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ONE LAST TOUR

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FEATURES

DEPARTMENTS

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From the Editor's Desk…

One of our primary goals with this publication is to present insights and practices from veteran pro audio practitioners, so it's with great satisfaction that I announce our newest team member, senior editor M. Erik Matlock.

Erik brings a wealth of hands-on experience to the position, having worked in a wide range of roles in pro audio for more than 20 years in a dynamic career that encompasses system design and engineering in the live, install and recording markets. He's also served as an author for several industry publications, including this one, as well as founding and operating his own blogs addressing technical and other interests. We're so pleased to add his considerable experience to our team as we grow and go in new directions.

On the Back Page of this issue, Erik offers a perspective on the choice and application of loudspeakers, based upon his own experiences and observations over the years. It's an interesting discussion, bolstered by some real-world examples. In the same vein, Karl Winkler steps up with an interesting discussion of his own, in which he considers the impact of technology on the pro audio profession.

Also in this issue, Mark Frink offers an excellent interview of longtime front of house engineer Ken Newman as he works on what appears to be the final tour by Barry Manilow. Meanwhile, Kevin Young shares his conversation with monitor engineer Dan Housel on the approach he's taking with Lionel Richie.

In addition, Craig Leerman provides a helpful look at the world of wireless systems and the RF situation, while Andy Coules offers an instructive take on recording in the live realm with limited resources.

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

Keith Clark

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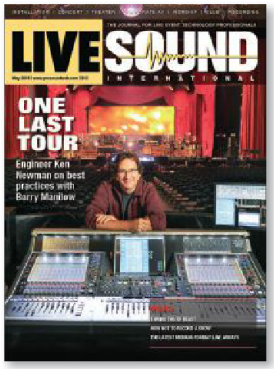

ON THE COVER: Engineer Ken Newman at the DiGiCo SD5 console he's using on "One Last Tour" by Barry Manilow. *(Photo by Steve Jennings)*

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Soundcraft Vi5000 & Vi7000

Digital consoles replacing the company's Vi4 and Vi6 models that offer optional 96kHz processing, upgraded channel counts, and hardware. Both offer the latest 3D Vistonics II and FaderGlow interfaces. A new local rack and active breakout box delivers simultaneous mixing of up to 128 inputs and 32 mono/stereo buses with up to 384 inputs and outputs. Sound quality is enhanced by low-noise mic preamps and 96 kHz 40-bit floating point processing, with the digital implementation of the BSS DPR901ii dynamic EQ adding to the channel processing. Effects contain 8 independent Lexicon multi-FX units and a BSS graphic EQ on every bus output. Both consoles also have an additional dedicated 64-channel MADI interface for Realtime Rack. Configuration via new encoder assign functions and shortcut keys are joined by a new extension to the Vi's patent-applied-for VM2 wireless system monitoring feature, with Shure ULXD systems now recognized alongside AKG'DMS800 and WMS4500 systems. ViSi Connect I/O expansion options are available (EtherSound, CobraNet, Dante, MADI and more) plus a choice of Cat-5e or optical fiber Stagebox connection. **www.soundcraft.com**

AKG D5 C

A dynamic directional microphone designed for live sound environments. It offers a cardioid polar pattern and utilizes the company's patented laminated Varimotion diaphragm. A dual shock mount reduces handling noise while an integrated pop filter reduces pops and wind noise. The spring-steel wire-mesh grille is designed to withstand the rigors of live stages. It is specified as being able to withstand SPL of up to 160 dB. An additional model, the D5 CS, is outfitted with a click-free on/off switch. **www.akg.com**

PreSonus SL-AVB-MIX

An Audio Video Bridging (AVB) audio networking option card for the company's StudioLive AI-series consoles. AVB enables synchronized, real-time, low-latency, streaming of audio across an Ethernet network. With this first phase of AVB support, StudioLive RM32AI or RM16AI rack-mount mixers can be used as a stage box and monitor mixer for any StudioLive AI-series front-of-house console. Note that StudioLive RM-series mixers already have the option card installed and simply require a firmware update to enable the AVB functionality. **www.presonus.com**

Eastern Acoustic Works (EAW) Redline

A series of powered loudspeakers for live and install applications that includes 2-way models RL12 (12-inch) and RL15 (15-inch) joined by the RL18S 18-inch subwoofer. All are outfitted with customized transducers. Both the RL12 and RL15 deliver consistent directivity via beamwidth-matched crossovers on 90- x 60-degree rotatable horns. Four-aperture ports bolster bass support. Transducers are driven by a 1,250-watt, fanless, class-D, Power Factor Corrected amplifier. EAW Focusing processing refines impulse response. Proprietary DynO processing optimizes the power transfer from amplifiers to transducers. Three user-defined voicing options allow a variety of uses without need of a computer or external processor. The RL18S subwoofer is cardioid-ready via rear-panel controls. Enclosures are made of 15 mm hardwood with proprietary RoadCoat treatment, joined by 18-gauge grilles with a magnetic Redline strip (removable). The enclosures also have symmetrical monitor angles to form left-right wedge pairs as well as integral road handles for portability. **www.eaw.com**

← JBL Professional VTX V20-DF

A down fill adapter to cover audiences more accurately in terms of VTX V25-II and V20 loudspeaker counts. The adapter frame is designed and engineered as an attachment of up to six V20 loudspeakers to larger V25 line arrays. It enables precise alignment and optimum line source array coupling of the vertical wavefronts of the V20 and V25 loudspeakers with minimum physical gap when attached, while offering four variable attachment angles (0, 3.5, 7.5 and 11 degrees) between the V20 down fill block and lower V25 enclosure. It can also be used as a pullback accessory when deployed as part of the compression suspension systems on VTX V25-II-CS or VTX V25-CS, and can also be used with VTX V25-II or V25 enclosures in fixed angle tension suspension mode. Modeling support for the VTX-V20-DF down fill adapter will be included in an upcoming release of JBL Line Array Calculator software and integrated into JBL HiQnet Performance Manager control software. **www.jblpro.com**

d&b audiotechnik D20 ↑

A 4-channel, 2-rack-unit power amplifier with a user interface and remote control and monitoring via Ethernet through the Open Control Architecture (OCA) protocol. All four independent channels provide up to 1,600 watts, each with two 16-band equalizers that offer parametric, notch, shelving and asymmetric filters and up to 10 seconds of delay. The user interface has a color touch screen along with a rotary encoder. It is enabled with an integrated web interface for use with a browser and can be controlled using d&b R1 Remote control software V2, which runs on Windows or Mac. R1 provides enhanced EQ functionality. The class D amplifier, which has a switch mode power supply and Power Factor Correction, is largely independent from voltage variations caused by substandard electrical sources. **www.dbaudio.com**

L-Acoustics X Series ↑

A line of coaxial loudspeakers for install and portable applications. Models include the X8 live monitor, X12 multipurpose enclosure and X15 stage monitor, outfitted with high-excursion neodymium drivers, ellipsoid directivity, laminar vented ports and up to 30 percent weight reduction. The X12 loudspeaker's ellipsoid directivity of 90 x 60 degrees makes it adaptable to front of house applications, and it's available with a range of rigging accessories and RAL custom color options for installs. The X15 stage monitor's narrow, 40- x 60-degree directivity helps in eliminating feedback potential. It's also outfitted with integrated risers that allow it to go from 35 to 55 degrees. The X Series comes as an addition to the compact 5XT fill loudspeaker launched two years ago. **www.l-acoustics.com**

dbx Professional goRack ↓

A loudspeaker management processor for optimizing portable PA systems that incorporates that incorporates the company's DriveRack processing in a unit that fits into the palm of the hand. It provides EQ, compression, AFS (Automatic Feedback Suppression), subharmonic synthesis for enhanced LF response, and more. Five buttons offer access to the most commonly used processing effects, and a large data wheel lets users dial in any of the unit's sound-enhancement effects, including 20 musical genre-based 31-band graphic EQ settings. The compressor is based on the design of the original dbx 163X. The goRack also includes dual dbx mic preamps, selectable mic/line input settings, and aux input as well as left and right channel volume controls. It connects between a mixer and an amplifier via its stereo inputs and outputs, which accommodate both XLR and 1/4-inch connectors. **www.dbxpro.com**

Yamaha TF Series

Three compact digital consoles, the TF5, TF3 and TF1, offering 33, 25, or 17 motor faders, respectively, along with 32, 24, or 16 rear-panel analog inputs. All include recallable Yamaha D-PRE preamps. There are a total of 48 inputs on the TF5 and TF3, with 40 on the TF1, including dual stereo analog/USB digital inputs and dual returns. Live recording features include up to 34 x 34 channel recording and playback via USB 2.0 and 2 x 2 with a USB storage device. Touch-Flow Operation combines elements of Yamaha's user interfaces with a touch panel system. In addition, there is 1-knob COMP and 1-knob EQ via a single rotary control on inputs for adjusting individual sources and on outputs for overall sound control. GainFinder facilitates gain setup for individual input signals. TF mixers also provide a range of input and output channel presets created in consultation several mic manufacturers as well as engineers. The input channel presets are made to match a span of musical instruments and voices. Output channel presets include parameters optimized for in-ear monitors and Yamaha powered loudspeakers, with variations to match different environments and room sizes. **www.yamahaproaudio.com**

NEXO ID Series

Currently comprised of the ID24 compact full-range loudspeaker and two low-profile subwoofers, the ID S110 (single 10-inch) and ID S210 (dual 10-inch). The ID24 incorporates twin 4-inch drivers in a V formation in combination with a compression driver offering two preset directivity options. A user-adjustable horn can be rotated without the need for tools via a switch on the rear panel to provide 60- or 120-degree HF coverage with the standard horn fitted. The ID24 is offered in three versions: ID24i for installation, ID24t for touring, and ID24c, an "a la carte" version that allows the user to custom configure. The asymmetric shape of the ID24 cabinet make it able to be mounted horizontally or vertically, and, using the correct accessories, it can be deployed for underbalcony applications. The ID S110 and S210 subs have a low-profile design that allows them to be deployed discretely in a range of locations. Sixteen ID24 cabinets can be powered by a single NXAMP4x1, the company's smallest powered TDController. **www.yamahaca.com**

K-array Concert Series

Includes the KH2, KH3 and KH5 self-powered full-range models joined by the KS5 self-powered subwoofer. All offer a very thin profile and are designed for a range of touring and install applications. The full-range models are equipped with the company's proprietary "slim array technology" (S.A.T), controlled by discrete DSP channels for detailed beam steering and operational flexibility. All components are driven by a multichannel class D amplifier. Each loudspeaker is equipped with an onboard touch screen with matrix control, with RS485 and USB connectivity provided for remote control. The KH2 incorporates two 8-inch neodymium woofers (with 2.5-inch voice coils) joined by two 1.4-inch compression drivers (also with 2.5-inch voice coils), while the KH3 is loaded with two 12-inch neodymium woofers (with 2.5-inch voice coils) and two 1.4-inch compression drivers. The KH5 has two 15-inch neodymium woofers (with 3-inch voice coils) and four 1.4-inch compression drivers, and the KS5 subwoofer houses dual 21-inch neodymium woofers with 5.5-inch voice coils. **www.k-array.com**

D.A.S. Audio Action 118A

A self-powered horn-bass subwoofer designed to provide extended bass response. A 3,200-watt (peak) amplifier drives a single D.A.S. 18LX long-excursion woofer incorporating a 4-inch voice coil for higher power handling. The Class D amplifier with switch mode power supply and DSP has an EQ switch that allows selection of a "deep" or "loud" response. I/O includes two balanced inputs as well as stereo filtered output connections for satellite systems. The variable crossover ranges from 100 Hz to 160 Hz and can be defeated to offer stereo "loop thru" connections. A gain control and polarity reverse feature increase user control over the system. The enclosure is made of birch plywood finished with a coating of ISO-flex black paint. Handles on the sides and the rear of the cabinet along with two rear-located wheels, enhance portability. **www.dasaudio.com**

Meyer Sound LEOPARD & 900-LFC →

A full-range line array and low-frequency control element, both self-powered, that are the smallest members of the company's LEO loudspeaker family. They incorporate newly designed class D amplifiers that help eliminate distortion while consuming less power and generating less heat. The onboard amplifier and control circuitry for LEOPARD are contained in a single, field-replaceable module. Proprietary QuickFly rigging with captive GuideALinks provides setting of splay angles from 0.5 to 15 degrees. Six LEOPARD and two 900-LFC enclosures can be flown using a 1/2-ton motor. A complete system approach can be attained with MAPP XT for acoustic prediction, Compass RMS real-time system performance monitoring, Galileo Callisto loudspeaker management, and the new MDM-5000 distribution module for routing AC power, audio, and RMS signals. **www.meyersound.com**

DiGiCo S21 →

A digital console incorporating the company's Stealth Digital Processing, using the same FPGA algorithms as the SD7 consoles, while the mic preamp design is the same as found with the 192 kHz SD-Rack. The S21, which operates at 96 kHz standard, provides 46 buses: 16 x stereo (32); stereo master (2); solo buses (2 stereo, 4 total), and 10 x 8 matrix (8). Forty flexi-channels (mono/stereo) are the equivalent of 80 DSP channels. There are also 16 flexi buses (also mono/stereo, 16 assignable 32-band graphic EQs, 8 FX engines (reverbs, delays, w/modulation and enhancer), 4 assignable DiGiTubes and 4 assignable multiband comps, and a compressor per channel and bus. The control surface is outfitted with 2 multi-touch screens, 21 touch-sensitive moving faders, 4 layers of banks of 10. Mic/line inputs total 24, joined by 12 analog outputs and two AES I/O (mono). **www.digico.biz**

Avid VENUE | S6L

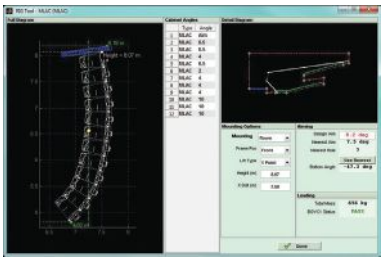
A modular, scalable live mixing system that incorporates the VENUE | E6L system engine to handle large channel and plug-in counts at low latency. All processing is at 96 kHz, with support for higher sample rates possible due to the amount of processing power available. The console's daylight-visible touch screen workflows, a AVB, Dante, MADI, Thunderbolt, and more. (Option cards are available as separate purchases and not included with core systems.) The control surface is available in three configurations and the E6L engine is available in two configurations: 192 processing channels, 96 mix buses plus LCR, and 144 processing channels, 64 mix buses plus LCR. **www.avid.com**

Allen & Heath Chrome Edition

Chrome Edition GLD-80 and GLD-112 digital mixers have been re-designed with a new metallic color scheme, while the AR84 and AR2412 AudioRacks have received a black finish. GLD Chrome firmware adds auto mic mixing capability and further additions to the processing suite, including new FX and channel-based compressor emulations. The automatic mic mixer (AMM) can be configured to work across all 44 microphone sources, allowing the user to select which inputs should be auto-mixed without the restrictions of a 16-channel insert based system. The AMM can be set to work in two modes. The firmware also includes a new DEEP plug-in architecture that allows users to select from a number of different processing units on every input and mix channel. Two new RMS-VCA inspired compressor models, the 16T and 16VU, are included as well. Integrated within the mixer's channel processing, all six compressor models can be selected on any of the input and mix channels on the fly. **www.allen-heath.com, www.americanmusicandsound.com**

Martin Audio Display 2.1.10

The latest upgrade to the software "brain" of the company's MLA Series of loudspeakers now fully supports all Compact and Mini models. Deployment options such as "ground-stack" and "flown" are selected at the start of the design process. The rigging view then provides detailed mechanical information to help construct and position the array. The Slice Import function from existing Display projects is now available, making it possible to build a database of designs that can be utilized again upon returning to the same venue. Display's optimization code has also been improved. Depending on specific system deployment, HF headroom has improved by up to 3 dB. Another enhancement is the inclusion of non-linear index points, making the prediction of how the array will perform faster to generate. The spacing of the reference points has also been optimized. And, Display now helps identify "shadow" coverage areas such as underneath deep balconies where there is no direct transmission path from array to listener due to physical obstruction. **www.martin-audio.com**



FBT MUSE 118FSA

A reversible and flyable active subwoofer designed for use with the company's MUSE 210LA full-range line array. It incorporates a hybrid-loaded 18-inch (3-inch voice coil) neodymium woofer, with a very large area laminar-flow port that helps eliminate power compression and port turbulence. Frequency response is stated as 33 Hz to 100 Hz. The sub is designed to suspend from the MUSE flybar facing either forward or rearward.

Also available is the MUSE 118FSCA, factory preconfigured as rear facing for cardioid applications. The transducer is driven by a class D power amplifier, with switch mode power supply, rated to deliver up to 1,200 watts (RMS). Onboard DSP provides eight presets. (A passive version, MUSE 118FS, is also available.) The control panel offers XLR input and link, volume, presets, delay, phase shift (0 to 180 degrees), and ground-lift. The enclosure is made of birch plywood coated with scratch-resistant black paint. **www.fbt.it**

← CADAC CDC six

A digital console based around further development of the company's advanced gesture-operation user interface. The traditional fixed physical controls have been replaced by a user interface, accessed via an optically bonded 23.5-inch 16:9 LCD touch screen, with digital encoders to the bottom and the right displaying the graphical user interface. A combination of touch screen and traditional encoders, for additional control of console features, enables "touch and swipe." The surface also offers stereo metering located to the left of each of the motorized faders, plus an individual full-color OLED display above each fader. It has a 64 input channel, 48 assignable bus design (as group, stereo group, aux, stereo aux or matrix plus LCR, monitors and talkback), with 20 touch-sensitive motorized faders. The CADAC mic preamps are matched with 24-bit/96 kHz Delta Sigma AD/DA convertors. There are four AES3 inputs and 4 AES3 outputs, 8 assignable mic/line inputs and 8 analog outputs. 4-band parametric EQ, 16 stereo onboard effects, dynamics, input and output delays, snapshot automation, 16 VCA groups with "VCA unfold" navigation, as well as comp/limiter, 4-band parametric and 31-band graphic EQ on all outputs. An integrated card for Waves MultiRack SoundGrid server is also provided. **www.cadac-sound.com**

↑ Mackie ProFXv2 Series

A re-designed range of mixers with a new preamp design and effects engine. Two additional models include the 4-channel ProFX4v2 and 30-channel ProFX30v2, bringing the total number of models to 6. The Vita preamps have a class-A front end, dual feedback stabilization and bias current optimization for low-noise performance. The ReadyFX engine employs floating-point DSP, providing a choice of 16 reverbs, choruses and delays. Multi-band GEQ is included for tuning mains and eliminating feedback from stage monitors. I/O varies by model, covering applications ranging from 4 to 30 inputs. Some models also offer built-in USB recording and playback. **www.mackie.com**

Radial Engineering Headload Prodigy ↑

A combination load box and DI for driving a guitar amp at a higher output in order to maximize tone yet produce a lower stage volume when needed. It employs cement-encrusted resister coils to covert the excessive power from the amplifier to heat and is able to withstand up to 100 watts RMS. In order to optimize for in-ear monitors, a dual band EQ on the front panel is included. To add connectivity options, two additional 1/4-inch outputs are provided with one that is post EQ, post JDX, the other pre EQ, pre JDX. These can be used to feed additional amplifiers, effects racks or recording systems, and adjusted using the front panel level control. The case is made of durable 14-gauge steel, with proprietary book-end design that creates protective zones around the switches and controls. One or two units can be rack-mounted into a single 19-inch rack space using an optional rack kit. **www.radialeng.com**

Yamaha Commercial Audio RSio64-D ↑

An I/O audio interface offering up to 64 inputs and 64 outputs with Dante/Mini-YGDAI card conversion while also providing router functionality. It's for use with R Remote software or Yamaha CL and QL Series digital consoles. Numerous input/output formats can be connected to a (Audinate) Dante network via four card slots that accept more than 30 different Mini-YGDAI cards. In addition to handling formats like AES/EBU and ADAT, it also accepts processing cards that provide capabilities such as Lake processors and Dan Dugan automatic mixing. Each Mini-YGDAI slot includes a sample rate converter (SRC) so that devices operating on different word clocks can be connected without causing noise or signal interruptions. Presets for seven routing patterns are built in, including routing between Mini-YGDAI cards. A rotary switch on the front panel allows direct selection of one-to-one Dante/Mini-YGDAI routing, branched Mini-YGDAI/Mini-YGDAI routing, and other options. **www.yamahaca.com**

Roland Pro AV XI-Expansion

Interface cards that extend the audio and video I/O capabilities of both the M-5000 digital console and V-1200HD production switcher. When used with the M-5000, the XI-SDI, XI-DVI and XI-SFP expansion interface cards add direct audio I/O capabilities for video workflows. The XI-SDI and XI-SFP cards both support 16 audio inputs from embedded SDI audio and outputs at 48 kHz plus video throughput. The XI-DVI card provides two audio I/O channels from HDMI signals at 48 kHz plus a single video throughput. The XI-REAC adds two REAC ports that each handle 40 audio inputs and 40 audio outputs. The XI-Dante card offers a primary and a secondary port, plus a control port, and supports 64 bidirectional audio channels at 48 kHz or 32 bidirectional channels at 96 kHz. The XI-MADI supplies two sets of connections, each offering BNC and optical connectors plus a video sync BNC connector. The XI-WSG adds three 48 kHz- or 96 kHz-capable SoundGrid Ethernet ports to the I/O capacity of the M-5000. **proav.roland.com**

Crown Audio XLS DriveCore 2 Series

Four 2-channel power amplifiers, including the XLS 2502, XLS 2002, XLS 1502 and XLS 1002, with power ratings from 350 to 775 watts per channel at 4 ohms and 1,100 to 2,400 watts at 4 ohms bridged. Harman's DriveCore class D amplifier circuitry combines power output with energy efficiency. Selectable input sensitivity is either 1.4 Vrms or .775 Vrms. A band-pass filter is provided on each channel, enabling more precise DSP tuning. Also included are Peakx limiters for loudspeaker protection; balanced XLR, 1/4-inch and RCA inputs; and touch proof binding post and Speakon outputs. The front-panel LED indicators (except the clip and thermal indicators) can be turned off if desired. A remote sleep mode trigger function reduces power consumption from a distance, and a security setting disables the front-panel menu buttons. All models are 2 rack spaces high. **www.crownaudio.com**

Lectrosonics SSM (Super Slight Micro)

An ultra-compact, full-featured wireless body pack transmitter that joins the company's Digital Hybrid Wireless Series. Dimensions of the all-metal housing are 2.52 x 1.48 x 0.57 inches and weight is 2.3 ounces, including battery. It is fully compatible with all current Lectrosonics receivers as well as several older models. SSM offers a wide tuning bandwidth of three standard Lectrosonics blocks or up to about 76 MHz, depending on the specific frequency range. RF power is selectable at 25 or 50 mW. It is available on four different frequency ranges for worldwide use. Tuning steps of 100 or 25 kHz can be selected in the menu, allowing for up to 3,072 frequency choices. The microphone connector is the industry-standard Lemo 3. An infrared sync port on the transmitter allows for quick setup with compatible receivers. A backlit LCD provides ease of setup in any lighting conditions. A micro-B USB port makes it possible for firmware updates in the field as features are added in the future. **www.lectrosonics.com**

Shure Wireless Workbench 6.11

The latest version includes inventory batch edits, enabling parameters to be modified for a group of devices or channels at the same time. Users can control configuration of naming, color-coding, and other parameters for multiple channels. The coordination workspace enables frequency calculations for subsets of wireless inventory before deployment. These can be saved and recalled for different locations or specific events within a production. A ranked list of compatible frequencies can be manually assigned to desired channels from this workspace as well. A monitoring interface displays all channels with a common channel strip layout. Channel strips can be customized using the Channel Strip Designer and arranged into various layouts. Multiple channel strip arrangements can be stored in views and used to monitor a customized group of channels.

Wireless Workbench 6.11 is compatible with the Shure Axient wireless management network and PSM 1000 personal monitor systems as well as UHF-R, ULX, and QLX-D wireless systems. **www.shure.com**

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FAST FORWARD

Can we be too up to date with technology?

by Karl Winkler

IT'S BECOMING increasingly difficult to keep up with technology. What used to be "merely" hardware now has a software component. Everything has layers, touch screens, and incredible power and flexibility, which is great. But the not-so-great part is that the learning curve is ever steeper.

And while there are standards for MIDI, USB, .WAV, Ethernet, MADI, and many other methods of keeping all this gear connected and communicating, each mixing platform, DSP box and loudspeaker management unit has its own proprietary file structure and operating system. In other words, live sound and all that it entails has become more complex than ever.

Meanwhile, most of us also have smart phones, tablets, laptops and PCs with multiple programs and apps, and using Wi-Fi, LTE, 4G and other ways to access the Internet. And we communicate with each other via SMS, Facebook, Twitter, Instagram, Skype, message boards, e-mail, and phone calls (via cell or good ol' land lines). We set up these devices and services, develop at least a working knowledge of each, and oh, by the way, need to keep track of passwords for all of them.

SHOOTING OURSELVES IN THE FOOT?

Of course, it's important to keep up with these things in a general sense. But how "up to date" does we really need to be? Let's start with the hardware.

What was once the occasional software update has turned into a near bombardment of update notices for our firmware (hardware), software and apps of all kinds. Most of the time, these updates are beneficial by fixing bugs, beefing up security, adding features or generally improving the user interface.

But sometimes, an update can "break" an otherwise working unit. We've probably all experienced this one. For me, it was when I was working on a DVD project on my iMac a couple of years ago and updated to the latest OS. I quickly discovered that not only had they stopped supporting DVD authoring, something else was broken so that my photos were rendered in strange colors. To finish the project I had to borrow a friend's Macbook that hadn't received the update.

And that's when I learned a valuable lesson (again): the latest version isn't always the best. In fact, it was basically folly on my part to do a major OS update like that during the middle of a project, and if I'd done any research beforehand, I would have known to expect problems.

DELIBERATE OBSOLESCENCE?

Now, not every manufacturer deliberately obsoletes its equipment or decides to negate interoperability with other equipment, but it does happen. And in our business, having the show go on is key. In fact, it's everything. To paraphrase the late Albert Lecesse of Audio Analysts, "priority one is to make sound, and priority two is to keep making sound." Sure, that fancy vocal microphone or "black box" processor may be really nifty, but is it suitable for

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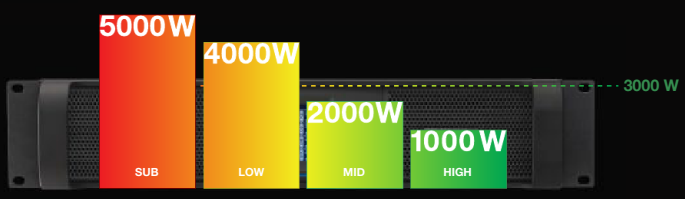

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the tough, demanding world of touring sound? Or even for the perhaps less demanding but still critical world of worship sound? One reason we're still using some "classics" in our daily work is because they keep working, day after day. Anything delicate, fussy, unreliable or otherwise non-conducive to this consistency will end up getting shelved in favor of devices that always work.

WHAT'S THE POINT?

There's another aspect to this, though. The best way I can describe it is that some of us tend to be, well, addicted to the latest technology, apps, social media or whatever it might be that's new. Sure, it's cool to be up on these things but where should we draw the line between professionalism and "cutting-edge" just for the sake of it?

Clearly this is a personal question, but sometimes we don't ask it enough. Certainly we have to know about the platforms on which we work, including the standard plug-ins, how to do updates, and what the critical updates are for our systems.

But beyond that, it's not usually wise to get caught up in an endless cycle of keeping up with the Joneses. It's one thing to be perpetually hip when you're a free-wheeling college student, but very much another thing when you're tasked with the responsibility to deliver quality results, night after night, with a complex and powerful sound reinforcement system.

What most training manuals for technical equipment said in the past was along the lines of, "Learn the basics then choose just a few of your favorite settings, and use just those settings until you get really good at it." This still applies today – if we can't be fluid and confident with some of the basic setups using cutting-edge equipment and systems, we simply need more practice. Forget the fancy stuff, at least at the beginning of the learning curve, and master the fundamentals.

POTENTIAL OR RESULTS?

One thing that I've found annoying over the years is an emphasis on potential at the expense of actually producing something of value. In other words, "Wow – look at all of the amazing features and all of the other whiz-bang things in this new box and what we can do with it!"

I love technology, ideas and "what ifs" as much as the next person (and maybe more). But in the final, clear-eyed analysis, we need to be producing consistently good results. This is what it means to be professional.

Where should we draw the line between professionalism and "cutting-edge" just for the sake of it?

The "wouldn't it be cool if" statements must be grounded in reality, and experience is what gives us the ability to determine what might be feasible or productive and what might not. It's one reason the old salts in pro audio are so hard to impress – they've seen it all and then some. And the best of them are the first to acknowledge something useful and beneficial – after they see it proven.

To me, the bottom line is that it's important to keep up and stay savvy about the latest technology, while at the same time keeping an eye on the real prize: productivity and professionalism. It is still – and will always be – the best path to success and to advancing our careers. ■

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A perspective of the stage and house system deployed for "One Last Tour."

Barry Manilow
One Last Tour

A conversation with engineer Ken Newman

by Mark Frink,
photos by Steve Jennings

Barry Manilow performing on the current tour with a Shure KSM9.

How does a world-class front-of-house engineer support a fastidious megastar who is very particular about every aspect of music production making his grand exit to touring? Barry Manilow is back on the road for "One Last Tour," and this time he really means it. Veteran engineer Ken Newman, who began working with Manilow in 1992, is mixing for the first time on a DiGiCo SD5 console and a Martin Audio MLA loudspeaker system, managed by Delicate Productions system engineer Phil Reynolds.

Will Miller, who Manilow "borrowed" from Josh Groban, mixes the artist's monitors on a dual-surface Yamaha PM1D console, while Francois Paré handles the band's monitors from the console's second surface via Aviom personal mixers and an assist from Tony Luna on stage.

Manilow is easily one of the all-time top-selling pop artists, with 25 consecutive Top 40 hits between 1975 and 1983 and worldwide sales of more than 80 million albums over his 4-decade career. Last year's My Dream Duets debuted at number 4, making it his 15th Top 10 album.

I caught up with the "One Last Tour" in San Jose at the Shark Tank (officially called SAP Center) for a conversation with Newman about his role at the helm and the unique combination of best practices that he's built for Manilow. He started out working for a local New Jersey sound company, coming off the road for Atlantic City house gigs at Resorts International and then the Sands, but then went on

tour with Shirley MacLaine for several years followed by a string of other top tier A-1 Audio clients like Ann-Margret, Liza Minnelli, Engelbert Humperdinck, Tony Orlando and Paul Anka.

Between mixing gigs in 1991, A-1 Audio sent him out as system tech for several of Manilow's "Showstoppers" dates, with Paul Dalen mixing. When the "Greatest Hits" tour was organized the following year and Dalen wasn't interested, he suggested Newman for the role. And the rest, as they say, is history, with Newman mixing Manilow's live shows for a dozen years until 2004's "One Night Live! - One Last Time!" in-the-round arena farewell tour.

On breaks between Manilow tours, he also mixed Chris Isaak, Julio Iglesias, Anita Baker, Stevie Nicks, Burt Bacharach and Sergio Mendes, and during Manilow's seven years of Las Vegas residencies at the Las Vegas Hilton and Paris Las Vegas, he worked for corporate staging vendor WorldStage. When Manilow went back on the road in 2012, he called on Newman once again, and now he's serving this final arena outing.

The Same Everywhere

The main loudspeaker arrays provided by Delicate Productions (Camarillo, CA) consist of 11 Martin Audio MLA modules plus a single MLD down fill cabinet per side. Side hangs are comprised of a dozen MLA Compact modules per side. Three MLX dual 18-inch subwoofers per side are used in a cardioid configuration, and front fills are W8LM mini line array enclosures.

"The biggest challenge on the Manilow show has always been gain-before-feedback on Barry's vocal mic, because he's not comfortable putting the mic close to his mouth, and he's not the loudest singer I've ever worked with. Those factors, combined with his desire for every aspect of his dense arrangements to be heard by his audience while keeping his vocal well on top of the mix, combine to make gain on his vocal mic a constant challenge.

"Barry also frequently talks about how the entire audience should hear everything, with no one being offended by the show being overly loud, but at the same time by the show being loud enough to be exciting and moving. So I guess you'd call that the challenge of even coverage."

MLA is a marriage of loudspeakers and software, where transducers are individually driven and optimized to deliver the summation of clear sound at the ears of the audience to meet the goal of even and smooth coverage across the entire coverage area. The system does this by controlling EQ and phase for individual transducers after modeling the physical listening area. After the DISPLAY2 software's optimization procedure is complete and the individual presets are loaded into each cabinet via the U-NET network, the system is 98 percent tuned at turn-on.

"The MLA system allows me to spend less time walking the room because I have confidence that coverage is uniform throughout. That allows me to concentrate on my main job, which is mixing the show and making sure the mix is everything Barry wants it to be, by listening to the previous show's multitrack file and refining the mix as if it was the actual show. Through headphones, of course, so as not to bother the rest of the crew that's working on setting up the show.

"MLA doesn't need much in the way of tuning, because we've already entered our target curve into the system's software. Optimization mainly consists of adjusting the delay times of the side hangs, subwoofers and front fills. It can be done in as little as 15 minutes, or if I want to really nitpick, it can take up to an hour.

"Also with MLA, the level from the front to the rear of the seating area changes considerably less than with a traditional speaker system, and the tonal quality is significantly more uniform. Those are two of the main aspects of MLA that I really love. Knowing that I'm not scorching people in front of me with up to 10 dB more level than

One side of Martin Audio MLA arrays flying on the Manilow tour.

at the mix position is very comforting. And knowing that people in the 'cheap seats' are getting the whole mix and at a comfortable listening level is so great."

Delicate Productions president Jason Alt adds, "It's a real privilege to be a part of Barry Manilow's 'One Last Tour.' I worked as a system tech with Ken on Manilow at A-1 Audio, and I know MLA gives him exactly what he has always wanted out of a system: great sound, accurate representation of the music, and what only MLA can deliver, control of a system that adapts to the environment."

Ken Newman (right) with Delicate Productions system engineer Phil Reynolds at the tour's DiGiCo SD5 console.

Something Old, Something New

The system has proven to be a good fit for his time-honed approach. "For many years, the zones I send to the drive rack are left and right outputs for the mains, mono for the side hangs, and front fill for the first several rows. I'm an old (Yamaha) PM2000 guy and that's where I defined my preferences," he notes. "These days I use five stereo groups mixed to matrix outputs: vocal, background vocal, band, rhythm, and playback. This allows me to adjust front fills to have more vocals, less rhythm or whatever's needed, and to insert processing or adjust parameters like EQ or compression on those subgroups.

Will Miller (above) and Francois Paré, who mix monitors on a dual-surface Yamaha PM1D console.

"When I first started working for Barry, he'd been through a few microphones and was using a Shure SM87 on a Vega wireless. One day Harold Blumberg (a friend and colleague) told me about a new mic that Shure would be coming out with, and offered me a wireless system with it. It was the Beta 87, and it made getting the sound Barry wanted on his voice much easier. We used the Beta 87 until I stopped working on the show in 2004, but when I came back at the end of 2011, I thought it would be good to use the latest, greatest mic Shure had to offer so we switched to the KSM9. I'm looking forward to trying the KSM9HS for hopefully more gain before feedback."

Newman is mixing on a DiGiCo SD5 for the first time. "When I came back in 2011, the last console I had used on Barry's show was an Amek Recall that was no longer supported, so I needed to switch to a newer digital console. I went through a lengthy process selecting the right digital console for my situation. The Soundcraft Vi6 fit that bill exactly, and turned out to be a fantastic console. At the end of 2014, when there was talk of another tour with additional inputs needed, I felt that despite the fact that I really loved many aspects of the Vi6, I had maxed out some of it's capabilities.

"There are so many great DiGiCo features, it's hard to say which ones are my favorites. The fact that I can put any channel anywhere on the surface and the ability to make any strip stereo. How great is that! For this show, it doesn't get much better. And then there's the fact that the fader banks are 12 faders (as opposed to 8) each, and they're switchable to different layers independent of each other. And how about that multiband compressor? I could go on and on.

Newman's personal outboard rack has a TC Electronic M5000, an Eventide H3000S and a TC M-One. "I learned long ago that making this show sound the way Barry wants is largely about reverb and effects. When I started, we had a rack of Lexicon PCM70s with my PM4000. Gradually I was introduced to different effects units over the years that I came to rely on as part of 'the Barry sound.' When I came back to the show at the end of 2011, I wanted to start with that basic palate of effects so that with both a different console and a different speaker system from what I had used previously, at least the effects were the same.

"I use one engine of the M5000 for

Barry's main vocal reverb and the other for strings, guitar or percussion reverb. The M-One is for drum reverb. This is really key to the show's sound. The H3000 is for Barry's vocal double and effects. I have a number of presets in each unit that are selected by MIDI output from the SD5 per snapshot. Some snapshots are only firing MIDI changes, to change, for example, the drum reverb, which is mostly snare, at the end of a song."

Innovative Approaches

Newman relies on a BSS Soundweb 9088 to solve two problems. "I'm a big fan of having my record mixer be separate so that it's completely independent of the main console. In the old days, we took a BSS DPR804 dual delay line, a little Yamaha M406 mixer and a Shure VP88 stereo mic to make show recordings with other artists like Paul Anka. Barry really wants the best representation of what it sounded like in the room, complete with room ambience and audience, so I simply continued using the same recording setup.

"During the seven years I wasn't working for Barry, I worked for a company that does primarily corporate events, and they used Soundweb as system processors with their analog consoles, so I got quite familiar with their capabilities. When I returned at the end of 2011, I combined the functions of the DPR804 and M406 into four channels of Soundweb. I have both a 'With Mic' recording output and a 'Board Mix' output, and also add a taste of compression and EQ as well.

"I also realized that when Barry's audience gets loud, it would be nice to mix on nearfield monitors, so I designed a little Soundweb configuration to take the same stereo mix feeding the main clusters and delay them and EQ the (Genelec 1032) nearfields to closely match the PA above 100 Hz, and now I

Newman's personal outboard effects and insert rack.

can hear more closely what's happening with the mix. I also want to listen to the console's solo output, and my personal preference is that the solo is not delayed, so the solo output of the console comes into two more inputs on the Soundweb.

"I built a custom GPI remote control that can switch between 'mix' and 'solo,' and set the delay and EQ each day in a control panel in Soundweb. I also added a level control and a mute switch on the remote, complete with a footswitch jack so that if I'm really getting into the mix on the nearfields, I can step on the footswitch and mute the them to check the room's sound.

"Barry wants what I call a 'With Mic' recording of each show, so I make a CD for him (on a Tascam SS-CDR200) as well as recording the same mix onto a USB stick so I can hear what he listens to. The same mix is sent to the video guys, who record it onto his show DVD. I also make a board mix recording onto a USB stick for the musical director, and I make a simple 4-track Pro Tools recording on my Mac Mini, with the board mix on two tracks and the VP88 stereo mic on the other two tracks. This way I can make a better-sounding mix of the show later, varying the level of the VP88 as needed. I also make a multitrack (Pro Tools) recording of each show for archival use and virtual sound check."

Extra Gain

Newman's outboard rack also includes a Shure DFR22 and a Rupert Neve Designs 5045. "In the ongoing quest for maximum gain-before-feedback, I've always been a fan of parametric EQ, so I insert a DFR22 into the stereo vocal sub-group that includes the vocal mic as well as its reverb and effects.

"I then 'ring out' the vocal mic along with its reverb each day, and hopefully get as much gain before feedback as possible. I call the 5045 a 'magic box' that gets me a few more dB of gain before feedback. I ring out the mic with the 5045 punched out, and then I just dial in a few dB of 'depth' on the 5045 for just a bit more gain before feedback. Magic!

"The Summit DCL-200 is another item that I used with an analog console that carried over to the new setup. Since I need so much gain on Barry's vocal mic, I like to get just a few dB of that gain from the DCL-200, along with a very subtle amount of compression when possible, while I'm at it."

With pressing matters at hand as show time approached, we wrapped it up, with Newman concluding, "There's a show scheduled June 17 at the Barclays Center in Brooklyn, where Barry grew up, on his 72nd birthday, so that should be quite the event. It's also currently shown as the last stop on the tour. I haven't yet been to the Barclays Center, but I hear it's quite large, so I look forward to MLA doing its thing and bringing the happiness of Barry Manilow to all of the audience members equally." ■

MARK FRINK is a regular contributor to Live Sound International and is available to mix IEMs this summer. He can be reached at livesound@markfrink.com.

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HOW NOT TO RECORD A LIVE ALBUM

A case study in breaking the "golden rules" of recording.

by Andy Coules

WHENEVER WE RECORD music there are certain "golden rules" that aid us in getting the end results we require. These rules are based on years of experience and have been handed down from engineer to engineer as a useful guide to the best practices. While they're not rigid, experience has taught us that it is wise not to break them if we hope to produce a recording of the required standard (i.e., to be released).

When recording a live concert we tend to rely heavily on these rules because we're no longer in the controlled environment of the recording studio. The needs of the recording are secondary to the immediate requirement of running the show and providing a pleasing live experience for the audience, not to mention the fact that there's only one take.

For a few years now I've been working as front of house engineer for a band that produces vibrant and energetic live shows, so I suggested trying to capture that energy by recording a live album. An upcoming European tour presented an opportunity but there was no budget available so I had to borrow what equipment I could in order to achieve this aim. The process caused me to pretty much break every one of the golden rules, yet the result was a releasable album.

Making The Math Work

The band consisted of seven musicians playing 10 instruments: full drum kit, bass guitar, acoustic guitar, electric guitar, charango (a 10-stringed South American instrument traditionally made from the shell of an armadillo), strum stick (a three-stringed instrument designed to be simple to play), violin, melodica, trumpet and clarinet.

Combine all of that with two additional floor toms (used for the encore) and three vocals, and there was a grand total of 27 inputs. We had a Korg D888 (eight-track, eight-channel, eight-bus) hard disk recorder, and this is where I broke my first rule...

RULE 1: Always record all inputs separately. Due to the inherently fleeting nature of live performance it's wise to record every single stage input on a separate channel because this enables maximum flexibility when it comes to treating and blending the individual elements into the final mix.

It also provides the ability to repair any mistakes and even overdub replacement or additional parts – believe it or not, very few live albums are "warts and all" unaltered depictions of the actual event. In the context of a live concert any wrong notes or mistakes are fleeting and soon forgotten, all part of the immediacy of live music. But if those same mistakes are captured on a live recording and replayed multiple times they quickly become glaringly obvious, jumping out every time you listen to it. So the ability to separate individual instruments and repair, if necessary, can be vital.

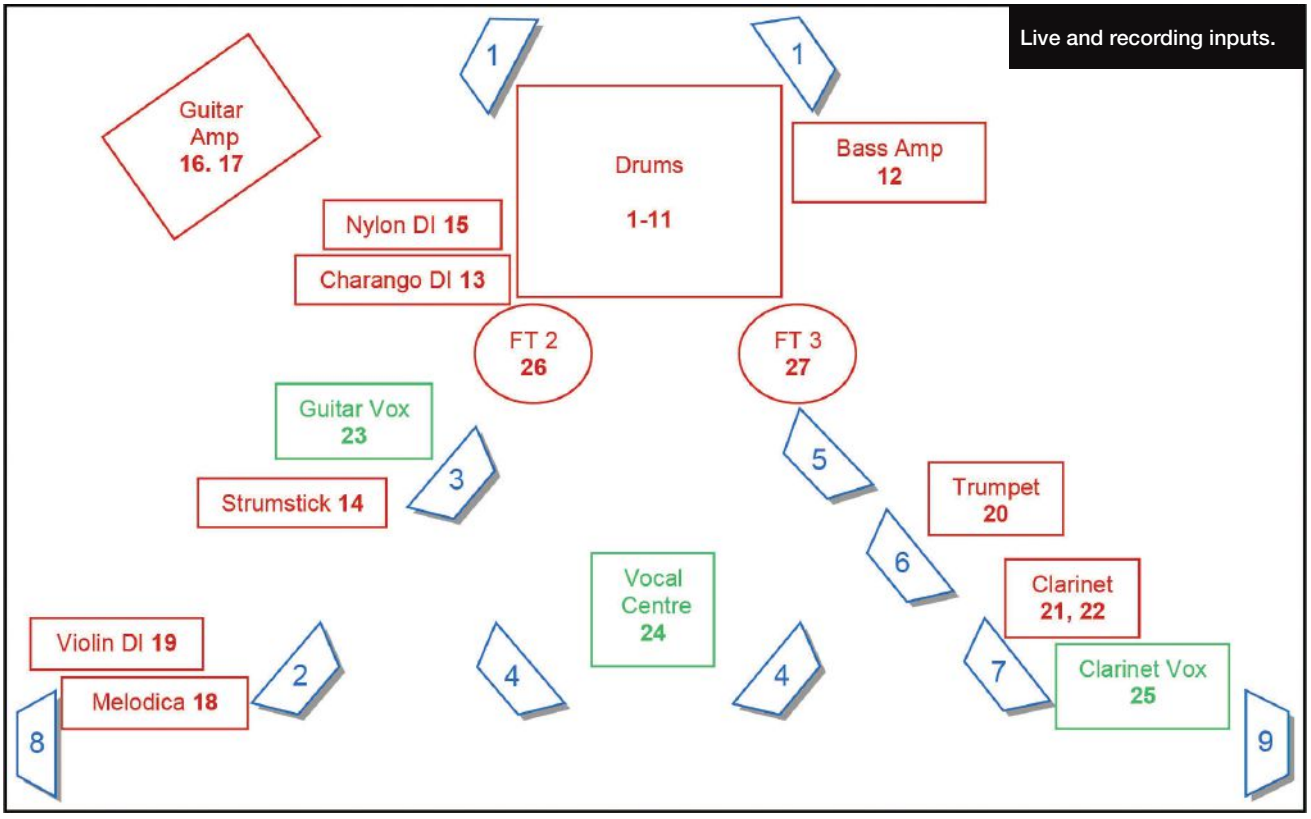
However, the recorder afforded no such luxury. I had to find a way to get those 27 inputs down to eight. The key was the realization that while there were multiple instruments on stage, there were still only seven musicians, and while they're a talented bunch, none of them have yet figured out how to play two instruments at once (Figure 1). All I had to do was assign one channel to each musician (while separating the lead and backing vocals), which resulted in the following track listing:

1 - Drums
2 - Bass
3 - Guitars (acoustic, electric, charango and strum stick)
4 - Violin (plus melodica)
5 - Trumpet
6- Clarinet (top and bottom microphones)
7 - Lead vocal
8 - Backing vocals

The hardest decision I had to make was to record all of the drums to one channel. This would be a major issue in the final mix, but there was simply no choice. A drum kit comprises multiple elements that cover the widest frequency range of just about any instrument (with the possible exception of the pipe organ). It's also a physically wide instrument that benefits from spatial separation of the individual elements via panning, an option which would now b e denied because I was recording all 11 inputs in mono on one track.

RULE 2: Always record the raw input signals. The common practice when recording a live show is to use a splitter to take a complete copy of all of the stage inputs and thus record them before they're relayed to the front of house/monitor consoles. This ensures capturing the raw signals unaffected by any processing employed for the live mix.

The reason for this rule is that the demands of the recording mix are quite different to those of the live mix. The live mix is comprised of those elements required to reinforce and enhance the ambient sound coming from the stage to



Live and recording inputs.

create a clear, precise and above all visceral experience for the audience. The recording mix is much more about presenting a coherent sound that attempts to reproduce the energy and excitement of the concert that will withstand repeated listening.

The live mix is also designed to work on a specific sound system in a particular room, whereas the recording mix needs to take into account the fact that the end result will be listened to on a wide variety of reproduction equipment in various environments.

The requirement of recording to eight tracks meant a splitter was not an option. For the three tracks with just one input (bass, trumpet and lead vocal), I was able to use the desk channel direct outputs to record an unprocessed signal. For the other five tracks, there was no choice but to use sub groups to combine the channels I needed, which meant those signals would be post everything (i.e., inserts, EQ and fader).

This required that I be acutely aware at all times of how any of the processing would affect the recording mix, and to try to seek a compromise between the demands of the two outlets. Thankfully my methods for applying EQ, gates and compressors are quite subtle so I was confident this would work; the key was being careful not to make any large or sudden moves.

RULE 3: Always employ redundancy and have a back-up recording system. You only get one take at a live recording, so any equipment failure that causes the recording to fail is catastrophic. A second recorder, ADAT machine or laptop can provide a handy back-up should the primary device fail, and this redundancy should be considered a vital part of the recording set-up.

However, this option wasn't available on this project, so I decided that the best option would be to record all 15 shows on the tour and hope that the best performances could be compiled into a coherent album.

The Korg D888 recorder used on the project.

RULE 4: Adequately capture the audience/ambient sound. Hearing the crowd respond to the live performance really helps the listener to immerse themselves in the playback and imagine they're present at the concert. Ambient microphones will also pick up some of the sound of the band and PA system propagating in the room which, when added to the dry multi-channel recording, can really add life and glue the whole recording together.

But because the recorder didn't have any additional tracks available, I came up with the simple solution of recording the audience sound separately using a stereo microphone plugged into my laptop. This meant I would have to manually synchronize the two recordings in the mix, but there really wasn't any other choice.

Necessity & Invention

By far the biggest challenge in the mix was to inflate that single track of mono drums into something resembling a fully fledged drum kit – it would sound weak if the drums weren't pumping.

The solution was to duplicate the drum track four times and EQ each of the copies differently: one low-passed to bring out the kick, one band-passed to bring out the snare, and a pair high-passed to act as overheads (one of which was delayed a small amount to create a pseudo stereo effect when panned). This enabled me not just to modify and mix the key elements of the kit, but also to apply reverb only where I wanted it.

The tracks featuring multiple inputs, which I had been forced to record post fader via sub groups, actually presented few problems. The fader moves done live translated well to the recording mix, balancing the instruments in the same way I had done live. Any wayward fader movements were easily corrected with fader automation.

The decision to record multiple shows in lieu of having a back-up system resulted in many hours of recordings to trawl through to find the best performances, but fortunately there were no issues with the equipment and we managed to capture every note of every show.

Synchronizing the audience tracks with the eight-track recording proved to be fiddly but ultimately straightforward. In most cases there were stick hits on the drum tracks that could be used to visually align them. I then fine tuned the alignment by panning the audience recording to one side and the eight-

track recording to the other so that when I listened on headphones it was much easier to identify and correct any timing discrepancies.

About halfway through the project I needed to go out on tour again but the mix needed to be finished, leading me to break one last golden rule.

RULE 5: Never mix exclusively on headphones. When we listen to music via loudspeakers, our ears don't just hear the signal from the nearest loudspeaker(s), they also hear signal from the further loudspeaker(s), which is slightly obscured by the acoustic shadow of our head. There are also reflections from the walls, ceiling and floor that combine in a complex way to create what we consider to be a natural sound.

But when using headphones, the left ear only hears signal from the left loudspeaker, and vice versa, which can make it harder to judge the sound and positioning of individual elements as well as the mix as a whole. This is why it's generally considered a bad idea to mix exclusively on headphones.

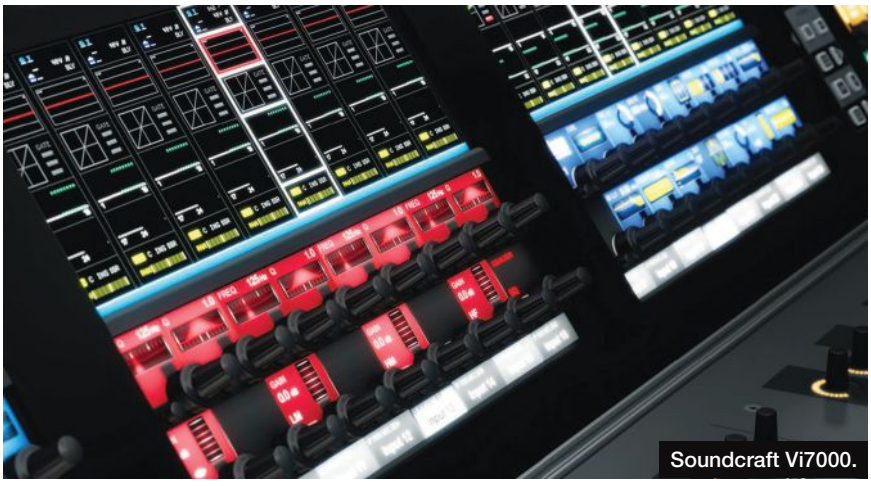
The requirement to finish the recording mix while on tour meant I had no choice but to use headphones. Therefore, I had to trust the EQ and panning decisions made when mixing on loudspeakers, using them as my reference points for making any further changes to the mix. Every time I made a major change, it was saved as a different incremental version so that I could constantly reference back to the original to make sure I wasn't straying too far from the path.

Different Path

The series of decisions forced by equipment restrictions ultimately resulted in spending much more time on the project than if I'd done it "by the book." A healthy combination of stubbornness and perseverance helped in achieving the desired result of producing a releasable album, but this project would not have been possible without modern hard disk recording technology and DAW software.

While I enjoyed the challenges, this is not a path I recommend. Let it be an object lesson in how not to record a live album.

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


Soundcraft Vi7000.

CONSTANT UPGRADES

Conveniently adding further capability to digital consoles. ***by Craig Leerman***

ONE OF THE NOTABLE aspects of newer digital consoles is that they can be upgraded, often to a significant degree, via new software and firmware that's usually available as a convenient download. It's a great way to garner even more capability without the need to invest in new console (or often, additional hardware), which is always a most welcome development.

There's been a significant amount of activity of late in this regard, and we thought it useful to offer a round-up of what's new, as well as perhaps overlooked upgrades, with digital console software and control app capabilities. As always, we encourage you to do further investigation, particularly on ProSoundWeb, where further details and direct links for download are provided. **New Allen & Heath** GLD firmware version 1.5 provides DEEP plug-in architecture that allows users to select from a number of different processing units on every input and mix channel. Two new RMS-VCA inspired compressor models, the 16T and 16VU, are also included. Integrated within the mixer's channel processing, all six compressor models can be selected on any of the input and mix channels on the fly, without burning valuable FX slots or adding latency.

In addition, Chrome firmware includes several additions to the onboard FX suite, including a new Stereo Tap Delay, with 2.7-second maximum delay time, split L/R beat fraction control, millisecond mode, and tap tempo functions. This is joined by a new Bucket Brigade delay emulating the non-linearity of solid state delay units, and Echo, an emulation of the classic tape echo system popular in the 1970s.

Recently released version 3.0 firmware for **Yamaha Commercial Audio** CL and QL Series consoles adds a new 8-band parametric EQ and real time analyzer. Specifically, an 8-band PEQ in the GEQ Rack and Effect Rack make it possible to select 8-band parametric EQ in the GEQ RACK and EFFECT RACK. Version 3.0 also provides four banks of enhanced user defined keys, 5.1 panning and monitoring for surround broadcasts, and a newly developed bus compressor for insertion in the stereo mix bus. In addition, Dan Dugan Sound Design automatic mic mixing already included in QL is now also provided for CL Series consoles, and because turnabout is fair play, Mix Minus features previously only available in CL are now also provided in QL consoles.

Meanwhile, the new StageMix v5.0 control app for several Yamaha Commercial Audio consoles includes a 61-band real-time analyzer that receives input from the built-in iPad microphone. This function is integrated with the PEQ/GEQ displays, allowing the engineer to move around the stage while checking for problem frequencies at various locations as well as use PEQ or GEQ to make appropriate adjustments on the spot. And new v2.0 for R Remote I/O racks provides built-in Dante networking, and the GUI has been revised to allow numeric entry of gain values as well as generally smoother, more efficient operation.

The new **Soundcraft** Vi7000 and Vi5000 digital consoles provide a wealth of capability, with Vi's patent-applied-for VM2 radio microphone status monitoring feature providing native control and monitoring of Shure ULX-D and QLX-D digital wireless systems as well as AKG DMS800 and WMS4500 sys-


Allen & Heath GLD Chrome.

Yamaha QL Series.

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Packed with exciting new features, the highly anticipated V3 software for our popular CL and QL Series digital mixing consoles is available now.

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- Dugan Automatic Mic Mixer for CL Series
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V3

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CL SERIES
DIGITAL MIXING CONSOLES
The standards stay, but innovation never ends.

DIGITAL MIXING CONSOLE
QL SERIES
QL5/QL1

YAMAHA
commercial audio

tems. This new integration with the console enables automatic Shure device discovery, identification, and mapping of each wireless system to the appropriate mixer channel. When that channel is selected on the console, all essential wireless parameters are displayed.

It also enables live monitoring of the channel's RF and audio metering, with the ability to adjust receiver gain from the console, much like trim adjustment for a wired mic. In addition, battery life status for re-chargeables and standard AA alkalines is supported. Both Shure systems use Ethernet connectivity to deliver system data to the consoles. Along with the new Vi5000 and Vi7000, a subsequent software update will support Shure wireless integration, including the current Vi1 and the Vi3000.

Along the same lines, new **Yamaha** TF mixers also provide a range of input and output channel presets created in cooperation and consultation with mic manufacturers such as Audio-Technica, Sennheiser and Shure. The input channel presets are made to match a span of musical instruments and voices, covering parameters such as head amp gain, EQ, dynamics and much more, right down to details like channel name and color. Output channel presets include parameters optimized for in-ear monitors and Yamaha powered loudspeakers.

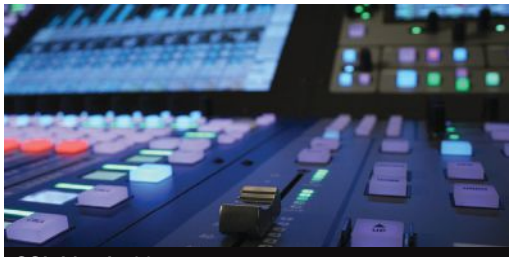
Soon to be released (Q2, 2015) Remote Control Software (RCS) for the now-shipping **Roland Pro AV** M-5000 console provides operation of the console from a computer (Mac/Windows). Connection can be made via USB or remote connector, allowing operation over a LAN. The GUI for the M-5000 RCS allows multiple windows, and includes support for high-resolution displays and other optimizations. This enables use of a second display for viewing even more windows, such as a large meter view of inputs and outputs.

The dedicated M-5000 Remote app (also due Q2) supports remote control from an iPad, with three methods of connection offered. When using the console's dock connector for iPad attachment, the iPad can be used to perform 2-channel recording and playback, and input sources and output channels can be assigned as desired. The GUI for the app offers full support of Retina displays for crystal-clear graphics.

The new **DiGiCo** S21 console offers exceptional capabilities, and this bodes well for future upgrades as well. Using a high power QuadCore SoC, associated with high bandwidth memory, the S21 connects to a low-power 484-ball array FPGA which in turn connects to fourth-generation control SHARC DSP, capable of not only controlling the FPGA, but offering the potential for additional processing in the future.

New HD mode firmware for **PreSonus** StudioLive AI-series mixers make them capable of recording and playing back audio streams at sample rates up to 96 kHz over the onboard FireWire 800 interface. The number of recording and playback streams is unaffected. HD mode limits cascading and output processing while retaining bus mixing, Fat Channel processing on every input channel, and one reverb and one delay processor. Because included Capture and Studio One Artist for Mac and Windows already record high-definition audio, no updates to these applications are required. In addition, the company's Nimbit online, direct-to-users marketing, promotion,

PreSonus StudioLive AI-series.


SSL Live L500.

and sales service now supports high-definition (24-bit, 96 kHz) audio.

The just-unveiled **Avid** flagship VENUE | S6L live mixing system utilizes the E6L engine to handle huge channel and plug-in counts at very low latency. All processing is at 96 Hz, with support for higher sample rates due to the amount of processing power available. Onboard plug-in processing is greater than previously available, and Pro Tools integration provides streamlined recording and playback functionality without the need for a separate audio interface. The new system is powered by the same VENUE software as other Avid live sound systems. I/O can be shared across multiple networked systems with Avid True Gain advanced gain tracking technology.

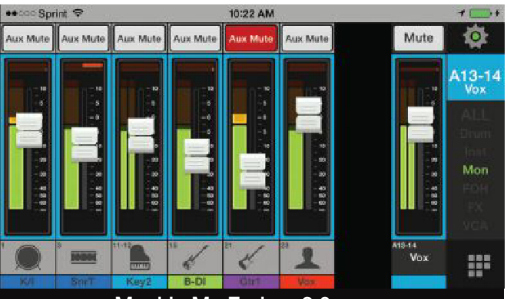
Recently released V3 software introduces more than 40 new features and updates for the **SSL Live** console range. With the software update, the L500 console increases from 192 mix paths to 256 with a doubling of effects processing power (depending on the effects selected). The L300 also increases from 128 to 192 mix paths. V3 also includes remote control software, console expander mode, new effects, enhancements to the solo system, user interface changes, an optional Dante card and Super Q. Super-Q is the next generation of the Query function, offering workflow flexibility from the touch of a single button.

In addition, new SOLSA (SSL Offline Setup Application) software allows creation and editing of SSL Live console show files on a laptop, desktop or tablet PC. Almost anything that can

be done on a console can be manipulated and configured using SOLSA.

Mackie recently announced the availability of new Master Fader v3.0, providing significant upgrades for the new DL32R wireless digital mixer with iPad control as well as DL1608 and DL806 mixers. New features include the addition of four subgroups and four VCAs. Subgroups can be stereo-linked and provide dedicated processing, while VCAs offer flexible control over groups of channels. Users can dial in the mix and get single-fader control over groups like drums, guitars and more. Also new is the overview screen, which delivers at-a-glance information for all input and output channels. In addition, digital trim has been added to each DL1608/DL806 input.

Also new from Mackie is My Fader v3.0, an upgraded version of the control app for its digital mixers. My Fader v3.0 allows on-stage performers to control their own monitors and provides engineers with



Mackie My Fader v3.0.

quick mobile control over a mix. The new version is headlined by a new updated user interface and an expanded feature set for additional control.

The **QSC** TouchMix Series just got a firmware upgrade to version 2.1 that includes multiple languages. In addition to English, the mixer's built-in info system and demo screens now include user selectable options for Chinese (Mandarin), French, German, Russian and Spanish. In addition, both TouchMix-16 and TouchMix-8 now offer password protected multi-level security access, programmable user buttons, expanded channel presets, and the ability to assign aux buses to the left and right main outputs, allowing the auxiliary to function as a sub group.

The TouchMix Series also now offers expanded Wi-Fi options (including wired connection to an infrastructure router) and an updated iOS app for remote control as well as personal mix apps for both iPhone and iPod Touch, which the user may limit or allow on a per-device basis. The new software also offers iPhone control of the record/playback transport controls, with Android support announced for the future. ■

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

Dan Housel at his Avid VENUE Profile console on the recent European leg of Lionel Richie's latest tour.

Purposeful Evolution

Refined monitoring for Lionel Richie on tour

by Kevin Young

Over Dan Housel's two-year stint as monitor engineer for Lionel Richie, he's sought constant improvement in presenting the iconic singer's classic, distinctive vocal signature. We spoke recently, just prior to the final show at London's O2 arena on the European leg of Richie's latest tour, about what he's been up to lately in that regard.

"I mix on digital consoles – I'm an Avid Profile user – but having learned on analog gear and having spent time working in the studio, there are certain things I miss about analog, a certain 'flavor' lacking on digital preamps that analog pres have," he notes.

It's something the LA-based engineer says that he's come to notice while running monitors for other acts in his 15-plus years in the business, including work with blues guitarist Keb' Mo', Filter, Pitbull and Maynard James Keenan's bands Puscifer and A Perfect Circle. During Richie's gigs, particularly on fly dates, Housel continues, he started considering how to add that same type of flavor, not only to better the overall sound for Richie and his band, but to allow him to compensate at times when Richie's approach to performance and sonic considerations collide.

EVALUATING OPTIONS

"We sometimes have struggles," he says, explaining that at times Richie experiences difficulty finding himself in the mix the way he wants, which is a function of his desire to connect with the audience. "He's a Shure SM58 user, which is – for lack of a better description – a very active microphone on stage," Housel continues. "He moves the mic around a lot, from in front of his face to down by his chest. But when I asked him about it, he looked at me and said, 'Baby boy, they don't want to see the microphone, they want to see my face'."

Housel mitigated the problem by backing off on reverb and turning on downstage wedges to provide additional presence to compliment Richie's in-ear monitors, and he also began looking for other ways to add more definition. Over time, he's floated a variety of other solutions with fellow engineers, including Avalon 737 mic preamps that he uses on Keenan's vocals during A Perfect Circle gigs while mixing on an Avid Venue SC48.

"The 737s are fairly commonplace, but I needed two channels because Lionel uses two microphones during performances, a hard-wired 58 at the piano and a wireless 58," he says, noting that carrying two of the units wasn't

a compact enough solution and adding, "They're a bit sensitive for travel."

The search also included BAE 1073 and Midas XL42 preamps. "I've seen the XL42 used for the same application, but they're discontinued now and the few people who are willing to part with them are asking a handsome price, so I did more research. The name Radial came up during a discussion with one of the Puscifer guys, who suggested the company's 500 series preamps."

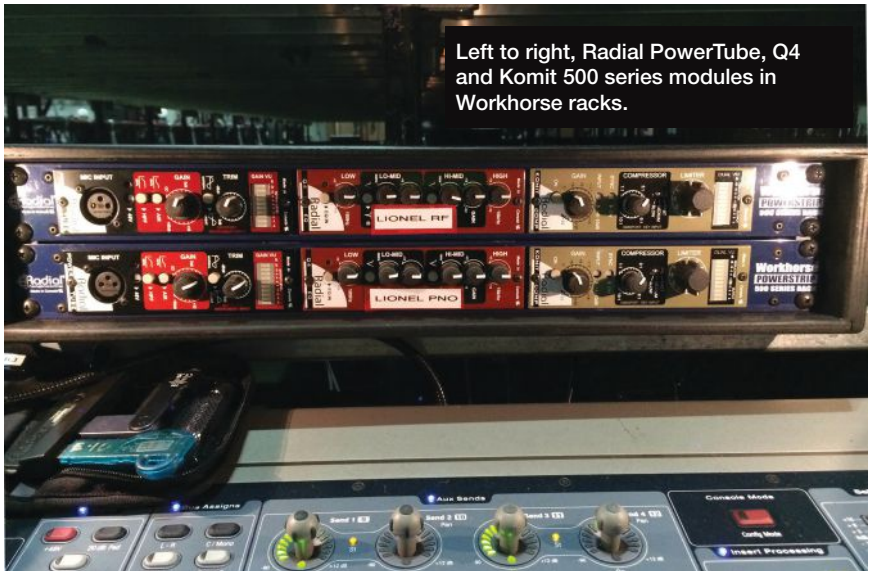
Already a long-time user of Radial Engineering direct boxes, Housel reached out to company president Peter Janis, who he'd met during his own time as a manufacturer's representative. "I tested out a Radial PowerPre along with a Q4 four-band EQ and Komit compressor/limiter during pre-production for the European leg," he says. "I could definitely hear that there was more separation compared to the console preamps, but I was still considering the more sensitive tube gear. I was sitting in my hotel with the PowerPre chassis in hand, researching the PowerTube and thinking that, if they're built the same as the PowerPre, they're going to be all right."

DEPTH & DETAIL

As it's name implies, the PowerTube incorporates a 12AX7 tube drive and a Jensen Transformer input. "To me it sounded like the first time I listened to a hi-def audio recording," he states. "It was just next-level separation, depth and detail. I also added the Q4, but I don't use EQ within Lionel's vocal signal chain, so that's bypassed. Again, he moves the microphone around a lot, so if I EQ for one mic position – at the chest or higher up – when it's in the opposite position, I'm down the river."

The 500 series rack form factor was another plus in addition to the sonic characteristics and durability, and as a result, dual sets of PowerTube preamp, Q4 EQ and Komit comp/limiter modules have achieved the desired result in Radial Workhouse 500 series racks that occupy just 2RU.

Richie noticed a difference immediately, Housel notes. "He pulled his in ears out, stopped the band, got on the mic and asked me what I'd done differently. I told him we spent a little money to make his vocals sound better and he started singing alone, then stopped and said, 'I'm gonna sue your ass for not having these before'."



Left to right, Radial PowerTube, Q4 and Komit 500 series modules in Workhorse racks.

Housel details the overall vocal chain for Richie: "We go out of the 58 and hit the split. Then rather than running into the stage rack for my Profile, we hit a cross patch that's also carrying all 10 channels of our Pro Tools rig." That, he adds, varies from song to song, mostly providing syncopated and rhythmic support – "shaker and tambourine, ear candy" – and goes directly into the front of house rack on line level inputs.

After the cross patch, the signal goes to a PowerTube, the bypassed Q4 and then to a Komit. "I set the compression at 3:1 on a slow attack cycle for the vocal so it really only grabs the latter portion of the vocal, but the compressor's output circuit is giving us another 6 dB," he explains. "With Lionel, the mic pre isn't necessarily your best friend, especially with that 58 – it's a bit of a fire hose, and there's a risk of more ambient than direct sound at times – so I try to get as much output gain in as many places as possible.

"I don't EQ his voice within the console either, with the exception of the high pass filter set at 160," he continues. "It's boomy at times, but I started inserting a McDSP MC2000 multiband compressor plug-in about a year after I started. Each compression band also provides some output level, so it does work like an equalizer in the respect that you can give it more or less gain within the signal chain, which helps make up for when Lionel holds the mic at his chest."

SERVING THE ETHIC

In addition to Richie, four of the members of Richie's five-piece band sing backup vocals, all using SM58s with the exception of Dino Soldo, who's on a Crown CM311 head-worn mic. "He plays all the wind and utility instruments so he uses the CM311 with a Shure RF system, which makes it easier for him as some of his vocal passages

come up just after saxophone sections. But the MC2000 is applied to everyone's vocal chain," Housel says. "That's all I use. Previously, I was using a Fairchild 660 plug-in stacked with an 1176 to 'peg' the final output for the vocals."

Housel had employed a similar setup with Keenan's bands previously, but the band noticed a bit of latency so he switched to the MC2000. "I didn't hear it, but it's the job of an engineer to work harder to produce a better product for people because we have such great tools available. It's a musician thing," he says, referencing his work as a guitar player in the past. "You want to keep learning and get better."

He's also made a variety of mic changes for the band in service of that ethic, including Sennheiser 904s on drummer Oscar Seaton's toms. "To me, it sounds like a Sennheiser 421," Housel notes. "It's got the boom and the transient attack, but unlike a 421 I can put it within a very small, confined area."

Kick in and out are handled with a Shure BETA 91A and beyerdynamic M88, respectively. Snare is captured with SM57 on top and a BETA 57 on bottom, with a second snare covered by an additional SM57. Neumann KM 184s are applied for hi-hat and ride, with AKG C414 XLS multi-pattern condensers for overheads.

Housel has moved from a combination of ribbon and condenser mics to a pair of dynamics – SM57 and beyerdynamic M201 – for guitarist Ben Mauro. "I like those for their frequency response, to my ear, in terms of what a guitar should sound like," he explains. "On the console EQ I use a lot of high-pass and low-pass to narrow in on what the actual guitar sound is without having to go in and boost, say, somewhere in the mid band from 600 Hz to 1 kHz. Low end isn't necessarily a guitar player's friend, but if there's too much high end you're going to have an endless battle between hi-hat, kick, snare and guitar, which is what many guitarists have blazing in their mixes."

A screenshot of the McDSP MC2000 compressor plug-in on Housel's console that's applied to everyone's vocal chain.

BIG COMPLIMENTS

There's also a good deal of direct (box) action on stage, the domain of more Radial gear. J48s handle acoustic guitar and bassist Ethan Farmer's rig, with JDI Duplex passive boxes for keyboardist/MD Chuckii Booker, keyboardist/sax player Dino Soldo and Richie's piano, a Nord Stage in a slim grand chassis.

"On bass I'm also using a Radial SGI re-amp module because we're a long distance from the receiver of the wireless to the stage, and he's got an octave pedal and a local tuner I've got to catch," Housel adds. "The RF receivers for his bass are all the way stage right. When I first came on, the tech had a long 1/4-inch cable that he was putting in a loom."

Audio-Technica ATM35 cardioid condensers are utilized with Soldo's alto and tenor saxes, with another SM57 for his soprano sax and an SM58 for his harmonica. The mic package is completed with four Sennheiser MKH 50s deployed in L/R and L/R/C to capture audience ambiance.

All performers wear Ultimate Ears in-ear monitors, ranging from UE 11s to UE 18s, with signal delivered via Shure PSM 1000 personal monitoring systems.

"I also carry six Clair 12AM active stage monitors for emergencies, all downstage and on a single mix," he says, noting that Clair Brothers is serving as the tour's sound company. "But in two years, I've turned them on maybe twice."

The stage is also outfitted with two Clair "bowtie" dual-18 subs, one per side, and eight CO8s, four per side, as side fills, receiving a mono mix of kick, snare, toms and bass. "I try to use as few sources as possible that might interface with Lionel's microphone when he wanders the stage," he says, "and I want to stay out of the way of our house engineer's mix, but while I'm mixing for a smaller audience, if they don't have a good show, he's not going to have a good show."

Housel still considers the monitor rig a work in progress and is mulling over other possible changes, including a potential switch to a DiGiCo console. The overall mantra is achieving a balance between what the performers are comfortable with while trying to make things better for them in the long term.

"I got lucky with Lionel and the band," he concludes. "My first gig was a fly date and I'd never met our tour manager, Glen Matthews, or the band. Then, our input list was mostly Shure products, which are known commodities to me, and their gain structure was a known thing to me. So we got the band on stage, I put up a quick mix, we went through a few songs, made a few changes, and the band came off after sound check and said, 'Sounds great. Are you a musician? Because it's really musical sounding.' And that's one of the biggest compliments I could ask for as an engineer." ■

Based in Toronto, KEVIN YOUNG is a freelance music and tech writer, professional musician and composer.

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TAMING THE RF BEAST

Best practices and useful technology in wireless world

by Craig Leerman

WIRELESS SYSTEMS are a key component in almost every facet of live entertainment production, especially concerts and corporate meetings and events. The demand continues to increase as the supply of available bandwidth is both shrinking and becoming more congested.

As a result, pre-planning and wireless frequency coordination are becoming more important, particularly as the FCC (Federal Communications Commission) is preparing to sell off more of the UHF spectrum where the majority of wireless microphones and monitoring systems operate (currently 470-698 MHz in the U.S.). And, our systems already share portions of the broadcast spectrum with ever-proliferating TV band devices (TVBDs, formerly known as white space devices, or WSDs).

Add the problems of intermodulation interference (intermod) into the mix, and as audio professionals, we really need to focus on frequency coordination and wireless system design. Intermod happens when two (or more) transmitter signals mix in an active device (transmitter, active antenna, active splitter, receiver) and produce additional frequencies above and below each original transmitting frequency. These intermod products occur at the same spacing as the two original frequencies were apart.

Let's say you need three wireless systems onstage – two mic systems and a guitar system. You select 498.000 MHz for one of the mic systems and 499.000 MHz for the other mic system. The spacing between the frequencies is 1 MHz, so don't place the guitar system on 497.000 MHz or 500.000 MHz because that's where the third-order intermod products will occur.

This is simple enough to figure out for a few systems, but every time another frequency is added, it must be coordinated with every other frequency in service. It gets pretty complicated on bigger gigs, and astronomically so on larger events where there can be hundreds of frequencies in use.

MODERN DEVELOPMENTS

There are several software programs that we can turn to for help with frequency coordination. Some are free, such as Sennheiser Intermodulation and Frequency Management software that provides rapid calculation of intermod-free frequencies, and Shure Wireless Workbench, with recently released version 6.11 also offering several new features to help manage wireless system performance over the network, from pre-show planning to live performance monitoring. RF Guru from Stage Research and IAS (Intermodulation Analysis System) from Professional Wireless Systems are two more programs that can help, available for a fee.

To make sure our wireless systems don't get stepped on by TV transmitters and other high-power users (and/or cause any interference to them), we can turn to several websites that can help us steer clear of problems. The Shure Wireless Frequency Finder is a free tool where users can input their location, with the program selecting frequency bands and offering information on know TV transmitters in that region. Electro-Voice and Sennheiser are two more manufacturers that offer this kind of online help. Along with intermod calculations, the aforementioned software programs also pull in data from the FCC on active TV stations.

One way that manufacturers have adapted to the shrinking spectrum is to make their systems more efficient, with several systems now able to operate within the same amount of bandwidth that used to be occupied by a single system. For example, the Shure ULXD

"Radial DI's do exactly what they are supposed to do: deliver great audio."

~ Paul Boothroyd

"The Radial J48 is one piece of kit that can be relied upon in my worldwide touring. It is not only reliable but delivers what it needs to do... and that is good audio."
~ Paul 'Pab' Boothroyd
(Paul McCartney, AC/DC, Paul Simon, Faith Hill, Pink)

"With the PZ-DI, we were able to integrate multiple piezoelectric devices seamlessly and with glorious results. Radial has once again proven they are a cut above the rest."
~ Brad Madix
(Rush, Marilyn Manson, Shakira, Def Leppard)

"Radial direct boxes make everything I put through them warm, punchy and clear. They are great DI's"
~ Chick Corea
(Electrik Band, Miles Davis, Return to Forever)

"The JDX gave me all the character and distortion without the worries of bleed and feedback. I was so surprised how close it sounded to the SM57, I had to double check the patch."
~ Jim Warren
(Radiohead, Arcade Fire, Nine Inch Nails)

"We struggled for years on Zombie to get clean audio from the video servers... the JPC solved it! I finally get clean CD quality and dont cringe at the noise levels. The JPC is great!"
~ Joel Lonky
(Rob Zombie, Goo Goo Dolls, P. Diddy, Billy Idol)

"Radial DI's provide the flexibility I need to perfectly match any situation I come across. I specify Radial exclusively for every tour I mix."
~ Dave Natale
(Rolling Stones, Joe Cocker Lionel Richie, Fleetwood Mac)

"The Radial JDX is almost too good to be true. The artists hear the sound they are playing and I'm not fighting mic coloration any more. I just plug it in & turn it up."
~ Jon Garber
(Rascal Flatts, Brad Paisley, The Band Perry, Chely Wright)

"On Santana tour, we have Radial J48's on guitars and JDI's on keyboards. Since January this year we have traveled all over and the Radial DI's have worked great!"
~ Rob Mailman
(FOH engineer - Santana)

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system can operate up to 47 active transmitters in one 6 MHz TV channel space (or 63 in one 8 MHz TV channel in high-density mode).

Another notable innovation is Lectrosonics Digital Hybrid Wireless technology, which utilizes a proprietary algorithm to encode the digital audio information into an analog format, which is then transmitted over an analog FM wireless link. The receiver employs high-end filters, RF amplifiers, mixers and detector to capture the encoded signal and a DSP recovers the original digital audio. This hybrid approach enhances immunity to noise without compromising spectral and power efficiency, as well as operating range.

Lectrosonics Venue receivers also work in tandem with System Designer software that includes a spectrum scanner providing a visual display of RF activity within the tuning range of the system to quickly locate clear operating frequencies. In addition, a walk test recorder generates a visual of RF levels during a walk test of a project site.

Frequency-agile systems that can detect problems and automatically (and seamlessly) switch both transmitter and receiver to another (open) frequency represent another effective approach. The new System 10 Pro wireless from Audio-Technica (*Road Test, LSI April 2015*) is a good example. The receiver and transmitter are actually transceivers that are constantly communicating with one another. If interference occurs, the units switch over to a clear frequency without a hitch.

FURTHER OPTIONS

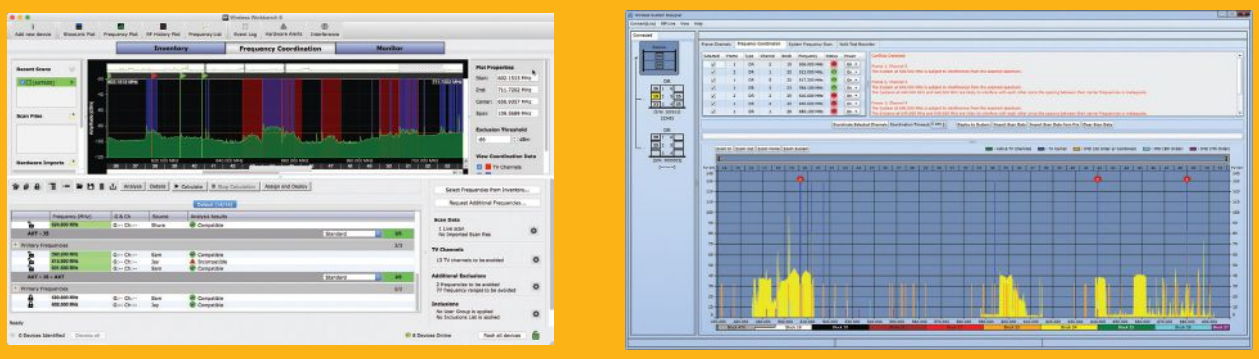
One way to avoid a lot of the headaches in the UHF band is to operate outside of it. Many years ago most (if not all) wireless microphones operated in the VHF (TV band channels 2-13) spectrum from 150-216 MHz. My company still has a few VHF systems that work quite well because the spectrum is pretty empty.

Radio Active Designs (RAD) recently introduced the UV-1G wireless intercom system, which can help in crowded UHF environments. The system offers up to 30 base stations and 180 belt packs in the same footprint as one base station and four belt packs that use traditional FM technology. And the belt packs operate in the VHF band, freeing up valuable UHF spectrum for wireless microphones and IEMs. The system uses proprietary Enhanced Narrow Band technology, a unique modulation scheme that is more spectrally efficient than current FM (Frequency Modulation) technology. Each channel in a UV-1G system has an occupied bandwidth of only 25 kHz.

A Sennheiser evolution wireless D1 instrument system

Several wireless manufacturers also offer systems that operate in the 2.4 GHz band, most commonly used by Wi-Fi. While it's certainly true that there's a lot of activity within the 2.4 GHz range, manufacturers have developed ways to make it work, including the aforementioned System 10 Pro that uses frequency agility to switch frequencies.

The Line 6 XD-V70 was an early professional caliber wireless system operating in the 2.4 GHz band. The system has proprietary technology called Digital Channel Lock (DCL) that distinguishes its own digital audio from any other third-party signal, including Wi-Fi. Up to 12 XD-V70 systems can operate in the band simultaneously, while the newer XD-V75 allows up to 14 systems.



At left, frequency coordination tab of Shure Wireless Workbench 6.11, and at right, Lectrosonics System Designer software that includes a spectrum scanner, walk test recorder and more.

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The Sennheiser evolution wireless D1 is a recent addition to the 2.4 GHz club, with transmitters and receivers that automatically pair and select suitable transmission frequencies, while multiple D1 systems can automatically coordinate themselves.

Systems are also coming on the scene that operate in the 1.9 GHz range, such as the recently released Sennheiser SpeechLine Digital Wireless system with Automatic Frequency Management technology that searches and can switch to a clear frequency if transmission is disturbed. In addition, network integration enables the system status to be remote controlled and monitored using the Wireless System Remote (WSR) app, AMX or Crestron.

ADDITIONAL ASPECTS

And that brings up another concept, which is monitoring. Particularly when using multiple wireless channels, this is a good idea, with many newer systems offering networking and the ability to link receivers together as well as remote monitoring of the system as a whole.

Soundcraft and Shure recently announced a new collaboration that enables native monitoring and control of select Shure wireless systems on Soundcraft Vi Series digital consoles. The new Soundcraft Vi5000 and Vi7000 digital consoles support Shure ULX-D and QLX-D systems as well as AKG DMS800 and WMS4500 systems.

There are some excellent third-party spectrum analyzers available that are also useful in keeping an eye on things. A spectrum analyzer monitors a range of the frequency band that is determined by the user. It sweeps across the range over and over, measuring the strength of the present RF signals and displaying the results. Users can see the background noise level as well as their own wireless system signals and any unwanted or unknown signals that may cause interference.

RF Explorer offers a compact, portable spectrum analyzer that's available in different models depending on the frequency bands the user needs to view. The unit can also interface with a computer for data storage, control and monitoring.

Kaltman Creations offers several handy tools along these lines, including the RF Command center that interfaces with a computer and the portable RF-Vue that interfaces with a tablet allowing the user to walk around and take measurements. The company also offers a nifty unit called the RF-id SOLO, a small frequency counter that can identify the exact frequency a transmitter is operating on as well as confirming power output, which helps in identifying and troubleshooting wireless problems both onsite and back at the shop.

Another way to reduce the potential for problems is to use directional antennas, especially for transmitting with IEMs. Directional "paddle-style" (a.k.a., log-periodic dipole array) antennas can provide up to 6 dB of gain, and Helical antennas can provide up to 10 dB. With receivers, they can reduce the amount of unwanted noise picked up in comparison to an omnidirectional antenna, and offer forward gain that helps when they're placed at a decent amount of distance from the transmitters.

BEING PREPARED

Earlier I mentioned the importance of pre-planning. Along with monitoring the airwaves during an event, it's really the key to a successful show involving wireless systems. Before every event where we're using wireless, we check to see what transmitters are in the area and if any other wireless systems will be in use. Because we do a lot of work in Las Vegas casinos, it's not uncommon for an in-house show in a theater that uses dozens of wireless to be located on the other side of the wall from a ballroom.

The portable RF-Vue from Kaltman Creations.

We also coordinate with the in-house A/V department and any other production vendors working in the building. Once we've selected clear frequencies that work well with each other (as well as a bunch of backup frequencies) we monitor the airwaves during the event, checking for potential problems.

Recently we were hired to provide some wireless systems at a general session. It wasn't huge, just eight wireless mics plus four stations of wireless intercom, but there were more than 200 frequencies in use on three floors of meeting space in the facility.

It could have been a disaster, but fortunately the event manager contracted a wireless coordinator, who made sure all of the frequencies for our event, and the other frequencies in use at the property, all "played nice" together. On our end, I monitored with a spectrum analyzer during rehearsals and on show day, and had a list of backup frequencies we could switch to if needed. With this little bit of attention to the details, we had a great event with no wireless problems over the entire week. ■

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

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GETTING IT DONE

Audio technology deployed a variety of ways. ***by Live Sound Staff***

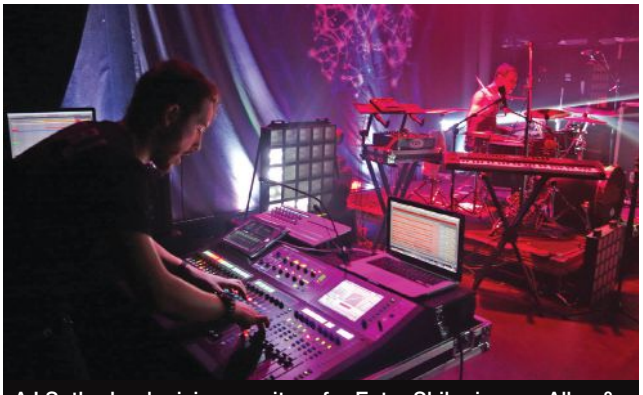
MONITORS FOR ENTER SHIKARI ON TOUR

UK post-core band Enter Shikari just completed North American and European tours, carrying an Allen & Heath GLD mix system to manage monitors. Chosen by monitor engineer AJ Sutherland, the package included a Dante-enabled GLD-112 mixer with an AR2412 and two AR84 remote I/O racks to manage five pairs of wedges and four stereo IEMs, plus the engineer's listen wedge and cue mix.

"I bought a GLD-80 when it first came out and have used it on various projects over the past three years. I think it nails the midscale console market," Sutherland states. "The IO, functionality and audio quality are suited to the demands of high level productions and it's small enough to fly around the world and fit on cramped stages.

"It's really helped the remaining band members migrate onto in-ears, and they seem to be enjoying discovering what is possible, plus the management can't complain about the price," he continues. "Being able to have the same console for each show makes my life a massive amount easier and the band's show a lot better. A&H is my first choice of console as they are so fast and easy to use, which is vital on such a high energy show."

The tour production was managed holistically so that all facets of the system – backline, audio, lighting and video – were integrated. Novation controllers and keyboards were positioned around the stage for any band or crew member to control the track, with time code flowing to lighting/video, and MIDI connecting everything from drum triggers and Kemper rigs to audio consoles and whammy FX pedals.



AJ Sutherland mixing monitors for Enter Shikari on an Allen & Heath GLD control surface.

d&b audiotechnik M4 wedges on stage, with V-Series mains serving the house, for Sara Bareilles on tour.

MAINS FOR SARA BAREILLES ON THE ROAD

Mix engineer Trey Smith chose a main system headed by d&b audiotechnik V-Series loudspeakers for the recent U.S. tour by Sara Bareilles, supplied and supported by Spectrum Sound of Nashville, with Chris Michaelessi serving as system tech. It was the first full production outing for Bareilles, with Smith noting that the smaller footprint of the system made carrying the rest of the production much easier.

Typically the system incorporated left/right V8 arrays with the wider 120-degree (horizontal) V12 cabinets at the bottom. The tour visited a mix of indoor theaters and outdoor sheds with capacities ranging from 4,000 to 10,000.

"At the biggest we flew six V-SUBs per side as well, and where possible had a 12-cabinet J-SUB array across the front of stage on the floor," he adds. "Sara is not the Foo Fighters when it comes to low end, but she is dabbling in some tracks, an 808 drum sample for example, and I like to mix big and round so we do need that weight in the low end.

"Those cardioid subs are great," Smith says. "Sara likes to pull her ear buds in and out depending on how she's feeling on stage and what song she is singing; that's why we also have d&b M4 wedges at all her singing positions: not having that low end throwing back on stage is a big help. Plus the directivity we could achieve with the sub arc was amazing."

AUDIO UPGRADE AT A HISTORIC CATHEDRAL

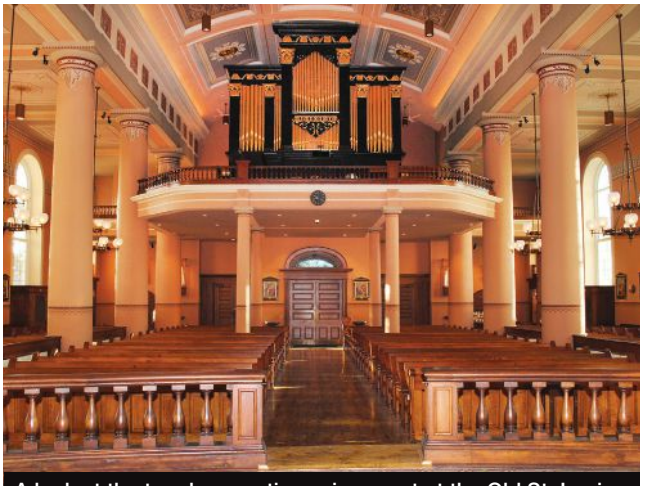
Consecrated in October of 1834, the Old St. Louis Cathedral was built in the Roman Catholic tradition of grand, highly reverberant worship spaces. The first cathedral built west of the Mississippi River, today it stands in the shadow of the Gateway Arch.

The archdiocese recently engaged Tech Electronics to address the sanctuary's ongoing audio issues, with the St. Louis-based firm choosing to implementing Iconyx digitally steerable column arrays from Renkus-Heinz. Specifically, the entire sanctuary is covered by two IC24-R-II columns, mounted beside two structural columns on the wall behind the altar. With a low-profile design, the columns integrate with the church's classic architecture, and a custom paint job helps them virtually disappear in plain sight.

"Any other solution involves hanging large, boxy speakers in the sanctuary, usually with acoustical treatment in the room," notes Tony Badamo of Tech Electronics. "In most cases, it creates a visual distraction without really addressing the intelligibility issues."

The two Iconyx columns are aimed in crossfire fashion to the opposite corners of the sanctuary. Using the system's RHAON software, Badamo created a series of audio beams to provide focused direct coverage throughout the room, while avoiding unwanted reflections off walls, balcony facings, windows, and the classic vaulted ceiling.

"Iconyx enables us to create a custom coverage pattern that is amazingly even in every seat," he states. "You can actually sit in a pew directly behind one of the huge structural columns and understand every word perfectly."



A look at the tough acoustic environment at the Old St. Louis Cathedral addressed by Renkus-Heinz Iconyx.

TRIPLE PLAY FOR A LEGEND OF DISCO

Soundcraft Vi consoles were at the heart of the action recently when one of the legends of the Disco era, Nile Rodgers, performed with his 9-piece band in a special one-off concert at London's IndigO2. Production manager/front of house engineer John Ryan and monitor engineer Marco Dellatorre utilized the house's two installed Vi6 digital consoles, while Eduardo Puhl, house engineer at Koko in Camden Town

Looking over front of house engineer John Ryan's shoulder as he mixes Nile Rodgers at London's IndigO2 on a Soundcraft Vi6.

where Vi consoles are installed, brought his Vi1 (and Vi Compact stage rack) to handle the broadcast and recording feed. The show was streamed live via Living Indie.

"We took analog splits from the other desk channels into the stage rack down, running optical fiber to the Vi1 which was located in a production unit just outside the venue," Puhl explains. As a back-up, he also recorded the show using the redundant port on the Vi1 MADI card to output to a MADI sound interface, sending that to a laptop running Reaper DAW software.

Dellatorre ran 24 outputs from the monitor desk, with parametric EQ on all IEMs and graphic EQ on all wedges, with no gates or compressors (and snapshot on "Let's Dance"). "I use the desk's preview feature during the show because I need to create a second snapshot for 'Let's Dance,'" he says. "On that song there are a lot of changes and even if I can create this snapshot during sound check, I prefer to create it on the fly because my mix approach is live and I follow the band mood every gig."

Ryan adds that digital mixers are an efficient solution in these types of applications: "As we are often tight on time, we travel with a series of show files. We loaded a previously saved file from a show in France and Matt arranged the layers the way I needed them, and within a few minutes we were ready for sound check. Ease of navigation on the Vi6 was just a matter of getting used to where things were on the surface."

QUAD STAGES IN SAVANNAH

Georgia's Savannah Music Festival selected a variety of Meyer Sound loudspeakers to reinforce all four of its principal venues for the first time, including the Trustees Theater and Lucas Theatre. The selection of Meyer Sound was driven by Chris Evans, the festival's chief audio engineer, who is responsible for system specification, tuning, and supervision of the audio crew.

Evans also mixed Mavis Staples using MINA line arrays at the Trustees Theater. "We had great results this year with MINA," he says. "The system has so much headroom that every engineer had more than they needed. It has made the Trustees a showcase venue." The 1,250-seat theater was supported by twin arrays of 12-each MINA loudspeakers, four 600-HP subwoofers, and four UPJ-1P VariO loudspeakers for fill.

Meanwhile, M1D line arrays were set up at the 1,165-seat Lucas Theatre for the Arts, while the 700-capacity Ships of the Sea outdoor shed was outfitted with M'elodie line arrays. At the 300-seat Charles H. Morris Center, artists such as the Barr Brothers and Buckwheat Zydeco were heard through UPQ-1Ps.

Other Meyer Sound gear included UPJ-1P, UPM-1P, and UPJunior VariO loudspeakers; 700-HP, 600-HP, and M1D-Sub subwoofers; and Galileo loudspeaker management systems with Galileo 616 processors. MJF-212A and UM-1P stage monitors were also used. As in prior years, systems were provided by Rock N Road Audio of Tucker, GA, with facilitation by Roy Drukenmiller and Robin Gang.

Meyer Sound M'elodie supplied by Rock N Road Audio for the Ships of the Sea outdoor shed at the Savannah Music Festival.

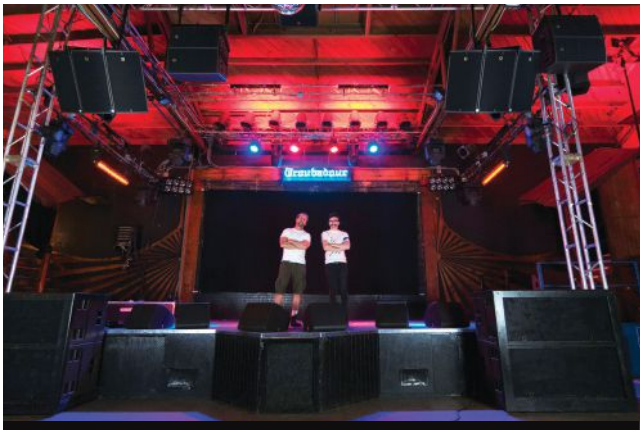
PHOTO CREDIT: ELIZABETH LEITZELL

SONIC RETROFIT FOR A LEGENDARY VENUE

The renowned Troubadour in West Hollywood has undergone a major upgrade of its house system, with Rat Sound of Camarillo, CA implementing a variety of L-Acoustics components. Showing no signs of slowing down after nearly six decades, the club was named by *Billboard* in 2014 as one of the top five must-play clubs in the country and by *Rolling Stone* the

year prior as the second best rock club in America.

Dave Rat and Paul Freudenberg of Rat Sound began discussing the upgrade with staff at The Troubadour three years ago. "With the venue's mix position located up in the balcony, the balcony being off center, and the room being both extremely wide and fairly shallow, this project was an exciting challenge," explains Rat. "We were very happy with what we were able to achieve with asymmetrical hangs of L-Acoustics ARCS II and ARCS Focus enclosures. Additionally, we got to do some fun custom work, including creating cages to protect the subs on custom rolling plates."



Dave Rat (left) and The Troubadour's Oscar J. Narro surrounded by the venue's new L-Acoustics gear.

PHOTO CREDIT: MADELINE KEYES-LEVINE

According to Freudenberg, "Dave and I examined numerous designs and orientations of different types of loudspeakers, then evaluated the designs using L-Acoustics' Soundvision. We ended up with a solution of six ARCS IIs flown in horizontal mode to cover the main floor and four ARCS Focus cabinets oriented in a vertical configuration, slightly offset to stage left from the positions of the main speakers to accommodate the offset balcony.

"The ARCS family's directivity fits the low-ceilinged environment very well," he adds, "and the 30-degree vertical coverage nicely addresses the balcony while also providing front of house with direct sound." A flown L-Acoustics 115XT HiQ coaxial loudspeaker provides center fill, while a 12XTi coaxial covers a small balcony section at stage left. Four SB28 dual-18-inch subwoofers groundstacked two per side down below deliver low-end reinforcement. ■

Point Source Audio CM-i3

Evaluating a new intercom headset. *by Craig Leerman*

INTERCOM SYSTEMS CAN BE an integral part of production work, depending on the type and parameters of the show or event. My company serves a lot of corporate events, where intercoms are quite important in following cues and communicating between departments to facilitate changes on the fly. We wear headsets linked to our intercom packs, and there can be confusion in listening to the system and communications simultaneously. So when Point Source Audio recently announced a new headset, I was quite motivated to check it out.

Billed as the world's first modular intercom headset, the new CM-i3 has a slim-line design that uses factory or user supplied earphones (earbud or IEM custom style), coupled with a microphone mounted on a compact headband. A user can configure the system as a single- or double-ear headset, and remove either earpiece (or both) on the fly rather than having to take off the entire headset to check environmental sounds or interact with others. The in-ear design helps to eliminate listening fatigue while also accommodating the easy replacement or changing of the earphones. And, the entire assembly weighs in at only two ounces.

Point Source Audio CM-i3 intercom headset. Note that it can be ordered without earphones for users who prefer their own.

The zippered carry case for the CM-i3.

The dynamic mic element offers a cardioid pattern, impedance of 200 ohms, sensitivity of -60 dB, and a frequency range of 300 Hz to 10 kHz. It's specified as having 12 dB of off-axis noise rejection. The company-provided EM-3 earphones are a single-ported driver type, have a sensitivity rating of 120 dB SPL (at 1 kHz) and impedance of 32 ohms. They're plugged in via a 3.5 mm TRS connector, with three sizes of eartips offered to fit a wide variety of users. (The headset can be purchased with or without earphones.)

Six different versions of the CM-i3 allow users to choose the model that is compatible with their particular intercom system(s). The CM-i3-4F (the model provided for this Road Test) offers a 4-pin XLR female connector and works with all Clear-Com compatible systems. Using the last two or three digits for identification, other models include the 4M for RTS mono systems, 5Ms for stereo RTS systems, 5F for Telex systems, 5Mm for Telex TR700/800 and TR80N systems, and PH with a 3.5 mm plug for Riedel as well as iPhone and iPad loaded with an intercom app.

DESIGN FACTORS

Out of the box I was struck by just how light the headset is. At only two ounces, it weighs less than one-quarter of the weight of my usual single muff headset (and it's considered a light one). The CM-i3 ships with a sturdy zippered carry case that's large enough to hold the headset as well as extra eartips and any adaptors that might be needed for using the headset with different systems. It also comes with a foam windscreen for the mic as well as an O-ring to keep the windscreen in place. (Nice touch!)

The CM-i3 frame is made of rubber covered wire and offers cushioned pads. It's very easy to adjust to get a nice snug fit. Once adjusted, the headset stays put but doesn't put a lot of pressure on the head, so there's no fatigue in wearing it. The rubber has paths that route the

cables to an exit at the rear of the unit. I prefer the cables running down my back instead of at the side of my head; they're more out of the way.

Both the mic and earphone cables are contained in an overwrap and are about four feet long, a good length for most uses. Point Source Audio also offers an expandable coiled cable (CM-EXT) that's 10 feet long when extended, along with a belt clip for an XLR connector (CM-BCP) to help in cable management.

I plugged the headphones into a laptop on my desk and listened to a few tracks. The headset felt so comfortable and the earphones sounded so good I wore them while doing some track editing for a client. An hour later it struck me that I was still wearing the headset – that's how lightweight and comfortable the unit is.

On my bench in the shop, I set up an intercom system with two belt packs to further evaluate the unit. The mic performs great and the system sounded better using the earphones than my standard single muff units. I had several folks on staff check it out as well, and every one of them commented on the quality of the audio. Satisfied that everything was working as it should, I took the CM-i3 out to a few gigs.

POSITIVE EXPERIENCES

At the first event, I was freelancing as an A2, wrangling wireless mics at a corporate gig in a casino ballroom. The single muff headset provided by the A/V company had seen better days so I swapped it out for the CM-i3. When presenters came over to my table to get wired up, I could simply remove one earpiece and talk to them without having to take off the entire headset. And again, it was extremely comfortable to wear. I had some of the crew try it out as well and received nothing but compliments.

The CM-EXT expandable coiled cable.

Next up was a variety show where I mixed monitors. It was great to be able to take out either earpiece and listen to the stage for a bit without completely removing the headset. I also liked how I could turn it into a single "muff" unit when an acoustic act was onstage, and put both earbuds in for isolation when it switched to a louder electric act.

Since I was seated the whole show I decided to place my belt pack next to the console. Fnding the cable to be just a little short for this application, I hung the pack back on my belt. However, the CM-EXT expandable coiled cable would have solved the length issue in addition to coming in handy for moving around, expanding and contracting to take up the slack.

The final field application was a corporate event, heavy on video program, where I mixed front of house from a riser in the back of the room. The client, director, video and lighting techs were also on the same riser. When they keyed their mics, the PA and video music flooded over the intercom system, hampering its intelligibility.

But due to the excellent rejection of the CM-i3's cardioid mic element, when I keyed my belt pack I injected far less of this noise into the system. The director commented about how great my mic worked and asked if I would get one for everybody to wear after a break. He was more than a little disappointed that I only had the single test unit with me.

And I must add that once again it was great to be able to use either earpiece and switch between them without having to take off the headset. I also discovered that I could wear my regular headphones over the CM-i3 when I needed to cue up and edit an extra audio track at the last minute. This meant I was able to get that task done without being off the comm system.

All intercom systems users need to check out the new CM-i3. Flexible and convenient to use, and with excellent audio quality, it significantly enhances intercom communications as well as the ability to hear the surrounding environment when needed.

U.S. MSRP: $400 with earphones. ■

Among a variety of options are three sizes of eartips, a mic windscreen with O-ring, and a belt clip accessory for XLR connectors.

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

Flexible Performers

Design factors of medium-format line arrays, and the latest models. ***by Gary Parks***

These days I'm surprised when I go to a concert or a performing arts center and don't see line arrays flown at the sides of the stage, often complemented with digital audio networking, remote system control and monitoring, and real-time audio monitoring of the array's response in the house. These sleek systems have come a long way from the huge piles of cabinets and horns that I remember from the mid-1960s when I first started going to shows.

Line arrays can provide many benefits, including more even audience coverage in terms of frequency response and SPL, control of vertical dispersion well into the lower midrange, improved sight lines, and ease of setup. They're designed to be flown and taken down quickly, often in "blocks" of individual modules, and to be flexibly adjustable to different curvatures – and a more limited quantity of modules can even be ground-stacked.

Line arrays vary in how amplification and signal processing are implemented, with many being self-powered and requiring only a line-level signal for each cabinet. The electronics in such systems are highly integrated into the overall design, and built for reliability – since they are typically inaccessible during the performance. Other manufacturers choose to provide dedicated external processors and amplifiers matched to the requirements of the transducers and enclosure design, or recommend particular third-party processors and amps.

Smaller-format line arrays, versus enclosures with 12-inch or 15-inch LF drivers, allow wider splay angles within the array elements, while still maintaining consistent coverage. This characteristic can be useful for covering smaller venues that have multiple levels, and can also help work around architectural structures like the edges of balconies. To help sound engineers achieve the most consistent results, many manufacturers complement their systems with predictive software that will calculate the expected performance of particular line arrays at differing splay angles and output levels, across a variety of frequency ranges and array lengths.

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

Even within the medium-format category, there's a lot of variety. Among the represented brands and models, the horizontal coverage angle varies from 80 degrees to 150 degrees from a single array column, with most ranging between 100 to 120 degrees. Some manufacturers offer cabinets with the same "footprint" with differing horizontal coverage, allowing the user to better "customize" coverage for a particular venue. Enclosure width varies from a bit over 23 inches to over 30 inches, and weight for each cabinet ranges from a bit over 30 pounds to over 100 pounds. Many are self-powered, and others have dedicated external processing and amplification.

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

The medium-format line arrays presented in this Real World Gear tour of recent models is not meant to be all-inclusive, yet covers a variety of manufacturers and design concepts based around LF components in the 8-inch to 10-inch range.

GARY PARKS is a pro audio writer who has worked in the industry for more than 25 years, holding marketing and management positions with several leading manufacturers.

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

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

Adamson Systems S10

www.adamsonsystems.com

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

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

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

TECHNOLOGY FOCUS

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

OF NOTE

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

KEY SPECIFICATIONS

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

EAW KF730

» www.eaw.com

Configuration: 3-way
Dispersion (h x v): 110 x 12 degrees
LF: 2 x 10-inch cone drivers, proprietary Phase Aligned
MF: 2 x 7-inch cone drivers, horn loaded
HF: 2 x 1.75-inch voice coil drivers, horn loaded
Frequency Response: 80 Hz – 20 kHz
Maximum SPL: 136 dB
Power: External biamp, 700 watts LF & 350 watts MF/HF
Rigging: Integral lightweight fly hardware
Size (h x w x d): 13 x 28.5 x 17.5 inches
Weight: 77 pounds
Companio n Sub: SB730 (dual 12-inch)

RCF HDL 20-A

» www.rcf-usa.com

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

K-array KH2

» www.k-array.com

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

d&b audiotechnik Y8/Y12

www.dbaudio.com

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

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

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

TECHNOLOGY FOCUS

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

COMPANION PRODUCT

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

KEY SPECIFICATIONS

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

NEXO GEO S830

» www.yamahaca.com

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

D.A.S. Audio Event 210A

» www.dasaudio.com

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

JBL Professional V20

» www.jblpro.com

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

FBT MUSE 210LA

www.fbt.it

Encompassing advanced technology and refined engineer¬ing, the new MUSE 210LA redefines the modern line array in terms of power, size, light weight, flexibility and ease of use. True modular-ity means that the self-powered MUSE 210LA is ready for any application, from a small installation using two enclosures to a large outdoor concert deploying 16 enclosures per hang.

Each modular MUSE element incorporates two 10-inch custom-designed woofers with high excursion voice coils and two custom B&C HF compression drivers with 1-inch throat. The waveguide, optimized via a series of simulations using BEM finite elements to eliminate distortion, has been designed to present an optimum load to the driver's diaphragm at 800 Hz, allowing the propaga-tion of an acoustic wave in the horizontal field up to 18 kHz.

The enclosure is equally rugged, practical and stylish, constructed of birch ply-wood and incorporating two die-cast aluminum handles and integrated hardware to suit any application.

TECHNOLOGY FOCUS

Class-D convection cooled bi-amplification delivers 600 watts and 300 watts (LF and HF, both RMS) with switch-mode power supplies. Onboard DSP proces-sor provides eight EQ presets, with adjustable HF level of +/- 5 dB for accurate ampli¬tude shading. Works in conjunction with EASE FOCUS 2 software.

OF NOTE

Onboard control panels provide XLR input and link, preset control, HF level control, high-pass filter and ground lift. Each cabinet can be splayed in 2-degree increments via simple pin adjustments.

KEY SPECIFICATIONS

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

VUE audiotechnik al-8

» www.vueaudio.com

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

QSC Audio WideLine-10

» qsc.com

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

dB Technologies DVA T4

» www.dbtechnologies.com

Configuration: 3-way
Dispersion (h x v): 100 x 15 degrees
LF: 1 x 8-inch neodymium cone driver
MF: 1 x 6.5-inch neodymium cone driver
HF: 2 x 1.4-inch neodymium compression drivers
Frequency Response: 80 Hz – 19 kHz
Maximum SPL: 128 dB
Power: Onboard 3-channel class D amplifier (420 watts RMS)
Rigging: Integrated hardware, adjustable
Size (h x w x d): 9.6 x 23.2 x 13 inches
Weight: 30.4 pounds
Companion Sub: DVA S09DP (15-inch)

L-Acoustics KARA

www.l-acoustics.com

With a design inspired from the K1 stadium system, KARA delivers a considerable number of improvements over the previous generation of line sources, particularly with regard to directivity control in the horizontal plane, transducer resources for increased operating bandwidth and coherence, vertical coverage capability, and extensive choice of operation modes to accommodate various LF contour requirements.

As a stand-alone system, its compact size and low weight complies with rigging and visual limitations. KARA can be deployed either as a compact distributed solution for delays and fills, a dedicated K1 down fill extension for stadium and arena concert applications, or as a FOH system thanks to its full compatibility with the SB18 extension.

KARA can be acoustically and mechanically modeled with SoundVision 3D simulation software. In addition, LA Network Manager software offers remote control and monitoring of the amplified controllers via an intuitive graphic interface.

TECHNOLOGY FOCUS

Wavefront Sculpture Technology morphs the spherical wavefront of the HF driver into a cylindrical, isophasic wave. Coplanar symmetry provides a coherent wavefront over the entire horizontal coverage at all frequencies. Array Morphing EQ adjusts the tonal balance to reach a reference curve or ensure consistency of the sonic signature with arrays of differing size.

NOTABLE APPLICATIONS

Starlight Theater, Kansas City
Webster Hall, NYC
Green Music Center, Sonoma, CA
Clowes Hall, Indianapolis
Cruzan Amphitheater, Miami
"The Last Ship," Chicago
Jack Johnson

KEY SPECIFICATIONS

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

Grund Audio Design GA-2021N

» www.grundorf.com

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

Electro-Voice XLD291

» www.electrovoice.com

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

Carvin Audio

» carvinaudio.com

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

Martin Audio MLA Compact

www.martin-audio.com

With superior sound control to ensure even coverage to every member of the audience, the award-winning MLA Compact brings MLA technology to the wide range of applications that do not require the full power and throw of the full-size MLA, or where a smaller, lighter system is called for.

MLA Compact shares the sonic attributes of the flagship MLA and is designed for medium-scale touring and fixed installations. Its compact size belies its output capabilities – a 12-box array can easily deliver full rock SPL in a 5,000-seat venue, while a 24-box array will approach the output of many "full-size" systems which have less efficient acoustic elements.

A desired house-curve is achieved right from power-up, while automatic, intelligent configuration and optimization eliminates trial and error in system set-up. MLA Compact delivers improved venue-to-venue, gig-to-gig consistency and repeatability, while artistic changes to balance at the mix position (or elsewhere) translate directly and accurately throughout the audience.

TECHNOLOGY FOCUS

DISPLAY software interacts with onboard DSP for highly accurate array optimization. "Fly-by-wire" software adjusts vertical coverage electronically to cope with changing conditions and last-minute changes in rigging height. "Hard-avoid" areas can be programmed in, and there's a leakage parameter to meet environmental noise constraints.

OF NOTE

Five dedicated class D amplifier channels per enclosure provide individual powering and DSP control of individual cells.

KEY SPECIFICATIONS

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

Montarbo RA16

» www.montarbo.com

Configuration: 2-way
Dispersion (h x v): 120 degrees; vertical array dependent
LF/MF: 2 x 8-inch neodymium cone drivers
HF: 1 x 3-inch voice coil, titanium compression driver
Frequency Response: 70 Hz – 20 kHz
Maximum SPL (peak): 130 dB
Power: External; LF – 800 watts/HF – 200 watts recommended
Rigging: Integrated hardware, adjustable from 0 to 12 degrees
Size (h x w x d): 11.1 x 20.7 x 17.3 inches
Weight: 39.7 pounds
Companion Sub: RA1815 (cardioid bass horn)

Ramsdell Pro Audio LA10-2

» www.ramsdellproaudio.com

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

Bose Professional RoomMatch

» pro.bose.com

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

Meyer Sound LEOPARD

www.meyersound.com

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

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

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

TECHNOLOGY FOCUS

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

OF NOTE

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

KEY SPECIFICATIONS

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

Alcons Audio LR16

» www.alconsaudio.com

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

Soundbridge 7208XY

» www.soundbridge.com

Configuration: 2-way
Dispersion (h x v): 80 or 120 x 10 degrees
LF: 2 x 8-inch neodymium cone drivers
MF/**HF:** 1 x neodymium EXO driver
Frequency Response: 68 Hz – 22 kHz
Maximum SPL (peak): 137.7 dB
Power: External; biamp or triamp
Rigging: Integrated rigging; rigged in pairs with woofers facing in toward each other
Size (h x w x d): 8.7 x 38.7 x 20.2 inches
Weight: 86 pounds
Companion Sub: 7000-Series (18-inch)

Turbosound Liverpool TLX84

» www.turbosound.com

Configuration: 2-way
Dispersion (h x v): 110 x 10 degrees
LF: 2 x 8-inch cone drivers
HF: 2 x 1-inch titanium dome compression drivers
Frequency Response: 70 Hz – 17 kHz
Maximum SPL: 128.5 dB
Power: External; 450 watts continuous, 1,800 watts peak
Rigging: Integrated hardware, adjustable in 1-degree increments
Size (h x w x d): 9.6 x 29.1 x 19.1 inches
Weight: 47.7 pounds
Companion Sub: TLX215L (15-inch)

Outline Butterfly C.D.H. 483

www.outlinearray.com

The Butterfly C.D.H. 483 incorporates some of the most sophisticated technology ever developed for a line array, with the 3-section element/enclosure designed to deliver in the most demanding applications.

The HF section is defined by a 3-inch compression driver coupled with a proprietary D.P.R.W.G. (Double Parabolic Reflective Wave Guide) device, which takes a circular planar (flat) wavefront emitted by the driver and transforms it into a rectangular planar wavefront at its output, keeping signal paths identical from every emission point of the source.

Four high-efficiency 8-inch mid woofers – two band-pass loaded and two reflex high-pass loaded by the sides of a waveguide with a 90-degree dispersion angle – are connected in parallel without any type of passive crossover, so can both be powered using just one amplifier. This enables the energy in the portion of band reproduced by both sections together to be doubled, making up for the lack of power in the mid/low frequencies typical of other comparably sized line array elements.

COMPANION PRODUCT

The compact Mantas line array is engineered with D.P.R.W.G. technology to match the acoustical signature of Butterfly, and it's also equipped with the same flypoints. As a result, it can be utilized with Butterfly arrays to deliver wider horizontal coverage (120 degrees) for front fill and down fill applications.

OF NOTE

The C.D.H. 483 cabinet shape fosters optimum coupling of array elements up to the highest frequencies, keeping the distance between the sources as short as possible and at the same time providing them with a continuous loading "baffle."

KEY SPECIFICATIONS

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

Alto Professional SXA28P

›› www.altoproaudio.com

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

BassBoss LA88

›› www.bassboss.com

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

ISP Technologies HDL 2208

›› www.isptechnologies.com

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

WorxAudio TrueLine V8

www.worxaudio.com

The TrueLine V8 is a 2-way, high-efficiency line array loudspeaker incorporating a large-format, 3-inch voice coil compression driver coupled to a stabilized proprietary FlatWave Former (wave shaping device) that delivers clear and penetrating high frequencies over a predictable and controlled coverage area. Dual 8-inch cone transducers coupled to the (A.I.M.) Acoustic Intergrading Module minimize comb filtering throughout the entire operating range.

The enclosure is constructed from sturdy multi-ply Baltic birch and is heavily braced for cabinet rigidity. Protecting the enclosure is a multi-layered, catalyzed polyurethane finish that is available in black or white. Anodized, aluminum TrueAim rigging hardware with 1-degree increments is available.

The V8 is available in black exterior, while the install version (V8i) is available in black or white exterior. Superior workmanship and transparent sonic quality make the V8 an exceptional tool for a wide range of demanding sound reinforcement applications.

TECHNOLOGY FOCUS

All control software can be linked via Ethernet to control the PXD-8080 Series amplifiers in the V8 powered versions, while EASE Focus 2 array software can be used when arraying multiple V-Series (or X-Series) line arrays.

COMPANION PRODUCT

The PXD Series Class D amplifier platform delivers 800 watts for the LF and up to 800 watts for the HF. Proprietary UREC power supply technology offers universal power mains for worldwide compatibility, and is regulated to deliver consistent power even under adverse conditions.

KEY SPECIFICATIONS

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

LD Systems VA8

» www.ld-systems.com

Configuration: 2-way
Dispersion (h x v): 100 x 10 degrees
LF: 2 x 8-inch neodymium cone drivers
HF: 1 x 1.35-inch voice coil compression driver
Frequency Response: 70 Hz – 19 kHz
Maximum SPL: 120 dB
Power: External, LF – 500 watts/HF – 150 watts (both RMS)
Rigging: Integrated hardware, adjustable from 0 to 6 degrees in 1-degree increments
Size (h x w x d): 9.7 x 29.1 x 18.4 inches
Weight: 60.6 pounds
Companion Sub: VAPS215G (dual 15-inch) & V218BG (dual 18-inch)

PK Sound VX10

» www.pksound.ca

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

SLS LS8800

» www.slsloudspeakers.com

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

NEWSBYTES

:: The latest news from ProSoundWeb.com ::

PEOPLE

Adamson Systems has appointed **Justin Stone** to the newly created position of U.S. account manager, responsible for establishing new accounts, servicing the existing U.S. customer base, and creating new business across the United States. Working directly with Adamson director of U.S. operations **Mick Whelan**, Stone is based in Omaha, NE. Prior to joining Adamson, he was the production manager for the Romeo Entertainment Group, and also spent time both as an audio technician and account manager for Audio Visions.

Full Compass Systems has announced that **Cristin Livezey** has expanded her role to vice president of finance and procurement; in addition to managing the company's financial accounting, she will now manage the purchasing department as well. She joined Full Compass as a corporate controller in 2008 and helped to drive initiatives resulting in record sales. In 2010,

she was promoted to VP of finance and has since been dedicated to deepening outside financial relationships, optimizing internal processes and controls, and managing cash flow to achieve profitable growth for the company.

Martin Audio has named **Lee Stein** as vice president, North America. Stein has more than 18 years of industry management, sales and marketing experience, including roles with Avid and Sennheiser. Martin Audio managing director **Luke Ireland** states, "The U.S. market for Martin Audio has grown considerably in the last five years thanks to the support and dedication of (director of U.S. operations) **Rob Hofkamp** and his team, and Rob has obviously been instrumental in developing the brand in the U.S. for the last two decades. Bringing in another heavyweight in Lee with his complementary industry knowledge and organizational skill can further enhance our strength in the U.S. and build upon these foundations to serve and grow our customer base in the next five years."

RCF USA has named **Tarik Solangi** as national sales manager. He joined the

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04 **DSP:** Powerful processing on all inputs and outputs that replaces racks of outboard gear

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:: News Bytes ::

company two years ago, providing technical support for live sound and line array products. Previously, Solangi owned his own production company and recording studio for more than 20 years, racking up more than 500 album credits as a recording engineer. He also previously worked with a variety of national acts and regional bands on tour.

PreSonus has announced **Raul Resendiz** as key accounts manager, where he is managing strategic account objectives with Guitar Center and Musician's Friend as well as working with independent representatives in the U.S. to increase brand awareness, customer and sales staff satisfaction, and sales. Resendiz has worked with Midiman/M-Audio and Focusrite in technical support, quality assurance, marketing, and product field training, and also served as a pro audio consultant.

COMPANIES

Sommer cable GmbH, a Germany-based designer and manufacturer of audio, video, and IT cables and related products, has opened a new subsidiary, **Sommer cable America**, based in Santa Rosa, CA. As a result, the company states that most of Sommer's more than 12,000 products can now be delivered within 5-10 business days to American customers. Newly appointed CEO **Martin Ucik** (pictured here) heads the North American team, supervising all sales, marketing, order processing, and support activities in the U.S., Canada and Mexico.

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Getting The Job Done

Sometimes there's more than one answer to a question.

by M. Erik Matlock

I WILL NEVER FORGET one particular service call.

A small church called me out to quote a new sound system. This room sat about 120 people, pretty standard size and shape for rural Georgia. They weren't happy with the existing system and wanted it replaced.

When I walked into the room, I saw a single, odd-looking loudspeaker mounted tight to the ceiling over the podium. The pastor and tech told me how much they hated the way it sounded. I wanted to find out how much of the gear worked and how much absolutely had to go.

Fairly quickly, I verified that only one wireless microphone system had a problem and everything else was fine. Someone had messed with the system, screwing up the house EQ. Basically they'd fallen victim to a psychotic knob turner. Nothing else was wrong, so I went ahead and tuned the system.

That single, odd-looking point source box lit up the room beautifully. The coverage was almost perfect, there was barely the need to EQ for feedback issues. The sound quality was very impressive. All in all, they had already purchased the ideal system for that room. It was a good design and fit their budget.

This experience has had an impact on my decisions to this day. That one little system shattered most of my pre-conceived ideas about how a system should be installed or look. It changed my perspective, challenging me to focus on actual needs and consider every option.

Later, when working in the live sound realm as "line array fever" spread across the land, I noticed that 9 times out of 10, our point source boxes were perfectly fine for the application. Outdoor concerts, most indoor venues, and various-sized events went off without a hitch using our trusted stacks. The ease of setup and minimal rigging was generally the best choice for "hired gun" contractors like us.

Traditional point source loudspeakers are where most of us cut our teeth in this business. We mixed through various types of heavy boxes and generally walked away pretty happy. We used the best tools available and made good things happen.

I bring this up because on occasion, the phrase "a line array is the only way to go" comes up. Granted, in many applications, line arrays are indeed an excellent choice. I remember when we finally added an Electro-Voice XLC rig to our inventory. It was magical. The level of control was so amazing. The sonic and coverage issues in many difficult venues that had dogged us for years were suddenly tamed. I quickly became a believer. But the front seats and side wing areas still required

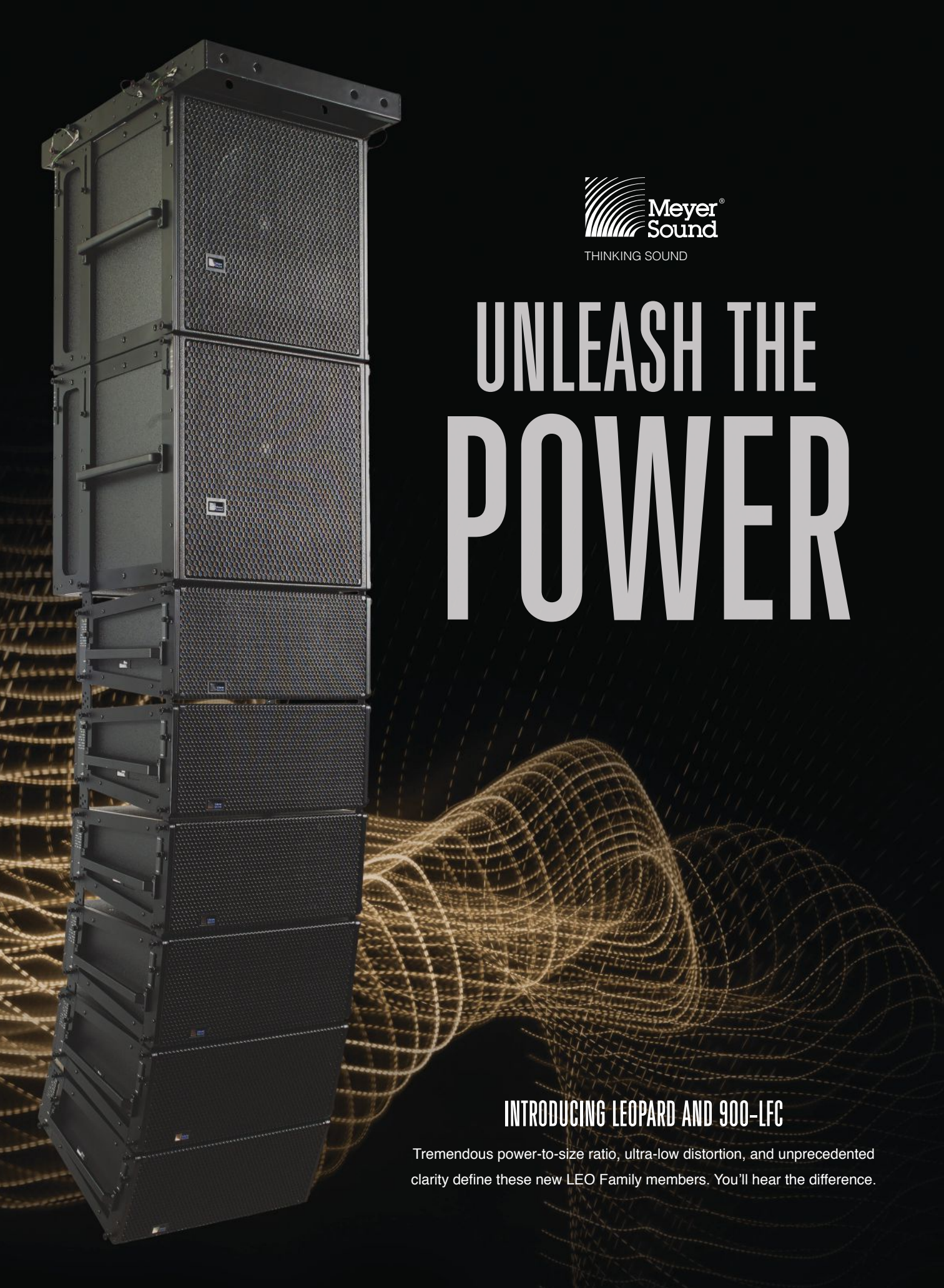
fill loudspeakers to bolster coverage, and point source boxes were the best option for us.

Loudspeakers are simply tools to accomplish specific goals. Particularly these days, we have excellent analysis and modeling programs that can quickly help us understand what's needed in terms of coverage and other aspects, and we can apply this knowledge to selecting the right tools and deploying them correctly.

There are more and more line arrays on the market, and most of them are really (really) impressive. In fact, a bunch of them are featured in this issue, and I encourage you to check them out. The technology is developing and converging to create remarkably cohesive, and in some ways almost automatic, true systems that can achieve amazing things. Yet somewhat under the radar are point source options that are just as outstanding. And underlying it all is our ability (or lack thereof) to use whatever we have to get the job done.

We all do it — the excitement of having the latest stuff can take us to the emotional level of a runaway five year old in a candy shop. But if we're going to practice stewardship and at least pretend to be responsible adults, we need to see things for what they are. There's a right tool for a given project, but so often, more than one. It's our responsibility to understand those tools, what they can really do, and get the most out of them.

M. ERIK MATLOCK is senior editor for Live Sound International and ProSoundWeb, and has worked in professional audio for more than 20 years in live, install, and recording.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