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Live Sound International (ISSN 1079-0888); USPS 011-619, Vol. 23 No.9, is published monthly by EH Publishing, 111 Speen Street, Suite 200, Framingham, MA 01701 USA. US/Canada/Mexico subscriptions are $60 per year. For all other countries subscriptions are $140 per year, airmail. All subscriptions are payable by Visa, Master Card, American Express, or Discover Card only. Send all subscription inquiries to: Live Sound International, 111 Speen Street, Suite 200, Framingham, MA 01701 USA. Canada Subscriptions: Canada Post Agreement Number 40612608. Send changes of address information and blocks of undeliverable copies to Pinney Bowes International, PO Box 25542, London, ON N6C 0E2. POSTMASTER: send address changes to Live Sound International, PO Box 56, Framingham, MA 01701. Periodical Postage paid at Framingham, MA and additional mailing offices. Reproduction of this magazine in whole or part without written permission of the publisher is prohibited. Live Sound International® is a registered trademark of EH Publishing Inc. All rights reserved. 2014 EH Publishing. Check us out on the web at http://www.prosoundweb.com.
Sonic harmony through a TFT touchscreen or the sophisticated d&b trilogy: ArrayCalc simulation software, R1 Remote control software and the pristine D80 amplifier make for quick access to all system settings.
From the Editor’s Desk…

On a recent Friday I climbed into our “seasoned” (old) minivan and pointed it toward Cincinnati, a city I’ve visited several times and thoroughly enjoy. The last time there, I heard a superb performance by the Cincinnati Pops Orchestra conducted by legendary film composer John Williams, and met a lot of really nice people in the process (October 2010 LSI).

The recent visit proved just as worthwhile in getting acquainted with Grant Cambridge, an up-and-coming audio professional, and checking out the first-ever day of the Buckle Up Festival. It was a pleasure talking with a bright member of the “younger generation,” and I encourage you to check out what he has to say in this issue about how he’s gone about building his sound company.

Meanwhile, a familiar name in these pages, Mark Frink, has contributed an interesting look at the audio approach for the Steely Dan concert tour, also obtaining the answer to the intriguing question, “What’s used on a Steely Dan tour to tune the PA?” In addition, Mark has contributed an insightful overview of approaches for monitor engineers to get ahead of the game.

Senior contributing editor Craig Leerman provides a primer on key organizational points for busy production companies, along with another installment in his popular Microfiles series on vintage microphones as well as a Road Test of a recently introduced loudspeaker line.

Danny Abelson continues the discussion with noted front of house engineer Dave Natale on his experiences, and be sure to take Ken DeLoria’s test on the Back Page. And as always, there’s much more. Enjoy the issue.

Keith Clark
Editor In Chief, Live Sound International/ProSoundWeb
kclark@livesoundint.com
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D&B AUDIOTECHNIK
Y-SERIES

Detailing a flexible new loudspeaker range.

by Live Sound Staff

THE NEW Y-SERIES from d&b audiotechnik provides a flexible, configurable option for small to medium applications. Y7P and Y10P loudspeakers handle point source duties, with the B6-SUB providing extended frequency response. In addition, two line source loudspeakers and a matched cardioid subwoofer – Y8, Y12 and Y-SUB – are designed for line array tasks.

The Y7P and Y10P loudspeakers share the same dipolar 8-inch driver arrangement centered on a 1.4-inch compression driver fitted to a rotatable CD horn, facilitating deployment horizontally or vertically. Dispersion characteristics of 75 x 40 degrees and 110 x 40 degrees (h x v), respectively, present several deployment options, individually as a full range system or in combination with other elements from the Y-Series, either ground stacked or flown.

A port design within both models extends LF performance down to 59 Hz. The B6-SUB incorporates an 18-inch driver built into a bass-reflex design, extending frequency response down to 37 Hz.

Meanwhile, the Y8 and Y12 utilize the same rigging design as their bigger brothers from the J- and V-Series, and they also share the same 80- and 120-degree horizontal dispersion characteristics. They can be suspended in columns of up to 24 loudspeakers, with splay angles from 0 to 14 degrees with a 1-degree resolution. Dual 8-inch drivers with neodymium magnets mounted in a dipolar arrangement around a 1.4-inch compression driver enables the Y8 and Y12 to offer a horizontal dispersion pattern that’s controlled down to 500 Hz.

The Y-SUB houses a forward facing 18-inch woofer and a 12-inch woofer radiating towards the rear, producing a cardioid dispersion pattern. Driven by a single amplifier channel, the Y-SUB can be ground stacked or flown at the top of a Y8/Y12 array.

Y-Series models assimilate into a workflow comprising the Array-Calc simulation software, the R1 Remote control software and d&b amplifiers. The process is designed to foster consistent and efficient results. Numerous line and point source loudspeakers can be combined within Array-Calc, providing a graphical representation detailing the coverage, level drop and safety aspects of a system setup in a given space.

The R1 export function transfers all configurations and settings into an R1 Remote control project file, taking into account any system specific functions. The R1 workplace presents the format for operating systems via the Remote network through CAN-Bus to access D6 and D12 amplifiers, as well as the D80, which can also be controlled through Ethernet using OCA protocols.

Y-Series loudspeakers are available in both portable/mobile and installation (Yi-Series) versions, differing only in cabinet construction and mounting hardware. Special color and weather resistant options, along with transport options and accessories, are available.
Morris Light and Sound out of Nashville, TN has been mighty busy these days. The team recently deployed a NEXO STM System for the Bayou Country Superfest in Baton Rouge, LA and the Florida Country Superfest in Jacksonville, FL. These star-studded events featured country’s finest heard through NEXO’s finest. Moving on, they then deployed the same system in 5 different configurations at the legendary CMA Music Festival back home in Nashville.

We caught up to David Haskell – Owner of Morris Light and Sound, to get his thoughts on the STM system.

“Having the flexibility and ability to configure and quickly deploy the STM for every option ranging from a small ground stack system to a full stadium rig with the same sonic results is both amazing and cost effective. STM is truly THE system for ALL audio applications.”

— David Haskell
CLEANING UP

Too loud? Maybe volume isn’t the reason.

by Karl Winkler

ONE OF MY PET PEEVES: there really is no excuse for loud, bad concert sound. It’s a topic I’m revisiting in light of Dave Rat’s comments in a recent issue (June 2014 LSI).

In particular, what piqued my interest was Dave’s statement that “painfully harsh, poorly mixed sound is always too loud.” His point is that yes, rock concerts are (and should be) loud, and even so, they aren’t as measurably loud as a NASCAR race or an NFL game. And I agree that sound level can be an important part of an experience.

But for many, sound that is “too loud” takes away from the enjoyment of the audience rather than enhancing it. So let’s examine some of the reasons that an audience might have that impression.

TURN THE TABLES?

First, of course, is the fact that perhaps

the sound really is too loud. Yes, I may be getting older, but I’ve found that in many cases, and this includes trendy bars and restaurants, that the volume level often makes me uncomfortable. I read an article recently providing one explanation for this phenomenon, that the idea is to “turn the tables” more quickly so patrons come in and order a few drinks but don’t hang around longer than necessary. The use of sound as a negative force is intended to increase profits.

O.K., but what about rock shows? This, of course, is where Dave’s point is most valid. Audiences want an experience, and loud sound is a big part of that. Heck, without body shaking bass, we’re not sure if we’re really even at a rock show and not just listening to an iPod. I remember an interesting demo given by the makers of ServoDrive subwoofers a few (ahem) years ago. Their point was that with a deep, extended bass response and fairly mild mids and highs, the sound would be perceived as “big” by the audience, yet people could still talk to one another.

But back to the point about volume... At big shows, I like it loud too, as long as it’s clean loud sound. Which brings me to my next point: what if the sound isn’t “clean”? In other words, what if it’s distorted?

I contend that distortion, as a result of poor gain structure, is the number one cause of sound being “too loud,” particularly to the novice listener. The simple reason is that distortion artifacts largely fall into the upper midrange of the human hearing system, right smack where our ears are most sensitive.

Distortion caused by bad gain structure usually happens because one gain stage (or device) is overdriving the next one in the chain. For instance, the problem can happen within a console, or between a console and a drive rack. It can also occur between a wireless micro-
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phone and the console input.

A common source of high-frequency distortion is vocal sibilance, due to the tremendous energy found in some vocal sounds, particularly “s” and “f.” Fortunately, today we have an array of tools to help combat this problem. Generally, it’s good to start with the right mic capsule for the vocalist’s voice, and to make sure the gain structure is super clean the rest of the way through. If there’s still is a problem, we have plug-ins to help.

**SMILEY FACE, ANYONE?**

Next up is poor system EQ; between it and distortion, we’ve identified 90 percent of the loudness problem. Again, behind the desk, everything is perfect. But in some of the overlapping loudspeaker zones, maybe there’s HF buildup, maybe joined by some phase issues.

I also think it’s far too common to add mids and highs to individual channels to “bring them out” when it might be better to cut something else instead. This kind of boosting on several channels can very likely result in overloading hearing while not really bringing about the improvement in clarity we’re seeking. In the process, we might be pushing the master bus, a subgroup or a matrix channel over the edge, resulting in distortion.

Then there’s the famous (infamous?)

**TASTE IN MUSIC?**

One of the more subtle effects that causes some audiences to claim that sound is too loud has to do with their familiarity with the particular music being presented. Most of my years touring were with the U.S. Air Force jazz band, the Airmen of Note. It’s an 18-piece big band that performs traditional selections (Glenn Miller, Count Basie, etc.) as well as modern arrangements and compositions.

After years of touring with them, I came to the conclusion that if the music is familiar to the audience, no one thinks it’s too loud. Case in point: many of our concerts featured traditional style for the first half and modern style for the second half. I don’t recall a single complaint about the volume from anyone for a first half. In fact, after an hour of this music, people approached me at intermission asking why the band didn’t play one song or another. Some claimed it was “exactly like when I saw them in 1943 during the war.”

But when the exact same group, with the same PA and same FOH mixer in the same hall, played the second half, complaints were commonplace. The difference? You guessed it — the audience was unfamiliar with the material.

To come full circle, I think familiarity is a big reason why people like it loud when they go to rock concerts. They want to see their favorite artists playing their favorite hits, loudly and with a big PA that says “THIS IS A BIG ROCK CONCERT.” And there’s nothing wrong with that, as long as it’s also clean sound, well mixed.

**KARL WINKLER** is director of business development at Lectrosonics and has worked in professional audio for more than 20 years. Reach him at karl1@karlwinkler.com.
As a FOH engineer and master of ceremonies, you want the entire audience, front to back, in tune with your mix. Every lyric, symbol, bass and high note in glorious definition.

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IN THE BALLPARK
A quick and easy method for presetting console input gain.

by Pat Brown

BOTH ANALOG AND DIGITAL mixing consoles have an input gain control ahead of the channel fader. The gain control’s job is to scale the input signal to an appropriate level. It’s setting is source-dependent, which means that proper setting requires a sound check that includes the vocalist and their respective microphone. There are three variables in play:

1. Talker level
2. Microphone sensitivity
3. Talker-to-microphone distance

It’s not always possible to set the input gain in advance, but there is a simple method for “ball parking” the setting without the “talent” being present. I use a small, battery-powered loudspeaker with internal pink noise source (Figure 1).

This device provides an acoustic reference level, and when maxed out, produces about 70 dBA-Slow at 1 meter, which is about the level of a "raised-voice" talker. The level at the grill is about 100 dBA-Slow (Figure 2), which would be about the level of a raised-voice talker with their lips on the microphone.

1. Place the vocal mic against the grill of the reference source (Figure 3).
2. Set the channel fader and main fader to the desired setting (usually at or near “zero”).
3. Adjust the input gain of the mixer channel to a strong reading on the mixer’s main meter, allowing some “summing room” for the other channels that will be added to the mix (about 10 dB is usually sufficient for mixes of 12 channels or less).
4. Repeat for the other vocal microphones.

I now have the input gain “roughed in” for a raised-voice talker with their lips against the mic. Now, we all know that the level presented by the talent will be different than this. If the level is lower, advance the input gain a bit to compensate. If the level is stronger, reduce it a bit.

Since a reference has been set, the input gain is compensating for the difference between “raised-voice” and “actual.” It’s much faster and less obvious than starting from scratch with a dead microphone. This technique is especially useful for wireless mics, since there are gain structure considerations ahead of the mixer.

I recently mixed for a local school play that used a pool of wireless mics of differing brands and types. Here is what
I did to setup each microphone.

1. Place the mic against the grill of the reference source.
2. Adjust the transmitter to produce the desired signal level at the receiver (yellow on the receiver’s meter) (Figure 4).
3. Adjust the mixer’s input gain as described above.

4. Repeat for each wireless microphone.

This optimized the gain structure of the wireless system, and produced the same level into the mixer (post input gain) from all of the wireless systems (a potpourri mixture of Shure and Lectrosonics wireless systems with three different types of headworn mics), including handheld and headworn units. Now any mic could be handed to any performer, and the input gain of the mixer quickly tweaked to compensate for their actual level relative to the reference.

To finish the system gain structure, I played a music track to “meter zero” on the console and adjusted the amplifier levels (in this case, powered loudspeakers) for the desired SPL in the house. Remember that meter zero is approximately what the multiple channels will sum to with all mics in use.

Of course, there are other ways to get there, but getting levels in the ballpark using a reference acoustic source greatly simplifies system setup.

PAT & BRENDA BROWN lead SynAudCon, conducting audio seminars and workshops online and around the world. For more information go to http://www.synaudcon.com.
Steely Dan’s Jamalot Ever After tour of North America kicked off in July in Oregon, and coming up on an impressive 56 dates, is wrapping up in late September in Port Chester, NY. Donald Fagen and Walter Becker are well known to both fans and pro audio folks for being demanding about sound production, so the question must be asked: “What mix of technology and expertise meets their demands for perfection on tour?”

Returning to mix Steely Dan is front of house veteran Mark Dowdle, who mentions that this tour is on the “Bucket List” of many live sound engineers. (Many pro audio journalists too.) “From the beginning of my career, I’ve always been inspired by their timeless music,” he simply states. His extensive credits include Elton John, Gloria Estefan, Fleetwood Mac, Tina Turner and Jackson Brown, to name just a few.

Dowdle is mixing on a 48-channel Midas XL4 console. “Donald and Walter’s requirement of using analog makes the XL4 the obvious choice,” he notes. The XL4 for the tour, supplied by Jim Sawyer Professional Audio Service (New Brighton, MN), is fitted with stereo channels for overheads, Leslie and Nord, plus six effects returns. “This was the first time I’ve had an XL4 with moving faders for the inputs as well as VCAs, which I use to maintain fader positions,” Dowdle notes. “It allows me to start with a preset from a previous show where I was happy with the mix.” At the console, he’s one of many who have recently adopted Crown Seating’s Stealth Chair, which applies ergonomic principles to provide a comfortable console operator’s chair.

Also new on the tour is the addition of a Martin Audio Multi-cellular Loudspeaker Array (MLA) main system, provided by On Stage Audio (OSA), which has offices in Las Vegas, Chicago and now Nashville. Dowdle also mixed Steely Dan on last year’s tour, using an MLA system at Ravinia Festival, north of Chicago, the first season of a new OSA installation highlighted by two 12-box arrays in the 3,200-seat pavilion.

Tour sound icon Robert “Nitebob” Czaykowski, Steely Dan’s road manager since 2007, has ample pedigree as both TM and FOH for Ian Hunter, Ace Frehley, New York Dolls, and Alana Davis, with beginnings mixing Aerosmith, KISS and Ted Nugent. He mentions the punch and power that horn-loaded systems provide in the low-mids, especially for guitar bands, that’s lacking in front-loaded line arrays. “I used most of Pink Floyd’s Martin PA for Aerosmith’s 1977 tour,”

Reelin’ In The Years

Concert sound for Steely Dan’s summer tour.

by Mark Frink, Photos by Steve Jennings

S
At left, Steely Dan performing on this summer’s Jamalot Ever After tour. Above, Donald Fagan (left) and Walter Becker doing what they do best. 

he recalls. “The six-week European headline tour I did with KISS in 1988 was with a Martin F2 PA.”

Nitebob introduced Becker to MLA at an OSA demo in New York’s Manhattan Center last October during the AES convention. “Walter heard it, was able to walk around the room and see what the coverage was like, and asked them to play different types of tracks through it,” he explains. “The coverage is really good.”

Steely Dan often plays theaters where the mix position is underneath a balcony. “I go out there every day and listen to it; that’s part of my gig,” he says. “The thing that really knocks me out about MLA is that you can actually control it so that it’s not splattering off a back wall or clustering up in the lower balcony.”

Do It Again
This year’s tour started as usual with band rehearsals at SIR in New York. “I was actually in the room with the band mixing on a set of Tannoy AMS 12 nearfield monitors, which I believe to be truth in listening,” Dowdle says. “At production rehearsal, the transition from the Tannoys to the MLA proved to be as accurate as I had hoped. I was very pleased, because the mix was in the pocket right off the bat.”

MLA is a powerful marriage of loudspeakers and software, where transducers are individually driven and optimized to deliver the summation of crystal clear sound at the audience ears, ensuring even and smooth coverage across the coverage area. The system does this by controlling EQ and phase for individual transducers after modeling the physical listening area.

Dowdle points out that MLA provides extremely even front-to-back SPL as well as evenness of frequency response throughout the listening area. “The coverage is very smooth, especially its shading,” he notes. “You can walk up on the PA in the front and it sounds just like it does in the back of the room.”

He also points to an improvement to the stereo field. “Everything is more defined, so that automatically translates into the stereo field being more discernable. MLA gives me dynamic range, clarity and definition so that I’m able to position and layer sounds in the stereo field and really hear where they all are.” He adds that the sound is extremely coherent and very responsive from a mixing standpoint. “You make a small fader move and it’s immediately noticeable.”

Dowdle has been surprised by the constant comments from the audience. “I’ve been mixing for a long time and usually nobody ever says anything. This particular tour, I’ve had more response from the audience than any tour I’ve ever done in my entire career, and it’s always been very positive and it’s always been very poignant. That’s in large part because of MLA allowing me to get it exactly how I want everywhere in the room.”

Fagan and Becker are known for being fastidious about sound quality. “Both have come out into the audience on a number of occasions and always been positive with their feedback and what was going on,” Dowdle says. “More often than not, Donald will come out and listen and his comment most often is ‘it sounds great,’ which is probably the highest compliment that I could ever receive in my career.”

Any Major Dude
OSA crew chief and MLA system engineer Martyn “Ferrit” Rowe worked with TASCO for years and then independently with A1 Audio, Electrotec, and PRG, mixing monitors for Judas Priest for 15 years and White Snake for 12 years. “Iron Maiden, Black Sabbath, Ozzy Osborne, big hair and spandex in the ’80s, it was probably me,” Rowe says. “I first met Mark [Dowdle] on Alice Cooper in
Europe in ‘82.” Many in pro audio also know Ferrit from his work with EAW and Martin Audio before joining OSA as director of engineering.

The tour travels in two trucks, carrying consoles and backline (including a Steinway grand) in one truck, and lights and PA in the second. “We’re carrying 26 MLA and 2 MLD down fills, as well as 18 MLA Compacts, plus 8 MLX subs and 6 W8LMD used as front fills,” Rowe notes. “The first 18 feet is 45 percent of the weight.”

The tour played arenas in Oklahoma City and New Orleans using 14-box MLA mains and 9-box MLA Compact side arrays, adding a few MLA at the LA Forum, courtesy of Delicate Productions (Camarillo, CA) “That was the only venue on the 56 shows that we had to add PA,” states Mitchell “Bubbles” Keller, fourth-year Steely Dan production manager, former Masque Sound live manager and previously PM for The Ramones and Iggy Pop. “It’s quite impressive that on a two-truck tour we can carry enough PA to do an arena.”

The rest of the itinerary ranged from sheds, theaters and casinos. “The smallest venue was Humphrey’s in San Diego, putting two subwoofers and ground stacking six MLAs” Ferrit says. “I’ve actually just done four shows in a row where we’ve done a single point hang with 10 MLA enclosures.”

As well as the flexibility and scalability of the system, he’s also keen to point out the simplicity of operation with MLA. “It’s like fly by wire; you tell it what you want and the software produces a custom preset for your system and the room. This isn’t auto EQ, you still have control over all the decisions that are being made, but the computer is doing the heavy lifting.”

Ferrit adds that MLA provides a considerable amount of consistency. “Once I’ve done the optimization procedure the system is 98 percent tuned at turn-on. Before MLA, I was a firm believer in multi-measurement [Rational Acoustics] Smaart. Now I can just place a measurement mic at the mix position, where you’re making your EQ decisions anyway, knowing that most of the room sounds very similar.”

**Green Earrings**

In addition to mixing the support act and flying MLA, OSA system tech Bob Alumbaugh is responsible for the tour’s mic inventory. As studio connoisseurs, Fagan and Becker are especially particular about microphone and outboard electronics selections, which were made primarily by Dowdle.

Earthworks SR40V condenser vocal mics have been the choice for Becker and Fagen’s vocals since their introduction in 2011. Background vocalists
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An AEA KU4 ribbon mic for horn solos located downstage center.

Cindy Mizelle, Carolyn Leonhart-Escoffey and LaTanya Hall sing into new Telefunken M81 dynamic vocal mics, which retains the M80’s original 5 and 10 kHz presence peaks without its rising HF response. “The M81 produces a smooth vocal range while maintaining an isolated, somewhat remote characteristic to the backing vocals so that I’m able to place them in the mix with pinpoint accuracy,” Dowdle says.

Michael Leonhart and Jim Pugh play trumpet and trombone into a pair Neumann U87 big-boy condensers. Walt Weiskopf’s tenor sax and Roger Rosenberg’s baritone sax are captured with two AEA R84 ribbon mics, angled to take advantage of their classic figure 8 polar pattern with its 90-degree side rejection. “The AEAs for the saxes in the lower register are awesome,” Alumbaugh states. “Combined with the clarity of the U87s used on bone and trumpet, the balance between them is great.”

Downstage center, an AEA KU4 supercardioid ribbon mic – a re-creation of a rare RCA KU3A used for Hollywood film scoring in the 1940s – is deployed for featured horn solos. The design employs a unique baffle to attenuate the rear lobe of a typical figure-8 ribbon. Monitor engineer Peter D. Thompson says the KU4 “sounds delicious and has great rejection qualities as well.”

My Old School

The approach for Keith Carlock’s six-piece Gretsch drum kit starts with a Telefunken M82 dynamic end-address mic inside the kick, employing it’s 350 Hz notch filter. “I’m able to reproduce the kick in ways that truly emulated the original Steely Dan recordings,” Dowdle says. “The mic’s filter switch and internal shock mount system are vital to its response and quality.”

The infamous Granelli Audio Labs G5790, a 90-degree modified SM57, is used on both snare and second snare. “The Granelli’s right-angle allows the mic to correctly address the snare drum, which is often compromised by hi-hats and drum hardware nearby,” he explains.

A Shure Beta 56A dynamic is applied for snare bottom, with Sennheiser e904 dynamics on the four toms. Telefunken’s new M60 small-format tube condenser handles hi-hat and ride cymbal. Dowdle toured with prototypes last year, which are an FET version of the AKG-designed class-A ELA M260 tube pencil condenser. “The transient response is magnificent and the audio is so crisp and clear that it seems almost too good to be true,” he says. “The M60 is the best high hat mic I’ve ever used.”

Telefunken C12 tube condensers serve as overheads. “Having used C12s in the studio, I knew I’d be pleased, but I had no idea what a huge difference they’d make,” Dowdle notes. “The drum sound is robust and truly amazing. I’m able to capture every nuance from not only the cymbals, but the rest of the kit as well. The result is a powerful, open sound with a pleasing air that emanates from the entire drum kit. They’ll have to pry these mics from my cold dead hands.”

Becker’s guitar plays through a Satellite Mudshark or one of several Dr. Z tube amp heads (KT45, RxES, Mazerati GT). Three double-12 cabinets are each miked with a Shure SM7B dynamic. The cabinets are removed offstage and gober’d with foam and packing blankets. Music director Jon Herington’s closed back dual-12 Celestion cab is powered by a Guytron GT-100 “F/V” head and is also miked offstage with a Shure KSM313 ribbon mic.

Keyboardist Jim Beard’s Hammond XR-3 into a Leslie 122 is captured with a Sennheiser MD421 dynamic on the low rotor and a pair of Shure KSM313s, while his Nord Stage keyboard is taken direct with a Radial J48 active DI. The Steinway D grand piano is outfitted with an Earthworks PM40T Touring PianoMic System, supplemented by a 3-bar Helpinstill pick-up feeding another Radial J48.

An API 2500 stereo optical compressor is inserted on the XL4’s main mix bus, which Dowdle notes that he
prefers whether using a digital or analog console. “I just give it light compression, just tickle it, and it brings the whole thing to a nice level where it’s very pleasing and extremely musical,” he elaborates. A second API 2500 is inserted on Fagen’s vintage Fender Rhodes, one of three (dubbed “Grace,” “Wilma,” and “Lucy”) carried on tour and lovingly refurbished by RetroLinear.

Twin Tube Tech CL-1B optical compressors are inserted for Fagen’s vocal mics, both when he’s sitting at the Rhodes or standing. Four more CL-2A dual-channel optical compressors are applied on backing vocals and the horn mics. Digital effects include a TC Electronic 2290 delay for “Black Friday,” a vintage AMS RMX16 reverb for drums, a TC M5000 reverb for horns, and a dual-machine Lexicon 960L for both main and backing vocals, along with an Eventide H3500 to fatten background vocals.

Peter D. Thompson, a 24-year veteran of Thunder Audio (Livonia, MI and Nashville) has mixed monitors for Steely Dan since 2007, and does the same for Squeeze and Bob Seger. Recently, he also mixed FOH for The Strypes. Thompson employs a Soundcraft Vi6 digital console for his mixes, delivered to 13 Meyer MJF-212A self-powered floor monitors, using singles for everyone except Fagen and keyboardist Beard.

“It’s nice to have stage monitors that sound like studio monitors,” he says. “I’m running all the mixes flat, other than some minor graphic EQ adjustments during the show to the vocal mixes. A majority of the equalization is made on the channels so everyone hears the same thing in each mix.” The sax players are provided with Meyer UPJ-1P compact self-powered loudspeakers on custom stands. Herrington, Leonhart, and Carlock receive in-ear mixes via Shure P6HW hardwired belt packs, and the drum IEM is supplemented with a pair of Meyer 600-HP dual-15-inch self-powered subs.

As with most FOH engineers of Dowdle’s vintage and background, Steely Dan’s catalog and Donald Fagen’s solo album *The Nightfly* are standards for tuning and evaluating sound systems. To answer the perennial question, “what’s used on a Steely Dan tour to tune the PA?” he replies that he uses Steely Dan live every day during sound check. But prior to that, he plays Thomas Dolby’s “My Brain is Like a Sieve” from *Aliens Ate My Buick* and Frank Zappa’s “Lucielle” from *Joe’s Garage*, which has a very natural sounding vocal. A Shure KSM9 condenser mic then helps in the fine-tuning of its response.

MARK FRINK is an independent author, editor, consultant and engineer and can be reached at LiveSound@MarkFrink.com.
THE ART OF LOGISTICS

Properly managing materials and equipment.

by Craig Leerman

I’VE WORKED IN THIS BUSINESS for well over three decades, and some things haven’t changed and probably never will. The promoter always wants more for less, the rider sent to me was the “old” one, and if the gear doesn’t work I don’t get paid. There’s little that can be done about the first two, but I can make sure my equipment stays in top shape in order to be able to pay the rent, staff, and bills.

Taking care of gear is an absolute top priority, making it far less likely to break down or malfunction unexpectedly at a show. In addition, keeping it clean and looking good inspires confidence from clients who are apt to pay more because the stuff looks and (probably) sounds better than the competition.

Whether touring the world or doing a gig at the nearby tavern, successful production companies master the art of logistics, which I define as the management of materials and equipment between the warehouse and the end user.

MAKE A LIST (OR THREE)

The first aspect to address is inventory management, or where to store the gear. One small system may be easier to keep together in a truck or trailer. (Just remember to safeguard against extreme temperatures.)

Most of us have more stuff than that, however, so we need to keep it in a garage, self-storage unit or office/warehouse. Devote dedicated floor areas for larger items and use shelving, cabinets, and/or drawer units for storage of smaller items. At my company, we repurpose old filing cabinets to store and organize microphones and a lot of small parts. The mics are kept in their factory cases or small foam-lined pistol cases, organized by type into different drawers along with stand adapters, drum claws and clamps, mic clips and DIs.

Next, think about preventative maintenance and figure out a maintenance schedule. (I’ve covered this in depth previously, most recently in the July 2013 issue, and it’s also available on Pro-SoundWeb.) Things like cables might need attention after each gig while other items, such as road cases, might only need a little attention once a year. We do a check of gear both as we set it up and then pack it up at each show, making a list of any items that may need attention back at the shop (like a bad caster) and marking any bad cables and separating them so they don’t get taken to the next gig by mistake.

When booking a show, we create an event equipment list, a listing of all the equipment and spares that are required for that particular event. There are software programs that do inventory management, or you can simply set up spreadsheets to list all items required for a particular system. It becomes our pull list for the gig, serving as a handy way to check off items as we gather them for the show and stage them in one spot for packaging and loading into the truck.

The events my company handles run the gamut from simple to complex, and we end up with a lot of smaller items that need to get to the gig. Instead of schlepping a lot of individual small cases, we use some larger road trunks that can carry all of the little items in...
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one package. To make things more efficient during load out, we label the large trunks as to their contents so stage hands can figure out which trunk each item goes into.

Some folks prefer trailers on smaller shows, while others use cargo vans or small box trucks. Larger shows typically require big box trucks or even tractor trailers. No matter, securing the load in the truck or trailer in a safe manner so it will ride well down the road is a must.

There are a few options for cargo retention, with truck straps being the most common (and best) option. The straps can hook onto D-rings and truck cargo rails, or they can be used with E-track, a metal track that has a series of slots that allow straps to clip at any place along the track. Packing blankets can be used to pad items that are not in cases to keep them looking good.

Truck load bars are also a common way to secure cargo for over-the-road commercial trucks. They come in two main styles – bars that simply clip into E-track, and bars with a ratcheting system to expand and wedge themselves between the truck walls. The ratcheting-style bars are not the best choice for cargo that’s on wheels, like road cases, because they rely on friction alone and can slip as the truck moves.

**MOVING & PROTECTING**

We do a lot of corporate gigs at venues with loading docks, so we prefer dock-height trucks, but for those who rarely or never encounter docks, then a truck with a lower deck might be the better choice because it places the truck’s center of gravity lower, making for a more stable ride. If you prefer ramps over lift gates (as we do), then a lower deck provides a more shallow ramp angle, making it easier to push heavy things into the truck. Lift gates are great in moving large, heavy items to the ground and back up, but they add some weight to a smaller box truck, lessening its overall carrying capacity.

Because we like a dock-height truck with a ramp, the ramp angle is pretty steep. A trick to get around this is to mount a 12-volt automotive winch in the box and use it to pull the heavy items up the ramp. Just don’t pull by the item’s handles or you might just pull them off. Instead, wrap a spanset or two around the item to distribute the force around the box, and hook the winch to the spanset.

Don’t forget to include truck and trailer maintenance on your equipment maintenance schedule. Regular lube and oil changes coupled with equipment and safety inspections help keep trucks in good shape and can also be useful in catching smaller problems before they turn into big ones.

To protect equipment and keep it looking good, consider investing in covers, cases and trunks. I like to think of them as an insurance policy that pays off bit by bit, every time we move gear. Sure, cases can cost quite a bit, sometimes even more than the items they carry, but every cost analysis shows they’re worth it in the long run.

Covers are normally used for loudspeakers, especially subwoofers because they’re often just too large to put in a road case. Many manufacturers offer covers for their products, and there are also several companies that provide quality padded covers and custom covers for just about anything you can think of.
pack dimension cases usually include stacking cups in the lid that allow a similarly sized case to ride securely atop another, with the wheels of the uppermost case prevented from movement in the recessed stacking cups of the case below.

Road trunk is the common term for larger, heavy-duty cases. They can be item-specific, like a feeder cable trunk, or more generic, loaded differently depending on what is needed at the gig. Some are outfitted with removable dividers that allow different compartmentalization options. (Nice trunks are sometimes referred to as “Cadillacs.”)

Cases may have a hinged lid or be of the “pullover” or “slipover” style, where most of the case, except for a small lower tray, is lifted off. This allows for removal, or use of the item while it stays in the lower rolling tray, and is popular for backline amplifiers, snakes on reels, and loudspeakers.

As the name implies, mic boxes are cases designed to secure and transport microphones, direct boxes and accessories. They commonly include foam inserts that provide specific protection and organization advantages. Work boxes offer drawers and storage areas to organize tools and supplies at shows. I keep separate work boxes for audio, lighting, and backline so the specific tools, parts and supplies needed for each area of production are present and easily accessible.

Cases for mixing consoles come in a variety of styles. Smaller mixers might be stored in briefcase-sized foam-lined satchels, while larger consoles often travel in cases with lift-off lids, with the console sitting the case bottom during use. A feature many mixing consoles offer is a doghouse, a compartment at the rear of the case that allows cables and snake fans to be pre-connected to the console and stored in the case when not in use.

**Riding the Rails**

Racks are specialized cases that house electronic components, held into place on rack rails. The standard rack rail dimension is 19 inches wide, and gear is designated by how many vertical spaces (or rack units) they use in a rack: single space (1RU) is 1.75 inches, 2RU is 3.5 inches, 3RU, is 5.25 inches, and so on. The equipment, shelves or drawers have “ears” that extend on either side of the front panel, and these allow the item to be bolted onto the rails that are normally tapped for a 10-32 thread bolt.

Racks may offer front rails only or an additional set of rear rails to facilitate supporting larger/heavier equipment like amplifiers. Shock mount racks suspend the rack inside an outer shell either by springs or surrounding the inner rack with foam. These offer more protection for fragile electronics.

Another rack variation is a mixer rack, combo rack or slant top rack that can house a rack-mount mixer on top, oriented in a comfortable operating position with space for additional rack equipment below. We use these styles a lot with our smaller mixers because they allow us to roll in and have everything needed all wired up in one unit – just pop off the lids and plug in the loudspeakers.

A more recent trend has manufacturers incorporating table legs into the removable lids of the racks, turning them into tables. In fact, I won’t buy a new mixer rack or workbox without a table option because it provides a handy place to set up computers and other items by the mixer without having to scrounge up at table at the venue.

Not every case and rack has wheels, so additional helpers like hand trucks and mover’s dollies (a.k.a., “skateboards”) are required on many gigs. Hand trucks must be magical as they have a habit of growing legs and walking off when we’re not looking, so we bring a bicycle lock to secure them to a large road case or lock them back in the truck at a gig.

Don’t forget to add cases, racks, hand trucks and dollies to your preventative maintenance schedule so they too stay in good working condition. A nifty trick for hand trucks that use pneumatic tires is to put sealant (a.k.a., “slime”) inside the tires to lessen the chances of air leaks.

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Senior contributing editor CRAIG LEERMAN is the owner of Tech Works, a production company based in Las Vegas.
RESOLUTION SOLUTIONS

Striving to reproduce what’s put into a sound system – nothing more, nothing less...

by Bruce Borgerson

Bruce Borgerson: What is high resolution?

JOHN MEYER: To discuss resolution, it’s much easier to start by looking at the video world. In projectors, we need a certain amount of light, measured in lumens, to hit the screen and cover it evenly. In audio, the equivalent is a system’s SPL level and even sonic coverage in the room.

But with projection, we also understand video resolution, which could be anything from VGA through 4K and beyond. There are also other ways of defining the image resolution, like bit depth and contrast ratio. Taking all of the specs together gives you a pretty good idea how well an image will be displayed for the viewer.

What is the equivalent of video resolution in audio?

Tools for audio system design and analysis have come a long way. We can now measure SPL and uniformity of coverage at different frequencies. However, unlike video, there is no commonly accepted methodology to holistically and neutrally report the quality of audio signals as they are reproduced for the listener, because in audio, you have to take into account the acoustic space, the audio source, and the behavior of the equipment.

Adding to the complexity is that resolution with sound is generally harder to define than visual. If you thought you had a 4K projector and you couldn’t read the credits at the end of the movie, you’d know you have a problem.

Are there any negative consequences of not having a common method to look at sound as a cohesive whole?

The problem is that all three elements, audio source, acoustic space, and equipment behavior, greatly impact the experience for the listener, and sometimes this gets lost when all people see is specs like SPL and coverage. Sound systems that have plenty of SPL and are implemented to specs could end up being completely unintelligible. It’s not the fault of the sound system designer or the speaker manufacturer. When you have loudspeakers that lack clarity or a reverberant room that lacks absorption, you cannot achieve the desired resolution regardless of how you tweak the sound system. And because there is no standard, there is no accountability for when people are disappointed by the result.

When the Montreux Jazz Festival was opening a new venue many years ago, we told them that they really needed to fly some acoustical material to bring the reverberation down from three seconds to around one second. The other option circulated was to use directional loudspeakers without acoustical treatment. We disagreed with that last option, as there were more issues involved than simply aiming the PA.

We went back and forth on it, and finally I said, “Look, what about the band on the stage? They are not going to be directional, and they will excite the room so much that it won’t matter what loudspeakers you have because it will be so loud in that space.” Well, they said they hadn’t thought of that.

So resolution in this case is not just the resolution of the input signal source?

High-resolution audio is a holistic view of looking at all the elements that impact the listener experience, but it still starts with the input signal, bit depth for dynamic range, and sampling frequency for bandwidth. Most new consoles now output 24 bit/96 kHz signals. We have the transmission capability and the storage capability to handle high resolution. But we need to look at sound reinforcement as a whole, from signal to loudspeakers to room acoustics.

After signal, the next step is to look at the qualities of the reinforcement equipment?

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image quality between a VGA projector and a 4K projector at the same level and brightness, the same concept should apply to loudspeakers as well. SPL and coverage specs only give you part of the picture. One key parameter in looking at loudspeaker equipment is what we call linearity – how accurately a system translates an audio signal, particularly when reproducing it at high levels. Linearity is particularly important in the parts of the system that remain in the analog domain. The industry has lost sight of that in recent years.

Back in the 1940s and 50s, there was a lot of emphasis on developing linear sound reproduction, because it was a challenge to make the different sounds of instruments come out distinct and intact, rather than being muddied by harmonics. In a linear system, if you put in a piano and a trumpet, you get the two back out, without anything added or taken away. If you want to add an effect, you can use plug-ins at the console level upstream. The linear system reproduces what you put in, nothing more, nothing less.

If you put five tones into a subwoofer, a linear system will reproduce only those five tones. In a non-linear system, the subwoofer will alter the original ones and add other tones, which you may or may not find pleasing. Non-linear loudspeakers are difficult to measure and predict, and this makes it harder for the user to attain a specific end result.

With all the new technology at hand, why do we still fall short?

Sometimes too much new technology can make a problem harder to solve. In a complex system, if somebody drops the ball upstream you’re stuck with the problem. We experienced that recently when our loudspeakers were installed in a large system with all-digital signal interfacing and networked distribution. We had severe distortion problems and discovered that the digital mixing console was overloading the network. And the people that put in the network had already left the site. We ended up fixing it ourselves, because nobody was checking to see that the whole system was working to that point. It made us realize that it doesn’t make any sense to put in high-quality loudspeakers if the digital system in front of it is overloading. On the other hand, why put in lousy loudspeakers at the end of a very high quality digital system?

We need to be systems-oriented. The whole thing has to work as a unit and be tested as a whole system. You can buy a standard test disk for your Blu-ray player, but in high-end sound reinforcement we don’t have an equivalent for that. It’s not part of our thinking.

What changes are needed if we want to achieve greater audio resolution?

The first thing is to set clear goals for the source signal, loudspeaker system response, and acoustic behavior of the space, and look at them in a systemic way. We also need to set standards and testing procedures that every manufacturer can apply and every user can understand.

Bruce Borgerson has been a freelance journalist and audio industry communications consultant since the 1980s. Visit ProSoundWeb to read the complete interview.
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A light summer rain was falling as I entered the grounds of the Buckle-Up Festival, a three-day, six-stage country music event in mid July held on the banks of the Ohio River in the Sawyer Point region of downtown Cincinnati. It’s a beautiful setting for this first-ever festival, and also served as the site of the Bunbury Music Festival the week prior that featured a variety of rock performances.

My first goal was locating Grant Cambridge, the managing director of Event Enterprises, which serves as the production company for both festivals. Eventually we both arrived behind the Main Stage front of house position, covered by a tent intended to provide shade from the summer sun but now serving a vital role in warding off the steady drizzle.

I’d not met Grant in person and was a bit surprised by the youthful looks of the fellow hiking up to me with a full backpack and ubiquitous comm radio, but he exhibited the demeanor of an old pro despite serving as the tech pivot point in support of nearly 80 live performances over three days, including headliners such as Willie Nelson, Eli Young Band, Alabama, Emmylou Harris, and The Band Perry.

Step By Step
Upon graduating from Ohio University in Athens in 2003 with degrees in audio production and music, Cambridge started working free-lance audio gigs in the Cincinnati area, where he’d grown up. The work began to come
more steadily, so he bought out a small local recording studio, essentially for the live gear that included a small PA and a couple of mixers.

A steady affiliation with the Mid-Point Music Festival, an indie music event held annually in late September, led to strong ties with festival organizer Bill Donabedian, who’s gone on to found Bunbury and now Buckle Up festivals. “It goes to show the value of business relationships,” Cambridge notes. “And even though Bill eventually sold the MidPoint festival, we’re still a vendor for them as well.”

He continued with a “day job” to supplement his income through 2009, eventually reaching a point where he could see making a go of working sound full time. It came down to having acquired consistent repeatable business combined with enough new prospects to make it a realistic pursuit. He marks January 1, 2010 as the official start of his full-time venture, christening it Event Enterprises.

Next came the process of building the business while still staying busy enough (and liquid enough financially) to pay the bills. Acquiring additional inventory was essential to the plan of becoming a full-service audio (as well as backline and lighting) provider and rental house, but he resisted the urge to go on a gear splurge, taking a more calculated approach.

“Believe it or not, the recession actually kind of helped our business,” he says. “Some folks were getting in over their heads on gear, so I kept my eyes open for opportunities.” For example, a church that had overextended itself led to the “right price” for a barely used compact main system comprised of NEXO GEO S8 line arrays, CD12 cardioid subwoofers and NX Series processors. And he smiles while recalling driving a box truck roundtrip to Nashville the day after Thanksgiving to pick up NEXO Alpha E full-range boxes as well as some processors.

By 2012, Event Enterprises occupied a small shop and began adding staff, as well as working with a host of free-lancers, to keep up with a growing client base.

**New Directions**

The Cincinnati market for production is showing growth, Cambridge notes, with a steady increase in festivals and street fairs offering live entertainment that requires production. The Major League Baseball All-Star game (and all of the festivities that surround it) is coming to the city next year, and there’s also a strong arts community with world-class symphony, ballet and opera, several theaters that host a steady schedule of live performances, and a new casino in the city along with several others within an hour of downtown.

Despite the optimistic outlook, Cambridge stresses careful planning. “As a business owner, I’m looking at how to make more from our existing inventory, and every decision to add inventory must be carefully considered. What to spend, how much to spend, what it’s going to mean over the long-term – these factors have to be analyzed,” he says, adding that his father, an engineer with a great head for numbers, has been exceptional in mentoring him on the power of the spreadsheet to carefully track costs such as depreciation and taxes.

It’s perhaps a bit of a different future than he imagined when coming out of college with two degrees but just a single (required) economics class under his belt. Yet the delicate dance between plying the audio trade and being informed on matters of commerce is essential for those seeking to also run the business.

“We’re in a simple supply and demand industry,” he states. “That’s what drives the decisions. The goal is to have every band and every engineer show up for every gig, look over what you’re supplying, and say “Great. We’ll sound good today.”

NEXO and Yamaha Commercial Audio have proven to fit very well into this vision, he adds. “They’re extremely supportive and very available, and we really appreciate it. You can easily reach
informed people on the phone, even on their cell phones at odd hours, when you have a question or a need. They’re really strong on training, and have even helped us in reconfiguring our amp/processing racks for different systems and scenarios.”

He also shares an anecdote about a time when a couple of processors malfunctioned the day before a gig, with the company shipping replacements overnight with no questions asked except a follow-up inquiring if the situation was back on track. “There are some other factors in play with Yamaha and NEXO, such as rider-friendliness, and the gear is very affordable in terms of what you’re getting for your money,” he says. “This type of 1-2-3 punch is what has gotten and held our attention.”

Walking The Grounds
Cambridge points to Yamaha consoles as fitting very well within his company’s worldview. “It’s exactly what you need. Solid, reliable, a load of functionality in a package that most folks are familiar with, and of course, great sound quality. Also efficient to pack, and kind to the budget.”

Event Enterprises has invested in Yamaha M7 and LS9 digital consoles, and continues to see a great return of them, and is now looking to the future with the CL and the new QL Series. In fact, every stage at both Buckle-Up and Bunbury had a Yamaha board for house and monitor mixing, accompanied by Rio stage boxes.

Cambridge notes, a bit wistfully, that he hasn’t had a chance to mix on one of the new QL consoles yet due to being occupied with management duties. “But the feedback from all of the engineers who’ve used the QL consoles here is that they’re digging them,” he adds. “The similarities and consistency with the other Yamaha consoles is great, while having the new effects packages and other advantages means more tools in a friendly, familiar package.”

Despite the misting rain that continued to fall (it thankfully ceased toward evening), we visited each stage to experience the various systems and take in several performances. The Amphitheatre Stage, sunken into a “concrete bowl,” was outfitted with the
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aforementioned Alpha E loudspeakers, and while they’re an “older” technology, they can still get it done.

Moving along, the Lawn Stage was flanked by the (also aforementioned) GEO S8 arrays, groundstacked on the wings next to the CD12 cardioid subs, while the performances on the humble Acoustic Stage were appropriately amplified with a couple of EAW KF Series loudspeakers. The River Stage, literally on the north bank of the Ohio River, presented another concrete bowl setting, with JBL VerTec arrays subcontracted from a local vendor flying left and right.

We next stopped by front of house at the Bud Light Stage, chatting with Nicholas Radina, who was providing mixes on a Yamaha QL5, feeding a PA with 10 NEXO GEO S12 modules per side, flown, with eight NEXO RS18 “Ray Gun” subwoofers placed equidistantly on the deck in front of the stage. All loudspeakers were driven by NXAMP (4 x 4) DSP/amplifiers. Another QL5 was posted stage right for monitors, with both consoles networked with Rio stage boxes.

“The new Yamaha QL5 is a joy to use,” Radina tells me. “A marked improvement in overall sound quality, and I also enjoy the premium effects and additional rack spaces and wonderful routing over the Dante network. Custom fader layers make visiting band engineers feel right at home. Well done, Yamaha.”

True Detective
In February of this year, Cambridge began talking with John Mills, VP of Morris Light and Sound in Nashville, about the NEXO STM system in general, and more specifically, about the possibility of supplying an STM rig to serve the Main Stage at both Buckle Up and Bunbury. Morris had added a large-scale STM system last year and deployed it for Kenny Chesney’s North American tour (August 2013 LSI).

“I wanted to do some ‘detective work’ on the system in a real working environment, see it up close — how it packs and comes off of a truck, how it goes together — all of the things you don’t get to see if you just go to a show,” Cambridge explains. “Morris is a known entity and I was quite confident the system would sound great, so I was also looking to network, define a partner to work with from a gear standpoint as well as learn from their considerable expertise.”

Those conversations came to fruition, with the Nashville company delivering an STM rig, S118 subs, and NXAMPs for both festivals. Specifically, main left and right arrays were made up of a dozen M46 main modules mated with B112 bass modules. The system concept enables building line arrays that scale up or down depending on the application, and in addition to the main and bass modules, the S118 subbass module, sharing the same footprint as the other two, can also be utilized in arrays.
Mills was also on hand to serve as system engineer, demonstrating assembly, answering questions, and performing final tuning. “I saw how easily John and the crew were able to get it dialed in,” Cambridge says. “That was impressive, as well as how far it could throw and how it sounds.”

Yamaha CL5 consoles were posted at front of house and monitors, networked with Rio stage boxes. Other consoles could be easily swapped in when requested by certain artists and engineers, and there was also an analog snake as a back-up.

“Several guest engineers have tracked me down just to tell me how great it all sounds,” he says. “With the STM rig, I really like the modularity, the way the individual modules can be scaled for any situation – big, small, unusual – whatever the gig calls for.

“There are a lot of efficiencies there. It’s very organized and packaged well. For a company like mine, this could be quite valuable, where every day presents a different type of gig. The flexibility is very attractive to a business of our type.”

I received the final word from John Mills: “The team at Event Enterprises are an amazing group. Grant’s a pleasure to work with, and as ever-changing as the details of a festival go, he’s always on top of finding an answer for us. He and his team are all very professional and work extremely hard.”

KEITH CLARK is editor in chief of Live Sound International and ProSoundWeb.
REALIZING EFFICIENCIES

A range of useful tips and techniques for monitor engineers.

by Mark Frink

WITH THE ADVENT of digital consoles, the monitor engineer’s ability to reset the entire desk with the push of a button is a powerful tool. However, there are many situations where the console must be adjusted from scratch or “zero initial data,” whether it’s a festival, support act or a one-off. Doing a little homework in advance can save valuable time.

Look at everything that must be taken care of on the console before the band is on stage: inputs and outputs named and patched, gain, high-pass filters and phantom power set, effects named, tweaked and patched, GEQs patched, matrix named and patched, user-defined or macro keys programmed, monitor cue on master fader, then store all of that as a starting scene.

By studying the artist’s input list and plot, and making some educated guesses (given the band’s musical genre), you can set up the inputs. If active instruments — like keyboards and guitars with on-board electronics — are plugged into passive DI’s, while active DI’s are used to amplify passive instruments that need gain, channel gain settings will be more predictable. Likewise check the pads on DI’s (and condenser microphones) before they go on stage.

If there’s a place at the end of the input list, or even somewhere in the middle, it never hurts to add one or two “oh by the way” (OBTW) channels into the console file, as invariably there’s something not accounted for in the band’s paperwork that’s been added along the way. If the piece of paper taped to the splitter and the file in the console don’t match, it’s tempting to fix it with a “soft patch” but I recommend “one-to-one.”

MEANINGFUL LABELS

Channel EQ can be quickly set with good starting positions for how you like to work using the console’s EQ library, which is a place to store EQ presets. Knowing the EQ filters that are the usual starting point for standard mics and typical applications, you can save the presets with meaningful names.

For example, you may cut some low mids and add some 1 kHz “click” on the kick drum using a (Shure) Beta 52, so by making a “Kick-B52” EQ preset, you can set that channel by just calling the preset to your selected channel with a touch of the screen or a click of the mouse. This is quicker than tweaking gain, frequency and Q for several filters. There may still be a need to tweak, but you get there faster.

Some consoles only allow a low-pass filter by stealing the highest EQ filter, turning its Q all the way down, then turning it on with the gain and adjusting its frequency. These can also be incorporated into presets, saving those encoder strokes as well.

Similarly, GEQ libraries can be used for specific combinations of vocal mics and floor monitors that are used frequently. After investing the time it takes to carefully adjust one wedge with a mic, that setting can be quickly copied into the graphic equalizers for a number of mixes with the same mic/wedge combination. Self-powered monitors are typically very consistent, allowing settings created for them to work well on many other suppliers’ wedges. Giving them meaningful names like “MJF-212wSM58” allows them to be saved for the next time that combination is deployed.

EQ libraries can also be used to adjust in-ear monitor equalization with individual musicians, allowing you both to explore making minor adjustments to their overall mix EQ to help their IEMs better match their hearing. By giving them meaningful labels, like “DaveIEM0915,” you can archive a record of daily IEM mix EQ settings in what can be an iterative process that gradually gets an IEM mix sounding better to that performer. You may even discover subtle changes to their hearing from the beginning of the set to the end, or from Friday night to Sunday night.

Obviously the same applies to effects libraries. I’ve heard very few reverbs that couldn’t benefit from a little parameter tweaking, including the EQ on their return channels. You might go ahead and tweak and load up the console with an entire
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assortment of application-specific effects from snare to fiddle to acoustic, so that when you want them, they're ready to go, as long as you have channels to return them.

At a minimum there should be a generic vocal and instrument reverb in your file, patched from a pair of aux sends and ready to go into a mix. Then replacing them with exactly what's needed can be quickly chosen from your custom effects library. When you save your entire console, all of these libraries are stored and carried on a USB key to the next gig.

Some consoles also allow making a preset out of an entire auxiliary send. One day percussionist Daniel de los Reyes told me his mix was a little off. I asked him when was the last show it sounded good to him, and he replied “Philly.” I was able to go back to my Philadelphia file, make a preset of his mix and paste it into the current file. Bueno, aquí estamos en Philly.

FOR DRUMMERS

Drum subs are intended to provide drummers with the low end on stage that’s missing from their IEMs or wedges. It’s often not enough to simply place the sub near, or even on the drum riser.

For maximum effect, it must couple with the drummer’s backbone, not just his ears, so placing it on a road case to get it up in the air a couple feet can make a big difference. If it’s double-18 sub, even better – point one at his rear end and the other at his shoulder blades. And watch him drool.

For a fraction of the price of a hard-wired IEM, a miniature mixer can be purchased for drummers that allows them to not only take a feed from the monitor desk, but also hear the click and tracks during the show, just like in rehearsal. And it doesn't need batteries.

TAKE THE CALL

Many shows and sets begin with a call on the walkie-talkie to alert the band’s crew. Most walkies have a 3.5-mm audio jack, and many consoles have a 2-track analog input that’s rarely used. (On an Avid Profile, it’s an RCA.) A short cable allows you to monitor the radio call to start the show while you have your ears in, and perhaps even put it in the band’s ears.

USE THE MATRIX

Every console has a matrix that goes largely unused in most stage monitor applications. Since any mix can be sent to a matrix, one use with in-ear monitors is to patch it to a spare IEM system, which allows any mix on the console to be quickly turned on at unity into a spare transmitter on a new frequency. It’s quicker and easier than re-patching XLRs at the back of a rack or desk. But for the downstage center “money” mix? Just put it into two transmitters with a pair of “Y” cables.

Another powerful use of the matrix is to add in talkback microphones to the cue mix so the engineer can always hear them. Instead of routing the cue send directly to its IEM transmitter, return it to two inputs to the matrix and then add in the talkback or “shout” mics from FOH, the drummer, the guitar tech, etc. When you fader-down the cue, you’ll still hear anyone talking. To accomplish this on many consoles, patch out of the desk and back in again with a pair of short XLR cables.

Inputs for additional shout mics may be in short supply in the console’s stage rack, especially if you start eating up channels for double-patched inputs (instruments played by more than one musician) and FX returns. A compact mixer, like a Mackie 1402, can provide some economy by mixing all of the shout mics down to one or two inputs.

If the console’s stage rack is also full, there’s often a dedicated talkback input where they can all be returned, but now the engineer’s talkback mic might have to go in the mixer as well. This also might be where the walkie’s audio input lands.

Leaving shout mics open clouds both the cue mix and the tech (roadie) mix with ambient stage wash. It’s important to manage these mics, which can be done with a switched mic or a “proximity gate.” A momentary foot switch that can be operated even while holding a guitar is even better for musicians and roadies.

The Hot Shot from Radial Engineering takes a dynamic mix and sends it out a second XLR when its momentary footswitch is pressed. These can be used with hard-wired dynamic mics at singing positions or simply by backline techs off stage.

For those times you need to be able to use a wireless mic for talkback, Radial’s Remote Relay is an A-B switch that can be placed on the output of a wireless receiver, similarly controlled from anywhere on stage by one or more momentary footswitches, connected by an XLR.

As with the Hot Shot, pressing the momentary switch removes it from its vocal input channel and puts it into a talkback channel. By panning that talkback channel to one side, it lets the performer know that it’s no longer in the PA. (Thanks Peter Janis!) =

MARK FRINK is an independent author, editor, consultant and engineer who has mixed monitors for numerous top artists. He can be reached at LiveSound@MarkFrink.com.
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KEEPING THE GIG
Opportunities to excel while earning artist trust.

by Danny Abelson

WE'RE CONTINUING OUR discussions with veteran independent touring engineer Dave Natale, this time focusing on strategies for keeping your front of house gig. Dave’s worked in the industry for decades and has demonstrated the ability to survive in very high profile settings. With that in mind, here are a few thoughts from Dave to consider for surviving the political minefield.

UNDERSTAND THE SITUATION
It may seem blatantly obvious, but many of us have encountered mixes where we flat-out doubt if the person mixing actually knows what their doing. While there may be the occasional politically-connected “imposter,” successful house engineers appreciate that getting the majority of decision-makers on their side is key to keeping your job.

According to Dave: “It’s totally subjective. It doesn’t matter if you think you can mix, it only matters how most of the band and management feel about your mixes. Maybe I can’t mix, but as long as just one more person on the team thinks that I can, as opposed to the ones that don’t, I’ll still have a job. Your client’s decision to keep you in that chair could be as simple as that.

“Here’s a totally different example. Readers who were fortunate enough to hear a Roger Waters show recently will probably agree with me that Trip Khalaf did a wonderful job at front of house. If you ask my opinion, I’ll tell you it sounded absolutely fantastic. In this case, maybe 80 percent of those in the know think he can mix, and perhaps 20 percent feel otherwise, but there’s no doubting he’s in his rightful place at front of house.

LEARN THE MUSIC
“Before a tour, your homework is to learn the music. You simply can’t mix well if you don’t know the cues, and that means knowing the music. I prefer not to learn a band’s music from the records, but rather in rehearsals. You can’t rely on the records because the band isn’t going to play it like that, so it isn’t going to sound like that. To me, this is self-explanatory.

“If your artist has a decent budget and can afford to rehearse for a week to 10 days, that’s a great opportunity to learn the music. For me, that’s plenty of time. There is no substitute for preparation. Initially at rehearsals, I have a legal pad and take a lot of notes, a page for each song. I’m writing like crazy, noting details, including solos, backing vocals, which keyboards are played on a particular song, and so on.

“One of the reasons I use a big pair of speakers in an isolated room during band rehearsals is that they really help me learn the music, particularly the sounds, and what the final product is supposed to sound like. It gives me a reference and some confidence going into my first show that I know the material and have a reasonable foundation to start with. This preparation is essential to making a strong first impression, and a good first impression is essential to retaining employment.”

KNOW THE PA
“You must understand what you can and can’t do with your PA. What are the limits, quirks, and dead spots? It’s really in your favor to know this. Your client
may ask you about something, and if you don't know the rig inside and out, you may not be able to answer the question.

**BE RESPONSIVE**

“When an artist asks you to do something, do it, and then follow up with them at an appropriate time to make sure they know you did it and are happy. You want it to be blatantly obvious that you’re listening, paying attention, understand English, have taken action, and understand that the artist is in charge.

“You’ll never get fired for fastidious attention to detail. The best way is to think of things that might become a problem, anticipate them, and solve them or have a solution ready and in your hand before they think of it or ask for it. Anticipate what the artist wants and get it for them. Doing this quickly is often the best way I know to keep a gig.

**STAY POSITIVE**

“A lot of keeping the gig is staying positive. Your personality has so much to do with it. If you can stay positive, and tell people ‘yeah this is going to sound great,’ they’re going to believe it’s going to sound better than the previous person. You can’t be blowing your own horn, because people don’t like that, but keeping a positive attitude helps a lot.

**WHO’S NAME IS ON THE TICKET?**

“Remind yourself that your role is just making it louder – you’re not the famous rock star. Always give credit to the band – it’s never about the engineers, it’s always about the artist. This is key to self-preservation.

“When people tell me how great a show sounded, I always reply, ‘I’ll be glad to tell the band when I see them backstage.’ Usually they miss the subtlety of that statement and say to me, ‘No Dave, it’s you and the great job you’re doing.’ I quickly remind them ‘No, it isn’t. Did the show sound like them? Yes? Well it’s because it is them.’ “It’s never about me. This is particularly true if you mix in the very Spartan manner I do, with little or no effects. Anything you hear really is the band, not something I’m doing at the console.

**Remind yourself that your role is just making it louder – you’re not the famous rock star. — Dave Natale**

**YOU’RE IN CHARGE**

“Remember that as the house mixer, you’re head of the sound department. Take responsibility for the mistakes that you and your crew make. Never, ever put the blame on one of your crew. They’re professionals, so they usually know when they’ve made an error.

“The best thing you can do is go straight to the dressing room and cop to the mistake. Even if they (the artist) don’t call you back there, by taking the initiative and admitting a mistake, you send a message they weren’t expecting.

“Regrettably, too many try to cover their asses or backpedal to protect their jobs. So when the artist encounters someone actually admitting a fault or mistake, it puts things in an entirely different and positive light, compared to many of the people they interact with. Typically artists aren’t used to those around them taking responsibility for their own actions. But when you voluntarily admit your mistakes, any concerns the artist might have about you usually just goes away immediately.

**DANNY ABELSON enjoys writing on the human elements that contribute to a great sounding show.**
WHEN EVERCLEAR FRONT of house engineer Derek Steinman talks about the gear he’s using for the Summerland Tour 2014, he insists that it’s difficult to get him to change things. Yet having said that, for this year’s iteration of the tour – headlined by Everclear and featuring performances by Eve 6, Soul Asylum and Spacehog – he’s carrying a number of different elements, including two Allen & Heath iLive consoles for FOH and monitors and various newly chosen microphones.

Steinman first got involved with Everclear when lead singer and guitarist Art Alexakis asked him to come out on the road for “one more tour” in the early 2000s. “That was 13 years ago,” Steinman says, laughing. “They’ve kept me busy ever since, and every year we do dates all over the world.”

Following the release of Invisible Stars in 2012, the band has headlined the Summerland Tour (during the summer months, naturally) with a variety of acts, offering a 90s nostalgia take. Typically Steinman wears multiple hats, sometimes also serving as production and/or tour manager, and this time out running FOH for Everclear as well as Eve 6, and occasionally, Spacehog.

MINIMUM & MAXIMUM
The move to the Allen & Heath iLive-T112 console was prompted by previous monitor engineer Tib Csbai, who used an iLive-T80 on monitors for Summerland 2013. “We were really happy with it,” Steinman says, “and this year after talking to Allen & Heath we continued to use it for monitors and moved to the T112 for front of house.”

A primary reason for the choice is that each show’s changeovers have to take place extremely quickly. “We’re vying for an older audience,” he notes, “whose tolerance for staying at venues for hours and hours isn’t high. Our show is three hours from the first note to the last note, and we have to keep changeovers to no more than 15 minutes to give audiences the maximum amount of music in a minimum amount of time.”

The consoles are particularly well suited to make that happen fluently every night, he adds. “We have W1 snakes connected to each band, everyone’s wired up, so we just use the quick connect and jump it over, switch the scene and we’re up and running.”

The tour marks monitor engineer John Riley’s first time out with Summerland. He’s mixing all four bands, utilizing the iLive-T80 to help keep things moving along. “Using the iPad app with the T80 to ring out monitors saves a ton of time, and although I didn’t have a lot of time on the board before the tour, I was able to build every band’s files on the offline editor, so from day one everything went smoothly.”

ONTOUR

SUMMERLAND 2014
Modifying the sound approach for a successful touring format.

by Kevin Young
Riley says, who delivers his mixes to in-ear monitors for Everclear and Eve 6, and wedges for Soul Asylum and Spacehog.

Steinman built the FOH files for all four bands with an eye to keeping the workflow as “analog” as possible. “I’ve configured the knobs on the upper deck so the compressor is always the compres- sor, EQ is always the EQ. Everything is the same – you see all the faders. I decided not to use layers on the recommendation of Dana Munroe (Allen & Heath rep in Oregon), who helped build the files, with the thought that sometimes a house engi- neer might have to mix the first band.”

With dedicated rotary controls, switches and meters, the console “drives very analog,” Steinman adds. “It’s clearly labeled, you push faders, grab your EQ, your comp, your gates, your preamp, whatever – that never changes – so it’s simple for anyone that doesn’t spend as many hours on digital consoles as some of us do.”

Both engineers point to the iLive’s Opto compressor as a “go-to” tool, which Riley uses on Freddy Herrera’s bass and Steinman applies on Alexakis’ vocals. “It has an (Universal Audio) LA2A setting in it that compresses really hard, but doesn’t get all ‘blankety.’” Steinman says. “I also use the onboard delays and chorus for Eve 6 quite a bit and they’re excellent.”

**STAGE HAPPENINGS**

The compact footprint of both consoles is also important given that Summerland hit roughly 40 venues of various descriptions this year. “We’re doing everything from sheds to places like House of Blues in Los Angeles, and the T112 is small enough that (regardless of space constraints), I can always be in a good mix position,” Steinman says.

“I’m also able to set up in little nooks in places if necessary,” Riley notes. “We’ve had situations, like Fremont Street in Las Vegas, where the monitor console is slightly out of the way, but with the iPad app I could see and oper- ate the console easily anyway.”

Steinman likes to keep the amount of production that the tour is haul- ing to a minimum, using house-pro- vided mains as well as house-provided wedges for Soul Asylum and Spacehog. Beyond consoles, the tour is car- rying Sennheiser ew IEM G3 wireless systems – six for Everclear and four for Eve 6 – and dedicated micro- phone packages for both bands. The Sennheiser rigs provide a cost benefit and consistency, he says, and specifying them stems from familiarity.

“We’ve also had a lot of luck in terms of not having to re-tune them often,” Riley adds. “I think this is the first time I’ve played in Las Vegas and not had to re-tune my ears.”

Everclear and Eve 6 wear a variety of Ultimate Ears IEMs. “If I could get every- one on ‘ears’ I would, but they want what they’re used to and I get that,” Steinman observes, also noting that the band’s amp cabinets aren’t isolated or turned around to face upstage. Still, the goal is to keep stage volume as low as possible.

Spacehog and Soul Asylum typically rely on a house-provided Shure package that Steinman describes as a “standard fes- tival mic pack,” while Eve 6 carries its own mics, which, with the exception of Tele- funken M80s for vocals, are also mainly from Shure. With Everclear, however, he’s testing out a variety of mics he’s previously used only for a brief time or not at all.

It’s his second year with European manufactured Lewitt microphones for drummer Sean Winchester’s kit. For kick in and out, there are two Lewitt
“Simply Awesome.”

“Lectrosonics gear is built like a Mack truck. We travel the world and in all the time we’ve been using this gear, I’ve never, ever had any issues. Lectrosonics’ durability is, in my opinion, unsurpassed.”
- Lorenzo Banda, Monitor Engineer, Foreigner

Pictured: Kelly Hansen, Foreigner lead vocalist with the HH transmitter.

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DTP 640 REXs, a dual-element mic that incorporates both a condenser and dynamic element. “It’s kick in and out on one head, and then I go inside the drum maybe 2 or 3 inches with the other one and angle it toward the beater,” he notes.

Riley elaborates: “The 640 is so phase-aligned that if you flip the polarity on one of them, it just disappears. Typically I have EQ patterns that I use, but with these mics I only take out a bit of 1K on the kick out, and then do little bits of rolling off here and there.”

Winchester also likes a KickPort system on the bass drum to clarify its signature and limit resonance and ring. It works well, Steinman says, but has an impact on mic placement. “I have to place mics a little further inside the drum than usual, because if you place one right at the hole, it sounds like you’re in a wind tunnel.”

Snare top and bottom are handled with a pair of Lewitt MTP 440 DM dynamic cardioids (the company’s version of a Shure SM57, he notes), while on hi-hat and overheads, it’s Lewitt LCT 340 condensers and, on toms, Lewitt DTP 340 TT dynamics.

**EXPANDING THE PALETTE**

“We also use the DTP 340 TT on Freddy’s Ashdown 4 x 10 bass cabinet, placed mid cone, almost like a guitar because I like the growl,” Steinman says. Additionally, the bass runs through a Countryman DI for backup and tone. The mic is high-passed and then mixed in, with compression, with the DI output: “I get plenty of smooth sounding low end from the Countryman, and a bit of attitude from the Lewitt.”

For Josh Crawley’s main keyboard, a Nord C2D organ, Steinman goes stereo with two Radial JDIs. “We had a Leslie cabinet for a while, but the direct emulation coming out of the Nord is pretty damn good and you don’t get all the stage noise with the Leslie.” On lead guitarist Dave French’s and Alexakis’ guitar rigs, Steinman is trying out Blue encore 100i dynamics. “We still have our Shure Beta 57As with us, but we’re happy with the Blue mics so far – they’re smooth sounding, the top end is detailed, but I’m still debating which mics are better for the band’s sound.”

There’s also been a mic switch for Alexakis’ lead vocals, from a Sennheiser e945 dynamic to an Earthworks WL40V hypercardioid condenser capsule on a
Line 6 XD-V75 digital wireless microphone system. Initially, both he and Riley were hesitant to make the change, but the wireless system provides emulations of various mics, including the e945. “It sounds good, has great off-axis rejection, and because it’s at 2.4 GHz, we never have dropout problems,” Steinman says.

Meanwhile, backing vocals have also changed, with Blue 100i dynamics in place of Shure SM58s, although he notes that the verdict is still out on which mics he’ll continue to use long term for this application. When it comes to the consoles, however, both he and Riley are definitive. “The Midas PRO 2 and Avid SC48 were the boards I’d specify for Everclear,” Steinman says, “and they’ve now been joined by the Allen & Heath iLive on our rider. Having a console other people can work on easily and sounds comparable adds value to me.

“The iLive does what it needs to without being complicated. Everything’s on the surface all the time, so it’s a quicker process than jumping back and forth between comp, gate and EQ,” he concludes. “I’m looking up as I’m mixing, which I find beneficial. It reminds me of analog mixing, and I feel like I’m more there mentally when I have my head up and I’m watching the band.”

Based in Toronto, KEVIN YOUNG is a freelance music and tech writer, professional musician and composer.
Two series, one family. Both represent the evolved sound quality and innovative functionality of today’s digital age while embodying the rich heritage behind the Yamaha name. The CL Series is comprised of 3 models with unique built-in features including Rupert Neve Designs Portico 5033/5043 EQ and compressor, Yamaha’s VCM analog circuitry modeling technology and Centralogic™ operation. The 2 models of the QL Series take the best of CL’s advanced features and combine a few additions such as a built-in auto mixer from Dan Dugan Sound Design to provide a simplified user-friendly all-in-one mixing experience. Connected by the Dante audio network, the CL and QL Series work seamlessly together to provide complete solutions for a variety of sound applications.
**THE SOUND ROCKET**

A look at the AKG D202E dual-diaphragm microphone.

by Craig Leerman

The majority of old microphones in my collection are for display, largely because they’re, well, old. Some require repair, others need an unusual connector or cable assembly to function. But the AKG D202E is different. It’s not in my working mic locker but I’ve employed it several times on stage, and may do so again, even though it’s the oldest AKG mic in my set.

The D202E is a cardioid dynamic handheld model with a twist – it employs two separate diaphragms, one for low frequencies and the other for high frequencies. The two signals are mixed together with a crossover network, with a full-range output signal sent down a standard XLR cable. The primary benefit that there’s little to no proximity effect, while also providing an almost uniform directivity over the majority of its frequency range.

The diaphragms are shock mounted in the head of the unit. The HF portion is at the front and the LF is at the rear, surrounded by large ports, with sound traveling up an acoustic tube through the body of the mic to reach the element.

**OPPORTUNITY REALIZED**

The entity that would become AKG Acoustics was founded in 1945 when two friends, Dr. Rudolf Görke and Ernst Pless, realized that there was a business opportunity in providing equipment to a growing motion picture industry in the aftermath of World War II. Deliveries to customers in those early days were often made via bicycle or wheelbarrow, and the first transaction was bartered, with the partners receiving butter, fresh meat and cigarettes as payment.

They formed AKG (short for Akustische und Kino-Geräte Gesellschaft, which means Acoustic and Cinema Equipment in English) in 1947, and began the manufacture of light meters, automobile horns, telephone handsets, and of course, microphones. The early DYN Series of mics were popular with broadcasters as well as live users, and in 1949, the company introduced its first headphones, the model K120.

The D12, which debuted in 1953, was one of the world’s first high-quality cardioid condenser mics. It also led to the company modifying its logo – three overlapping omnidirectional polar patterns were replaced by three overlapping cardioid patterns, a design that’s still in use today.

Another innovation came in 1970 with the BX20, a portable studio reverb unit utilizing helix springs with long time delays that closely mimicked the reverberation characteristics of large concert halls. Four years later the company was awarded its 1000th patent. The company went public in 1984, introduced its first wireless microphone system in 1991, and by 1993, occupied a large, efficient production facility in the Vienna suburbs. Also in 1993, Harman International purchased a majority interest in AKG, and one year later, acquired the rest of the stock. The company thrives to this day.

**DISTINCT STYLE**

AKG introduced the first 2-way cardioid microphones in 1966, including the model D202E. Company advertising at the time touted it as “especially suited for recording work in TV, motion picture and broadcasting studios” and “recommended for picking up percussion instruments in front of loudspeakers.” The distinctive rear ports, resembling the fins seen on space rockets, led to the nickname of the “Sound Rocket.” (And that era was the “Space Age.”)

I haven’t tried any of those applications, but as noted, have utilized the...
The AKG D202E a few times, particularly for large symphony orchestras. With its flat, full-range pickup (even off-axis), it works quite well with string and horn sections, and does a nice job when applied for a harp. The lack of proximity effect is great for spoken word, but probably not optimum for a singing vocalist, especially one who works the mic close for extra low-frequency emphasis. Another potential problem is the hand covering the rear ports, which could compromise the ability to pick up anything below 500 Hz or so.

The sintered bronze grill is rugged and acts as its own windscreen/pop filter. The body is made of a non-reflective black high-impact ABS polymer. The specification sheet notes, “Outdoor use, even in cold environments, is facilitated by the housing with low thermal conductivity.”

A 3-position bass roll-off switch offering 0, -7 and -20 dB positions at 50 Hz is located at the rear, to help reduce rumble and handling noise. Earlier versions were available with DIN and XLR connectors, with later units offering XLR only.

From a sonic standpoint, the AKG D202E is solid, especially considering how long ago it was designed. And it looks very cool, which is always a positive in my book. I’ll probably use it again with great results, particularly when I need to mike up a symphony.

CRAIG LEERMAN is senior contributing editor for Live Sound International and is an avid collector of vintage microphones. Read about more of his mics on ProSoundWeb by searching "Microfiles."
ADAMSON SYSTEMS
E219
Inside a new dual 19-inch subwoofer joining the Energia line.
by Ken DeLoria

THE NEW E219 SUBWOOFER joins the steadily growing line of Adamson Systems Energia Series (“E Series”) components that kicked off with the full-range E15 (15-inch) full-range line array module, later joined by the more compact E12 (12-inch) full-range line array module and E218 (dual 18-inch) subwoofer.

An Energia system is holistic and cohesive, as opposed to an assembly of individual, unrelated elements. All loudspeakers and subwoofers are expressly designed to work with networkable Class D amplification, DSP, cable and power distribution, software integration of control, and 3D simulation and diagnostics. There are solid engineering reasons for providing turnkey rigs for touring, one-offs, festivals, and permanent installations. Approaching “system technology” as a whole helps assure a consistently high level of performance, delivers a known, quantitative reliability index, and offers the ability to provide software updates that future-proof the investment.

Specifically, all loudspeakers are powered by Lab.gruppen PLM 20000Q amplifiers with integrated Lake digital signal processing and Dante networking capability. Before shipping, the processors are equipped with a selection of optimized frame presets that cover the most common array and subwoofer configurations, while also accommodating differing array design characteristics. The concept has proven to be quite successful in the marketplace, with Energia systems now a staple in the inventories of sound companies around the world, regularly deployed to serve a wide range of top-tier tours, festivals, and live events.

LIGHTER & TIGHTER
Extensive customer input indicated that the line would benefit with the addition of a musically exciting low-frequency experience, which served as the genesis of what would become the E219. It can also be viewed as a superior successor to the T21 subwoofer, a dual 21-inch design developed for the legacy Y-Axis Series. To the point, the E219 was conceived to be smaller and lighter while capable of delivering high output and exhibiting improved efficiency in its intended bandwidth.

As the name implies, the dual cone drivers have been scaled down to 19 inches, a key factor in reducing enclosure size and weight. Further, the smaller, lighter drivers are able to produce a punchier, tighter LF sonic quality than their 21-inch predecessors while also displacing “more air” than 18-inch drivers.

The 19-inch format has proven to be an ideal middle ground between the two. Directly compared to an 18-inch cone, it provides additional piston area for increased power transfer, but only a very small increase in mass, hence the rapid and impactful response characteristics.

The design specification called for the E219 to be optimized to perform in the bandwidth heavily occupied by modern musical styles, 30 Hz to 60 Hz, with available power output of more than 140 dB SPL (at 1 meter) from a single enclosure. To achieve these goals, a focus on obtaining uniform impulse response played a strong role in the R&D effort, resulting in a whole lot of computer modeling time.

Impulse response tells us a lot about an audio system’s performance, revealing time-domain characteristics that cannot be characterized by other means. Impulse response is generally described as a short duration time-domain signal, often annotated as h[t] for continuous-time systems (also called time-invariant), or h[n] for discrete-time systems (time variant).

The net effect is measured at the output of a system when an impulse is applied to the system’s input. In other words, impulse response describes the reaction of the system as a function of a specific period or “slice” of time. And because most music and virtually all speech are time-domain dependant for accurate reproduction, the impulse response of a given system is an accurate means of identifying that system’s characteristics.

Why is this useful? It allows us to measure the system’s transient response and phase response, and the data can be readily converted to frequency response plots by applying a Fast Fourier Transform (FFT). Since the impulse function contains all
frequencies, it accurately defines the response of a linear time-invariant system for all frequencies. Figure 1 shows how the E219’s impulse response translates to improved directional control and musicality, particularly when used in cardioid configurations. As is evident, the improvement in directionality and overall pattern control is at least an order-of-magnitude greater with the Adamson algorithms, and even more so in the 40 Hz plot.

SHAPE MATTERS

The E219’s long-excision SD19 drivers, manufactured by Adamson, employ Kevlar cone material, dual 5-inch voice coils and neodymium magnetic assemblies (Figure 2). Adamson pioneered the use of Kevlar with the MH225 loudspeaker, introduced back in 1987, and it’s at the heart of the company’s proprietary Advanced Cone Architecture design topology.

Made possible through the use of Kevlar, which has a higher Young’s modulus (low mass / high rigidity) than a standard paper cone, the driver has a lighter and stiffer cone assembly with faster acceleration and deceleration for a tight, punchy sonic quality, greater power handling before cone breakup, lower distortion, and improved durability.

The coil is centered by means of dual silicon spiders that serve to control X-Max, the maximum tolerable excursion before damage. The dual voice coil system, known as Symmetrical Drive Technology, is reinforced by a dual spider design providing added stability under high-exursion demands.

The SD19’s dual 5-inch voice coils provide exceptional power handling, determined through rigorous lab testing, as well as a reduction in power compression under severe operating conditions.

Additionally, the bass-reflex E219 enclosure, with the drivers front loaded, utilizes a unique tangential flow venting system that reduces harmonic distortion by minimizing air turbulence, thus fostering higher maximum SPL at lower distortion levels. The E219 is designed to function as a “normal” subwoofer, while a system comprised of E219s will provide true cardioid output by physically reversing one or more of the enclosures to face rearward in respect to the forward facing enclosures (Figure 3) and selecting the appropriate DSP settings, which include a unique all forward-facing “End Fire” preset that enhances multi-band rear cancellation without smearing impulse response.

Long a proponent of cardioid subwoofer deployment, the Adamson design team developed Convertible Cardioid Technology that was initially seen in the SpekTrix subwoofer in 2003. The experience gained from that project led to the development of a series of advanced DSP algorithms for Energia subwoofers.

Standard presets include regular non-directional usage plus three commonly used directional configurations: two enclosures arranged front-back (FB), three enclosures arranged front-back-front (FBF), and enclosures (quantity dependent upon how many are stacked) configured in an end-fired array (EF66) (Figure 4). All three configurations offer improved directional control in comparison to “standard” sub arrays, with the EF66 offering the greatest directionality by using advanced filters that achieve wide-band cancellation with only two sources – while maintaining impulse response integrity.

In a standard end-fired configuration, the null frequencies are determined by spacing and amount of sources, but Adamson’s proprietary algorithms are intended to provide wider rejection with only two sources, thus allowing the user to achieve maximum rearward rejection from a minimal footprint. All cardioid presets are stand-alone, with no extra delay or phase reversal required to take advantage of the benefits of cardioid low-frequency propagation.
THE FULL PACKAGE

As with all Energia enclosures, the E219 is constructed of marine grade birch plywood, measuring 23.5 x 55.8 x 35 inches (h x w x d) and weighing 249 pounds. Boxes can be ground-stacked or flown using modular rigging frames made of a combination of aircraft grade aluminum and high-carbon steel. The integrated rigging permits a 0 or 3-degree box-to-box angle. The reason to splay subwoofers at 3 degrees in most application scenarios is to not block HF coverage of the lower boxes in a steeply curved main hang when flown side by side.

When landing an E219 array of enclosures set to 3-degree splay angles, the cabinets will automatically collapse to 0 degrees. No physical manipulation needs to be done. Transport carts that hold up to three E219s are supplied as standard with each turn-key system.

Lab.gruppen PLM 2000Q amplifiers supplied with Energia systems offer 4 channels in a 2RU package, supplying 5,000 watts per channel at 2.2 to 3.3 ohms, and 4,400 watts per channel at 4 ohms. A Universal Regulated Switch Mode Power supply (100 to 240 volts) allows the amplifiers to operate properly anywhere in the world, while PFC (power factor correction) significantly reduces AC current draw on the mains, as well as parasitic noise on the power grid.

The onboard Lake processing provides crossovers, EQ, delay and protective limiter settings. It can be controlled via the front panel, a Lake Controller on a tablet, or addressed, controlled, and vital signs monitored via Ethernet or WiFi. Additionally, Dante networking capability is built-in and offers a fully redundant hardware configuration that switches to AES, analog, or secondary Dante signal sources if signal loss is detected.

Lake processor presets are included for the Energia line as well as all other Adamson loudspeakers. The latest preset library (v2.6) includes new tools to adapt the system’s response in an intuitive manner, allowing users to bring focus back to the music and not get sidetracked by technical terminology. And, all presets are equipped with impedance fingerprints in order to verify the transducer and cabling status before and after a show.

E219 sub interface with the amplifiers by means of high-grade cables terminated with Neutrik NL8 Speakon connectors. While only two wire pairs are used for an E219, the use of the 8-pole NL-8 makes pin-swap output connections possible to maximize efficient amplifier deployment. In addition to the expected paralleled I/O connectors, a dedicated NL-8 output-only connector allows four E219s to be powered on a single NL8 cable.

Figure 4: FB (front/back) configuration (top), and EF66 end-fired configuration.

AUTOMATING QUALITY

To aid in the task of efficiently planning and achieving an optimal Energia solution for each specific application, Adamson developed a 2D/3D modeling suite called Blueprint AV (Figure 5). It provides an accurate means of predicting directional performance via an easy-to-use application, allowing the designer to visually observe how a system will perform in a given venue on the computer screen – before deploying it. Parameters that can be manipulated include array size, cardioid configuration, individual gain and delay, coverage patterns, and splay angles.

E219s can be used as the only subwoofers in an Energia system or combined with smaller E218s, a recommendation regularly made for larger events and providing a complementary tonal character that adds to the punch and impact of the LF band. A typical configuration might consist of E219s on the ground with E218s flown with full-range line array modules. Rigging fittings are fully compatible with Energia’s E-Frame rigging. Additionally, the frame works as an adapter frame for underhang; that is, suspending E15 or E12 full-range enclosures beneath E219 or E218 subs.

The new E219 subwoofer furthers the ability of the Energia family to serve the specific needs of high-end touring sound and festival applications. The plug-and-play mentality consumes very little time setup time while insuring an optimum result, freeing up the sound team’s time for other critical duties. That said, by simply specifying a system without the portable road racks, a range of configurations are equally at home in meeting the needs of fixed installations such as performing arts centers, medium and large houses of worship, sheds, and arenas.

KEN DELORIA is senior technical editor for Live Sound International and has had a diverse career in pro audio over more than 30 years, including being the founder and owner of Apogee Sound.
PRESONUS HAS BEEN rather busy the past few years in introducing a number of live sound products, including StudioLive Active Integration (AI-series) loudspeakers. The range consists of three full-range models and a companion subwoofer.

Partnering with noted loudspeaker designer Dave Gunness and Fulcrum Acoustic, the AI-series makes use of Fulcrum’s Temporal EQ (TQ) DSP algorithms to incorporate processing that includes high-pass and low-pass filters, parametric EQ, delay, finite impulse filters, and temporal (time domain) filters.

In addition to the processing, the AI-series offers wired or wireless control for configuring, tuning and monitoring networked loudspeakers. Users can control DSP input level, a level, a high-pass filter, a group 31-band graphic EQ, 8-band parametric EQ (total of 10 bands), 8 notch filters, mute, delay, and signal polarity.

Custom settings can be stored onboard the loudspeakers for recall later from the back panel without a computer or tablet. An included USB Wi-Fi LAN adapter enables wireless networking, and wired communications are provided via a card with an etherCON port. Dante networking is also provided via an optional card.

IN THE BOXES

Full-range models are all 3-way designs, with an 8-inch midrange driver and 1.75-inch titanium compression driver in a coaxial configuration, joined by a single 12-inch (model 312AI) woofer, a single 15-inch (315AI) woofer, or dual 8-inch woofers (328AI). Transducers are driven by 2,000 watts of Class D amplification (1,000 watts LF - 500 watts per 8-inch woofer in the 328AI - and 500 watts each for MF and HP).

On the back, there’s an IEC power cord socket and power switch, combo XLR/TRS line input with volume control, XLR microphone input offering 12-volt phantom power with volume control, master level control, XLR mix output, USB port, DSP preset and network control and monitoring areas, and the etherCON control card. Selectable DSP contours include “normal” for main system duties, “LBR source” for low bit-rate source audio, and “floor monitor,” with a user setting layer on top of the selected contour. A 100 Hz roll-off switch is provided, as are signal, limit, clip and thermal indicator lights. A network area allows the user to configure and monitor WiFi connectivity.

18sAI companion subwoofers have an 18-inch cone driver driven by a 1,000-watt Class D amplifier. Processing includes dynamic limiting and excursion limiting. There’s also a pair of TRS XLR combo jacks for inputs, a pair of XLR outputs for jumping to the full-range boxes, and a single subwoofer volume knob. DSP settings include “normal,” “extended LF” for a bigger bass sound, and a “user” customizable preset.

Both full-range and sub cabinets are constructed of plywood, have heavy-duty metal grills, and include side handles. Full-range models also have M10 flypoints and a dual-position pole mount with straight forward and 10-degree down angles, as well as a 50-degree angle on one cabinet side for monitoring applications (Dual-position pole mount is not included on the 315AI.) The sub has a pole socket on top, and note to use a 31.5-inch PreSonus or third-party pole – a longer pole could cause instability.

Each of the 328AI loudspeakers evaluated for this Road Test have a stated frequency response of 59 Hz – 22 kHz (-6 dB) and a maximum peak SPL of 133 dB. They measure 29.1 x 14.3 x 15.7 inches (H x W x D) and weigh 51 pounds. The stated frequency response of the 18sAI sub is 32 Hz – 110 Hz (-6 dB) and maximum peak SPL is 135 dB.

At 21.8 x 24 x 26 inches (H x W x
D) and 94 pounds, the subs may be a bit bulky for a single person to pick up and move around without help. But with a handtruck I had no problems easily moving them, and PreSonus also supplied me with a platform dolly (D18s) that’s finished the same as the loudspeakers and has heavy-duty locking casters.

GETTING ACQUAINTED
In unpacking the two 328AIs and two 18sAIs, the first thing I noticed was the nice look and solid feel. The grilles have a built-in logo that glows blue when cabinets are powered on. The finish is really rugged, like a truck bed-liner, and looks great. The handles are comfortable and the cabinets sport nice rubber feet.

The loudspeakers ship with locking IEC cables (a great feature) as well as a USB plug-in wireless adapter for using StudioLive Room Control software. The software works with both PC and Mac, and is also available as an app for iPad. Adjusting parameters with the software is very easy, similar to using a remote with a digital console. You can even store a setting in the loudspeaker for later recall, and also monitor things like temperature and clipping.

Setup was easy. Everything is well labeled and I had a stack up and running within a couple of minutes. The 328AI was immediately impressive — I’m a big fan of coaxial designs, and this mid-high coax is certainly among the best I’ve ever heard. The pair of 8-inch woofers puts out plenty of bottom end, especially considering the compact size of the box. Adding the 18sAI turns the rig into a powerful 4-way system that sounds fantastic.

Stacking the 328AI over the sub is easy for a single person, but for me, the pole height is a little short, with the center of the coax only about 65 inches from the floor. I prefer to get my cabinets up in the air a little higher, especially the horns.

VARIETY OF USES
It was time to deploy the loudspeakers at some gigs. The first was in a small auditorium with a female singer performing with tracks. I was going to use the AI-series as the mains but had already contracted to provide a larger system (even though it was overkill), so instead used a single 328AI on stage in the side monitor position. I engaged the “floor” setting and set it directly in front of the singer, who had told me she didn’t really move around and didn’t need two monitors. For both voice and music, it performed quite well, and after the show, the performer said it was the first time she’s ever been totally happy with her monitor sound.

Next up was a company awards luncheon in a large banquet room with curved walls of windows. The program was background music followed by a few speeches and videos, and then an awards presentation. I placed the 328AIs on a couple of my taller pipe and base stands and used the 10-degree down-angle stand socket to focus output on the audience while keeping energy off the curved wall surface. This proved really handy, not to mention effective. Even though the mains produced more than enough bass for this gig, I placed a single sub in a corner for a little extra bottom end. Sonic performance was again stellar with both music and voice.

A few weeks later, the AI-series seemed like a perfect system to serve as a DJ rig for a corporate party — and it was. It’s easy for a single person to deploy the 328AIs pole-mounted on the subs, and in this case, the height was perfect for focusing energy on the dance floor. Further, the built-in dynamic and excursion limiting, established ahead of time, came in handy when the DJ decided to put a “smiley face” on his EQ and turn everything up to 11.

Finally, I deployed the rig for a trade show party at a casino ballroom with a live band. The 328AIs were mounted on my pipe and base stands next to the subs, and it was a breeze delivering enough output to blanket the entire room with full-range music and announcements. In addition, a pair of 312AIs (single 12-inch LF) served as monitors, and were a perfect complement to the mains. Both the keyboard player and female vocalist commented how nice they sounded.

If you’re seeking a portable rig that’s excellent in terms of form, function, and every other phase of the game, put the PreSonus StudioLive AI-series at the top of your list.

U.S. MSRP: 328AI — $1999.95; 312AI — $1899.95; 315AI — $2099.95; 18sAI — $1899.95

Senior contributing editor CRAIG LEERMAN is the owner of Tech Works, a production company based in Las Vegas.
OVER THE PAST THREE OR SO YEARS, many of the major suppliers have introduced digital wireless microphone systems, with some offering models from entry level to professional, and others remaining at the higher end. Additionally, one of the earliest entrants into digital wireless, Line 6, released a new generation of its platform last year.

Digital systems represent a paradigm shift, bringing with them new technologies, often greater immunity to interference, typically excellent and uncompressed frequency response, and sometimes the ability to pack more channels within a smaller slice of spectrum.

All of this activity is taking place as the “traditional” UHF spectrum resources are being re-purposed for use by mobile broadband carriers and other licensed and unlicensed wireless users. We won’t know the exact future of the 600 MHz band until at least mid-2015 when the FCC is slated to hold an “Incentive Auction” for mobile broadband telecommunications. (For more about this situation, see August 2014 LSI and ProSoundWeb.)

Regardless, the mission will be doing more with less. One of the solutions for packing more channels into less spectrum, employed by many of the newer digital wireless makers, is lower power transmitters – often with variable output levels such as 1, 10, or 20 mW. While this approach can somewhat diminish the maximum range of these systems, closer placement of receivers and the application of remote and/or directional antennas can compensate.

The diminished overall RF level generated by lower power transmitters can make it easier to increase channel density, decrease intermodulation, and allow frequency reuse by offering less potential interference among systems operating in nearby locations, such as at festivals with multiple stages, conventions, or closely spaced theatrical venues. Longer battery life is an additional advantage.

Adding to these benefits, programming transmitters is also more foolproof, with receivers performing environmental RF scans, selecting groups of compatible frequencies, and communicating them to the transmitters through IR or RF links. In some cases, both the receiver and transmitter settings can be monitored and changed via networking software, and most of these wireless systems can be networked. Audio can be output in the digital domain, so that the conversion to digital audio at the transmitter may remain unchanged until it reaches the loudspeaker system.

Many of the latest digital receivers cover a wide bandwidth, ranging from about 150 MHz to over 300 MHz, so a single receiver “split” is all that needs to be carried on a tour; most of the analog units cover approximately a 25 MHz spread. The front-end filtering is also more consistently and precisely tightened, allowing for closer spacing of channels.

Recent wireless mic systems that fall into the professional/touring class offer rack-mountable receivers, rugged transmitters, the ability to incorporate centralized networking and control, excellent audio quality, resistance to interference, antenna combining, and other desirable features to set the standard in wireless performance.

It will be exciting to see what comes next, both in terms of the technology in general and developments that will help meet new challenges. The circuitry is likely to get even faster, with the ability to predict and proactively make the necessary adjustments to keep the audio signals flowing. So far, product engineers at the various companies have creatively addressed the situation with solutions that sound great and perform quite well. I’m still amazed at how far it has come in the past couple of decades.

Enjoy this Real World Gear look at a range of wireless microphone systems for live applications.

GARY PARKS is a pro audio writer who has worked in the industry for more than 25 years, including serving as marketing manager and wireless product manager for Clear-Com, handling RF planning software sales with EDX Wireless, and managing loudspeaker and wireless product management at Electro-Voice.
The System 10 and System 10 Stompbox provide high-fidelity digital wireless that’s extremely easy to set up and use. Operating in the 2.4 GHz range, far from TV and DTV interference, both systems offer instantaneous channel selection and provide three levels of diversity assurance (frequency, time and space) for clear, natural sound.

The standard System 10 features a stackable receiver (ATW-R1100) equipped with a balanced XLR and an unbalanced 1/4-inch output jack with level control. The System 10 Stompbox receiver (ATW-R1500), designed for the guitarist, is a rugged, pedal board-mountable unit with foot switch for muting or toggling between outputs, two switched TRS balanced 1/4-inch output jacks, and an output mode selector.

### TECHNOLOGY NOTES
Digital 24-bit/48 kHz wireless operation; 2.4 GHz range – completely free from TV interference; three levels of diversity assurance: frequency, time and space; multi-pairing function to link a receiver with up to eight transmitters; automatic frequency selection; instantaneous channel selection, sync, and setup.

### KEY SPECIFICATIONS
- **RF Transmission:** Digital
- **Frequency Band:** 2.4 GHz ISM
- **Frequency Agility/RF Search:** Automatic
- **Diversity Scheme:** Frequency, time and space diversity
- **Receiver Formats:** Tabletop (ATW-R1100); Stompbox (ATW-R1500)
- **Receiver Outputs:** Analog
- **Audio Bandwidth:** 20 Hz – 20 kHz
- **Audio Sampling:** 24 bit/48 kHz
- **Latency:** < 4 ms
- **Transmitter RF Power:** 10 mW
- **Tx/Rx Communication:** Duplex

### SYSTEM 10 MODELS
- ATW-1101: UniPak Transmitter
- ATW-1101/G: Guitar System
- ATW-1101/H: Headworn Mic
- ATW-1101/H92: Mini Headworn Mic
- ATW-1101/H92-TH: Mini Headworn Mic (beige)
- ATW-1101/L: Lavalier Mic
- ATW-1102: Handheld Mic
- ATW-1501: Stompbox System

### TRANSMISSION SPECIFICATIONS
- **Shure ULX-D**: Transmission is digital, with a 24-bit/48 kHz sampling rate and an audio bandwidth of 20 Hz to 20 kHz (+/-1 dB). The system operates in the UHF band, offering several splits between 470 to 932 MHz with a receiver bandwidth of 64 MHz. Dante digital and analog outputs are provided, with either two or four receivers in 1RU and a half-rack, single-channel unit also available; the receiver has scanning and channel selection functions. Transmitter power is switchable in three steps – 1, 10, and 20 mW; using the lowest power in high-density mode enables up to 47 channels to operate simultaneously within one 6-MHz television channel. Ethernet ports are provided for networking, using Wireless Workbench software or AMX/Creston for monitoring and control. Includes switchable 256-bit encryption.

- **Lectrosonics Venue**: Transmission is digital hybrid, meaning digital audio on an analog carrier, with 24-bit/88.2 kHz audio encoding and an audio bandwidth of 32 Hz to 20 kHz (+/-1 dB). Multiple 25.5 MHz splits are available between 470 and 768 MHz, and the 1RU master chassis can contain up to six receiver channel modules. The receiver has scan and spectrum analyzer functions. The receiver diversity technology includes choice of OptiBlend Ratio Diversity, antenna phase-switching, and frequency diversity. Transmitter power is switchable between 50 and 100 mW on U.S. models. Both RS232 and USB ports are provided for networking, using LecNet2 software.
AKG DMSTetrad
www.akg.com

With 24-bit, 48 kHz audio coding, DMSTetrad is a digital wireless system delivering uncompressed studio-quality transmission and a linear frequency response for quality vocal and instrumental performances. Operating in the 2.4 GHz frequency bandwidth, it’s outfitted with proprietary Dynamic Frequency Selection (DFS) that insures that the cleanest frequency bands are selected automatically. Both time and antenna diversity further signal transmission that’s rock-solid and drop-free.

The entire DMSTetrad digital system includes the DSRTetrad receiver, DPTTetrad pocket transmitter, and DHTTetrad handheld transmitter that’s available with AKG’s D5 acoustics or DHTTetrad P5 with standard dynamic capsule. Two sets are available – DMSTetrad Vocal Set with DHTTetrad P5, and DMSTetrad Performer Set with DPTTetrad as well as a C111 L earhook mic and the MKG L instrument cable.

OF NOTE: The DMSTetrad receiver can be expanded to work with an additional three transmitters, and it also includes a mixer – all four channels can be mixed down directly to a balanced XLR sum output; detachable antennas can be mounted elsewhere for improved signal stability; 128-bit AES standard encryption prevents tapping of the audio signal, also making the system well suited for high-security conferences in addition to live performance applications.

TECHNOLOGY FOCUS

Proprietary DROCON (DROpout CONcealment) further protects against dropouts, requiring just two milliseconds to simultaneously process all four of the DMSTetrad system’s channels, and with only marginal impact on the receiver’s power consumption.

Sennheiser Digital 9000  >>  www.sennheiserusa.com

Transmission is digital, with a 24-bit/44 kHz sampling rate (also switchable to sampling rates of 48, 88, and 96 kHz), and an audio bandwidth of 20 Hz to 20 kHz (+/-0.5 dB). The system operates in the UHF band, offering one split between 470 to 798 MHz with a receiver bandwidth of 328 MHz. The 4RU receiver contains eight receiver channels, with scanning and spectrum analyzer functions. AES3 digital and analog outputs are provided. Transmitter power is switchable in three steps – 10, 25, and 50 mW. Ethernet ports are included, using proprietary WSM software for monitoring and control. Switchable proprietary encryption, and both standard and long-distance transmission modes are incorporated.

Carvin UX1000-MC  >>  www.carvin.com

Transmission is analog, with an audio bandwidth of 50 Hz to 19 kHz. The system operates in the UHF band, with one 24-MHz split available between 638 and 662 MHz. The half-rack, true-diversity receiver features RF scanning capability and the ability to select compatible frequency groups, with analog outputs. Transmitter power is variable, 10/50 mW. Both the transmitter and receiver have 960 user selectable channels, which can be grouped into four groups. Links to system components via unbalanced 1/4-inch or balanced XLR connections.
Transmission is analog, with an audio bandwidth of 50 Hz to 15 kHz (+/- 2 dB). The system operates in the UHF band, with two 28-MHz splits available. The half-rack, true-diversity receiver is frequency-agile with proprietary one-touch ClearScan RF scanning capability, and the ability to select compatible frequency groups. Transmitter power is switchable between 5 and 50 mW for the beltpack, and is 30 mW for the handheld. The handheld also accommodates a variety of interchangeable microphone heads.

Transmission is digital, with a 24-bit/44 kHz sampling rate, and an audio bandwidth of 10 Hz to 20 kHz (+0/- 2.5 dB). The system operates in the 2.4 GHz band, offering one split with a receiver bandwidth of 75 MHz. The half-rack receiver provides scanning and selection of compatible frequencies. The diversity scheme includes both spatial and frequency diversity, with each transmitter sending out the same signal on two different frequencies. Transmitter power is switchable between 3.3 and 10 mW. Transmission is encrypted, and both the handheld and bodypack transmitters incorporate EQ and microphone capsule modeling.

Transmission is digital, with a 24-bit/44 kHz sampling rate, and an audio bandwidth of 20 Hz to 20 kHz (+0/- 2 dB). The system operates in the UHF band, offering several splits between 480 to 760 MHz with a receiver bandwidth of 64 MHz. SPDIF digital and analog outputs are provided, with two receivers in 1RU; the receiver has scanning and channel selection functions. Transmitter power is switchable between 10 and 50 mW. Ethernet ports are included for networking, using proprietary RCS2.Net software to control up to 64 channels. Switchable 256-bit encryption and receiver EQ presets are provided. (Both dual-channel ACT-828 and single-channel ACT-818 pictured here.)

Transmission is digital, with a 24-bit/44 kHz sampling rate, and an audio bandwidth of 20 Hz to 20 kHz. The system operates in the UHF band, offering one split with a receiver bandwidth of 319 MHz. Analog outputs are provided, with two receivers in 1RU; the receiver has scanning functions. Transmitter power is switchable between 10 and 50 mW. Switchable encryption is incorporated in the transmission. Networking is IP-based, using proprietary Chameleon software, and can be monitored and controlled from computers, tablets, and smart phones.
Transmission is analog, operating on two 24-MHz splits in the UHF band. The half-rack, true-diversity receiver is outfitted with RF scanning capability as well as the ability to select compatible frequency groups. Both the receiver and the transmitter are synthesizer controlled via Phase Locked Loop (PLL) for stable RF signals. Transmitter RF output is 50 mW. The handheld transmitter offers interchangeable mic heads. An optional amplified Antenna Distribution System (ADS 4) is available, allowing up to 4 systems to be run off a single pair of antennas.

Transmission is digital, with a 24-bit/48 kHz sampling rate, and an audio bandwidth of 20 Hz to 22 kHz. The system operates in the UHF band, offering three splits between 470 and 698 MHz with a receiver bandwidth of 72 MHz. AES3 digital and analog outputs are provided, with two receivers in 1RU; the receiver has scanning and spectrum analyzer functions. Transmitter power is switchable in three steps – 1, 10, and 50 mW. A 2.4 GHz bi-directional link connects the transmitter to the receiver, so that changes can be made via software to both Tx and Rx, and the transmitter can even be put into “sleep” mode and awakened remotely. Ethernet ports provide networking using Wireless Studio 3.0 software for monitoring and control. Switchable proprietary encryption is incorporated.

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Meeting Audio Challenges On The South Lawn Of The White House

JBL VTX V20 arrays helping to meet the difficult coverage challenge at the White House.

◥ Contracted by the United Service Organizations (USO) for the sixth year to integrate the sound at the 4th of July concert on the South Lawn of the White House, live production company Event Tech deployed JBL Professional VTX line arrays, Crown Audio amplifiers and dbx Professional signal processing to meet the unique demands of the venue while delivering quality audio. Pitbull gave an hour-long performance for the 1,200 U.S. servicemen and service-women that were present, as well as their families and White House staff.

Because of a gradient on the South Lawn, it was a challenge to balance the sound between the two ends of the stage; stage-left was elevated two feet above the ground, and stage-right was elevated five-and-a-half feet above ground. To provide coverage of 280 degrees (horizontal), Event Tech used four stacks of six ground-stacked JBL VTX V20 loudspeakers and one VTX S25 subwoofer per stack, as well as six VTX G28 subwoofers per side in cardioid arrangement. Two VTX F12 and two VTX F15 loudspeakers were deployed for front and out fills.

Four VRack 4x3500HD and two VRack 12000HD amplification systems powered all loudspeakers. A dbx DriveRack 4800 delivered the signal processing and routing for the overall system. The system was configured and controlled with JBL HiQnet Performance Manager.

“We did a lot of pre-work at our office with the VTX rig, because we wanted to make sure that the coverage was precise and make it drop off 300 feet from stage left, so that the sound did not penetrate the White House,” says Jeremy Meyers, account executive and audio systems designer for Event Tech. “I was very impressed with the output from these small V20 boxes, given their relative size compared to those from other manufacturers that I have worked with in the past. The detailed coverage from JBL’s patented waveguide and the precise frequency output of the D2 drivers allowed us to accomplish our goal.

“We were also impressed with the VTX F12 and F15 fills, which had amazing output, even when compared to the V20 boxes,” he adds. “Thanks to our experience in live corporate events and the incredible caliber of these JBL loudspeakers, our veterans received a terrific 4th of July gift.”

PEOPLE

◥ Erik Miehs has joined Riedel Communications as service engineer for Australia, where he will boost the company’s ability to support existing customers and equipment rentals while contributing to the continued overall growth in sales and rentals across the region. He brings more than 10 years of experience working in audio and pro sound to the position, most recently serving as a communications and broadcast engineer at the Sydney Opera House.

◥ Mark Rahilly has joined Shure as a U.S. senior sales manager, covering the retail market in the east and southeast, based in Boston. Previously he worked with Harman Professional as a territory manager, and also worked for 10 years with Shure sales rep firm Richard Dean and Associates.

◥ DPA Microphones has appointed Christopher Spahr, Pedro Rocha and Leonardo Romero as area sales managers for the eastern U.S., western U.S. and southern U.S./Latin America/Canada, respectively. The company has also promoted Shan Siebert to general manager of the Longmont, Colorado-based office.

◥ To further support its Constellation acoustic systems, Meyer Sound has appointed Mac Johnson as Constellation program manager and Melody Parker as associate acoustic engineer. In his new role, Johnson drives all aspects of the company’s Constellation efforts, from lead development to design, sales, and commissioning of systems. He’s been an integral part of the Constellation team since 2011, when he transferred from the company’s education program. Meanwhile, Parker provides project support for Constellation and Libra acoustic image systems, and was formerly an acoustical consultant at Charles M. Salter Associates and a music analyst for Pandora.
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Plenty Of Wireless & DSP For *The Color Purple*

A recent production of *The Color Purple* at the Hillbarn Theatre (Foster City, CA) saw sound designer Alan Chang of San Francisco-based Coral Canopy deploying a combination of 23 Lectrosonics SMV and SMQV miniature beltpack transmitters as well as LMa beltpack models for wireless microphones.

On the receiving end, he utilized five Lectrosonics Venue receivers outfitted with a combination of VRS and VRT receiver modules configured for narrowband, wideband mid, and wideband low performance. A Lectrosonics UMC16B UHF multi-coupler was also on hand for RF antenna distribution. This equipment was used in conjunction with DPA 4061 and Countryman B3 microphones.

“The Color Purple is a 3-plus hour-long musical, which entails very precise battery management that no other manufacturer offers,” Chang explains. “One such example is Lectrosonics’ remote capability to sleep the transmitters for battery conservation by playing a series of tones and beeps from the RM Remote or from the LectroRM app on a smartphone.”

For signal management, Chang used a Lectrosonics Aspen Series SPN1612 16-input/12-output digital signal processor to handle the majority of the music and vocal performances, plus some sound effects cues. He also placed a Lectrosonics DM812 8-input/12-output digital automatic matrix processor into service for sound-effects driven loudspeakers. And, a Lectrosonics DNT0212 Dante network processor handled sound-effects playback through two networked computers (an Apple MacBook Air and a Mac Mini), also providing reverb processing and enhancement for the vocals.

Managing Multiple Ins & Outs At Edinburgh Castle

This summer’s Commonwealth Games in Glasgow (Scotland) were preceded by Live at Edinburgh Castle, a concert televised live on the BBC from this unique location that dominates the Edinburgh skyline. The line-up for the event was extensive and included Culture Club, OneRepublic, Kaiser Chiefs, and Motown legend Smokey Robinson.

Various acts were backed by a 12-piece house band and the BBC Scottish Symphony Orchestra. To keep up with the fast-paced nature of the high-profile event, two SSL Live consoles were provided by Britannia Row Productions. The event was headed by Brit Row project manager Lez Dwight, with Stefan Krista managing the inputs and cross-patch from scores of 12-way sub-snakes. Nahuel Gutierrez operated an SSL Live for the 75-member orchestra, generating group stems for front of house, monitors, BBC broadcast and 16 personal monitor stems for the orchestra.

Niccolo Antonietti used a second SSL Live console to mix monitors/IEMs for the house band, along with guest vocals, guest backline and Pro Tools tracks, totaling 60 outputs for 30 stereo mixes. Both SSL Live consoles were located remotely with only a BBC program shot to watch.

Antonietti’s console handled the house band, made up of drums, bass, guitars, two keyboards, percussion and three backing vocals, as well as director Mike Stevens’ keyboard and guitars. He also controlled the handheld microphones for guests, ambient microphones, intercoms, BBC feeds and 16 channels of Pro Tools, bringing Antonietti’s desk to 120 input channels.

The two SSL Live consoles also had a MADI connection between them to feed stem subgroups back and forth.

“The orchestra was using personal monitors, so I sent them band stems, like guitar, rhythm section, keyboards and vocals, as well as Pro Tools stems,” Antonietti explains. “I then received a premix of the orchestra and of all of the other channels that I couldn’t get on my desk.”

Over on the orchestra’s SSL Live console, Gutierrez used channel delay to synchronize the orchestra’s inputs. “The delays made everything sound a lot fatter and we loved it,” he says. “We measured the distances on stage and then put natural time delays from back to front, so percussion had the biggest delay and the first violins and cellos up front had the shortest delay.”
Registration is open for SynAudCon’s three-day Sound Reinforcement for Technicians (SRT) seminar in Dallas on October 6-8. SRT instructor Pat Brown provides a multimedia presentation along with interactive “hands on” exercises. The course covers the theory behind how systems work, and demonstrates how to use instrumentation to look “under the hood” to troubleshoot systems. On day three, system tuning is presented in a technical yet practical way. Find out more and register at www.synaudcon.com.

Full Compass Systems was a prominent participant in the 13th annual “Opera in the Park” concert for the city of Madison, a summer tradition that had another record turnout with an estimated 16,000 people in attendance. Many staff hours were donated by a team of Full Compass employees including CEO and founder Jonathan Lipp, who provided the front-of-house mix. “This is one of many events we sponsor to promote music in our community,” states Lipp, “It’s so vitally important for people to experience the gift of music and we will do everything we can to make sure they have that opportunity.”

Seattle-based PNTA has joined the Yamaha Commercial Audio dealer network, recently adding the new Yamaha QL5 digital console to its growing inventory. PNTA installs and repairs equipment and supports both non-profit and private customers through its event services department, providing production requirements for events of all sizes.

Audio-Technica U.S. recently presented Roseville, CA-based The Farm Technical Sales & Marketing with the President’s Award for its work in representing the company’s products. The award recognizes a manufacturer’s representative for outstanding commitment and dedication during the A-T 2013/2014 fiscal year. Philip Cajka, A-T U.S. president and CEO, presented the award to John Hood, principal of The Farm.

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Cher’s Dressed to Kill (D2K) tour, scheduled to resume in Albany, NY in mid-September, is traveling with three DiGiCo SD7 consoles supplied by Black Box Music of Berlin, Germany, the live sound production provider for the tour.

The DiGiCo consoles are operated by Dave Bracey at front of house, with Jon Lewis mixing monitors for Cher and Horst Hartmann supplying monitor mixes to the seven-piece band, dancers and technical crew. The D2K tour, Cher’s eighth solo venture, reunites the three engineers, as well as system tech Ulf Oeckl, other crew members and some of the musicians, who all worked together on P!NK’s 140-plus-date The Truth About Love tour, which wrapped in early 2014.

Bracey, who made the transition from DiGiCo’s D5 to the SD7 for Massive Attack’s 2008 European tour, reports that he typically does a lot of programming in the console. With Cher going through nearly a dozen costumes during the 18-song show, the D2K tour features numerous film and video montages while the star is offstage changing.

Bracey notes that some of the audio for the archival footage is less than pristine: “You have to process it in a way that makes it intelligible in the room,” he says. “So every interlude between every song needs its own snapshot. Some of the archival movie clips have such disparate audio signatures that only some well-adjusted multi-band comp and a channel of dynamic EQ would make the whole segment intelligible. The SD7 offers so much that I am yet to use a plug-in.”

The three SD7s are on an optical loop that allows the three engineers to share anything on the network, including video and communications. “That gives us the ability to throw something to each other at the click of a button, as opposed to having to go and find some XLR cables and patch it,” says Lewis. “If Dave suddenly needs her talkback mic, he can get it. If I need Dave’s effects returns, they’re there. It’s very quick and easy. And the ability to send video feeds around the system is great as well.”

“Even simple things like text chatting during the show,” Bracey chimes in. “If there’s something that’s not worth disturbing someone over, you just send them a message and they can answer in their own good time.”

Leading UK-based audio/production company SSE Audio Group has been appointed a Soundcraft Vi Select Dealer, and immediately purchased one of the first Soundcraft Vi3000 digital consoles off the production line, sold through Harman’s UK distributor, Sound Technology.

The new Vi3000 had its first major outing at the recent Isle of Wight festival where it was used by SSE regular Chris Courtney to mix monitors for Travis. Courtney had previously used a Vi6 for the band, but for this festival run, the reduced size and lack of rack requirements made the Vi3000 a more attractive proposition.

Take Me Downtown, the third headline tour for Lady Antebellum, was supported by Sound Image (Escondido, CA and Nashville), which provided an L-Acoustics K1 line array system. For most of the group’s performances, Sound Image flew left and right main PA arrays of 14 K1 plus four KARA downfills, each flanked by six flown K1-SB subs. Out fills were comprised of 12 KUDO per side, while four SB28 subs per side groundstacked below in cardioid configurations anchored the low-end reinforcement. Four more KARA – two per side – were additionally deployed as stage front fills.

The Pasadena (CA) Symphony and POPS was recently supported by exclusive audio provider Complete Production Rentals (CPR) with VUE audiotechik al-8 line arrays for a performance at the LA Arboretum. Based in nearby Westlake Village, CA, CPR has been the symphony’s sound resource for more than a decade. The final design included dual al-8 arrays consisting of 16 elements per side, with additional near field coverage delivered by four al-4 line array elements suspended below each al-8 array with VUE’s al-8-ufb “combo array” transition bar. Eight VUE V6 Systems Engines provided power and processing for the al-8 arrays, while a pair of V4 Systems Engines handled the smaller al-4 elements.
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1. Do you show up on time?

   Yes.
   Go to question 2.
   No.
   Try getting one of those “work from home” jobs where you make $5,000 a week stuffing envelopes. Good luck!

2. Do you like to work with a wide range of personality types?

   Yes.
   Go to question 3.
   No.
   Pursue a career in broadcast, where you can stay in the broadcast truck all day and night.

3. Do you like music?

   Yes.
   Go to question 4.
   No.
   Get a gig with Justin Bieber. Or become a truck driver. They usually sleep during the show.

4. Does the thought of letting down 60,000 fans because you didn’t plan the sound system properly disturb you?

   Yes.
   Become a surgeon.
   No.
   Become a politician.
   Maybe.
   Most of the fans will still get good sound, right?

5. Is life on the road attractive to you?

   Yes.
   Get that tool kit and gig bag together!
   No.
   Seek work in theatre, broadcast, studios, clubs, or other fixed locations. Alternatively, practice saying this phrase: “Would you like fries with that?”

6. Do you like using microphones?

   Yes.
   Consider a career in aviation.
   No.
   Buy a few anyway. The mic companies need you, and shiny mics can look quite fetching in a display case.

7. Can you function optimally after 48 hours without sleep?

   Yes.
   You’re hired. Can you get a commercial driver’s license too, please?
   No.
   Consider a career as a mattress tester.

8. Do you like Marriott and Hilton hotels?

   Yes.
   Consider a job in the hospitality industry.
   No.
   Good. Do you like Holiday Inn?
   Yes.
   Consider working as a night auditor.
   No.
   Good. Do you like Motel 6?
   Yes.
   You’re hired.

9. Do you prefer analog or digital consoles?

   Analog.
   Don’t call us, we’ll call you.
   Digital.
   Don’t call us, we’ll call you.
   Whatever you have in stock.
   Good. You’re hired.

10. How will you feel when a tour manager/artist/promoter tells you’re lousy at your job?

    I’d feel bad.
    Look for an easier gig like parole officer or personal assistant to Leona Helmsley.
    I couldn’t care less.
    You’re a natural, kid! Welcome aboard.

If you answered “Yes” or “No” to more than any five questions, you’re ready to enter the lucrative world of professional audio.

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