Also in most indoor venues, we have control over the room temperature, and for outdoor venues we can find out with a fair degree of accuracy what the temperatures will be throughout the show. If we were to accumulate and process this information, is it truly out of our realm to add it into our system tuning process and actually predict what the venue will sound like at show time?

Most of us are aware of what happens in a crowded venue using a sound system stacked just above head height. Basically, when the audience arrives, all of the high frequencies go away for anyone more than a short distance from the system. This issue is primarily caused by the sound refracting upwards when it encounters the heat generated by the audience. We know this while the prediction software (that I'm aware of) does not. So the software is not capable of showing us the advantages of flying the system higher to reduce this issue.

A similar issue happens with flown systems for outdoor events. During the day with the sun out and the ground warmer than the air, the refraction of the high frequencies upward can cause the sound to be duller and can significantly increase the coverage area. Later into the night, after the air has cooled and the temperature differentials have stabilized, the system gets brighter and the coverage area reduces. I encounter this phenomenon at multistage festivals with coverage areas that are significantly changing.

GETTING IT HANDLED

So out of all this, here are a few things:

1. The sound of an acoustic space is dynamic and constantly changing due to numerous factors. I've primarily focused on the thermal aspect here, but other factors include wind and diffusion/absorption by humans, as well as attenuation of frequencies over longer distances due to humidity.

2. Prediction software will not deliver correct results if the real-world environment is different than the prediction environment.

3. It's important to properly tune and align a system, but it's perhaps more important to prepare for what the acoustical space will sound like when people arrive and the environment changes.

Here are a few things I do to be prepared for and deal with constantly changing acoustics. First, knowing that the venue sound is constantly changing, I follow a "use the correct EQ" strategy. The channel EQ is to make the mic/instrument combo sound correct for headphones and recordings. Environmental venue changes have almost no effect on channel EQ so they stay set. I use the system processor to fix any issues with the sound of the loudspeaker system and array configuration. This is where I attempt to correct disparities in coverage distances, along with mechanical changes to the array positions. I use a house EQ as a hands-on control to compensate for the constantly changing venue sound. Quick, easy, and accessible as the cool air makes things brighter, I can compensate. As the warm audience heats up a cold room, I can add back in the frequencies that needed to be reduced earlier in the set.

For more than a decade I've been carrying a long-range digital infrared thermometer to measure temperatures through the venue. I work closely with the production manager and venue staff to create a consistent thermal environment. Usually, this involves shooting for a median venue temperature of 78 to 80 degrees (F). If the venue is colder, the differential between air temp and human temp tends to cause issues. If the median temp is much higher, there's risk of excessive heat for the audience in the upper balconies. The more consistent an acoustic space is thermally, the more predictable it will be.

I avoid hanging a single line array per side if there's potential for wind. Line arrays tend to be highly susceptible to wind gusts blowing the sound away from the audience. Hanging two arrays per side with some overlapping coverage greatly reduces this issue. Also, flying the PA higher tends to reduce the impact of heat issues radiating from the ground or from warm humans. And I take sound check system EQs with a grain of salt. The EQ from a previous good-sounding show in a similar venue is a much more trustworthy place to start.

Finally and most importantly, remember that a properly tuned system is one that performs correctly and sounds great from the moment the show starts and stays sounding great and covering well to the very last note. A perfectly tuned system in an empty room is about as useful as trying to serve a banquet dinner on a waterbed. LSI

Dave Rat *heads up Rat Sound Systems, a leading sound reinforcement company based in California, and has also been a mix engineer for more than 30 years.*

ENHANCING A STRANGE BREW IN AUSTIN

STRANGE BREW, Lounge Side in Austin, noted for hosting an eclectic calendar of acts ranging from local favorites to artists like Patty Griffin, Christopher Cross, and the Glenn Miller Orchestra, recently added QSC K and KW Series loudspeakers to its house and monitor systems. With its high, arching ceilings, intimate atmosphere, and stellar acoustical qualities, the room has a capacity of 160.

New QSC KW122 monitors in service at Austin's Strange Brew, Lounge Side.

"We've been an all-QSC venue everywhere since we opened, including outside on our patios and in the coffee house." states venue found Scott Ward. "Our guest artists — and it seems like everyone else too — have always raved about our sound right from when we first opened the doors," says Ward. "Our recent upgrade actually represents the second incarnation of QSC sound system we've used on the Lounge Side."

The newly upgraded house sound system incorporates a pair of KW Series loudspeakers flown left and right above the stage. A center fill hang residing between these two elements incorporates another single KW153 buttressed on each side by a K8 loudspeaker. Two KW181 subs extend the low frequency response of the rig, and as a final complement, a K10 loudspeaker is employed for side fill. And on stage, there's now four KW122s and a single KW152 for the drummer. Four K12s are also employed on an as-needed basis.

"We had a really big show the first night the loudspeakers were up-and-running," Walker adds. "I walked the room and stopped at different spots, and the sound was phenomenal. The upgrade actually took our original QSC system there to another level in terms of fidelity, clarity, and coverage."

DIGITAL MIXING THAT'S ALL WET IN THE UK

AN ALLEN & HEATH dLive digital mixing system is being employed by front of house engineer Steve Pattison on Wet Wet Wet's UK headline tour, which recently included a date at London's O2 arena. Supplied by SSE Hire, the dLive S7000 control surface is partnered with Allen & Heath's DM64 MixRack to handle an on-stage line-up that comprises the Scottish four-piece's traditional backline, multiple vocals and horn section.

Steve Pattison with the Allen & Heath dLive control surface he's utilizing for Wet Wet Wet.

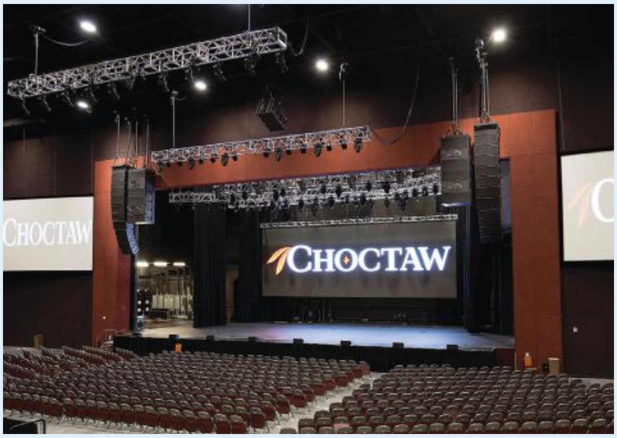
"It really feels like dLive is a step on from the iLive system that I've enjoyed using with various artists over the years," says Pattison, who's resume includes a five-year stint with Amy Winehouse, Ellie Goulding, Texas, Glasvegas and Royksopp). "It's very fast to use, despite the amount of control and features it has, and it sounds amazing.

"There's a very 'widescreen' feel to the sound," he continues, "small changes to the EQ or pan positions have a very obvious effect, for example, and really help things sit in a mix easily. And the big touch screens, which you can pinch and swipe like a smartphone, make it very easy to use in the heat of a show."

Project manager Pete Russell from SSE Hire looked after the tour and comments, "We see dLive as a real game changer product for Allen & Heath. It has some great features for a console at this price point and is something we were happy to add to our hire inventory as a result."

NEW OKLAHOMA VENUE GETS BIG STEREO SOUND

AIMING TO BECOME A regional destination for concert entertainment, Choctaw Casino Resort in Durant, OK has opened the Grand Theater, a multi-purpose venue outfitted with one of the largest Meyer Sound LEO Family installations in North America. Since opening with a sellout show by Aerosmith, the venue has hosted acts such as Kiss, Kid Rock, Boyz II Men, and Kenny Rogers.



Meyer Sound LEO arrays flanking the stage at the new Grand Theater in Durant, OK.

The LEO loudspeakers were chosen for the 3,000-capacity venue by Las Vegas-based consulting firm Coherent Design, which provided the sound design. System integration for theatre audio, along with all other AV systems in the broader resort expansion, was contracted to FBP Systems.

Coherent Design's David Starck specified a main system of eight LEO-M over four LYON-M line array loudspeakers per side, four LYON-M loudspeakers as center fill, and two flown arrays of six-each 1100-LFC low-frequency control elements in a cardioid configuration. Eight JM-1P arrayable and eight UPJunior VariO loudspeakers supply out fill and front fill, respectively.

Four MJF-212A and 12 USM-1P stage monitors provide onstage foldback, while two UPQ-1P and four JM-1P loudspeakers, one 600-HP subwoofer, and two 1100-LFC elements provide side and drum fill. Drive and optimization is provided by a Galileo Callisto loudspeaker management system with four Galileo Callisto 616 array processors.

"Listening to LEO is really like listening to a very high quality stereo system, with that level of imaging and detail," reports Doug Ebey, audio–video manager for the theatre. "What impresses me most is that we can get full, in-your-face concert sound and still stay within our preferred level limit, which we cap at 100 dB. Everyone has been thrilled with the performance of the system."

CONCERT SOUND IN EL PASO THAT'S ON THE MOVE

THE SPEAKING ROCK ENTERTAINMENT CENTER in El Paso, TX recently combined a Stageline SL260 mobile stage with a Martin Audio MLA loudspeaker system to bring quality sound to outdoor concert events, water parks, and multiple stage festivals. The mobile system consists of seven MLA and one MLD (down fill) cabinets per side, with four MLX subs ground-stacked in front of the stage.

"I like the versatility, audio quality, and the power behind each box which means I don't have to hang a lot of cabinets," says Scott Brown, the venue's production manager. "I can rig what we purchased and know the systems will easily fill the venue without killing the PA. The sound quality alone – the fullness, richness and intelligibility – is unbelievable.

"And MLA's control is very helpful because when we do the outdoor events, we're surrounded by residential neighborhoods and now we can make sure the sound doesn't bleed past the audience area. We don't get any complaints, but the audience still gets the full show."

Assembling one of the Martin Audio MLA arrays for the mobile system at Speaking Rock Entertainment Center in El Paso.

A RING OF FIRE FOR CEELO GREEN ON TOUR

DBTECHNOLOGIES DVX DM12 TH active stage monitors were deployed by front of house/monitor engineer Art Merriweather for CeeLo Green's recent "Love Train" U.S. tour. He notes that there were no instrument amplifiers on the stage but volume was still quite high.

Art Merriweather and one of the dBTechnologies DVX DM12 TH stage monitors on tour with CeeLo Green.

"This is hip-hop," he says. "They want it loud so they can feel it. And nobody on the tour likes in-ear monitors, so we've got powered wedges for each musician and side fills at stage left and right."

With only four DVX DM12 TH wedges on the tour, Merriweather moved them around from show to show. The DJ was given a "Texas headphones" setup, while the percussionist used a pair of them plus a drum sub, and the sax player was happy with a single wedge.

"Everyone was impressed," Merriweather reports, "and the wedges were also perfect for CeeLo's 'ring of fire'." The "ring" is a stereo pair of monitors in front and another stereo pair to his left and right. Vocals and a DJ track feed the front pair and sax and percussion feed the side pair. "It's an awesome setup," he notes. "I added a touch of reverb and the DM12s were as loud as CeeLo wanted, with plenty of headroom so we had no issues despite the stage level."

A CONSOLE GOING BOLDLY WHERE NO MAN…

STAR TREK: THE ULTIMATE VOYAGE, produced by CineConcerts, is bringing 50 years of Star Trek to concert halls, with a tour that launched in Florida in January playing at 1,500-8,500 seat venues in more than 100 North American cities.

For the audio production, the producers tapped Specialized Audio Visual (SAVI) of Clifton Park, NY, which provided a Yamaha CL5 digital console for the front of house mix, joined by with two Rio1608-D and one Rio3028-D input/output boxes.

The Yamaha CL6 serving Star Trek: The Ultimate Voyage with system tech Chris Dietrich (left) and front of house engineer David Hoffis.

The tour includes a 33-piece live symphony orchestra, all members of the Czech National Symphony with special solo instruments, that performs music written for the franchise while iconic Star Trek film and TV footage is simultaneously beamed in high definition to a 40-foot wide projection screen behind the musicians.

"We chose a Yamaha CL5 because of the need for a compact, but powerful, front of house console with a high channel count and the ability to network the entire system on a Dante network," states Patrick Ostwald, rental manager at SAVI. One of the Rio boxes is used for video playback to take track inputs, while two more are on the stage for the live stage inputs. Auvitran Audio Toolbox units are located in the amplifier racks to provide Dante to AES outputs for the amplifier inputs.

"The flexibility of the Dante network combined with the perfect number of inputs, DCAs, and matrix outputs, makes the Yamaha CL5 the perfect console for what we need on this tour," states front of house engineer David Hoffis. LSI

SITUATIONAL AWARENESS

Identifying common live mix mistakes, and what you can do about them. *by Andy Coules*

As a sound engineer, you spend years honing the ability to subtly appraise a mix. You learn to zoom in on each individual element and then zoom out to the whole mix, rapidly making decisions about what is and isn't working so that you can tweak various parameters to bring those individual elements into one glorious whole. After a while it becomes sub-conscious, you're not even aware of it, it's just part of what you do. The only problem being that once you've acquired this ability, it's hard to turn it off.

Most of us got into sound because we love music, so it's inevitable that we'll end up at a gig as a member of the audience at some point. It's not easy having no back stage privileges -- no free beer, no artisanal catering, and only being allowed to occupy the space between the stage and the front doors for the duration. What can make this experience even worse is being exposed to common live mix mistakes. Here are five that I've recently encountered at gigs, as well as some suggestions for how they can be avoided.

Big Roomitis. This particular affliction can be observed in medium to large venues where bands suddenly find themselves thrust into larger rooms than they're used to playing in, often as a result of being the support band to a more established act. This often means the band can afford to bring their own sound engineer along for the first time, and they'll often engage someone who's cut their teeth in small rooms. So when greeted with a large PA in a large space, the engineer struggles to cope and produces a big messy mix.

Now obviously we all need to step out of our comfort zone every now and then to up our game and take things to the next level, so a little bit of thought can help ease the transition. The key thing to realize is that in small venues you're not really mixing the whole sound; there will always be a large amount of ambient sound coming off the stage. The obvious candidates are drums and amplifiers but bear in mind that the monitor system will also be generating a fair amount of level (unless the band's using in-ears, of course). So in effect, a big chunk of the mix is already there and you're just adding a top layer of sound which (hopefully) gives the mix elements the definition they need to stand out and work together.

However, as soon as you get into a bigger room with a large PA, the stage spill is much less of a concern so you're required to build the entire mix from scratch. This is where a lot of engineers fall down because if you're doing the same thing as in those smaller venues, the mix lacks body, so you try to compensate with extra level and end up with that big messy mix. The key to avoiding this problem is to adapt your approach to the mix to suit the room and the system you're mixing on.

Unknown Desk Syndrome. The great thing about analog consoles is that once you've figured out how to work one of them, you can pretty much operate them all, it's just a question of scale. Unfortunately this maxim does not extend to digital consoles. Each manufacturer has its own unique way of addressing the digital paradigm, which can make it difficult for engineers to get to grips with all of them. Hence you turn up at the show, often as a support band, with a limited sound check and thus a limited amount of time to figure out a decent work flow on an unfamiliar control surface. I know from painful experience that if you don't know the console it can slow you down and really hamper the ability to mix the show as you normally would.

The trick here is to do your homework. This starts weeks in advance of the gig/tour. Find out which console you'll be using, and if it's not one you've used before, contact the manufacturer to see if you can get some training. Many of them offer regular free training session and some are even happy for you to pop along to their facility to practice.

If this isn't possible or is geographically inconvenient, then the next best thing is to download the offline editing software for the console and become familiar with the basics. Most offline editing software resembles the desk itself so you can get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a head start or at the least gain a better understanding of the workflow. The act of pre-programming a show file can also help you get used to the console, and most desks allow the creation of custom fader layouts that enable organizing the faders in a familiar configuration, which can greatly help speed up your operation.

Big Bottoms. We've all been to gigs where the mix is pretty good overall, except for some reason the bottom end is uncomfortably loud, with the kick drum and bass instruments grossly out of proportion and threatening to overpower the whole mix.

Modern sound systems provide us with prodigious amounts of bottom end which, if deployed properly, helps to enhance the visceral experience for the audience. However, too much bass is just as likely to shake free the fillings in their teeth or induce intense nausea, so it's important to remember that just because there's lots of bottom end available doesn't mean you have to use it. Some people don't like to high pass bass instruments but a high-pass filter at 30-60 Hz can really tame a wayward kick drum or bass instrument and help bring the whole mix together.

Another key is to be aware of how the bottom end sounds at front of house and how that differs from the point where the subs focus. This point might not be obvious due to the acoustics of the room so it's always a good idea to walk around as much as possible during the sound check. Try to ascertain the point at which the bottom end is most prominent, compare that to how it sounds at FOH, and adjust accordingly.

Where's The Snare/Mandolin/Tibetan Nose Flute? This can apply to any specific instrument that is curiously absent

from the mix. I witnessed it most recently at a reasonably high-profile gig in London where the snare drum couldn't be heard at all.

At first I thought, "he hasn't noticed it yet, he's focusing on the top line, he'll get to it eventually" but as the gig drew on the snare failed to trouble the mix whatsoever. I even went down to stand by the mix position to see if it was a quirk of the venue acoustics, but there was no snare at that location either. I could just about hear it acoustically in the room (snares are pretty loud after all) but I was pretty certain it was completely absent from the PA. Eventually, after about 20 minutes I noticed it tip-toeing into the mix, as if it were sneaking in late and didn't want anyone to notice.

The best way to ensure everything is present and correct when the show starts is really very simple: just perform a full line check every time. It doesn't matter if nothing has been moved or changed since the sound check; doing a full line check shortly before show time is a good habit to get into. It takes mere minutes but can help prevent a multitude of mishaps, not to mention the unwelcome sight of stage hands scampering around, trying to find the fault, while the band has already launched into the set.

It's also a good idea to perform a review of all active channels every song or two, just to make sure that everything continues to be present and correct.

Wandering Microphones. This is more of a problem I've witnessed in small venues where the equipment is less than perfect and the crew less than attentive, but I thought it worthy of a mention as it can affect all gigs.

I've always been a strong believer that the key to getting a good sound is capturing the sound properly at the source. The right microphone in the right position can not only help you get the signal you want while rejecting those you don't, it will also require less EQ and channel processing to work in the mix. However, mics can easily move during the show, either because they're knocked or because they weren't fully secured in the first place, and when you're all the way over at FOH, you might not immediately notice that the mic has moved or that the sound it captures has changed.

The way to avoid this is first, make sure all mics are properly secured in place; tighten every joint of every stand and give them a little push to make sure they stay put. It's also a good idea to make sure the musicians and stage techs are aware of the correct positioning so that they can also spot and correct any wandering mics during the show. This can be particularly useful during the quick changeover of a festival show when you're busy out front and the gear is all being wheeled in on rolling risers (which is a prime time for mics to wander).

Well, that's enough for this time. In my next article, I'll be presenting another five mistakes recently encountered. And don't be discouraged – we've all experienced these (and more) over the course of our careers. The key is avoiding repeats... 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.*

YOUTH MOVEMENT

Mix engineer, system tech and sound company owner Scott Ciungan.

by Kevin Young

UNDER THE STEWARDSHIP of Scott Ciungan, Burst Sound and Lighting has grown substantially since he took over the company in 2013. But when the 23-year-old first started working at the Detroit, MI-based production firm in 2005, he was still in high school. It was a weekend gig where he played the role of "shop kid," cleaning cables and anything else that needed it. Soon enough, however, that changed as he became increasingly involved with the company, eventually serving as then-owner Brian Johnson's right-hand man.

From its founding in 1996 through the 2000s, the original Burst was a production provider for the underground dance music scene, so when Cuingan got his start, he was primarily working raves in abandoned buildings located in and around Detroit. While that sounds like a recipe for some "sketchy" situations, it actually proved quite instructive.

"There was the 24-box (EAW) KF850 rig having to go up three flights of stairs, as well as dealing with bad power, but I was never involved in a police raid or anything like that," he says. "Still, it was eye opening; going from never doing that before to standing in front of a system that's the loudest thing you've ever heard with 1,000 people on hand, so you learn the ropes."

A main system supplied by Burst at the 2015 Electric Zoo Festival in New York.

FALLING UPWARD

Born and raised roughly 30 minutes outside of downtown Detroit on an island in the Detroit River called Gros Ile, Ciungan has been hanging out in the city since his early teens. "So I basically got my troubled youth days out of the way pretty early on and now I'm able to focus on running a company," he notes, adding that his work with audio came about as much by circumstance as anything else: "I just kind of fell into it, liked it and it progressed from there."

By his senior year of high school, he was working at Burst full time. "It's funny. I was actually answering client phone calls in the middle of math class. Then after high school, I moved downtown and got real busy." In addition to running Burst, Johnson was also serving as the theatre manager at the high school, which is where the connection of the two was made. And even though he was handling sound and lighting for plays and other events at the school, Cuingan still wasn't sure what he wanted to do with the rest of his life.

"It wasn't like I went to shows and said, 'I want to do this forever.' I wouldn't be here without Brian," he expounds. "If I'd never met him I wouldn't know anything about pro sound and lighting, or even have had an introduction to it."

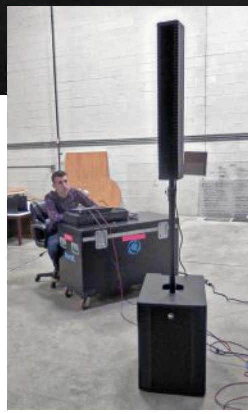
Ciungan's education in production was completely hands on. "I was just thrown to the wolves," he says, laughing.

"Brian and Larry Palmer – who owned another sound company around here, LCP Audio – were good friends. I started working with Larry part time as well, and between him and Brian it was pretty much, 'Here's a mixing console. There's the band. Make it happen.' From there, through trial and error, I figured it out and went from surviving to it becoming more of an artistic pursuit."

He continued learning the ropes, growing into a day-to-day role as Johnson's health ultimately took a turn for the worse. "Brian had been doing this for 15 years, and after a 10-year battle with cancer he was ready to close the book on the whole thing, but I asked him to give me a shot to see what I could do."

Burst served its first festival in the summer of 2011, the year Ciungan graduated high school. "I was actually advancing the festival through my finals," he notes. By age 21, he acquired the company after negotiating an asset purchase agreement with Johnson's family.

"Brian had been sick for a long time and the plan was for me to take over, but things happened quicker than we'd all hoped. He passed away in January 2013, and his family had no interest in the entertainment industry, so I bought Burst's assets and contracts and started my own company, Burst Sound and Lighting LLC, later that year."



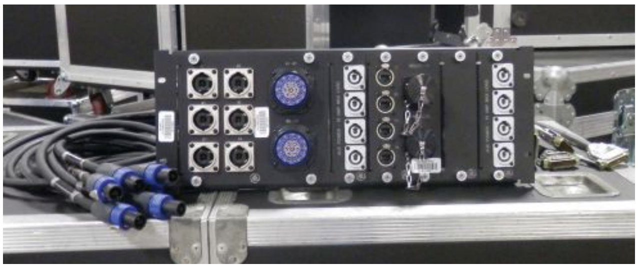
Ciungan evaluating an QSC TouchMix mixer and an RCF EVOX column loudspeaker and subwoofer in the Romulus shop.

MULTIPLE HATS

"I was an audio guy, a systems engineer. That's what I really like. I can also get by as a programmer now as well," Ciungan says. But the first couple of years of ownership were a whirlwind of other responsibilities, where his roles included selling and advancing shows and events, system tech, prep and de-prep, truck loading and unloading, scene changes, and business management – handling billing, payroll and keeping the books.

He remains Burst's sole managing member, but recently there's been some restructuring and rebranding, as well as bringing in solid, silent financial partners to help the company move forward. It's paid off in a big way, with Burst expanding its reach in the festival market as well as a big push in national touring, corporate events, and television and film applications.

"Our focus in past years has been festivals," he explains. "We have a team that really understands how festivals work and can execute them well. They're a different animal from a tour or a corporate show, and now I'm shifting a lot more responsibility to my team, which allows me to focus on progressively pursuing new ideas and better ways to do things."



A prototype of the panel with a Socapex module designed for a d&b systems.

The festivals the company currently works with include internationally known events such as the Electric Forest and Electric Zoo fests, Dlectricity, Movement and the Pemberton Music Festival in British Columbia. "Those are the main ones," Ciungan notes, "but we also do regional events and our roster is still expanding; doubling, maybe even tripling."

Roughly 80-percent of Burst's business is now done on a national level, with clients from out of state and Canada, all around North America. "Regionally, on the other hand, we're kind of a dry hire, rental house supporting other lighting and audio companies around here," he adds.

SIGNIFICANT GROWTH

Ciungan recently moved the company to a 35,000-square-foot facility in suburban Romulus, MI. It offers 28-foot trim heights and 3,500-square-foot pre-visualization suites so lighting designers can come in to run through and program their entire shows. There's also an overnight break room that programmers can sleep in if they're working late, along with a post-production audio suite to mix down live recordings, set up mixing consoles, go through show files and more.

The new digs also include a 3,000-square-foot continuing education space in which Ciungan plans to host Smaart classes, system optimization courses, lighting controller training, and rigging workshops. "Anything that we can do to keep our staff on their toes and stay up with progressive technology," he says, adding that the facility acts as the main preparation area for the festivals and tours the company works.

Burst has also just opened up sales offices in Los Angeles and Nashville, along with an office/warehouse in Las Vegas, stepping up its overall presence and sales force in the national market. All of that growth, though it had been in the works for some time, took place in January of this year.

"The Vegas location came into play to tailor (services) to TV and film, and to take advantage of a tax incentive increase the state of Nevada had just announced," he explains. "It will also be home to a corporate division as well. We have a great team there, people who've been in the area a

Every relationship we've developed, even from little gigs, can lead to our next world tour, so we treat everyone as a potential client.

long time and have a lot of experience in the field. We're also making a big push on international touring as well, with new staff members that come from production management backgrounds who understand the logistics and moving parts of having gear and a crew on the road."

It's been a frenetic path to tread over the recent past, fraught with risk but also the potential for big rewards. "I may be a bit over-ambitious, but maybe not. I just figured 'go for it'," Ciungan continues, "We moved out of our original space into a 10,000-square-foot building and wondered what we were going to do with all of that space. Then a year and a half later we moved to this 35,000-square-foot space.

"Over that time I developed some really good relationships with festival producers and tour managers. Cheap Trick was actually our first touring act and that relationship has led to introductions to others. They're touring this summer with Heart and Joan Jett, and their production team has become really good friends of ours. They've signed off on our quality of work and how we do things, and that's been incredibly important to our growth. But every relationship we've developed, even from little gigs, can lead to our next world tour, so we treat everyone as a potential client."

CONTINUING STRIDES

The company's growth was entirely organic in the early days. "When I took over, I didn't sit down and write out a five-year business plan," Ciugngan notes. "Now I do, of course, but to this point it's developed on an as-needed basis."

He's taken a relatively conservative approach to the growth of the company's equipment inventory: "We try not to blindly buy gear in the hopes of renting it out. We carefully analyze client needs and current market demands, and tailor our inventory to fit that."

His first significant investment was an Adamson E Series concert rig, and that's proven to pay off, with Burst maintaining an ongoing relationship with the company. Other components, such as Avid VENUE digital consoles, have been around for several years and continue to fill valuable roles.

Burst also has a fledgling division called Mod Works that is developing a line of interconnect panels and modules. An example is a patent-pending panel with a Socapex module designed for a d&b audiotechnik system with a standard W-2 and powerCON in/out module, developed to make assembly faster and bring down cost down. "We have a line of different panels and modules that will be released within a month or so," he adds. "At this point the designs are done and we're just optimizing them."

The company's staff has grown from Ciungan and a couple of independent contractors to 10 full-time employees and approximately 100 independent contractors who serve where needed on a regular basis.

"Burst went from a small regional company doing raves and the occasional festival in state to handling high-profile festivals and national touring acts, supported by multiple offices," he says. "It's been pretty crazy and we have to hold on at times, but I have a big dream and ambition to go with it – and I guess I have a lot of energy, too – but with that growth there have been mistakes. I haven't been a CEO for 20 years so there are always things that I'm going to learn."

As for being the driving force behind a company's national growth at an age when many are just trying to figure out what to do with their lives, Ciungan concludes, "I think I have an easier time with it because the growth has been so organic. I've been used to a heavy workload for a long time. It was kind of a natural thing that happened, so there hasn't really been any shell-shock from being in a position like this." LSI

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

CAPTURING THE KIT

Miking and related factors for quality drum sound in the mix.

by Joe Shambro

In the world of live sound, certain topics are guaranteed to draw fellow engineers into a multi-hour discussion that ends in no agreement and a hefty bar tab. Drum miking is at (or near) the top of the list. Further, every situation is different. Sometimes it's best to just go with a pair of overheads and a kick mic, while at the other end of the spectrum, sometimes the situation calls for individual spot mics for all 48 inputs on the drum riser.

Further complicating the picture are important aspects in addition to actual mic selection and placement. Sure, that's got to be "right" but it has to be combined with aspects such as processing and the overall scope of the mix. With that in mind, here's a walk-through of my general workflow for attaining quality drum sound, appropriate for the style of music (and reinforcing appropriately for the space).

KICK MICS

Other than buried and unintelligible vocals, I can't think of a bigger source of eye-rolling among engineers than lousy and/or inappropriate kick drum sound. This can range from a 110 dB reproduction of a basketball hitting a gym floor with no true low-end characteristics to a super-hot signature that overwhelms the entire low end. Where to start?

First, let's think about aesthetics. Most pop, rock, and R&B music benefits from a kick sound that's somewhere in the sweet spot between the previously mentioned tonal atrocities; a nice, solid, controlled low end with appreciable attack on the high end. A great way to attain it is with a dual-mic approach; a mic inside the drum to capture the attack of the beater hitting the head and another mic located outside to fill out the low end punch.

Inside is typically handled with a boundary-type mic with high SPL capability and a relatively full frequency response (at least 20 Hz extension on the low end and above 10 kHz on the high end). The Shure BETA 91A and Sennheiser E901 are two common models for this purpose. Some engineers favor the signature of the vintage Shure SM91 in comparison to the newer models due to its different capsule and preamp.

Outside, the choice is usually a dynamic mic designed for low-frequency extension. There are a lot of options and several variables, such as the drum itself, tuning, and aesthetics. Most commonly I see the Sennheiser E902 (or the lower cost E602), Shure BETA 52A, and Audix D6. All can provide a tight, punchy low end with nice high end extension, and if need be, can stand on their own without an inside mic.

Other popular options in this regard include "old school" designs such as the Electro-Voice RE-20 and Sennheiser MD421. Another option is the Audio-Technica AE2500 dual-element mic with both cardioid condenser and dynamic capsules. The dynamic element helps delivers the aggressive attack of the beater while the condenser captures tonality of the shell. I'm also a big fan of the Heil Sound PR40 (and the kick-specific version, the PR48), a large-diaphragm mic that shows exceptional LF response. (In fact, I've used PR40s to record space shuttle launches several times with plenty of headroom to spare).

KICK WORKFLOW

My general approach with kick starts with the inside mic. I bring it up in the mix, gain it (or pad it) appropriately, and start by gating the channel. (Very rarely do I come across any drum that's tuned well enough to get by without using a gate.) Next, a high-pass filter is applied around 85 Hz, since the punch will come from the outside mic, and this is followed by a deep cut in the 200 to 250 Hz range to clear up any "boxiness."

I also generally find that a couple of cuts in the 800 Hz to 1 kHz range helps keep the key vocal areas of the mix from being cluttered by the kick's punch, and if necessary, a spike at around 4 kHz can add attack if there's not enough of the beater. Finally, before moving on to the outside mic I throw a bit of delay on the inside channel, just enough to push the location of the inside mic to the location of the outside mic. This aligns the two mics, a difference that can be clearly heard if the tops and subwoofers of the PA are aligned properly.

Attention then turns to the outside mic. If the system is running aux-fed subs, once again I spin up the gain and do a minor bit of tweaking, usually adding a gate with a fast attack, mid-fast recovery, along with some minor EQ cuts around 200 Hz and 1 kHz. Then I take the mic out of the tops to see how well it plays with just the subs and the inside mic. If it's a particularly hefty PA, I'll add the outside mic to the tops, but if it's a more analytical, linear system, I've found that sometimes the additional low end in the top isn't as well translated, adding smear and clutter.

After that's done, I usually play with the gate release time and range settings to dial in the desired "punch." Shaping the sound to fade out exactly when you feel is contextually appropriate goes a long way to a kick sound that sits well in the mix.

From there, compressing the kick is a matter of taste and necessity. Generally, yes, you'll be compressing – and it's up to you if you want to do a parallel compression situation or compress on the individual channel strip. Chances are, sort of compression will be needed to get it to sit exactly where you want it in the mix. I generally start with a compressor at a 3.5:1 ratio, quick attack and recovery for the inside, and a slower compressor at around 2:1 to round the edges off the low punch, as needed. Sometimes, compression isn't needed – I'm looking at you, excellent jazz drummers – but in most iterations of pop and rock music, it helps.

Smaller clubs present different issues, especially when the stage doesn't have much depth.

SNARE MICS

Working our way down the console, snare is the next part of the equation. This is done with a single mic or more frequently, with two – one on the top to capture the initial "crack" and another on the bottom to capture the sound of the snares against the bottom head, which give the drum a lot of it's personality. It's also useful with drummers who go back and forth between hitting hard and playing subtly.

Solid dynamic mics that can handle high SPL are favored for snare. The Shure SM57 and BETA 57 are considered solid choices, favored by many because they handle SPL well and graciously deal with the abuse being struck hard by drumsticks. Other options I see (and frequently use) include the Telefunken M80, beyerdynamic M201, and Heil PR22, depending on the drum and the drummer.

Also getting increasingly more use are miniature mics that clip right on the drum shell, such as the DPA d:vote 4099D and Audio-Technica ATM350 condensers, both of which stay well out of the way and can be flexibly positioned via goosenecks. Many of these choices can also be applied to bottom snare, but I tend to prefer condensers for this application – the pre-polarized nature of the design means that it's better at picking up subtleties, and many condensers now handle high SPL.

A classic approach on snare and toms.

SNARE WORKFLOW

Often I choose to gate the top snare mic while keeping the bottom mic open or gated very lightly with a very low range. This lets the soft notes blend through while only opening up the top mic for a more sizeable "crack." (Remember to always flip the polarity on the bottom mic as well.)

Compressing the top is usually necessary, and the common snare sound most often heard in pop and rock music these days is the product of a 4:1 ratio, quick attack, quick-medium release. But the individual method and amount of compression is mostly chosen based on the "flavor" you're going for.

MICS FOR TOMS

Toms can be captured a variety of ways, and most require a little EQ finesse along with a good gate and compressor. My personal favorite tom mics are the convenient clip-on Sennheiser E904 (and the less expensive E604), along with the Shure BETA 98A and the Heil PR28. Any mic with solid SPL handling should be up to the task, and clip-ons make setting things up very convenient.

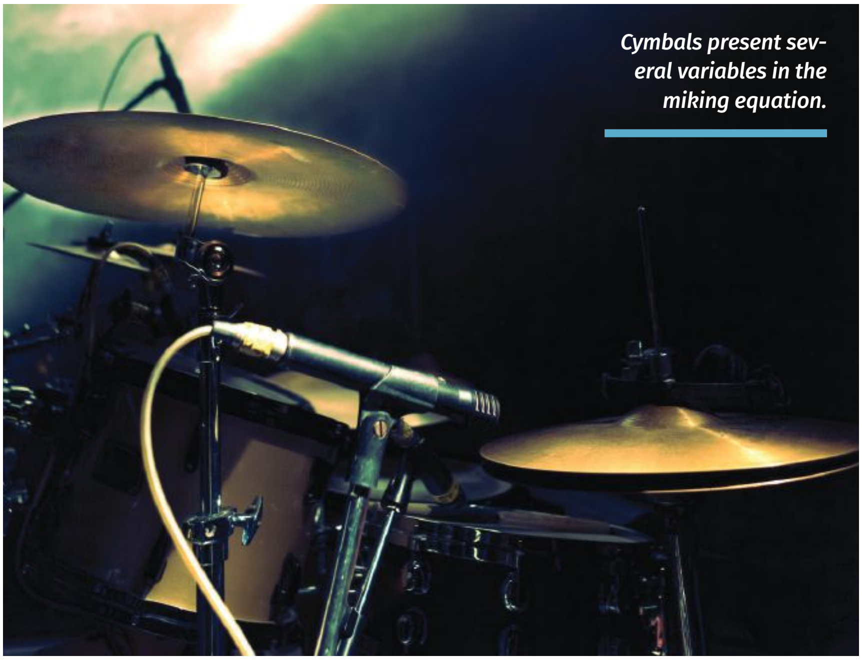
With floor toms, sometimes I like to experiment with a larger-diaphragm mic. The LF response of this drum matches up better with a mic suited for kick, and if there's space on the riser (and the drummer doesn't mind the bulkier mic), I almost always use this approach.

TOMS WORKFLOW

I start by gaining up my toms – placed into a group for compression – and placing a gate on them. I'll increase the gain and play with the hold and release times to get the duration of a tom hit I'm looking for, and then EQ to taste. Some mild compression, around 3:1, usually helps give body and substance to tom fills.

HAT, RIDE, AND OVERHEADS

Miking cymbals is something I tend to be very opinionated about. Not because I'm particularly passionate about certain techniques or mics, but because in my observation, at least, some engineers chronically overdo it, particularly in smaller rooms. Sometimes we forget that we're doing sound reinforcement: that magic "r" word means that we're there to provide extra horsepower to the items that need it – but not everything.



Cymbals present several variables in the miking equation.

My general rule is that if the venue's capacity is less than 500, I don't place overheads on a kit. Cymbals are exceptionally loud in small rooms, and generally there isn't the stage depth to get out of the vocal mics' range of pickup. Unless I've got a drum shield at my disposal, it's not necessary to mic the cymbals because of their natural projection. On the other hand, I almost always mic hi-hat, even in smaller rooms. It's such an important part of the mix that I want to have the control to bring it in when needed.

There are two primary choices for miking cymbals. First is the traditional overhead placement to the left and right of the kit, eyeballed to a reasonably equal distance on either side and placed high enough to capture cymbal sound as well as a small bit of the kit's overall ambient personality. The second approach is to spot-mic each cymbal closely. There's also the option of combining both techniques.

Generally, I place a set of condenser microphones – any reputable, reasonably priced brand will do – as an overhead pair, and then spot-mic the ride cymbal underneath the bell. I prefer an SM57 on ride, as I find its personality provides a pleasantly "dirty, trashy" sound to the bell, which augments the condenser's cleaner, more defined sound.

FINISHING IT UP

I group overheads into a stereo channel, rolling up the high-pass to about 500 Hz, with light compression to taste. Sometimes additional EQ attention is required around 2 to 5 kHz to remove some drum noise. Overall, overheads usually require minimal processing; just turn them up until you can hear them and they sit nicely in the mix, and you're good.

Processing drum channels from this point is a matter of personal preference. I usually build parallel compression groups with kick, snare, and hi-hat grouped together into a stereo group, along with a stereo group for toms, and another group for any additional instruments that end up on the drum riser. I'll create another stereo group as an unprocessed dry group, and put the overheads into this group after processing on their individual channel strip.

One more thing: making sure your buses are process-aligned is very important. I addressed this in the previous issue (*LSI March 2016*) in my focus on plugins. (The article is also available on ProSoundWeb.) So if you're unsure how to do this, check it out. LSI

Joe Shambro *is a free-lance live sound engineer and audio technology writer based in St. Louis.*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LOUDSPEAKER ADVANCEMENT

The evolution of large-scale sound system optimization, part 1. *by Bob McCarthy*

EDITOR'S NOTE: This is the first in a multi-part series. We present the first installment here, with others to follow in subsequent issues and available on ProSoundWeb.

We live in the present and plan for the future. Every once in a while it can be interesting to look back at our history and see he we got here. This puts things in perspective and helps us see what's coming.

The essential challenges to tuning a sound reinforcement system, a process we now call optimization, haven't changed in 40 years, and neither have the laws of physics. But the tools and techniques of the trade have changed dramatically, albeit incrementally, over time. As a person who has been in professional audio for 40 years, here's my perspective of how it was and the journey to the present.

Let's take a walk to front of house and meet our modern sound system. We see an engineered loudspeaker system and a multichannel digital signal processor loaded with all types of filters, delays, and more (maybe too much more). There's an acoustical analyzer ready to guide the optimization process and an experienced operator with a step-by-step plan for system tuning.

Not too long in the past, none of these things would be found there. Just 35 years ago we would likely find a collection of various custom loudspeaker cabinets built in the rental company's shop. The extent of the "signal processing" would be an analog crossover with fixed filter slopes and a graphic equalizer. At best we would find a primitive analyzer that no one trusted and little, if any, methodol-

An "old school" column loudspeaker, the Shure Vocalmaster, which was introduced in 1967 as the first complete system (mixer, amplifier and loudspeaker). Legend has it that the Beatles used this at Shea Stadium concert (1965). It was not, but the system on the field was not much more powerful than the Vocalmaster shown here. We're using it for our "Let it Be" concert on the roof of a dorm at Indiana University in 1975.

A 1979 concert with FM Productions at the Greek Theater in Berkeley. We are struggling with a custom 3-way box that has an HF horn tacked on the top of it (a primitive attempt at time-alignment). The box had minimal handles cut into it and an "L-track" screwed onto the side for rigging.

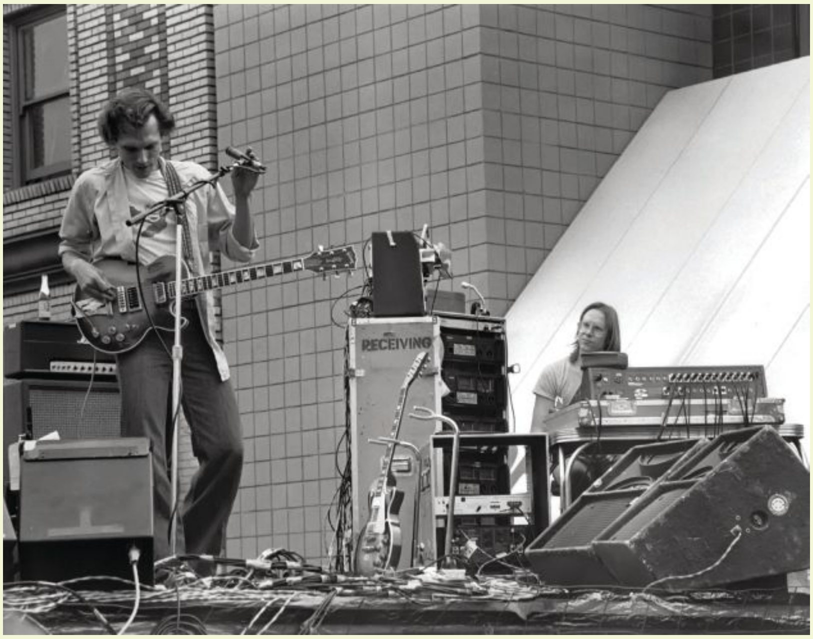
CREDIT: CLAYTON CALL

ogy or scientific process. All these aspects of our sound system have evolved in the period since, each in their own way and together as a whole.

I've been an eyewitness to the evolution of these tools and techniques into their present mainstream forms, as were countless other engineers. We can explore this history together by following five primary threads: the evolution of loudspeaker systems, signal processing, acoustical analyzers, analysis systems, and the methodology of optimization.

This is a personal perspective, not an authoritative history (which would be an entire book in itself). Every veteran of the pro audio industry has played some part in our collective progress, for which I am grateful. I encourage any and all to share their journeys with us and enrich our knowledge and respect for our past.

The central thread in my story is a 32-year relationship with Meyer Sound's Source Independent Measurement (SIM). This was the first analysis system capable of measuring the acoustic response of sound systems during a concert using the music as the test signal. The analysis system approach originated in SIM has helped lead to the analysis programs that are now so common that most live sound systems use some version of it today.



Jorma Kaukonen at the 1980 Berkeley Street Fair. This era marks the beginning of the transition from custom and user settable loudspeakers to systems (processed). I was mixing monitors and had the chance to compare the new "processed" system, the Meyer Sound Ultramonitor (on the left) to a pair of conventional "do it yourself" loudspeakers. The single UM-1 was much clearer and more powerful than the old school pair. There was no going back.

CREDIT: CLAYTON CALL

STONE KNIVES & BEAR SKINS

It may be difficult for the modern engineer to visualize the crude tools of the 1970s. The only commercially available professional loudspeaker systems were targeted for fixed installations such as cinemas and "public address" systems, and weren't the least bit ready for trucking, stacking or flying. The enclosures were not ruggedized with internal (or external) steel framing or integral rigging.

Popular music concerts needed portable power on a scale beyond the imagination of the loudspeaker manufacturers.

We all knew that you could either have high power or high fidelity but not both.

Rental companies innovated in developing mobile touring boxes while several manufacturers transitioned into the role of supplying the driver components inside them. At the time I entered the pro touring market, systems were described by which rental house made the boxes and the maker of the drivers, i.e., "a Showco system with JBLs." Nobody (at least that I knew of) was touring arenas with off-the-shelf boxes from a manufacturer. That was for high schools, cinemas, and "Hair" on Broadway.

Loudspeakers did not fly. They came out of the truck and stacked on the stage. The only ones that did fly were designed by consulting firms, usually comprised of a big pile of horns and a few woofers that we termed "flying junkyards." You can still see a few of these gathering dust in the ceilings of old arenas.

It got loud in front of our ground-stacked systems. It wasn't "hi-fi," it was rock 'n' roll. It was supposed to be rough and edgy. We all knew that you could either have high power or high fidelity but not both. The tradeoff in favor of power was simply accepted as normal as long as it got loud. Really loud.

The arena rigs I toured with in the 70s while at Showco and FM Productions were 4-way systems with 15-inch woofers, 12-inch woofers, horns and tweeters all in separate boxes. We ground stacked them on the stage and pointed them in the general direction of the audience.

The main system shown here for the Grateful Dead at the Greek Theater in Berkeley (1984) had minimal subdivision, entirely driven by a single channel of equalization with no delays or relative level adjustment. The standard optimization options of a modern system (macro and micro delay, relative gain adjustment and separate EQ for the uppers/lowers/side fills/etc. were not available options at that time.

CREDIT: CLAYTON CALL

The processing and amplifiers for the system at Orchard Hall in Tokyo, Japan in 1988 or so. This system had an abundance of subdivision: left/right, upper/lower, inner/outer, and so on. The amplifiers are in the left rack, and single channel loudspeaker processors are in the rack on the right.

For bigger shows, the stacks were taller and wider. Frankly, one of the key sound design principals was making sure it didn't fall over. Other companies, most notably Clair Brothers, made single boxes that were complete 4-way systems, but again, the same quantity and stacking principles applied.

I remember the first time we flew an arena system. We stacked the loudspeakers into a big steel basket with a plywood floor and up it went. For real. More "sound system in a freight elevator" than "flying system" but the progress toward level uniformity was amazing. It did not have to be insanely loud in front to be stupid loud in back!

The signal processing of the time was comprised of a 4-way crossover that drove the whole system, and simple limiters/compressors. Phase alignment was mostly just talk, since we had no tools to control it (no delay lines) or quantify it (no analyzer). This left us with the visual version: physically lining up the boxes as best we could. Again, the "don't let the speakers fall over" principle came first.

Level adjustments were done at the crossover or at the amplifiers, the latter being a notoriously difficult method to obtain consistent results. Typical level settings for power amplifiers in that era were "2 clicks down" or "3 o'clock." We had very little idea what the actual gain values were or how two different models related to each other. (By contrast, the modern amplifier can be software controlled in precise settings read in dB.)

MAKING PROGRESS

The concept of the engineered system barely existed in the 1970s. Custom was king and users were expected to be able to add their preferred touches to the settings of the crossover and other electronics. Engineered systems, by contrast, fixed these parameters to precise settings selected for the particular combination of drivers, horns and enclosures.

This doesn't sound radical now, but it sure was then. My first encounter with an engineered system was the rental company McCune Sound's JM-10, designed by a young John Meyer. I stood there stunned as one of my long-held beliefs was shattered: yes, you could have high power and high fidelity at the same time in an arena.

The 1980s were a time of tremendous evolution in loudspeaker technology. For one thing, the dinosaur boxes with separate lows, mids, horns and tweeters gradually became extinct. Basically the industry settled toward two box types: "full range," i.e. 2-way or 3-way boxes that covered from 70 Hz on up, and subwoofers for 100 Hz on down. Manufacturers also began to make ruggedized enclosures ready for touring. The engineered systems up to this time had been kept as exclusive and proprietary by the rental companies. We now began to see this approach become non-exclusive as the manufacturers embraced it.

The integration of multiple components into a single box with fixed physical characteristics set the stage for the next level: the introduction of "processed" systems. Loudspeakers were sold as a "system" with dedicated loudspeaker controllers that provided fixed crossover settings, amplitude and phase response correction, and limiters. All of these parameters were pre-optimized for the specific driver/box combination.

It was the beginning of the "plug and play" paradigm that has become the industry standard today. It may be surprising to know that this was highly controversial at the time. Many engineers felt that the manufacturers were limiting their options for optimization by locking out parameters such as crossover frequency and slope, etc.

Part of the resistance was fear of the unknown, since what was going on inside the proprietary controllers was mysterious. Another part was the feeling that manufacturers were taking away a part of their tool set. Many engineers took pride in their personal crossover setting

A highly subdivided center cluster in a symphony hall with extremely complex 360-degree seating (and only one hang point). The front mains have upper/middle/lower subsystems with inner/outer. There are three subsystems for the sides and another for the rear. A total of nine channels of signal processing (EQ, level and delay) were needed. Each cabinet has independent pan and tilt to help facilitate optimal vertical and horizontal aim and splay.

A modern era line array system (and the author) at the Appel Room, Jazz at Lincoln Center in New York. The main system is subdivided into four channels of signal processing (sub +3), and the various fill systems have independent processing as well.

skills and did not want to surrender that control to others.

It took a long time for them to realize that a complete box with fixed, known parameters is better engineered in a manufacturer's research lab than on the job site. The benefits of fully engineered systems over the custom recipe from Joe's Garage became so apparent that the custom boxes of the Wild, Wild West era rode off into the sunset.

BRINGING IT TOGETHER

Modern professional systems all have fixed pre-optimization. We buy "systems," not just loudspeakers, and expect them to be fully engineered with optimized crossovers, pre-aligned frequency response, appropriately scaled power amplifiers, and dynamic protection. They come in two varieties: self-powered and those with external amplifiers containing dedicated presets. End users expect to deliver a line level (or digital) signal to the system that in turn results in predictable output.

Each level of system evolution enables an evolution in optimization. How could we optimize a crossover in the old approach where horns and woofers were stacked up next to each other? Even with the best analyzer and digital processing there's no sensible solution to that challenge (except a dumpster).

The standardized linear loudspeaker system opens the door to optimization. The sensitive crossover points have a fixed solution, which creates a known element that can be utilized to build arrays. We can learn lessons about the coverage pattern, aiming, splay angles, compatibility with other models and more. Results can be predicted in advance because the response of the speaker is standardized.

It's personally embarrassing to think of how little I knew about loudspeaker array behavior back in those early days, but then again, it's a subject complicated enough for an entire book. Even a simple, small array of identical elements is very complex, but the outcomes are predictable. Known elements with known spacing and angular orientation will yield predictable results. This holds true with standardized engineered loudspeaker systems, but good luck with a system built in your garage and/or with wild parameters such as unmatched loudspeakers and processing.

Evolved loudspeaker systems exist at all power scales, running from Bambi to Godzilla and everything between. We see similar coverage patterns from 2-way systems whether they incorporate 15-inch or 5-inch drivers. They cover nearly the same frequency range, with the "big boys" reaching maybe an octave lower (9 versus 8 octaves).

The difference in power scale is gigantic though, which allows us to use a size proportional approach to achieve similar results. Recall the old school design principle: bigger venues use bigger stacks of the same stuff. The new paradigm is proportional scale: the same quantity of boxes can be used for a small or large venue, but we proportionally scale the elements up in size/power.

We define loudspeaker systems as coverage shapes and power scales. We can get 80-degree by 50-degree dispersion in all sizes, which makes it a scalable building block for both coupled and uncoupled arrays. After all, one person's main is another person's front fill. The modern line array is engineered to couple in large quantities in the vertical plane, but again the scalar paradigm applies.

This provides a picture of loudspeaker system evolution. Next time I'll focus on the signal processing that drives them. LSI

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

HAPPILY EVERY AFTER

A method for setting up a foolproof festival patch.

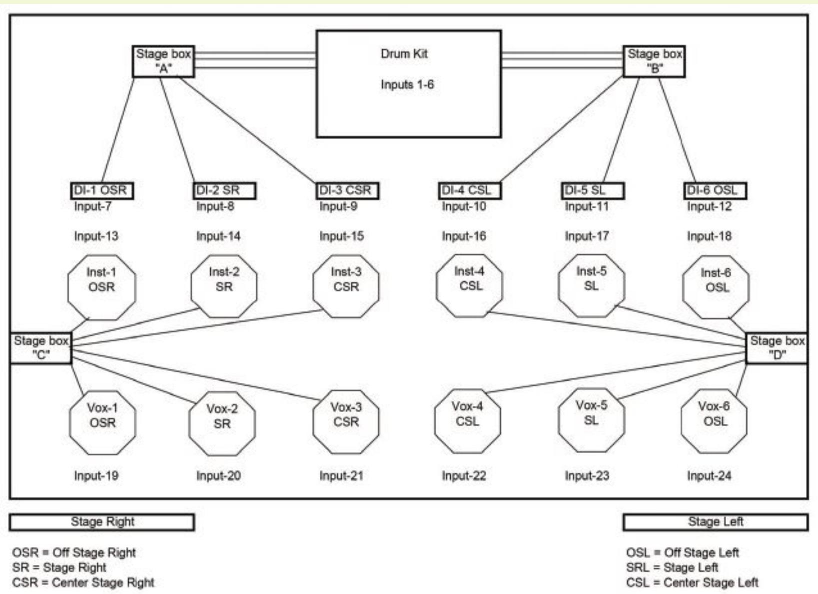
by Ike Zimbel

When I was coming up through the ranks at a sound company in the early 1980s, I did a lot of folk festivals. Within a few years I went from mixing a small satellite stage (with monitors from front of house) to main stage monitor mixer and then to main stage FOH mixer. During this process I transitioned from running my own show on the small stage to being an active participant in every changeover as the monitor engineer to watching helplessly from FOH while changeovers took way longer than I felt they needed to.

	Zimbel Audio Productions: Microphone Input List						
	Project Name:	Festival Patch, 24 Ch.					Date: 2016
Ch #	Input	Mic/DI	Stand	Sub-Snake #	Mon. IP#	48v	Notes:
						Y/N	Mics and DI's listed are suggestions only.
1	Drum-1, Kick	MD-421 / M-88 / B-52	SWB	A-1	1	N	
2	Drum-2, Snare	Senn E-604 / SM-57	SWB	A-2	2	N	
3	Drum-3, Rack Tom(s)	Senn E-604	Clip	A-3	3	N	"Y" cord if needed.
4	Drum-4, Floor Tom	Senn E-604	Clip	B-1	4	N	
5	Drum-5, SR Overhead	AT-4050 / akg C-3000	HWB	B-2	5	Y	"X-Y" config. These are not just cymbal mics.
6	Drum-6, SL Overhead	AT-4050 / akg C-3000	HWB	B-3	6	Y	"X-Y" config. As above.
7	DI-1, Offstage Right	J-48, CAI Type-10	n/a	A-4	7	Y	Active DI's preferred.
8	DI-2, Stage Right	J-48, CAI Type-10	n/a	A-5	8	Y	
9	DI-3, Center Stage Right	J-48, CAI Type-10	n/a	A-6	9	Y	
10	DI-4, Center Stage Left	J-48, CAI Type-10	n/a	B-4	10	Y	
11	DI-5, Stage Left	J-48, CAI Type-10	n/a	B-5	11	Y	
12	DI-6, Offstage Left	J-48, CAI Type-10	n/a	B-6	12	Y	
13	Instrument-1, Offstage Right	SM-57 / AE-2300	RWB	C-1	13	N	NB! Keep a few condenser mics available for "specials".
14	Instrument-2, Stage Right	SM-57 / AE-2300	RWB	C-2	14	N	
15	Instrument-3, Center Stage Right	SM-57 / AE-2300	RWB	C-3	15	N	
16	Instrument-4, Center Stage Left	SM-57 / AE-2300	RWB	D-1	16	N	
17	Instrument-5, Stage Left	SM-57 / AE-2300	RWB	D-2	17	N	
18	Instrument-6, Offstage Left	SM-57 / AE-2300	RWB	D-3	18	N	May need to be used for a vocal occasionally.
19	Vocal-1, Offstage Right	E-835 /Beta-58	RWB	C-4	19	N	
20	Vocal-2, Stage Right	E-835 /Beta-58	RWB	C-5	20	N	
21	Vocal-3, Center Stage Right	E-835 /Beta-58	RWB	C-6	21	N	
22	Vocal-4, Center Stage Left	E-835 /Beta-58	RWB	D-4	22	N	
23	Vocal-5, Stage Left	E-835 /Beta-58	RWB	D-5	23	N	
24	Vocal-6, Offstage Left	E-835 /Beta-58	RWB	D-6	24	N	

A sample festival patch input list.

I was also exposed to various festival management's attempts to organize the changeovers, which ran the gamut from having a really good stage manager to trying way too hard to "simplify" everything by using color codes exclusively. (Seriously, even to the exclusion of numbers, as in, "What channel is that mic in?" "It's in red." "Really? So what input is that on the snake?" "Red.")



A sample stage plot that depicts a clear, logical approach.

Through all of this, I noticed an important truth: No matter who was on the stage, and no matter what instruments they were playing, they all had to stand (or sit) somewhere on the stage. So I thought, "Why not put the mics where the people are going to end up?" And we all lived happily ever after. I'll explain how that works shortly, but first a few guiding principles and features.

1. All inputs always appear in the same order at each mixing console.
2. That order is always from left to right as viewed from FOH (i.e., stage right to stage left when you're on stage).
3. This puts all faders under your fingertips, and in line-of-sight with the stage.
4. All inputs are clearly labeled for their position on the stage, using existing

and well-known terminology like "1, 2, 3, 4 and "Left" and "Right."

MAKING IT HAPPEN

Here's how to set this up for a 24-input festival stage. Remember, it's a scalable method, so for most typical club stages, you can go smaller, and for concert stages, you can go larger. First, the requirements:

1. At minimum there needs to be four identical sub-snakes (in a pinch you can use the main snake head as one of the four, or on a smaller stage, two 12-pairs instead of four 6-pairs). In this case it's six to eight channels each.
2. Drum kit microphones, to taste. I usually favor kick and two overheads with "whatever works" for snare, toms, hi-hat, etc.
3. Six DIs, preferably active ones and preferably identical.
4. Six identical instrument mics on identical stands.
5. Six identical vocal mics on identical stands.
6. Some quality cloth tape and a Sharpie to label the DIs and the mic stand bases in this sequence:

DI-1, Off Stage Right (OSR)
DI-2, Stage Right (SR)
DI-3, Center Stage Right (CSR)
DI-4, Center Stage Left (CSL)
DI-5, Stage Left (SL)
DI-6, Off Stage Left (OSL)

Inst-1, Off Stage Right (OSR)
Inst-2, Stage Right (SR)
Inst-3, Center Stage Right (CSR)
Inst-4, Center Stage Left (CSL)
Inst-5, Stage Left (SL)
Inst-6, Off Stage Left (OSL)

Vox-1, Off Stage Right (OSR)
Vox-2, Stage Right (SR)
Vox-3, Center Stage Right (CSR)
Vox-4, Center Stage Left (CSL)
Vox-5, Stage Left (SL)
Vox-6, Off Stage Left (OSL)

Now drop a stage box in each corner of the stage. Each box is going to have three of each kind of input in it, so:

Box-A, Upstage Right, Drums 1-3, DIs 1-3
Box-B, Upstage Left, Drums 4-6, DIs 4-6
Box-C, Downstage Right, Inst 1-3, Vox 1-3
Box-D, Downstage Left, Inst 4-6, Vox 4-6

As you can see from the stage plot graphic that accompanies this article, as soon as you place a musician anywhere on the stage, they'll be very close to a vocal mic, an instrument mic and a DI.

USING IT AND FAQ

Like anything that really works in this business, this is a system, in this case a system for setting up, sound checking and presenting multiple, diverse and probably unknown acts in a minimum amount of time – and in front of an audience. How it works:

When allocating inputs, work outwards from center stage. So for a solo performer with a guitar and backing tracks for example, use Vox-3 (or 4, doesn't matter), Inst-3 and/or DI-3 for the guitar, and DIs 1 and 2 or 4 and 5 for the tracks. If time allows, the unused vocal and instrument mics can be struck off to stage right and stage left.

Based on this example, it's easy to see that with a duo, you would just use the two center vocal mics (3 and 4) and instrument channels – and so on – as the acts get larger. It's also really easy to allocate channels in advance if you're ever lucky enough to get stage plots from the acts ahead of time.

Q&A #1: Why are the stage boxes side stage instead of down the center? So that the unused mics and stands can be easily struck off to the wings if they're not needed, without leaving their cables all over the deck.

Q&A #2: Why only label the mic stand bases and not the cables? The intention is that the mics never come off the stands, and their cables never get un-plugged, so usually labeling the stand and the stage box is sufficient. In practice, the mics might get un-plugged one at a time if you're cleaning up the wiring on a break.

Q&A #3: What happens if a band has, say, seven singers and there's only six vocal channels? Simply use the last instrument mic on a taller stand if needed. Conversely, you might occasionally need to use a vocal mic for an instrument.

Q&A #4: Why did you specify active DIs? My rule-of-thumb is active instrument = passive DI and passive instrument = active DI. But if you have to choose only one, active DIs are better because they're usually fine with active instruments (i.e., synthesizers, drum machines etc) and they're better for truly passive instruments (i.e., ones with no electronic amplification such as a stock P-bass, and even some keyboards like a Rhodes 73, Clavinet, etc). Why are they better? Because active DIs have the really high input impedances that these instruments need to see as a load to sound their best.

Q&A #5: What happens if artists bring their own mics? Plug them in to the allocated vocal channels. We're here to serve. This system helps narrow down a bunch of variables, like input gain and high-pass filter settings on most of the inputs, most of the time – which makes it a whole lot easier to accommodate special requests when they come up.

Q&A #6: What happens if a channel is not being used? Mute it and pull the fader down. The advantages of each channel keeping the same basic setup (gain, HPF, 48-volt on/off, assignment, etc.) far outweigh those of having all the channels in a pretty little row. LSI

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

A CONSOLE TO MEET THE PHANTOM'S STANDARDS

THE FINNISH NATIONAL OPERA (FNO) was given the rare opportunity to stage a completely original production of Andrew Lloyd Webber's *Phantom of the Opera*. However, the audio system installed at the Helsinki Opera House could not match the vision of sound designers Sakke Kiiskis and Stanley Lönnquists, prompting the installation a DiGiCo SD7T mixing console.

Sound designers Petri Peltovako (left) and Lari Angervo with the DiGiCo SD7T with EX007 extender wing at the Helsinki Opera House.

It marks the first time that *Phantom* has been staged in Finland. The production includes a 40-piece orchestra and 44 singers with microphones (some of the choir is not miked), with the console also needing to handle a wide range of other productions, ranging from classical opera pieces and musicals, to modern dance with 7.1 backing tracks.

The updated audio system design and install (with support from the venue's sound crew) was provided by Santtu Sipilä, who worked with Reima Saarinen on the design for Finnish National Opera, specifying a DiGiCo SD7T with EX007 extender wing and six SD-Racks on twin optical loops, plus a Waves SoundGrid system.

"The main challenge was to fulfill all the needs of a modern opera house," Santtu states. "The venue works in two shifts and there are usually two productions per day in the main hall. For example a classic opera piece will be performed in the morning, then a musical with fully miked orchestra in the evening.

"To accommodate this kind of workload, which also requires backing tracks, effects and virtual sound checking, we use every channel on the SD7T. Both optical loops, Waves and the EX007 are in everyday use. One of the many advantages of the SD7T is that on the larger musicals we can have two operators on the console – one for the orchestra and one doing the vocals."

A SYSTEM FOR THE GETTYS IN 20 CITIES

CTS Audio provided Harman's JBL Professional line arrays driven by Crown Audio amplifiers for a recent 20-city tour featuring Irish musicians Keith and Kristyn Getty joined by a large number of celebrated folk musicians playing seasonal classics. The Brentwood, TN-based company, which has supported Getty tours for the past five years, deployed a main system incorporating VTX V20 line arrays, VTX S28 subwoofers, and VT4886 arrays for fills, all powered by Crown I-Tech 12000 HD amplifiers.

A perspective of the Gettys in concert, with JBL VTX arrays left and right, on the recent tour.

Jon Schwarz, lead system designer for CTS Audio, explains that with the large number of acoustic instruments on stage and the demands of rich, multilayered harmonies, it was imperative for CTS Audio to achieve a high degree of clarity for vocals and instrumentation. "The VTX S28 is a really musical subwoofer, which worked great in this instance," he says. "The VT4886 is a low-profile loudspeaker that we used to fill in areas at the front of the stage. With its size and output, it meshed perfectly with the rest of the system in terms of tonal characteristics, clarity and volume."

"This was a very unique tour," he continues. "We had a large group of folk musicians on a single stage with a lot of vocal harmonies. We really had to get the sound just right. Harman Professional Solutions gives us so many options to make even tough setups work, and the results speak for themselves."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The Black Eyed Peas
The Black Keys
Blake Shelton
Bob Dylan
Bon Jovi
Bonnie Raitt
Brent Mason
Boston Pops
Brad Paisley
Bruce Hornsby
Bruce Springsteen
Bruno Mars
Bryan Adams
Buddy Guy
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Cannibal Corpse
Casting Crowns
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Chayanne
Chris Cornell
The Chieftains
Chuck Rainey
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Clint Black
Coldplay
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Deadmau5
The Decemberists
Def Leppard
Derek Trucks
Dave Bottrill
Devin Townsend
Diana Krall
Dimmu Borgir
Disney
Dixie Chicks
Dolly Parton
Don Ross
Doobie Brothers
Dave Rideau
Dream Theater
Duran Duran
Dwight Yoakam
The Eagles
Ed Sheeran
Elton John
Eminem
Emmylou Harris
Enrique Iglesias
Eric Clapton
Eric Johnson
Erykah Badu
Evanescence
Eight Day Sound
Fall Out Boy
The Flecktones
Fleetwood Mac
Flogging Molly
Florida Georgia Line
Foo Fighters
Foreigner
Frank Filipetti
Franz Ferdinand
Garbage
Gavin DeGraw
Genesis
Gino Vannelli
Glen Ballard
Godsmack
Gomez
Goo Goo Dolls
Good Charlotte
Gov't Mule
Grand Ole Opry
Green Day
Gregg Allman
Guns N' Roses
Gwen Stefani
Hall & Oates
Herbie Hancock
Hoobastank
Hedley
HAIM
Havok
Imagine Dragons
Iron Maiden
James Taylor
Jamie Cullum
Jamiroquai
Janet Jackson
Jars of Clay
Jason Mraz
Jeff Beck
Jennifer Lopez
Jerry Douglas
Jason Aldean
Jason Derulo
Jimmy Buffett
Joe Bonamassa
Joe Chiccarelli
Joe Jackson
Joe Nichols
Joe Satriani
Joe Walsh
Joey DeFrancesco
John Hiatt
John Jorgenson
John Mayer
John Patitucci
John Petrucci
Justin Bieber
Justin Meldal-Johnson
Josh Groban
Journey
Juanes
Justin Timberlake
Kaiser Chiefs
Kanye West
Kasabian
Katy Perry
K.D.Lang
Keb' Mo'
Keith Urban
Kellie Pickler
Kenny Edmonds
Kenny Chesney
Kenny Loggins
The Killers
Kings of Leon
Kitaro
Korn
KT Tunstall
Lady Antebellum
Lady Gaga
Lamb of God
Lenny Kravitz
Leo Kottke
Leonard Cohen
Linkin Park
Little Big Town
Luther Dickinson
Macy Gray
Marcus Miller
Mariah Carey
Marillion
Mark Egan
Mark Knopfler
Mark Tremonti
Maroon 5
Marty Stuart
Matchbox 20
Meatloaf
Megadeth
Melissa Etheridge
Metallica
Michael Bublé
MØ
Mötley Crüe
Mumford & Sons
Muse
Nathan East
the National
Neil Young
Nelly Furtado
Nickelback
Nine Inch Nails
OneRepublic
Pat Metheny
Paul Boothroyd
Paul McCartney
Paul Simon
Peter Gabriel
Pink Floyd
Placebo
Porcupine Tree
Prince
The Prodigy
Queensrÿche
Radiohead
Randy Bachman
Randy Brecker
Randy Travis
Rascal Flatts
Red Hot Chili
Peppers
Rival Sons
Rickie Lee Jones
Rihanna
Ringo Starr
Robert Plant
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Rod Stewart
Roger Hodgson
Roger Waters
Rush
Ryan Adams
Sam Roberts
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Sarah McLachlan
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Shakira
Shania Twain
Sheryl Crow
Shinedown
Sigur Rós
Slash
Slayer
Slipknot
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Steely Dan
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Tony Levin
Tony Maserati
Toots & the Maytals
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U2
Usher
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Victor Brandt
Victor Wooten
Vince Gill
Weezer
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Two deployments of Renkus-Heinz arrays in the NYC region by Taylor Productions.

NEW YORK-STYLE MIXING & MATCHING

LONG ISLAND-BASED Taylor Productions has been utilizing Renkus-Heinz rigs as a key element in serving a full roster of concerts, festivals, and other events. Gigs range from Phil Collins to Jay and the Americans at medium to large venues including Lincoln Center, the Rainbow Room, MetLife Stadium, and Nassau Coliseum.

The company's main rig is a PNX102LA line array system, joined by multiple other PNX and CFX-Series rigs. With several R-H options to work with, Taylor Productions is able to mix and match cabinets to meet the specific demands regardless of the scale of the production.

"We'll pull out the PNX102 rig if the venue is large enough, because I just love the sound of that system," states company president Bill Taylor offers. "Often for more mid-sized venues we'll set up a nice CFX101 system - it's plenty loud enough, and sounds wonderful. If we're doing a smaller gig like an outdoor cocktail party or a jazz fest – something that requires great fidelity without a whole lot of sound pressure – our PNX82 rig is a perfect fit.

"I love the fact that you can rotate the horns if you want to. Having that kind of pattern control is essential for outdoor shows where you need to maintain control of sound levels in the neighborhood," he adds. "Often we'll use our PNX82s as front fill for a theater or concert show. They're really great for that kind of coverage. If we want tighter control on the bottom end we'll use our DNX212 subwoofers. They deliver such a tight, punchy low end, with a nice long throw." LSI

GIG SAVERS

Key tools that make it all work, no matter what.

by Craig Leerman

Barrels and cable adapters that the author carries to shows and events. There are dozens more of each type in his company workboxes. Note the green ID tape to help prevent these items from "growing legs."

The majority of live shows and events present interconnect challenges. We need to interface our gear with installed house systems, recording or broadcast trucks, A/V companies, rental gear, and even equipment from our own inventory that has different connectors.

My company's inventory, for example, consists of various analog and digital consoles, some of which have 1/4-inch TRS (tip/ring/sleeve) connectors for outputs, while others have XLR connectors and still others are outfitted with some combination of both. As a general rule, I don't carry 1/4-inch TRS cables to shows so instead rely on adapters to interface mixers, processors, amplifier racks and powered loudspeakers.

And on freelance gigs, it's not uncommon to have to hook up equipment from multiple rental houses and combine them into a working system. This necessitates carrying an assortment of adapters if I want to have audio come show time.

Adapters are available in three main styles, including barrel, box and cable. Barrel adapters are compact and usually just interface one connector type to another, but larger barrels may house internal components or even external switches.

Box adapters offer more space for connectors as well as internal components like transformers, and usually provide for multiple connections/channels. Most are made of metal that shields against electromagnetic interference (EMI). An example is a common direct box (DI) that converts a high-impedance signal with 1/4-inch input to a low-impedance mic level XLR output.

Cable adapters have one connector type (or gender) at each end, with a length of cable in between. They can also be of the "Y" or "splitter" type where two cables exit a single connector. Let's take a look at the different types of adapters that are common in live audio, as well as other interfaces and test devices that I've found to be essential in 30-plus years of doing this work.

Barrel Adapters. I carry an assortment to interface between 1/8-inch TRS (common for headphones), 1/4-inch unbalanced 2-conductor, 1/4-inch TRS balanced 3-conductor, XLR and RCA. The ones I use most often interface 1/4-inch TRS with XLR.

Cable Adapters. They're handy, allowing direct connection to gear without needing a separate cable. I've hand-made an assortment of 1/4-inch TRS to XLR short cable adapters that come in handy to interface gear between these two common output connections.

Headphone Adapters. For using 1/8-inch TRS headphone, in-ear monitor and ear buds with a standard 1/4-inch output jack.

Barrel Turnarounds. I carry a bunch of XLR male-to-male and XLR female-to-female barrel turnarounds to every gig. They're mainly used to turn snake channels from a send to a return or vice versa, but they also come in handy when a stage hand has inadvertently run a very long XLR cable in the wrong direction.

Transformer Barrels. These come in two types. One type converts an XLR mic level into a high-impedance 1/4-inch plug, great when you need another mic input but have used up all the channels with mic preamps in a small console. Simply plug one of these into the mono input of a stereo channel, and presto, extra mic input.

The other type sports a 1/4-inch jack and converts high impedance signal into a mic level via a male XLR. It's basically a "mini DI" and comes in handy when a band shows up with 15 1/4-inch keyboard and sampler/sequencer outputs and there's only 12 DI boxes in your workbox.

Mic to Line Bridging Barrels. These contain a transformer that coverts mic level to line level, or line level to mic level. using XLR input and output connectors. They're quite useful for interfacing the line level output of a console in a breakout room with the house system that only has a mic level input connector, or when interfacing a line level aux output with a camera's mic input.

DI Boxes. An industry staple for converting high-impedance instrument signal into low-impedance mic signal. DIs come in two basic varieties: active and passive. Active boxes contain electronic circuitry that requires power from a battery or phantom power from a console. Passive boxes use a transformer to convert the signal and do not require power. Many DI boxes offer ground lift switches and "loop-through" outputs so they can be inserted between the instrument and the stage amplifier. Single channel units are the most common, but DIs with multiple channels are available.

A representative sample of a variety of different box-type adapters.

Isolation (ISO) Transformers. Available in both barrel and box form and using a "1 to 1" transformer that breaks the physical connection in an electric signal path, removing any hum and buzz caused when two components are plugged into different power sources or when using a cable that has picked up EMI. Many of the box units are available in multiple channel versions, and some offer a variety of connection options so they also act as interface adapters in addition to isolation transformers.

I keep a variety of ISO transformers in my kit because I do a lot of corporate work where to my equipment interfaces with the house system and/or broadcast and recording trucks, which are all on different power sources and therefore more susceptible to hum and buzz issues.

Ground Lifts. Many times an ISO transformer isn't needed and a simply lifting of the ground at the input side of the connection solves the noise issues (in a balanced connection).

PAD Adapters ("PADs"). They pad or reduce signal by a preset or adjustable level.

Pin 2 To 3 Adapters. Before the industry adopted Pin 2 as Hot as the standard, some equipment used Pin 3 as Hot. I still have some older amplifiers that use the Pin 3 configuration and occasionally run into rental or installed gear with the same thing. These adapters simply swap Pin 2 and Pin 3, solving any polarity issues.

Splitters. Boxes or cables for splitting a signal into two or more streams. The most common use is with intercom systems. Instead of needing to daisy-chain every comm belt-pack, a splitter can be used to send "home runs" to the different tech areas.

Combiners. Boxes that combine two or more signals into a single signal. Some DI boxes have a combine feature that allow joining two line level signals into a single mic level signal. This comes in handy for taking a stereo signal from a computer but there's a limited number of inputs or snake channels.

Barrel Joiners. Technically turnarounds, but they're designed to join cables that use the same gender connector at each end, like RCA or 1/4-inch and SpeakON loudspeaker cables. There are also loudspeaker level splitters available for joining multiple cables to a single amplifier output.

Network Switches. With Dante digital networking becoming the norm for signal transport, I carry a few 4 port Gigabit switches with me that allow me to more easily set up and route signals across a show network.

Routers or WAPs (Wireless Access Points). I carry a small wireless router that interfaces my iPad with various consoles, giving me the ability to walk around and mix remotely with an app.

GOING FURTHER

Even with a workbox full of adapters and turnarounds there still may be connection issues, so here's a roster of test equipment to help troubleshoot and find the problems.

Cable Tester. They can be dedicated to testing a single type of connector, or multi-units that can not only test different types of connectors, but cables with different connectors at each end. I use a CBT-500 multi-unit from Hosa Technology and an HCT-BD testers from RapcoHorizon.

"Fox and Hound" (a.k.a., Send and Receive) Cable Testers. These have two parts with one half (the fox) sending out a signal and the other half (the hound) sniffing out the signal. They're great for checking out cables that have already been run

The author also carries these meters and test devices to gigs.

at a gig, or cables that have been installed in buildings. I carry Rat Sound Tools models for XLR, NL4 and 1/4-inch. The Rat Sound XLR receiver unit can also be used to test for phantom power.

Qbox. Made by Whirlwind, this is a nifty mic and line audio test unit. It includes a microphone, a loudspeaker, a test tone generator, outputs for headphones, a 1/4-inch jack for line-in and XLR ins and outs. It can be used to generate a tone to test an input, check dynamic mics and cables, and listen to a signal to verify that it's working.

MORE THAN AUDIO

Faulty audio cables aren't the only culprits at gigs. Power issues are also responsible for a lot of headaches, and as a result, I carry some power related test units to help determine and address these problems.

Battery Testers. A dead battery can be a show stopper so carry a unit that can check AA, AAA and 9-volt batteries.

Outlet Testers. Sometimes called "cube" or "3 light" testers, these compact units plug into a wall outlet or extension cable and can tell you if it's working, has a ground, and is wired correctly. However, they can't identify a "bootleg ground" unsafe outlet. *(For a great deal more information, see "Shocking Situations" by Mike Sokol in LSI February 2016 and on ProSoundWeb.)*

Non-Contact Voltage Testers. Also called Tic Trace Testers, they sense voltage when in close proximity to an outlet or cable. By using one in conjunction with a Cube tester, you can verify an outlet and ground are wired correctly. I carry one made by Greenlee so I can quickly check if an outlet or power cord is working.

Breaker Finders. They work on the Fox and Hound principle: plug the sender unit into an outlet and move the probe tip of the receiver unit over the breakers in the breaker box. The receiver will identify what breaker that outlet is connected to. I use a unit from Sperry as the sender can also double as a Cube tester.

AC Multi-Meters. The previously noted power meters can only tell you if an outlet or cable has power. An AC meter is needed to read what voltage that power is, especially when using generator power. A generator set at the wrong voltage can destroy gear. I prefer Fluke "Clamp Style" multi-meters because they can also tell me how many amps I'm pulling on each feeder leg of a distro.

Utility Mixers. While not an adapter or tester, I wanted to sneak this one in because it's important. I carry at least one small mixer to every gig to serve as a multipurpose problem-solver. It can be used to convert signal from mic to line level, line level to mic level, as well as pad, split, combine and meter signal. And it can also serve as a invaluable backup if the main console goes down. *(Find out many more applications in "Utility Mixers," easily accessible on ProSoundWeb.)*

It's not just audio gear that have interconnect/power issues, so I also carry an assortment of non-audio adapters, including XLR 3-pin to XLR 5-pin for lighting DMX, Edison to stage pin power, video adapters and barrels for VGA, HDMI and Composite or SDI (BNC and RCA connectors). Being able to solve almost virtually any problem along these lines insures that the show happens, my clients are happy, and I get their repeat business. LSI

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

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KLANG TECHNOLOGIES

A 3D Approach To In-Ear Monitoring.

by Mark Frink

The KLANG:fabrik Dante-enabled digital processor that's at the heart of the system.

The problem with in-ear monitors (and headphones) is that our ears hear one side of a stereo mix in isolation from the other ear, while natural binaural hearing provides each ear with some sound arriving from the far side of the head, slightly delayed, out of phase and filtered. Without the additional natural cross-feed, a stereo IEM mix is heard "between the ears" instead of around them, and also contributes to hearing fatigue.

Instead of processing an entire stereo mix at once, as do Head-Related Transfer Function (HRTF) plugins, a new approach from KLANG Technologies provides a 3D approach that individually adds HRTF processing to each mix's inputs, binaurally panning them across each stereo mix. It can even add height processing for a third dimension.

Because of this individual processing, KLANG can then rotate these mixes in orientation with the listener's head. While KLANG inputs are processed into 3D stereo mixes, pre-KLANG gain, high-pass filtering and EQ as well as any dynamics and effects would first take place in a front of house or dedicated monitor console.

KLANG (German for "sound") Technologies is based in Aachen, Germany, with North American distribution by Group One Limited. The KLANG ecosphere consists of three main products. KLANG:fabrik is the digital hardware processor. KLANG:app is the controller for creating 3D in-ear mixes from mono and stereo channels or stems. KLANG:vector consists of a 9-axis Wi-Fi motion sensor that can further allow each 3D mix to rotate and compensate for the listener's head orientation.

THE HARDWARE

KLANG:fabrik is a Dante-enabled digital processor (U.S. list: $5,990) that operates at 44.1 or 48 kHz with 24-bit precision. The 2RU chassis, weighing 11 pounds, sports an 800 x 600, 5-inch color touch screen and a USB port. A pair of Neutrik EtherCon Dante connections on the rear panel provide up to 64 audio inputs and up to 64 outputs. Additionally there are four ADAT-compatible optical inputs and outputs that each provides eight audio channels, for a total of up to 32 inputs and 32 outputs.

fabrik also provides eight pairs of line level (+ 4 dBu nom., + 22 dBu max.) analog outputs for direct connection to IEM systems. Digital input to analog output latency is specified as 3 milliseconds. The unit's DSP has an I/O trade-off, currently limited to 192 total cross-points. It can process 48 inputs for four users, 32 inputs for six users or 24 inputs for eight users. Since there are only 32 possible ADAT inputs and eight pairs of analog outputs, using Dante allows 12 inputs for 16 users, helpful for orchestras, or up to 56 inputs for thee users.

Alternatively, processed mixes can be returned to a console (via ADAT or Dante) for any post-KLANG mix processing, and more non-KLANG processed inputs – including click track and shout mics – can be added without increasing the number of KLANG inputs. Returning KLANG to a console also allows the engineer to monitor pre-processed inputs along with post-processed mixes. Input channel counts can be reduced by making sub-groups of each musician's inputs to be processed as individual stems. Fader bumps and rides on the console by the engineer are possible and can be helpful.

KLANG:vektor, the motion tracker, sends updated head-movement information to fabrik in real time. It can be used as either an in-ear cable set with integrated sensor or as an almost invisible unit attached to in-ear monitors, and transmits wirelessly at 2.4 or 5 GHz. (vektor is not yet shipping.)

KLANG:app GUI allows remote control of 3D panning.

THE SOFTWARE

KLANG:app (free) is used with a Wi-Fi router, with versions available for Android, Windows and Mac OS X. The app has three modes, one each for musicians, technicians, and administrators. Musician mode simply allows one user to mix and position their pre-defined inputs.

In order to get their "heads around HRTF" (pun intended), users should download the app onto a tablet and check out the Demo Mode. It includes a couple of short, looped 16-track live performances by a 7-piece German band named Shubangi that can be re-mixed and re-panned on the tablet while listening with a pair of headphones or IEMs, going from mono to stereo and 3D.

In addition, i3D Mode demonstrates how KLANG can rotate a mix in orientation with the user's head. Technician and Administrator Modes provide a row of buttons across the top of the tablet screen, allowing techs to cue each mix, adjust its settings and even "copy and paste" from one mix to another. Both modes also allow each mix to be re-configured. Administrator Mode further allows technicians to assign motion trackers and re-name mixes.

THERE'S MORE

The series also includes KLANG:quelle, a 4-channel Dante headphone amp that comes in two different form factors. A 1RU rack-mount version ($1,750) can be AC-powered or via Cat-5 power over Ethernet if PoE switches are used. Its front panel has four TRS jacks, four volume controls and four +/- 12 dB output switches, plus four pairs of XLR line level outputs on the rear. A compact version ($1,450) is housed in a third-rack-width chassis without the four pairs of XLRs.

As noted, KLANG is available in the U.S. from Group One Limited (www.g1limited.com). Find out more about the technology at www.klang.com/video, including a demo of the 3D audio capabilities. LSI

KLANG:vier (German for "four") is a lower priced 1RU version of the fabrik processor ($3,390). With fixed 24 x 5 routing, it's intended more for studio applications. It has 24 inputs from either Dante or three 8-channel ADAT inputs, and provides line-level outputs on a DB-25 connector for four stereo mixes. These are also duplicated on four TRS headphone outputs on the front panel that have the same +/- 12 dB output switches as the 4-channel headphone amp, but no volume pots. A fifth mix is available via Dante.

***Mark Frink** (livesound@markfrink.com) works with the Jacksonville Symphony and is available this summer for IEM mixing.*

RCF M18

Evaluating an ultra-compact digital mixer and interface.

by Craig Leerman

The RCF M18 digital mixer and MixRemote app.

The new M18 digital mixer from RCF is billed as an integrated solution for musicians, but it also has enough features to satisfy more discriminating audio folks. The 18-input mixer is controlled via iOS running the MixRemote app and is equipped with an internal dual-band, dual-antenna Wi-Fi access point.

Eight discrete mic preamps are joined by 10 line inputs, including two that are switchable to Hi-Z for use as guitar or bass inputs. Outputs include stereo mains, six aux sends, and a stereo headphone output that can be configured for cueing or for personal monitor mixing.

Every input channel is outfitted with a preamp, gate, compressor, and a selection of three types of EQ. There are also three internal effects on their own FX sends, including reverb, delay, and modulation. The aux outputs have a parametric EQ, while main left/right outputs sport a 31-band stereo EQ. Up to 16 insertable plugin multi-effects supply a range of processing, as well as amplifier and cabinet modeling. There are also mastering processors, including a Valve Warmer, Exciter, and Maximizer that can be inserted before the main outputs.

A dual footswitch jack and MIDI I/O ports support remote control, allowing the user to switch between instrument presets or to change FX settings. RCF has squeezed a lot of features into a small box that measures only 13.5 x 7.2 x 3.5 inches.

STRAIGHTFORWARD SETUP

Before even getting to the hardware mixer, I grabbed an iPad and downloaded the MixRemote app from the iTunes store. There's a cool offline demo mode where you can get familiar with the software. It's easy to navigate, and it's designed so that all page and channel functions can be reached with a couple of touches – an essential convenience if performing.

Satisfied that the app was functioning correctly, I moved on to the mixer and took it to my bench. The first thing I noticed was that the unit does not require a special external power supply; the onboard internal supply just needs a common IEC cable (included). It's one less thing for users to worry about.

Optional accessories include rackmount kit, large rubber bumpers for better protection, and a backpack case to store the mixer and a tablet. Made of steel and plastic, the chassis appears rugged and should survive well on stage or in transit, especially with the optional bumpers. All connectors are located on the rear panel, which makes for a neat appearance when the mixer is on stage. (Also note that the optional rack ears allow the mixer to be mounted with connectors facing front or rear.)

Connectors include six XLR inputs, two combo TRS/XLR inputs with Hi-Z capability plus eight TRS inputs, 10 TRS inputs, six TRS aux outputs, two XLR main outputs, and a 1/4-inch stereo headphone output. There are also MIDI input and output DIN jacks, a 2.0 USB, 1/4-inch TRS footswitch jack, and an RJ45 network jack. The external antenna mounts on the rear, right above the power switch. A power cord jack, main output signal lights, and phantom power signal lights complete the rear panel.

Hooking up the mixer was simple, and my iPad found the network on its own. Once connected to the mixer, the user can go into the settings network menu and choose either 2.4 GHz or 5 GHz bands, select channels, and also set up a security password if desired. The settings menu is also where phantom power is enabled (in blocks of four), MIDI channels are assigned, footswitch modes are configured, inputs are linked for stereo operation, and outputs are routed. There's also a neat feature that lets the user change the standard headphone mode from PFL to a personal monitor mix.

INSIDE THE APP

The software is intuitive, with eight button tabs at the top of every screen. The first is for Faders, bringing up eight at once. The user can swipe left or right to access different faders, or use forward and back arrows at the lower corners of the screen to access different banks of faders.

At the left of the Fader screen are three orange FX buttons and six green aux send buttons. Pushing a button changes the screen to sends on faders, with the fader caps orange for FX sends and green for aux sends. Standard input fader caps are white and the master fader is red. No matter what Fader screen is up, the master fader is always available to the right of the bank of eight faders.

Inputs have PFL and mute buttons, a pan control, and a tool icon that opens up access to that channel's preamp, HPF, gate, compressor, and EQ. Three EQ choices are provided: Standard, a 4-band semi-parametric; Vintage, offering a specific sound print on all four bands and emulating vintage analog equalizers; and Smooth, a flexible section with a choice of three smoothness settings for HF and LF. The 100 factory presets are optimized for different instruments and voices.

The next page button is for Effects, providing three separate FX units that include a reverb, delay, and chorus/flanger. Users can choose from a variety of nice-sounding effects, and 70 presets are available. Next is the Outputs button, which brings up six aux faders and the headphones fader. (All of these outputs have Standard PEQ). Phones is the next button, prompting a dedicated page with just the headphones fader and PEQ. Switch the phones to personal monitor mixing and the page displays sends on faders for mixing.

The Play/Rec button is next in line, providing playback controls and file management for playing files from USB drives. (The M18 supports all of the most common audio file formats. Also note that stereo recording from the mains will be available in a future upgrade.) The next page button is Meters, showing metering for every input, output, send, and return. It's followed by the Load/Save button, which allows users to take snapshots of settings and recall them later. The last button is the previously mentioned Settings.

NOT SO COMPLICATED

The software is intuitive to navigate and all of the controls are easy to use, even with one hand. The mic preamps sound quite good, and unlike some products in this price range, the processing and effects are sonically pleasing and the presets can serve as a solid starting point for a quality mix, with is especially helpful for working musicians as well as less-experienced audio techs.

A few more features were designed with exactly that in mind. Two high-impedance inputs accept guitar and bass input directly. Toss in a few channels with multi-effects units (delay, chorus, flanging, tremolo, tube emulated overload, distortion, pitch shifter and amplifier modeler), and the M18 can eliminate the need for stage amps and instrument floor pedals.

Further, the modeling is quite good and offers 11 guitar and bass amps, 11 speaker enclosures, three types of mics (including an emulation of a Shure SM57, naturally), and three locations for mic placement. Users can also change the patch order of the effects and save presets.

In addition, there are "easy" modes for some of the EQ parameters. And, several iPads can access the mixer at the same time, allowing multiple musicians to mix their own stage monitors from the aux sends or phones mode.

The connectivity suite on the M18's rear panel.

FINAL EVALUATION

The M18 can easily handle most of the low-input-count events that my company serves, so after updating to the latest firmware, I took it to a few gigs. The first was a concert fundraiser for a private school, with inputs consisting of a podium mic, wireless mic for a singer (and backup), two for music tracks (and two for backup). The main stereo outs fed the PA, and one aux was used for a stage monitor.

The singer really liked the sound of the reverb I chose (Small Bright Room), and it was easy to send effects to the monitor with the dedicated FX return faders in the aux send page. While I played her tracks and the walk in/out music from my laptop, I could easily have used the built-in player if there had been enough time to load the song files via USB.

Next was a charity auction with two wireless systems (one for backup), an announce mic, and a computer to supply music cues. This time I was able to load the song files on a thumb drive, so I used the M18 to play music during dinner. The room was really large so I decided to test the wireless network. At both 2.4 GHz and 5 GHz, there were no problems with connectivity at a distance of more than 300 feet. It even worked when I was moved out to the back hallway.

Finally, we don't handle a lot of DJ work, but I happened to get a call from a buddy who needed a larger system to DJ a party. We provided it, along with the M18. Again, the mixer worked flawlessly, and all I did was sit in the back of the room and keep an eye on the levels with my iPad. (Hey, someone has to do the tough gigs!)

The RCF M18 is highly recommended for anyone looking for a great-sounding, full-featured, ultra-compact mixer. And an even a smaller unit, the M08, will be available soon.

U.S. MSRP: $999 LSI

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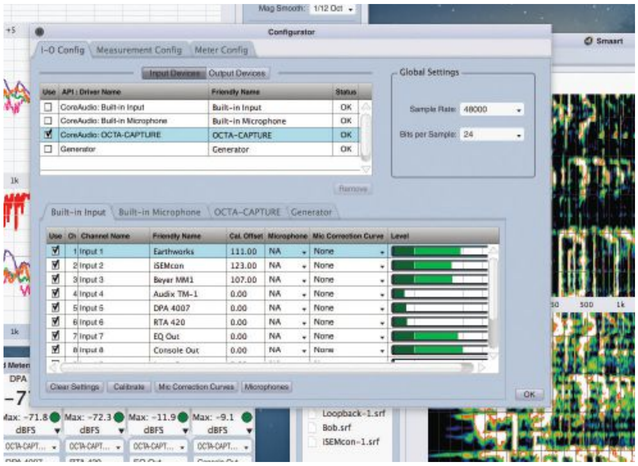
*Senior contributing editor **Craig Leerman** is the owner of Tech Works, a production company based in Las Vegas.*

Smaart v8

The evolution of an acoustic test and measurement platform.

by Jamie Anderson



New Smaart v8 provides streamlined measurement configuration.

As an industry, professional audio has fully embraced analyzers as part of the work process and is continually developing new, better and often unforeseen ways to implement them. Concurrently, since the release of Smaart v7 (in 2010) we've learned a tremendous amount about what features are important to users in a multi-measurement capable environment, and how they integrate Smaart into the workflow of their daily audio engineering tasks.

With that in mind, new Smaart v8 offers greatly expanded control of the software environment while simultaneously providing easier and more streamlined measurement configuration, control and data handling. It breaks out of the single-window, fixed-GUI world to allow users significantly increased control over their software environment, allowing them to much better able to adapt and expand Smaart to match their specific applications.

Whereas the jump from Smaart v6 to v7 was a complete re-invention/re-build of the Smaart platform, the transition from v7 to v8 is probably better described as an evolution, an expansion and refinement of our current Smaart software environment. (Very much like the progression from SmaartLive v4 to v5 to v6 – or for you really old dogs, v1 to v2 to v3.)

v8 basically comprises the same measurement capabilities as v7 – the same measurement engines, parameters and settings, the same basic plot types, and retains most of the same command structure and paradigms. The primary difference between the two versions, at least initially, is that v8 expands the capabilities of the interface and the ability of users to manipulate it.

And, in conjunction with this expansion, high priority has been placed on improving and streamlining the measurement configuration processes and data handling. Examples include new multi-window capability, a new tab-based interface, added functionality for use on tablets and touch screen computers, computer-to-computer API capability, and more.

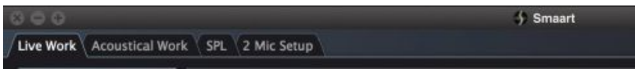


Figure 1: The new Tab-based interface that's a key upgrade to v8.

The measurement configuration process has been updated to provide an easier, more intuitive workflow, adding automatic measurement creation and management functionality to streamline configuration and support a "build-on-the-fly" process that is often an operational necessity.

A new program workflow take users through device selection/configuration and automatically creates Spectrum measurements. In addition, device and measurement configuration is managed through a single window. Input metering is now right in the I-O Config, convenient for checking inputs before even creating measurements. Further, creating a Transfer Function measurement now automatically creates the accompanying Spectrum contributors to it.

TAB-BASED INTERFACE

In v7, measurement configurations were organized and controlled using separate groups of defined Spectrum (single-channel) and Transfer Function (dual-channel) measurements. Switching the active Spectrum or TF groups controlled what measurements were actively configured and available. But many users expressed a desire for an interface where one could configure and switch between multiple workspaces, so we have expanded the basic "measurement group" paradigm from v7.

The result is a new Tab-based interface in v8 (***Figure 1***) to organize and control the measurement environment that includes both the active measurement configuration as well as the GUI configuration – v8's tabs use mixed groups, groups that can contain both Spectrum and TF measurements, and also retain plot view, interface layout and data information. Note that while v8's Tabs can contain mixed groups, the control and data bars still reflect only the type of measurement, Spec or TF, as indicated by the active plot (as in v7).

Switching between Tabs allows users to quickly configure and re-configure Smaart's measurement environment, with the fundamental restriction that only one Tab can be active in a given window.

BASIC LAYOUT/FUNCTIONALITY

The basic v8 interface is a familiar one for anybody coming from v7, though with some tweaks and enhancements that make this GUI more customizable. Each of the four Interface bars (Tab, Control, Command and Data) can be shown or hidden. Additionally, the Data bar can be expanded horizontally to allow for longer trace names, and the SPL/Clock readout can be hidden from the Control bar.

Figure 2 shows v8's Data bar (left), Command bar (bottom) and Control bar (right). Each can be toggled in and out using their triangular show/hide widgets located in the margin between the interface bar and the plot area. Meanwhile, the Interface Bar configuration is stored individually per Tab. Specifics for each include:

Tab Bar. This area of the interface shows the window's configured Tabs and can be used to select the active Tab. The bar is can be hidden from the view menu or with the hot-key.

Control Bar. This area shows all engines of the active plot measurement type available in the active Tab. It also includes the Window's SPL Meter (which can be toggled in/out), as well as the Signal Generator and view controls. The Control bar can be shown or hidden with the hot-key or with its triangular hide/show widget.

Command Bar. This area holds 10 buttons that trigger specific hot-key commands. These buttons are user configurable from the command menu and can be set to activate any hot key command. The command bar can be shown or hidden using the hot-key [U] or its triangular hide/show widget.

Data Bar. The top section contains the APL (Active Plot Legend, more on this later), while the lower section contains Smaart's Data Library. The data bar can now be re-sized horizontally by clicking and dragging it horizontally from its right-most border, and can also be shown/hidden using the hot-key or its triangular hide/show widget.

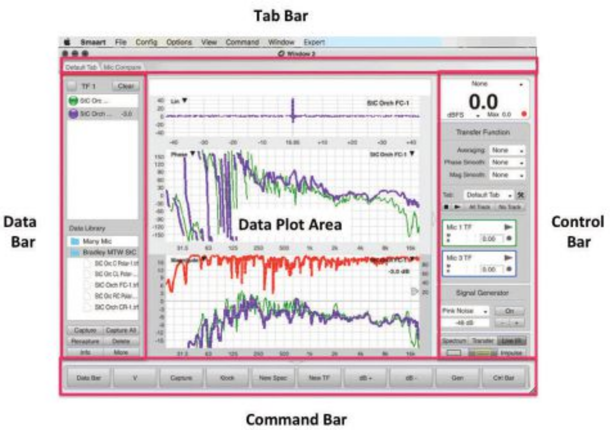


Figure 2: The four Interface bars make it easier to customize the GUI.

Figure 3: Broadband Meters window.

FURTHER UPGRADES

A wide range of other functional and interface facilities have been addressed in v8, many of them driven by our users. Let's have a look at some of the highlights.

Multi-Window Capability. There's now the ability to drive multiple windows, each containing its own set of tabbed workspaces. It's like being able to run multiple sessions of Smaart simultaneously. Multiple (multi-tabbed) measurement windows can be configured and simultaneously run across multiple displays, with the limits only being display space, and CPU/GPU capacity. v8 also includes some dedicated, special-purposed windows – most notably, the broadband meters and the API Client Window.

APL (Active Plot Legend). A new window that details the contents of the active/selected plot, showing the names, colors and trace offsets of live and static data traces, as well as providing the controls for hiding/showing the traces.

Improved Data Storage. For static data, we've overhauled the trace storage process to allow for better direct access to the larger file structure of captured traces, without cluttering up active memory space trying to keep everything loaded in RAM in case it's needed elsewhere.

Broadband Metering. Another user-requested feature is the ability to continuously view input signal levels, as well as monitor parameters such as SPL and LEQ for multiple inputs simultaneously. This capability is now provided with the Broadband Meters window (***Figure 3***), a mode-less window that can be configured to monitor any of Smaart's input signals via a simple grid configuration process. There's a lot more in terms of metering capability, including the ability to choose to display each input's SPL readout, Input Meter, or both (***Figure 4***).

IO Control. Access and console for the Smaart IO gain, as well as phantom power, are now integrated directly into the v8 interface.

Smaart-to-Smaart API. This integrated function allows any copy of v8 to act as a host to any other copy of v8, meaning that users can create a rig with one computer acquiring the input signals and performing the measurement computations, while another acts as a client and remotely accesses the measurement data. This capability then allows the v8 measurement environment to be spread, not just across multiple monitors, but across multiple computers.

FACTORY DIRECT

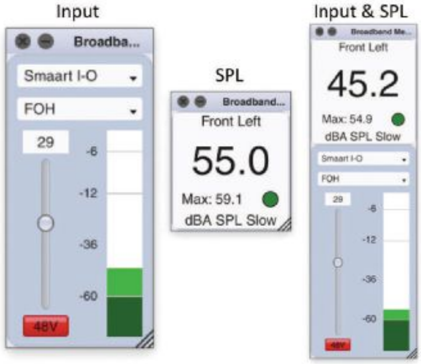


Figure 4: Further metering capabilities.

CONSTANT EVOLUTION

The bottom line is that even as we expand Smaart's complexity with new resources and facilities, we're also focused on making it easier to use as a tool. That's the primary goal of this measurement configuration process: to deliver an simpler, more intuitive workflow by adding automatic measurement creation and management functionality wherever possible to streamline configuration and to support a quick, intuitive process that is often an operational necessity for many of our users. Along with this, we've added features to support operation on tablet computers and even stretch Smaart across not only multiple monitors, but also multiple computers.

And of course, in the "under the hood" and "not-readily-visible" department, v8 also includes updates to the software's development environment as well as the licensing system and options. While not particularly sexy (although who doesn't find increased speed, functionality, stability and security sexy?) these updates were overdue and bring with them a lot of new capabilities and options for users, both now and in the future.

(EDITOR'S NOTE: new Smaart v8 is available for download at rationalacoustics.com, with options for current and new users clearly defined.) LSI

***Jamie Anderson** is a founding member of Rational Acoustics and has been teaching and working in the field of sound system engineering, measurement and alignment for more than 20 years.*

CONSISTENT COVERAGE

Medium-format line array designs and a look at the latest models.

by Live Sound Staff

The definition of which characteristics make a line array medium-format, as opposed to small- or large-format, is somewhat arbitrary. Is it how wide or high each element is, how much it weighs, how many components each houses, the diameter of the components, or how loud it gets? For this overview, we've based the selection on the size of the largest LF driver within the array, considering those with 8-inch to 10-inch cones to be medium-format.

Even within the medium-format category, there's a lot of variety. Among the represented brands and models, the horizontal coverage angle varies from 80 degrees to 150 degrees from a single array column, with most ranging between 100 to 120 degrees. Some manufacturers offer cabinets with the same "footprint" with differing horizontal coverage, allowing the user to better "customize" coverage for a particular venue.

Enclosure width varies from a bit over 23 inches to over 30 inches, and weight for each cabinet ranges from a bit over 30 pounds to over 100 pounds. Many are self-powered, and others have dedicated external processing and amplification.

Most of these systems use a pair of cones to cover the lowest frequencies, and often will roll off the upper frequencies of one LF driver while allowing the other to cover the midrange. HF is covered by a compression driver, or occasionally a ribbon driver, with pattern control via a horn or waveguide with a narrow vertical coverage angle. Thus a 3-way system is effectively created, with the coupling of the two cones effectively creating a larger LF radiating surface.

Lesser-scaled line arrays, in comparison to enclosures with 12-inch or 15-inch LF drivers, allow wider splay angles within the array elements, while still maintaining consistent coverage. This characteristic can be useful for covering smaller venues that have multiple levels, and can also help work around architectural structures like the edges of balconies.

The following Real World Gear tour of recent models is not meant to be all-inclusive, yet covers a variety of manufacturers and design concepts. Enjoy a look at the latest medium-format line arrays. LSI

RCF HDL 20-A
www.rcf-usa.com

Configuration: 2-way
Dispersion (h x v): 100 x 15 degrees
LF: 2 x 10-inch neodymium cone drivers
HF: 1 x 3-inch titanium compression driver on a custom waveguide
Frequency Response: 55 Hz – 20 kHz
Maximum SPL: 135 dB
Power: Onboard class D amplifier, DSP controlled input section with selectable presets
Rigging: Integral hardware with adjustable splay angles
Size (h x w x d): 11.5 x 27.7 x 17.5 inches
Weight: 64 pounds
Companion Sub: SUB Series (8006-AS, 8005-AS, 8004-AS)

L-Acoustics KARA
www.l-acoustics.com

Configuration: 2-way
Dispersion: 110 (h) x 10 (v) degrees, maximum inter-element angle
LF: Dual 8-inch neodymium cones
HF: 3-inch neodymium titanium driver
Frequency Response: 65 Hz – 20 kHz
Power: Bi-amped with LA8 or LA4X amplified controller
Rigging Angles: 0 to 10 degrees (8 angles)
Size: 9.8 (h) x 27.7 (w) x 19 (d) inches
Weight: 57 pounds
Companion Sub: SB18 (single 18-inch)

Martin Audio MLA Compact
www.martin-audio.com

Configuration: 3-way cellular drive
Dispersion (h x v): 110 x 10 degrees
LF: 2 x 10-inch neodymium cones, Hybrid slot-horn loaded
MF: 2 x 5-inch neodymium cones, horn loaded
HF: 4 x 0.7-inch neodymium drivers, horn loaded
Frequency Response: 65 Hz – 18 kHz
Maximum SPL: 135 dB
Power: Onboard 5-channel class D; DSP & networking
Rigging: Suspension of up to 24 enclosures
Size (h x w x d): 11 x 31 x 19.7 inches
Weight: 109 pounds
Companion Sub: DSX (dual 18-inch)

d&b audiotechnik Y8/Y12
www.dbaudio.com

Whether for mobile deployment or fixed installation, compact Y8 and Y12 line array loudspeakers provide flexible and configurable solutions for a broad spectrum of performance needs. Intended applications include houses of worship, theaters and conference centers, clubs, trade shows and auditoriums for audiences of 50 to 2,000.

Y8/Y12 offer 80 and 120 degrees horizontal directivity, respectively, with remarkable dispersion control down to 500 Hz. The mechanical and acoustic loudspeaker design permits columns of up to 24 cabinets, which can comprise Y8 and/or Y12s as well as the dedicated cardioid Y-SUB.

Both models utilize the same patented 3-point rigging as the larger V-Series and J-Series for scalable, reliable and efficient solutions, supported by an extensive range of transport options and loudspeaker accessories. Y-Series Yi models are specifically designed for permanent integration and differ only in cabinet construction and mounting hardware.

TECHNOLOGY FOCUS: *Utilizing sophisticated horn geometry and an advanced bass-reflex port design, both Y-Series line and point source loudspeakers deliver full bandwidth capabilities with an extended LF output. In addition, a custom waveguide and new HF driver for Y8/Y12 models provide the renowned high directivity and smooth HF of the V-Series.*

COMPANION PRODUCT: *Y7P and Y10P point source loudspeakers also offer wide and narrow dispersion options (75 x 40 and 110 x 40 degrees, respectively) and share the same directivity and dispersion control toward low frequencies. They include 2 x 8-inch drivers in a dipole arrangement with a 1.4-inch driver on a rotatable CD horn.*

KEY SPECIFICATIONS:

Configuration: 2-way
Dispersion (h x v): 80 or 120 degrees; vertical is array dependant
LF: 2 x 8-inch cones, dipole arrangement
HF: 1 x 1.4-inch-exit driver with a wave transformer
Frequency Response: 54 Hz – 19 kHz
Power: d&b amplification (D6, D12, D20, D80)
Rigging: 0 to 14 degrees in 1 degree steps
Size (h x w x d): 10 x 24.8 x 14.8 inches
Weight: 45 pounds
Companion Sub: Y-SUB (cardioid)

NEXO GEO S830
www.yamahaca.com

Configuration: 2-way
Dispersion (h x v): 120 – 80 (adjustable) degrees; vertical array dependent
LF: 1 x 8-inch neodymium cone driver
HF: 1 x 1-inch-throat compression driver
Frequency Response: 67 Hz – 19 kHz
Maximum SPL: 128 dB
Power: NEXO NXAMP amplifier/controller
Rigging Angles: Adjustable from 5 to 31 degrees (logarithmic steps)
Size (h x w x d): 16.8 x 10.8 (w) x 11.8 (d) inches
Weight: 28.6 pounds
Companion Sub: CD12 (dual 12-inch)

JBL Professional V20
www.jblpro.com

Configuration: 3-way
Dispersion (h x v): 105 x 0 – 12.5 degrees (inter-enclosure angles)
LF: 2 x 2261H 10-inch cone drivers
MF: 4 x 2164H 5-inch cone drivers
HF: 3 x D2415K compression drivers
Frequency Response: 60 Hz – 20 kHz
Maximum SPL: 133 dB SPL (MF)
Power: BSS OmniDriveHD V5 processing for use with Crown I Tech HD or VRack
Rigging: Angle Stop Mechanism (ASM) suspension allows tension or compression suspension
Size (h x w x d): 11 x 35.9 x 15.8 inches
Weight: 88 pounds
Companion Sub: S28 (18-inch) and S25 (15-inch)

FBT MUSE 210LA
www.fbt.it

Configuration: 2-way
Dispersion (h x v): 90 degrees x 10 degrees
LF: 2 x 10-inch cones, bass-reflex
HF: 2 x 1-inch-throat B&C drivers on waveguide
Frequency Response: 55 Hz – 20 kHz
Maximum SPL: 135 dB
Power: Onboard class D, switch mode; onboard DSP
Rigging: 0 to 10 degrees in 2-degree steps
Size (h x w x d): 11.6 x 25.6 x 16.7 inches
Weight: 83.7 pounds
Companion Sub: MUSE 118FSA (single 18-inch)

RWG Spotlight Listing

Meyer Sound LEOPARD
www.meyersound.com

The LEOPARD linear line array loudspeaker system is the smallest and most versatile member of Meyer Sound's flagship LEO Family. LEOPARD and its accompanying 900-LFC low-frequency control element are designed to create an exceptional listening experience across a wide variety of applications from rental to install and rock 'n' roll to classical.

Boasting tremendous power-to-size ratio with ultra-low distortion, an array of six LEOPARD and two 900-LFC loudspeakers can be flown using only a 1/2-ton motor. Each LEOPARD loudspeaker is optimized for an array of six cabinets or longer, with default low-mid array compensation for utmost simplicity out of the box. It also features class D amplifiers that consume less power and generate less heat.

With its exceptional precision and headroom, the patent-pending LEOPARD reproduces the audio source with extraordinary accuracy, captivating audiences with both power and subtle musical detail.

TECHNOLOGY FOCUS: *LEOPARD can be driven by the Meyer Sound's Galileo Callisto 616 array processor, which provides matrix routing, alignment, and processing for array components. Meyer Sound's RMS remote monitoring system provides comprehensive monitoring of system parameters from a Mac or Windows-based computer.*

OF NOTE: *The onboard amplifier and control circuitry are contained in a field-replaceable module. QuickFly rigging with captive GuideALinks allows easy setting of splay angles. LEOPARD can also serve a variety of fill roles with LEO-M and LYON systems.*

KEY SPECIFICATIONS:
Configuration: 2-way
LF: 2 x 9-inch cones, Hybrid slot-horn loaded
HF: 1 x 3-inch driver coupled to CD horn through patented REM manifold
Frequency Response: 55 Hz – 18 kHz
Power: Self-powered class D amplifier
Rigging: Captive GuideALinks provide splay angles from 0.5 to 15 degrees.
Size (h x w x d): 11.1 x 26.9 x 21.6 inches
Weight: 74 pounds
Companion Sub: 900-LFC (single 18-inch)

K-Array KH2
www.k-array.com

Configuration: 2-way
Dispersion (h x v): 110 x 10 degrees (preset dependent)
LF: 2 x 8-inch neodymium cone drivers
HF: 2 x 1.4-inch neodymium compression drivers
Frequency Response: 70 Hz – 19 kHz
Maximum SPL: 136 dB
Power: Onboard class D amplification; DSP controlled, digital steering
Rigging: Integrated hardware, adjustable
Size (h x w x d): 9.8 x 25.3 x 8.3 inches
Weight: 62 pounds
Companion Sub: KS5 (dual 21-inch)

WorxAudio TrueLine V8
www.worxaudio.com

Configuration: 2-way
Dispersion (h x v): 120 x 10 degrees
LF: 2 x 8-inch neodymium cones
HF: 1 x 3-inch voice coil titanium driver
Frequency Response: 65 Hz – 18 kHz
Power: PXD Series amplifier platform recommended
Rigging Angles: TrueAim Tour rigging adjustable in 1-degree increments
Size (h x w x d): 10.5 x 28 x 18 inches
Weight: 108 pounds
Companion Sub: TrueLine TL118SS & TL218SS (single and dual 18-inch, respectively)

Renkus-Heinz VARIA VA101
www.renkus-heinz.com

Configuration: 2-way
Dispersion (h x v): 90 (also 60 & 120) x 7.5 (also 15 & 22.5) degrees
LF: 1 x 10-inch cone driver
HF: 2 x 1-inch neodymium drivers on proprietary Tuned Conic Diverter waveguide
Frequency Response: 60 Hz – 20 kHz
Maximum SPL: 126 dB
Power: Class D biamp (500 watts LF, 250 watts HF) with integrated RHAON networking; passive version also available
Rigging: Articulated hardware provides adjustment from 0 to 7.5 degrees
Size (h x w x d): 13 x 23.7 x 15 inches
Weight: 64 pounds
Companion Sub: VA15S (15-inch)

Adamson Systems S10
www.adamsonsystems.com

The new S10 is a 2-way, full-range line array enclosure ideal for a wide range of mid-size portable and install applications. It's loaded with two 10-inch ND10-LM Kevlar neodymium cone drivers joined by an NH4TA2 1.5-inch-exit HF compression driver.

A wave shaping sound chamber produces a slightly curved wavefront with a nominal dispersion pattern of 110 by 10 degrees (h x v). The chamber exhibits increased vertical response with minimal sacrifice of HF energy in the far field. Patent-pending Controlled Summation Technology further eliminates low-mid lobing. The S10 offers a maximum peak SPL of 141.3 dB, quite notable for such a compact enclosure.

The cabinet is made of marine grade birch plywood as well as aircraft grade steel and aluminum. The rigging system incorporates Adamson's proprietary Slidelock rigging technology for exceptionally easy setup and strike. An install-specific version (S10i) is also available.

OF NOTE: *The S10 (and companion S119 sub) are designed to be driven by the E-Rack unified rack solution, available in 8-channel and 12-channel configurations with Lab.gruppen PLM 12K44 amplifiers (with Lake processing) and supplied with a 20-port Ethernet switch to route dual-redundant Dante and control signal. Blueprint AV software is included.*

KEY SPECIFICATIONS:
Configuration: 2-way
Dispersion (h x v): 110 x 10 degrees
LF: 2 x 10-inch Kevlar neodymium cone drivers
HF: 1 x 1.5-inch-exit compression driver
Frequency Response: 60 Hz – 18 kHz
Maximum SPL (peak): 141.3 dB
Amplification: Designed for use with E-Rack
Rigging: Proprietary SlideLock rigging system allows angles to be set prior to lifting
Size (h x w x d): 10.4 x 29 x 20.7 inches
Weight: 60 pounds
Companion Sub: S119 (single 19-inch)

TECHNOLOGY FOCUS: *Controlled Summation Technology brings the LF drivers as close together as possible while symmetrically outwardly splaying them, which increases usable frequency range while decreasing summation at the crossover point, reducing interference. The LF drivers are also recessed behind the exit of the HF sound chamber so as to not limit their size and shape. Delay aligns the LF and HF.*

QSC Audio WideLine-10
www.qsc.com

Configuration: 2-way
Dispersion (h x v): 140 degrees; vertical array dependent
LF/MF: 2 x 10-inch cone drivers
HF: 1 x 3-inch compression driver on proprietary slot waveguide
Frequency Response: 55 Hz – 18kHz
Maximum SPL (peak): 133 dB
Power: Biamp or triamp; external amplification
Rigging: Integrated hardware, adjustable from 0 to 10 degrees in 1-degree increments
Size (h x w x d): 10.8 x 27.4 x 20.7 inches
Weight: 83 pounds
Companion Sub: WL218-sw (dual 18-inch), WL118-sw (single 18-inch)

VUE Audiotechnik al-8
www.vueaudio.com

Configuration: 3-way
Dispersion (h x v): 90 x 10 degrees
LF: 2 x 8-inch neodymium cone drivers
MF: 4 x 4-inch Kevlar neodymium cone drivers
HF: 2 x 1-inch-exit Truextent beryllium-diaphragm compression drivers
Frequency Response: 75 Hz – 18kHz (+/-2.5 dB)
Maximum SPL: 136 dB
Power: External VUE V6 Systems Engine
Rigging: Integrated hardware, angles selectable in 1-degree increments
Size (h x w x d): 10.2 x 29.4 x 17.5 inches
Weight: 76.6 pounds
Companion Sub: al-8-sb (single 18-inch)

Clair Brothers i208
www.clairbrothers.com

Configuration: 3-way
Dispersion (h x v): 120 x 10 degrees (90 degrees h also available)
LF: 1 x 8-inch cone driver
LMF: 1 x 8-inch cone driver
HF: 1 x 1.4-inch-exit compression driver HF waveguide
Frequency Response: 60 Hz – 20 kHz
Maximum SPL: 132 dB
Power: Amplifier recommended at 1,000 to 1,300 watts, 8 ohms
Rigging: Integral bimodal rigging with angle adjustments between 0 and 10 degrees
Size (h x w x d): 9.2 x 28.9 x 23.9 inches
Weight: 62 pounds
Companion Sub: iS118 arrayable subwoofer

RWG Spotlight Listing

EAW Anna | *www.eaw.com*

Anna brings all of the benefits of Adaptive Performance to mid-sized applications requiring excellent sound quality, high output and precise coverage, as well as the ability to quickly adapt that coverage to any venue geometry.

Anna's smaller footprint and lighter weight make it ideal for permanent or temporary use in sheds, theaters, clubs, mobile staging and corporate AV. Anna arrays also interlock seamlessly with Anya arrays to horizontally expand Anya-based systems for out fill or delay. Both Anya and Anna also integrate perfectly with Otto (EAW's Adaptive subwoofer) to form a cohesive one-family solution for any application.

With the combination of Anna's ingenious enclosure design and the power of Adaptive Performance, users can be confident that the system will provide spectacular results at every seat in any venue, for touring or permanent installation.

TECHNOLOGY FOCUS: *Anna enclosures include 14 built-in amplifier and processing channels, independently powering and processing each loudspeaker component. Resolution 2 software controls the processing of each acoustic cell individually to generate the ideal coverage pattern for the venue while minimizing the impact of the venue's acoustics. The end result is extremely high fidelity, output and coverage control.*

OF NOTE: *Anna modules hang straight, without any vertical splay, and Resolution 2 software adapts performance to produce custom-tailored, coherent full-range coverage pattern to perfectly match the venue in seconds.*

KEY SPECIFICATIONS:
Configuration: 3-way
Dispersion (h x v): 100 degrees x Adaptive
LF: 2 x 10-inch cone drivers (proprietary Offset Aperture loading)
MF: 4 x 5-inch cone drivers (on proprietary Radial Phase Plugs and CSA apertures)
HF: 8 x 1-inch-exit compression drivers on proprietary horn
Frequency Response: 45 Hz – 18 kHz
Maximum SPL (Unadapted): 130/136/141 dB (LF/HF/MF)
Power: Self-powered
Size (h x w x d): 11.3 x 40 x 23.6 inches
Weight: 135 pounds
Companion Sub: Otto

Outline Butterfly C.D.H. 483
www.outlinearray.com

Configuration: 3-way
Dispersion (h x v): 90 degrees x 7.5 degrees
LF: 2 x 8-inch neodymium cones, band-pass loaded
MF: 2 x 8-inch cones, horn loaded
HF: 1 x 3-inch-exit neodymium drivers on Double Parabolic Reflective Waveguide
Frequency Response: 110 Hz – 18 kHz
Power: 800/120 watts (LF & MF/HF, continuous RMS)
Rigging: - 0 to 7.5 degrees, 0.125-degree increments
Size (h x w x d): 9.4 x 29.6 x 23.6 inches
Weight: 75 pounds
Companion Sub: C.D.L. 1815 (cardioid)

Electro-Voice XLD291
www.electrovoice.com

Configuration: 3-way
Dispersion (h x v): 90 x 10 degrees
LF: 1 x DVN2080 8-inch cone driver
MF: 1 x DVN2080 8-inch cone driver
HF: 2 x ND2S 2-inch compression drivers
Frequency Response: 75 Hz – 18kHz (-3 dB)
Maximum SPL: 144 dB
Power: External, TG Tour Grade amplifiers recommended; LF/MF – 200 watts, HF – 80 watts (both RMS)
Rigging Angles: Integrated hardware, adjustable
Size (h x w x d): 9.9 x 28.6 x 14.5 inches
Weight: 48 pounds
Companion Sub: Subwoofer: XS212 (12-inch) & XLC215 (15-inch)

Bose Professional RoomMatch
www.pro.bose.com

Configuration: 2-way
Dispersion (h x v): Numerous coverage pattern choices
LF: Dual LF10 10-inch cones
HF: Six EMB 2 (2-inch voice coil) compression drivers on continuous-arc diffraction-slot manifold
Frequency Response: 60 Hz – 16 kHz
Maximum SPL: 139 dB (HF)
Power: Bi-amped, 500/150 W (LF/HF, long-term), PowerMatch Series recommended
Rigging: Integrated hardware, adjustable; optional frame accessories
Size (h x w x d): 16.9 x 39.1 x 23.6 inches
Weight: 123 pounds
Companion Sub: RMS215 (dual 15-inch)

dBTechnologies DVA K5

www.americanmusicandsound.com
www.dbtechnologies.com

Equipped with top-notch features and engineered for maximum versatility, dBTechnologies' DVA Series brought the benefits of line array technology to a much wider range of users. Now the Italian manufacturer has taken that ground-breaking technology to a new level with the DVA K5.

It incorporates a class D 500-watt (RMS) Digipro G3 digital amplifier that makes it capable of reaching up to 129 dB SPL. Thanks to FIR filtering technology, a new dual rotary encoder DSP user interface allows extremely precise system tuning.

The module's electronics and components are housed in a sturdy polypropylene box, reinforced with an internal anti-vibration metal structure which drastically reduces any resonance, allowing the cabinet to achieve an excellent acoustic response.

TECHNOLOGY FOCUS: *The double rotary encoder provides a separate tuning for LF correction (coupling) and MF/HF correction (high-angle midrange loss and air absorption compensation due to throw distance).*

OF NOTE: *The unique design, selected components and high-end amplification technology come together to deliver an exceptional weight-performance ratio: each system weighs just 31 pounds. The newly designed front grill has an integrated raincover.*

KEY SPECIFICATIONS:

Configuration: 3-way
Dispersion (h x v): 100 x 15 degrees
LF: 1 x 8-inch neodymium cone driver
LMF: 1 x 6.5-inch neodymium cone driver
HF: 2 x 1.4-inch voice coil compression drivers
Frequency Response: 70 Hz – 19 kHz
Maximum SPL: 129 dB
Power: Self-powered (class D)
Rigging: Integral rigging with 1.5-degree angle adjustments
Size (h x w x d): 9.6 x 23.2 x 13.1 inches
Weight: 31 pounds
Companion Sub: DVA KS10, DVA KS20

Alcons Audio LR16

www.alconsaudio.com

Configuration: 2-way
Dispersion (h x v): 90 x 10 degrees
LF/MF: 2 x AMB8 8-inch cone drivers
HF: 1 x RBN601 6-inch ribbon driver on "morpher" waveguide
Frequency Response: 70 Hz – 20kHz
Maximum SPL: 131 dB
Power: External Alcons ALC controller/amplifier; LF – 560 watts, HF – 70 watts (both RMS)
Rigging: Integral hardware, adjustable 0 to 15 degrees in 1-degree steps
Size (h x w x d): 8.9 x 30.5 x 17.1 inches
Weight: 63.9 pounds
Companion Sub: LR16B (dual-15-inch)

PK Sound VX10

www.pksound.ca

Configuration: 2-way
Dispersion (h x v): 100 x 10 degrees
LF/MF: 2 x 10-inch neodymium cone drivers
HF: Dual 4-inch planar wave drivers
Frequency Response: 110 Hz – 18 kHz
Maximum SPL: 137 dB
Dispersion: 90 degrees horizontal, vertical array dependent
Power: Onboard class D amplifier, 2,500 watts total for two cabinets (passive also available)
Rigging: Quick Fly hardware, adjustable from 0 to 7 degrees in 1-degree increments
Size (h x w x d): 12.6 x 28.1 x 15 inches
Weight: 63.5 pounds
Companion Sub: CX800 (dual 18-inch)

Grund Audio Design GA-2021N

www.grundorf.com

Configuration: 2-way
Dispersion (h x v): 100 degrees; vertical array dependent
LF: 2 x 10-inch neodymium cone drivers coupled to a dual asymmetrical fiberglass horn
HF: 2 x 1-inch compression drivers
Frequency Response: 50 Hz – 18 kHz
Maximum SPL: N/A
Power: Rated at 600 watts (RMS)
Rigging: Integral hardware, adjustable
Size (h x w x d): 12.2 x 37 x 13 inches
Weight: 65.5 pounds
Companion Sub: GA-L15 (single 15-inch)

RWG Spotlight Listing

TURBOSOUND FLEX ARRAY TFA-600H
www.music-group.com/brand/turbosound

The TFA-600H is a compact trapezoidal 3-way enclosure combining a Dendritic HF device and a patented midrange Polyhorn in a single physically aligned waveguide with equal path length, ensuring a phase-coherent wavefront at the horn mouth.

Neodymium drive units are used throughout in order to achieve the compact cabinet's exceptionally low weight, making it convenient to transport, handle and rig. In addition, the drive units are symmetrically arranged within the enclosure, which contributes to the smooth and consistent horizontal and vertical coverage.

The enclosure has both vertical and horizontal flying systems integrated into the cabinet. This flexibility of use is made possible by the rotatable mid/high section, making it possible to address the majority of sound reinforcement applications with only one type of loudspeaker enclosure.

TECHNOLOGY FOCUS: *The TFA-600H is switchable tri-amp/bi-amp. It includes a passive crossover network between the MF and HF drive units for bi-amp operation that can be bypassed for full tri-amp operation.*

OF NOTE: *The horizontal or "A" rigging mode is used to create flown or ground-stacked array configurations, while the vertical or 'B" rigging mode is for single box and virtual point source applications. A flight-cased trucking system allows three boxes to be pre-rigged and transported together.*

KEY SPECIFICATIONS:
Configuration: 3-way
Dispersion (h x v): 75 x 16 degrees
LF: 2 x 10-inch cone drivers (carbon fiber loaded, inside/outside wound coils)
MF: 1 x 6.5-inch cone driver
HF: 1 x 1-inch neodymium compression driver (aluminum dome)
Frequency Response: 80 Hz – 20 kHz
Power: Switchable bi-amp/tri-amp operation
Rigging: Integral rigging with angle adjustments between 0 and 16 degrees
Size (h x w x d): 12 x 28 x 22 inches
Weight: 93.5 pounds
Companion Sub: TFA-600L

Ramsdell Pro Audio LA10-2
www.ramsdellproaudio.com

Configuration: 2-way
Dispersion (h x v): Option of 120, 90 or 60 x 10 degrees
LF: 1 x 10-inch neodymium cone driver, vented
HF: 1 x 1.75-inch voice coil neodymium compression driver
Frequency response: 65 Hz – 18 kHz
Maximum SPL: 121 dB
Rigging: Integrated hardware, 0 to 10 degrees in 2 degree increments
Power: Active and passive versions available; optional bi-amp
Size (h x w x d): 15.9 x 21.7 x 12 inches
Weight: 29 pounds
Companion Sub: LA12-S (single 12-inch)

Carvin Audio TRX3210
www.carvinaudio.com

Configuration: 3-way
Dispersion (h x v): 100 x 10 degrees
LF/MF: 2 x 10-inch neodymium cone drivers
HF: 2 x 1-inch-exit Mylar compression drivers on a PurePath lens system
Frequency Response: 80 Hz – 18.5 kHz
Maximum SPL: 131 dB
Power: External, 500 – 1,000 watts recommended
Rigging: Captive flypoints, optional SureFly rigging
Size (h x w x d): 11.5 x 23.5 x 14 inches
Weight: 50 pounds
Companion Sub: TRx118 (single 18-inch), TRx121 (single 21-inch)

DAS Audio Event 210A
www.dasaudio.com

Configuration: 3-way
Dispersion (h x v): 90 degrees; vertical array dependent
LF: 1 x 10Mi4 10-inch cone driver
MF: 1 x 10Mi4 10-inch cone driver
HF: 1 x M-75 3-inch compression driver
Frequency Response: 70 Hz – 20 kHz
Maximum SPL: 134 dB
Power: Onboard 3-channel Class D amplifier (180 watts continuous/360 W watts peak per channel)
Rigging: Integrated rigging, angles adjustable
Size (h x w x d): 10.6 x 28.7 x 14.4 inches
Weight: 74.8 pounds
Companion Sub: Event 218A (dual 18-inch)

BassBoss LA88
www.bassboss.com

Configuration: 2-way
Dispersion (h x v): 120 x 8 degrees
LF/MF: 2 x 8-inch neodymium cone drivers, horn loaded
HF: 2 x 1.7-inch compression drivers on isophasic waveguide
Frequency Response: 50 Hz – 18kHz
Maximum SPL: 129 dB
Power: Onboard digital dual-channel amplifier (1,500 watts)
Rigging: Integrated hardware, adjustable in 1-degree increments
Size (h x w x d): 10 x 25 x 19 inches
Weight: 55 pounds
Companion Sub: Motive Horn & Profundo Series

ISP Technologies HDL 2208
www.isptechnologies.com

Configuration: 2-way
Dispersion (h x v): 100 x 10 degrees
LF/MF: 2 x 8-inch neodymium cone drivers
HF: 1 x 2.6-inch compression driver
Frequency Response: 68 Hz – 16kHz
Maximum SPL: 134 dB
Power: Onboard 3-channel DCAT amplifier (850 watts RMS total)
Rigging: Integrated hardware, adjustable from 1 to 10 degrees in 1-degree increments
Size (h x w x d): 9.1 x 24 x 19 inches
Weight: 62 pounds
Companion Sub: HDL118 (18-inch)

Alto Professional SXA28P
www.altoproaudio.com

Configuration: 2-way
Dispersion (h x v): 100 x 7.5 degrees
LF: 2 x 8-inch (2-inch voice coil) ferrite cone drivers
HF: 2 x 1.4-inch neodymium compression drivers
Frequency Response: 77 Hz – 18 kHz
Maximum SPL: 125 dB
Power: External; LF – 400 watts/HF – 75 watts (both continuous)
Rigging: Integral hardware with 20 degrees of tilt (maximum)
Size (h x w x d): 10.7 x 24.4 x 16.7 inches
Weight: 48.1 pounds
Companion Sub: SXA18P (18-inch)

SAE V2208P
www.saeaudio.com

Configuration: 2-way
Dispersion (h x v): 100 x 3 degrees
LF: 2 x 8-inch cone drivers
HF: 1 x 1.73-inch voice coil compression driver, horn-loaded
Frequency Response: 80 Hz – 18 kHz
Maximum SPL: 121 dB
Power: Self-powered (Class D)
Rigging: Integrated flying system with angle adjustments between 0 and 12 degrees
Size (h x w x d): 11.2 x 26.4 x 11.2 inches
Weight: 46.4 pounds
Companion Sub: V1212P LF module, V1218P subwoofer

VTC Elevation Series EL210
www.vtcproaudio.com

Configuration: 2-way
Dispersion (h x v): 90 x 10 degrees
LF: 2 x 10-inch neodymium cone drivers
HF: 1 x 1-inch-exit neodymium compression driver on Paraline lens (and both HF and LF components work with Synergy Horn design)
Frequency Response: 55 Hz – 20 kHz
Maximum SPL: 128 dB (LF)/135 dB (HF)
Power: External amplification, processor controlled. (1,200 watts LF/160 watts HF continuous power rating)
Rigging: Integral rigging with angle adjustments between 0 and 10 degrees
Size (h x w x d): 15.3 x 28.4 x 19.5 inches
Weight: 89 pounds
Companion Sub: ELS212, ELS218

SLS LS8800
www.slsloudspeakers.com

Configuration: 2-way
Dispersion (h x v): 110 degrees; vertical array dependent
LF: 2 x 8-inch cone drivers
HF: 1 x PDR1000 ribbon driver
Frequency Response: 72 Hz – 20 kHz
Maximum SPL: 129 dB
Power: External amp; recommended controller settings and amp power ratings
Rigging: Integrated hardware, adjustable from 0 to 10 degrees
Size (h x w x d): 9.6 x 28.25 x 13 inches
Weight: 60 pounds
Companion Sub: LSB8115 (single 15-inch)

T.J. Smith is the new president and general manager at **EAW**. Previously he held a variety of leadership roles with the Harman organization, with recent assignments including four years as general manager of the company's operation in Shenzhen, China, and management of the signal processing brands based in Salt Lake City.

The addition of Smith combined with the retirement of **Kenton Forsythe** has facilitated the transition of **Jeff Rocha** to the role of EVP of strategy and business development, where he is focusing on a product roadmap as well as driving growth in core vertical markets.

Jeremy Lommori has been promoted to head of technical sales and support for **Riedel North America**, overseeing the company's system and support engineering team. Based in Burbank, CA, most recently he served as a systems consultant at Riedel, and has also worked as live production manager at Mars Hill Church and as senior production engineer at Azusa Pacific University.

Dom Harter is the new managing director at **Martin Audio**, charged with helping to continue to grow the company's worldwide business. Previously he served as both director of R&D as well as sales director with Turbosound before joining Harman's Mixer Group, where he held several positions, including global sales director/VP for the mixer business unit.

Firehouse Productions recently added 96 **L-Acoustics** K2 enclosures to its inventory, in addition to 36 LA8 amplified controllers housed in a dozen LA-RAKs. Shared between the company's Red Hook, NY and Las Vegas locations, the new gear is already scheduled to be part of the L-Acoustics systems that will accompany Radiohead at Madison Square Garden and headline tours later this year by Florence & The Machine and the "Rock, Paper, Scissors" double bill of Peter Gabriel and Sting. LSI

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PROBLEMS & SOLUTIONS

Optimizing personal monitoring for electric bass players and related issues. *by Mike Sokol*

Recently an attendee of one of my workshops e-mailed about problems he's having in providing a bass guitar mix over a personal monitoring system utilized by a 4-piece band, all of whom wear in-ear monitors.

The system has one transmitter feeding four receivers, receiving a simple split-track mix with all vocals on one channel and all instruments on the second channel. (Note that the bassist is plugging directly into the snake via an active DI box without any amplifier or processor on stage.) The band is complaining that the bass guitar sounds "fuzzy" in all of their headsets. Here are some recommendations.

- Check to make sure the instrument output isn't overdriving the input of the active DI box. If the bass sounds fuzzy or distorted in the PA, or while soloing that channel on the console with headphones, then engage the 20 dB pad. Active DI boxes (and yes, passive models as well) can be overdriven and distorted by active guitar pickups. Monitor for this at the console with a good set of headphones.
- Many bass guitarists play quite aggressively (pop and slap), which usually is compressed and limited by their on-stage amplifiers or processors. However, a direct connection into a DI (passive or active) doesn't round off these dynamic peaks, which can cause distorted or fuzzy sound anywhere in the signal chain that doesn't have sufficient headroom to deal with it. Integrating a Line 6 Bass POD or SansAmp bass module can provide an elegant solution.
- IEMs sometimes can't generate enough low-end energy for bassists to monitor themselves properly. If there's a lot of bass leaking back to the stage from a loud PA, then the IEMs can provide the mid-high frequency detail. But with a nice 85-90 dB SPL mix in the room, there may not be enough bass for the player to "feel" what he/she is playing. A solution can be adding a low-frequency "thumper/shaker" such as the ButtKicker Concert, Prosound 429 from Clark Synthesis, or the TST429 Platinum from Sensaphonics, which can be mounted on a small platform for the player to stand on. The result: instant bass that's felt without anyone else hearing it.

Also, a very small bass amp with a direct out might be a good idea, as long as the volume level is kept to a low roar. Any small practice amp should do the trick; let the main PA make bass for the room.

- Combining everyone's personal monitoring into a common transmitter that feeds multiple receivers is less expensive but it's also a lot less "personal." This approach can work when everyone needs a similar mix (multiple backup singers, for instance), but it's generally not a good idea with IEMs fed by wildly different instruments. Just as is usually the case with floor monitors, each person/player wants to hear his/her own instrument predominantly, plus a lower-level mix of what's needed for cueing.

It's better to utilize individual channels or transmitters so each player can receive a tailored, custom mix. SPL levels in the IEMs should be much lower for each person, and without overdriving the transducers within the earpieces as players try to hear what they're playing over what everyone else is playing.

This also holds true for floor wedges. Supplying independent mixes on six different wedges will end up being much lower in volume in the long run since each artist gets to hear primarily themselves plus a few cue instruments. With the recent advent of many options of powered wedges available, this is considerably more cost effective than it used to be.

- Finally, be aware that it's very dangerous for musicians to wear only one earpiece on a loud stage. It's very difficult to hear how loud the SPL level is coming into the one ear via the earpiece while the other ear listens to the band. Artists cranking it up in only the one earpiece can do extensive hearing damage in just a few shows.

In fact, some monitor engineers won't run sound for anyone who insists on using only one earpiece for fear of a lawsuit if the artist goes deaf. So be very careful if anyone in a band you're working with wants to try this sort of thing. LSI

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


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