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25th ANNIVERSARY

INSIDE

WORKING IN CONCERT

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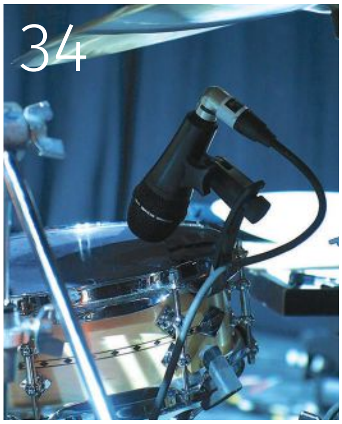

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From the Editor's Desk

IT'S ALL ABOUT "bigger better best" – until it isn't. We celebrate grandly scaled systems and the cutting-edge technologies that help drive their development, and that's a highly interesting facet of professional audio.

Yet there's another reality, one in which many working day-to-day in the field find themselves on a regular basis, or at least occasionally: sometimes the gear is being asked to do more than it was designed to do, and/or sometimes the parameters of a gig shift dramatically, and within mere minutes prior to show time. And that's when knowledge, experience and a deeper understanding of the craft of sound reinforcement matters most.

For more than five years now, Craig Leerman has served as a vital resource to readers of *LSI* and ProSoundWeb in contributing a wealth of articles focused on the nuts and bolts of making the show work, no matter what; the ability to take what's available and employ it to maximum effectiveness. He's at it again in this issue with two pieces doing just that in outlining ways to maximize consoles/mixers in challenging circumstances and also presenting a wide range of microphone strategies, that, as the article title suggests, are "beyond the norm."

Elsewhere, Pat Brown steps up with an in-depth focus on the length of FIR filters, while on the Back Page, Merlijn van Veen looks at the ability (and lack thereof) to detect phase shift. Kevin Young provides the scoop on the unique approach taken with synthpop band Chvrches in the live realm, and he also shares a look at the career of top front of house engineer Kyle Hamilton.

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

Keith Clark

Keith Clark
Editor In Chief, Live Sound International/ProSoundWeb
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ON THE COVER: Mix engineer Paul Gallagher with a DiGiCo SD11i compact console, and more, on tour with Chvrches. *(Photo by Steve Jennings)*

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Meyer Sound Galileo GALAXY

A loudspeaker processor platform that is fully networkable, with multiple units sharing 24-bit/96 kHz multi-channel audio via AVB (Audio-Video Bridging). Maximum input count has been increased from six to eight for implementation of 7.1 systems. FPGA-based processing with 96 kHz floating point resolution adds increased dynamic range, a lower noise floor, and low latency of 0.6 ms (analog in to analog out). Also includes a delay matrix for integration with Crestron and other third-party controllers, a word clock input on the AES3 version, and improved equalization tools. Three versions of GALAXY are available. ***www.meyersound.com***

Allen & Heath dLive v1.2

A firmware update for dLive consoles that includes Director, a multi-platform editor and control software for Mac or Windows OS that allows engineers to prepare shows offline and then mix the performance from a laptop or Windows tablet. It can be used as a supplement to a dLive surface or with a MixRack as part of a surface-less system. V1.2 also provides preamp modeling to the DEEP Processing suite of embedded plugins on all 128 channels, with no added latency to the signal path. Other additions include a surround sound 5.1 main mode plus eight dedicated mute groups on top of the 24 DCA groups. Multiple PAFL buses can now be configured when two or more operators are sharing the same dLive system. V1.2 is available as a free download from the company website. ***www.allen-heath.com, www.americanmusicandsound.com***

Yamaha PX Series

A line of power amplifiers with four new models: PX10 (2 x 1200 watts), PX8 (2 x 1050 watts), PX5 (2 x 800 watts), and PX3 (2 x 500 watts), each incorporating a newly-developed class D amplifier engine that concentrates all necessary DSP functions into a single custom LSI chip. Included are onboard PEQ, crossover, filters, delay, and limiter functions that can be set via the LCD display. Both basic and advanced setup modes are provided, as are eight programmable presets that can be saved to a USB thumb drive. The Config Wizard assigns optimized loudspeaker settings that match various system configurations. Intelligent protection functions help insure reliability, as does proprietary D-CONTOUR multi-band dynamic processing. Inputs are both XLR and TRS, while outputs are binding post, SpeakON, and phone. ***www.yamaha.com***

d&b audiotechnik TI 316 QSC Q-SYS

A software plugin for the Q-SYS platform from QSC enables control and monitoring of d&b amplifiers and loudspeakers from any Q-SYS Core. Power on/off, mute, gain, preset recall and other amplifier controls can be provided from any Q-SYS user control interface (UCI) hosted on any number of control devices – and vice versa – with status information. Q-SYS also allows the status logging of all audio connections to amplifiers and the operating state of connected loudspeakers to be surveyed. ***www.dbaudio.com***

L-Acoustics Kiva II

A compact modular line source that is rated to deliver 6 dB more maximum SPL in comparison to its predecessor, with impedance stated as 16 ohms. The enclosure has also been made more rugged. Kiva II incorporates the company's Wavefront Sculpture Technology (WST) that fosters long-throw capability and even SPL from the front to the back of the coverage area. Its coplanar transducer arrangement and a new K-shaped coplanar transducer configuration generates symmetric horizontal coverage of 100 degrees. Each Kiva II unit weighs 31 pounds and is equipped with flush-fitted rigging. ***www.l-acoustics.com***

QSC SPA Series

SPA2-200 and SPA4-100 power amplifiers are rated to provide 2 x 200 watts or 4 x 100 watts per channel into 8- and 4-ohm outputs, with the ability to bridge channels for 70- and 100-volt capability. They are ENERGY STAR compliant with fast and quiet power-up circuitry. Half-rack in size, they can be mounted under tables, wall-mounted behind displays, or placed side-by-side in a credenza rack. ***www.qsc.com***

Radial JDX Direct-Drive

Simulates the sound of a guitar amplifier while doubling as a direct box, also eliminating variables such as room acoustics and resonance. The unit offers a 1/4-inch guitar input and passive thru-put to feed a stage amp as well as a dedicated tuner output that is buffered to eliminate loading on the pickup or noise from the quartz clock. Signal is passed through a series of filters modeled after the sound of a Shure SM57 microphone in front of a Marshall 4-12 cabinet. Two more settings add Marshall tube head characteristics, with a second adding Fender Twin combo tones. Outputs include 1/4-inch guitar level output and balanced XLR connections. The output includes 180-degree polarity reverse and ground lift switches. The unit can be powered with a standard 9-volt power supply or pedalboard power brick. ***www.radialeng.com***

WorxAudio Technologies XL1

A two-way line array loudspeaker incorporating a 1.4-inch exit compression driver coupled to a stabilized proprietary FlatWave Former that provides 160-degree horizontal dispersion. Dual 8-inch cone transducers are coupled to a proprietary Acoustic Integrating Module (A.I.M.). Powered versions include 2-channel class D model PDA-1000 power amplifiers with integrated DSP, Dante networking, and the WorxControl loudspeaker management system that provides alignment delay, limiter, and compressor. The amplifier can be protected from outdoor elements with the optionally available rain shield. The XL1 is available in black or white plywood enclosures with a multi-layered catalyzed polyurethane finish. ***www.worxaudio.com***

Audio-Technica ATUC-50

A wired discussion system with 24-bit/48 kHz uncompressed digital audio, multi-mode operation, 12-band feedback suppressor, the ability to record directly to a mass storage device on the front of the control unit, and interpretation channels. A choice of 17- and 23-inch gooseneck microphones is offered. Employing standard Cat-5 and above, a complete system can support up to three ATUC-50CU control units, and 150 ATUC-50DU discussion units can be connected in either daisy chain or ring topology, controlled from any web browser with no need for additional software installation. Configuration is via Web Remote Control, with settings able to be stored as presets for recall. ***www.audio-technica.com***

Shure MicroFlex Wireless Expansion

Includes the MXWNCS2, a dual-channel version of the existing 4- and 8-channel charging stations. Connecting via Ethernet, the charging station links docked microphones to Access Point Transceivers. Compatible with MXW1, MXW2, and MXW6 wireless transmitters, the charging docks can be installed on a wall or podium using the included mount bracket. Networked connectivity facilitates remote battery status monitoring and 1-touch transmitter linking to the MXWAPT2 dual-channel Access Point Transceiver or wireless mic routing hardware. Updated firmware (v4) integrates these new components and enables users to establish a backup mic that operates on the same audio channel as the main mic. In addition, a remote pairing function links spare Microflex mics that may be needed to any access point with a single control system command. All units are Dante enabled. ***www.shure.com***

VUE Audiotechnik al-12

A line array designed to integrate seamlessly with other elements in the series due to proprietary Acoustic Linearity principles that are supported by VUE's CST (Continuous Source Topology) as established in proprietary HF waveguides in previous models. Each module includes dual 1.4-inch-exit compression drivers with proprietary Truextent beryllium diaphragms as well as two 12-inch LF neodymium drivers flanking six MF neodymium drivers with Kevlar cones. The new VUEDrive V3 Systems Engine is rated to deliver 9,000 watts total burst power (all channels) as well as provide integrated DSP for loudspeaker management, array element configurations, network remote capabilities, analog, AS digital and Dante networking, along with user-definable input EQ and delay. ***www.vueaudio.com***

Roland Pro AV XI-WSG

An expansion card for the company's M-5000 consoles operating under the OHRCA (Open, High-Resolution, Configurable Architecture) platform. It has three ports for connection to Waves SoundGrid Servers, allowing for low-latency, redundant audio connection and direct connection to a PC for MultiRack SoundGrid control without need of an external network hub. The card accommodates 64 inputs and 64 outputs for integrating Waves plugin processing, multitrack recording, and playback. ***www.proav.roland.com***

Crown Audio DriveCore DCi-DA

Five power amplifier models that are Dante networking enabled. They also include DSP with JBL Professional tunings, network control/monitoring, and the ability to drive 2-/4-/8-ohm, 70-volt and 100-volt loudspeaker loads out of each channel without the need for an external transformer. DCi-DA amplifiers are also compatible with the HiQnet Audio Architect software application and are available in 8|300, 8|600, and 4|1250 configurations. ***www.proav.roland.com***

Sennheiser A1 Frequency Variant

A new frequency variant for the evolution wireless ew 100 G3 and ew 300 IEM G3 series systems developed in light of forthcoming changes in spectrum allocation in the U.S. and Canada. Specifically, systems in both series are available in an additional A1 frequency variant for operation between 470 MHz and 516 MHz. ***www.sennheiserusa.com***

PreSonus StudioLive AR US

Hybrid mixers available in 8-, 14-, and 18-channel versions, all equipped with a USB 2.0 audio interface to capture all input channels, plus the main mix, to a Mac or Windows PC, with 24-bit, 96 kHz quality. Recording and editing are fostered with included Capture software and Studio One 3 Artist DAW. An onboard stereo SD recorder is supplied for recording the main mix without a computer and to play up to 32 GB of MP3 and .WAV files. PreSonus Super Channel is also included, allowing connection of four stereo analog and digital sources. Additional capabilities include Bluetooth 4.1, Class A mic preamps, two instrument inputs, 3-band semi-parametric EQ, pan, mute, PFL solo, and internal stereo effects processors with 16 presets, along with an optional footswitch controller. ***www.presonus.com***

Electro-Voice X1-212/120 & X2-212/120

Additional members of the X-Line Advance line array series provide 120-degree horizontal dispersion for use in wide coverage applications or as nearfield elements when positioned below 90-degree modules in either passive or bi-amplified mode. Also included are integrated rigging, wavefront-shaping circular hydra (X1), pin diffraction hydra (X2), and proprietary FIR-drive settings. The new models are compatible with all grids and dollies currently available for the 90-degree versions. IRIS-Net loudspeaker settings, EASE files, EDS documents, and LAPS 3.3 are also in the package. ***www.electrovoice.com***

Bose Professional ShowMatch DeltaQ

Array loudspeakers with the company's proprietary DeltaQ technology that allows changes in directivity or "Q" for each module in the array to match total array coverage to audience areas and distance, and with phase coherency. Designed for both permanent and portable applications, enclosures include field-changeable waveguides in compact enclosures. Full-range models incorporate four EMB2S compression drivers with two 8-inch neodymium woofers. Vertical coverage options include 5, 10 and 20 degrees. Integrated rigging is rated for up to 24-box arrays. A matching-width single 18-inch subwoofer is also available, along with rigging accessories. ***www.pro.bose.com***

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Ashly Audio digiMIX24

A 24-channel digital mixer with integrated signal processing and effects, iPad remote control integration, and control software. The 24 inputs include 16 class A mic/line inputs); there are also up to 14 mix buses, eight aux outputs, eight channel inserts, 33 channels of dynamics and EQ processing, and two stereo effects processors with reverb, delay, chorus, and more. The control surface pairs with a 7-inch LCD color touchscreen, intelligent metering, and motorized faders. System management software allows users to save, load, or copy mixer setups. Optional field-installable cards provide either Dante connectivity and or a USB interface for multitrack recording and playback. ***www.ashly.com***

EAW RADIUS

A loudspeaker family that includes 8- and 12-inch two-way point source loudspeakers in two horn patterns, a double 8-inch articulated line array, 12- and 18-inch subwoofers, and soon, a 12-inch coaxial stage monitor. All incorporate the EAWmosaic iOS-based application, an integrated platform for prediction, control and monitoring, wirelessly, at any location in a venue. Also included are proprietary EAW Focusing and DynO, along with Dante networking. Models are driven by class D amplification and include large LCD navigation screens that provide on-box access to all functions. ***www.eaw.com***

DPA Microphones MMP-G & GSM4000

The MMP-G is an active cable that's a preamplifier with active drive that facilitates d:dicate capsules connecting to wireless systems. It's available with cable extending from the rear or from the side. The GSM4000 is a gooseneck shock-mount with a clip at the end where a d:dicate capsule can be mounted with a preamp connected to it. ***www.dpamicrophones.com***

Lawo mc² Remote App

An upgraded version of the app for iOS that enables remote control of many of the functions of the company's HD-core-based systems. In addition to controlling a mixing console channel's fader and mutes, and monitoring the corresponding display and labeling, the app allows control of all source- and target-related settings. Users can also load snapshots and pre-configured functions, including selection of monitoring sources, setting of crosspoints, or triggering GPIOs for Red Light or start/stop control of external devices. The upgraded app is available for free download at the iTunes App Store. ***www.lawo.com***

JBL Professional CBT Series Expansion

The CBT1000 full-range column loudspeaker and CBT1000E extension offer adjustable vertical pattern selections for aligning coverage to the geometry of a room. Selections include four "pattern up" and four "pattern down" coverage angles for 16 different vertical coverage combinations. In its asymmetrical settings, the CBT1000 directs more sound toward the farther listeners, helping to counteract sound level differences between near and far listeners. A patent-pending horizontal gradient waveguide provides tapered coverage to reach the front and rear corners of the room while minimizing over-splash. It incorporates a dozen 6.5-inch drivers and 24 tweeters. Continuous output is stated as 130 dB, with 3,000 watts of power handling. ***www.jblpro.com***

Lab.gruppen LUCIA Expansion

Three 1RU models have joined the series, including the D 40:4L, D 20:4L and D 10:4L. All offer four channels, with stated total output power ranging from 4,000 watts (D 40:4L) to 1,000 watts (D 10:4L). They're available with Lake Processing and the company's Rational Power Management (RPM) technology designed to provide flexible allocation across all channels to help ensure efficient and rational use of amplifier inventory without the need to reduce channel count or total power. ***www.labgruppen.com*** LSI

AESTHETICS VS. ENGINEERING

Getting all of the elements of a mix in harmony.

by Karl Winkler

A book I recently read and reviewed for the AES Journal got me thinking about the blend of art and science we face regularly in professional audio. The book, "Audio Production and Critical Listening; Technical Ear Training" by Jason Corey (Focal Press/CRC Press), is a very comprehensive work and covers every conceivable approach to ear training for mix engineers.

As with music, I believe that ear training is important in the audio world. Corey covers everything from EQ to dynamics processing to reverberation to the basics of microphone techniques. The book also includes a CD with software for ear training exercises.

Probably the most important concept is what the author calls "Isomorphic Mapping." He explains that this is a process of "mapping technical and engineering parameters to perceptual attributes; to assist in the linking of auditory perceptions with control of physical properties of audio signals." Frankly, I recommend this book to anyone in the audio business, but perhaps most importantly for those just starting out.

In the early 1990s, when I first became involved with sound reinforcement, a set of CDs by Dave Moulton called "Golden Ears" took a similar approach in terms of presenting audio examples of different frequencies, filters, compression, etc., so that the listener can learn these things "offline" rather than by trial and error.

Before that, a teacher in some of my audio classes insisted that we do a listening exercise at the beginning of each class. We listened to recordings and took notes on the instruments and voices in the mix, their relative levels, any effects we could hear, the acoustic space(s), and anything else that could be detected. All of this training has come in quite handy over the years.

ESOTERIC HEIGHTS

With all of the opportunities for training available, the question becomes "what do we do with them?" As with any tool in the toolbox, the results are usually more about our experience, skill level, and perhaps most importantly with any art form, aesthetic choices. In other words, do the tools influence the way we work, or does the way we work influence how we use the tools? I think the short answer is "both."

Something I've come to understand over the years is that the really top people in any field become specialists. Sure, it could be said that a good rock engineer can probably mix any rock band and do at least a passable job. But what about the vast differences in sound and style between, say, folk rock and metal?

And as great as those differences are, there are even more significant disparities between those genres and other types of music. Is a good rock engineer capable of mixing jazz properly? Classical? Bluegrass? Each of these genres has its own written and unwritten rules, and some of them are found even in obscure corners of the planet.

For instance, I recently attended a wedding in downtown Santa Fe (NM), just off the old Plaza in the center of town. The entertainment for the dinner and dancing was a bluegrass group. True to form, they set up a single microphone and gathered 'round to play and sing. When a particular musician had a solo or a lead line, he stepped closer to the mic. And the results were predictable: the group blended themselves amazingly well, and everything was crystal clear. Did it sound like the record? No, but it did sound like a live bluegrass band with all the energy and excitement you'd expect.

I still don't know how those musicians can play guitar, banjo, mandolin, bass and fiddle, all while singing 2-, 3- and 4-part harmonies! I suppose one answer is that they're specialists and have focused on a narrow niche of music probably for decades. Could these same musicians play other styles? Perhaps, but probably not nearly as well.

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TOP DOWN OR BOTTOM UP?

I've long believed that before we can make really good (and valid) aesthetic choices when mixing, we must have a foundation of understanding, or perhaps it's better described as a "proper set of appropriate tools." The material in Corey's book and the accompanying software files can certainly add to the toolbox.

But then the pertinent question arises: what to do with these tools? This is "bottom up thinking," and to understand the answer, we must look at how we approach our work with audio. To avoid becoming the hammer for which everything appears to be a nail, let's start with the love of the craft.

What kind of music do you like? If you could pick any band or ensemble to work with, who would it be? The answers to these questions are "top down thinking," in other words, taking the 10,000-foot view and seeing the entire picture. Without this view, we might be laying bricks but won't know what we're building.

So if, for instance, Metallica is your dream band, you probably already know nearly every song. And to figure out how that material was mixed, you might employ the critical listening skills and isomorphic mapping discussed above. From there, you could figure out how to improve the mix over what was presented originally. Right? That's a good start.

Or say you prefer classical music. You'd want to study up on the work of Fred Vogler, Jack Vad, John Newton, and Michael Bishop, to name a few. Jazz would be no different, with Rudy van Gelder, Tom Jung, and Jim Anderson coming to mind. All of these great recordings, and the really great live shows, are mixed by specialists in their field. They have all studied the craft in a highly focused manner. Perhaps they didn't start out aiming at one particular niche, but getting there is what allowed these people to rise to such heights of excellence.

WHERE DOES THAT LEAVE US?

It doesn't make sense to try to be everything to everyone. As I recall, it was Aesop who said, "In trying to please all, he had pleased none." So focus is important, but flexibility and going and growing with the flow should be part of the process, too.

And, as always, I heartily recommend engaging in training and education (or "bricklaying" as it's sometimes called) so that the proper foundations are in place for good aesthetic choices based on a proper, deep and always growing understanding of the fundamentals. With that in mind, get started with reading an audio book and listening to some great music! LSI

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

MAKING IT WORK

Maximizing console layouts and other aspects of getting more with less.

by Craig Leerman

Several years ago my company was hired to provide production for a fashion show in a ballroom. In advancing the show with the producer, the audio portion looked to be fairly easy, with just a few wireless microphones for the presenters and music playback provided by a DJ. Outputs were supposed to be the feeds to the left and right main PA stacks, along with a send to the video company recording the show.

Add a board mic and our own music for walk in/out and PA testing, and we calculated needing eight inputs – three for the wireless systems (one spare), two for the DJ input, two for the CD player and one for the board mic. So we decided to use an eight-channel mixer with four mono and two stereo inputs, and to also bring a six-input backup mixer. Back then, analog consoles were the norm so we also needed a front of house rack that included some EQs, compressors, and effects units.

The ballroom could be separated into two spaces by an air wall. The show was to be held in half of the room, and if sold out, the wall would go away and more seating would be added. In case that happened, we added a few portable loudspeakers and a delay unit to my truck pack. On the day of the show we set up the system and were testing it when my guy John ran over with some bad news: there would be a six-piece band instead of a DJ.

The band didn't bring any additional PA or monitors, so John and I took stock of what we had. There wasn't enough time to get any more gear. First, the six-channel mixer would be pressed into service as a sub-mixer to the eight-channel model. Next up was transferring FOH closer to the band because we didn't have a big enough snake. The original plan was to place the wireless receivers at FOH and simply run a few long XLRs to wherever the DJ set up, so we didn't bring a main snake, only a long return snake to feed the amp racks to the mains. We did, however, have some short stage snakes in the cable trunk that would help bridge the gap.

The band consisted of two keyboard players, a guitar player, a bass player, a drummer and a percussionist. There were four keyboards between the two players and we only had three DIs, so to link the fourth, we patched two long 1/4-inch guitar cables together and ran it directly to a 1/4-inch input on one of the mixers. Other solutions included placing a kick drum mic on a folded-up jacket inside the drum (we were also short on mic stands), deploying the extra loudspeakers as monitor wedges, and using gaff tape to secure a mic to a stand that didn't have a clip. Many adapters and barrels also played key roles.

Fortunately we had just enough gear, and were sufficiently creative, that it all worked out. But I learned some valuable lessons from that show, chiefly to always expect the unexpected and to bring extra gear just in case. And those who have worked in the pro audio biz for even a short amount of time likely have similar stories.

DOING THE HOMEWORK

On most shows there's usually enough "rig for the gig," but particularly on freelance gigs, I frequently run out of inputs and/or outputs because of last-minute changes and thus have to improvise. Here are some of the things to attempt to avoid the problem and deal with it (because sometimes it's just inevitable).

First, advance the gig, especially with the right people. A promoter may hire you and provide a rider that may or may not be current, but only the artist's production people really know what they need. So call the production contact and get the scoop on what's really needed.

Next, bring extras for every piece of gear and cable. I work a lot of high-end corporate gigs where there are meetings and seminars during the week, and usually a band or comedy concert at the end of the week. The pressure is on during the meetings because the company has paid a lot of money getting everybody together and the presentations have to be perfect. Same goes for the entertainment, who are often A-list performers.

At this level there are usually spares for everything, including loudspeakers and consoles, but on smaller gigs there may not be enough funds to justify a second large-frame console as a backup. However, fit a spare smaller console into the budget. I carry a backup "utility mixer" to every show. It used to be a small analog unit, but now it's a compact 16-channel digital mixer with output EQs, processing, effects and onboard multi-track recording to USB.

Not only can a utility mixer substitute for a FOH or a monitor console if either fails, it can also act as a sub mixer, as a distribution amplifier for outputs, as a press mult, or even as a fancy adapter for converting one type of signal or connector to another.

ANALYTICAL APPROACH

Sometimes we arrive at a gig and find ourselves looking at a console (that we didn't specify) without enough inputs or outputs for the event. Still, many consoles have more than just channel inputs, and these "extra" inputs can be used to our advantage. For example, some models have sub group inputs that allow a line-level input to route through the subgroup fader and on to the main mix and/or the matrix. These could be used for walk in/out music from a computer, the outputs from a utility mixer, or even for line-level sends from a keyboard.

Another section on consoles that may house an input is the matrix. Again, these will be line-level and can be used as above. Smaller mixers often include a set of RCA jack I/O for recording and playback. With adapters, these RCA line-level inputs can come in handy for interfacing with a computer for walk in/out music or video playback audio. While these inputs don't have any EQ or processing, most computer audio programs do feature basic EQ that can be used instead.

Stereo returns offer another path into the console for line-level signals. Many smaller consoles offer a stereo return volume knob allowing the ability to set and ride levels during the show.

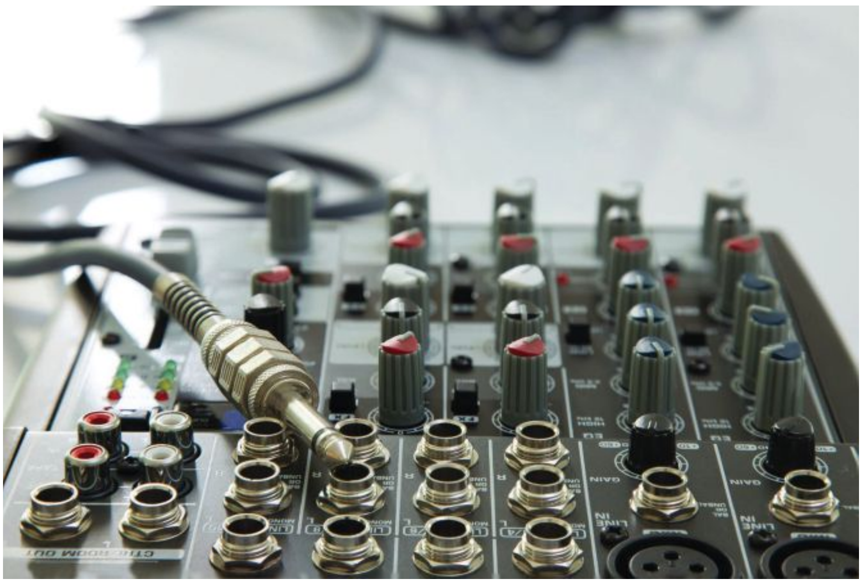
Meanwhile, a way to get extra mic inputs into a console is to utilize transformer adapters that convert a mic level signal to a line-level signal, plugged into the stereo input channels. While basic barrel adapters are not on par with most consoles' built-in mic preamps, they're usually good enough, especially when you're stuck. Most stereo channels allow for a mono signal (usually using just the left input) and the input might get routed to just the left mains. To get the input to both left and right (L+ R) mains, route the stereo channel to a subgroup or matrix and then send those outputs to the L+R mains. Also, using both the left and right inputs on stereo inputs can work well for drum overheads and stereo signals like keyboards.

SENDS EVERYWHERE

Let's move on to the output side. Some events, again including corporate events, demand audio in many locations, including main, fill and delay loudspeakers as well as ties into the lobby and other pre-function spaces, sends to overflow rooms, feeds to video world, teleprompters, backstage areas, dressing rooms and even a send to the intercoms.

Most of the newer digital consoles offer up a slew of onboard and remote network or stage box outputs that can be assigned as needed, but smaller digital and analog consoles can run out of outputs quickly. Here's another area where utility mixers come in handy, for the reasons noted earlier.

Recently while freelancing a smaller event, one more output was needed, and I was just about to break out the utility mixer when I noticed the main desk sported an AES stereo output. The rental loudspeakers had AES inputs, so I borrowed some 3-pin DMX cable from the lighting folks (because AES requires 110-ohm cable) and ran the mains off the AES outputs. That freed up two of the assignable XLR outputs, giving me a



It's always a good idea to keep a utility mixer handy to fulfill a lot of potential needs.

You do carry plenty of adapters at every gig, right?

spare for the show.

Other places to find "extra" outputs on consoles include the subgroup outs, matrix outs, mono out and the control room out. Just make sure you don't solo anything when using the control room outputs on any console because only the soloed item will then appear on that feed. I've also used channel direct outputs in a pinch. These work well when video world wants a feed of just the podium or presenter's main wireless.

And if you're using an outboard hardware processor or EQ, many offer both TRS and XLR connectors that are wired in parallel. While not ideal, both output connectors on an EQ can be employed to get another send.

CUT THE FOOTPRINT

Another way to reduce the amount of inputs is to reorganize and consolidate. For example, at a meeting, unplug the backup podium mic and clearly label the cable end and keep it next to the console, ready to go in case of emergency. Same goes for the spare wireless system. Just be sure to label the ends of the cable clearly so you can grab it quickly and plug it in if/when needed.

Drums are a big aspect to take a look at. Are the overheads really required in a small room or can the cymbals be heard clearly in the audience listening area? Maybe only a single overhead or ride cymbal mic is required.

Further, does the snare really need two mics? A bottom mic indeed does increase the "snap" and provide tonal options, but is it required? Same with a second kick drum mic. I prefer using a two-element mic on kick, and many folks swear by a second large-diaphragm mic outside the drum to get more tone from the resonant head and body, but you can live without it for one gig. A single large-diaphragm dynamic can capture a great sound from most kick drums.

Early in my career nobody owned a console large enough to provide each tom with a separate mic channel. Placing a mic between the two rack toms was the norm for years and in an emergency it works fine. In fact, I occasionally still use one mic on two toms, especially in smaller rooms where the drums don't need a lot of amplification.

Another way to consolidate drum mic channels is to area-mike the kit with just a few well-placed mics, as is done frequently in the studio world. These techniques work particularly well with jazz or world music, especially in smaller rooms or if it's a quieter show. A three-mic setup might have one mic placed on the kick as usual, another positioned by the hi-hat but pointed so it also picks up some snare drum, and the third mic as an overhead to get the entire kit.

An alternative for a jazzy drummer who uses brushes might be to move the hi-hat mic closer to the snare or position it overhead to pick up more snare. These techniques also work well for percussion setups, i.e., using a single mic to capture two or more hand drums like congas or bongos.

FURTHER STEPS

There are additional ways to pare down an input list. Bass guitar DI and a mic on the amplifier? Choose one. Keyboards can be dropped from stereo to mono, and a large keyboard setup can be sub-mixed with a utility mixer down to a minimum of channels. Vocals can also be consolidated with backup singers sharing a mic, preferably one with a cardioid pattern.

Some musicians may only play a second instrument like an acoustic guitar or mandolin for one song per gig. It may be possible to use the same channel for both

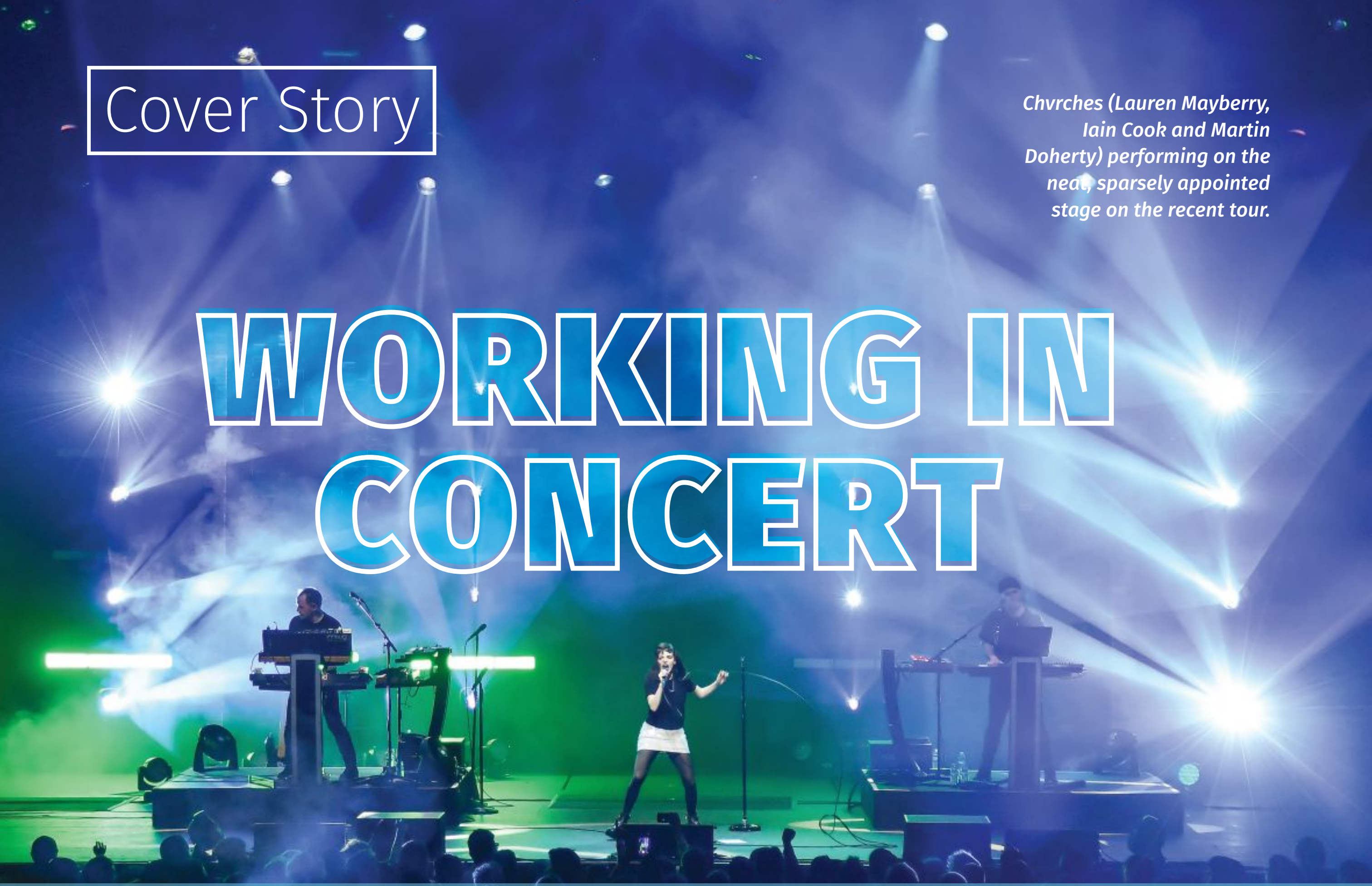
A 3-way transformer-isolated mic splitter is handy for sending a single balanced audio input to three unique destinations.

instruments and take a snapshot of the settings so they can be recalled when the switch is made to the second instrument. In addition, combiner boxes can be used to route two mics to the same input channel of a console. These boxes work well for mics on the same instrument like a piano, drum toms, and mallet instruments like xylophones or marimbas where there's need to "share" the channel EQ.

The next time a surprise jumps into your lap at a gig, keep these ideas in mind. Many shows can be squeezed into a smaller footprint, and if done judiciously and correctly, clients and audiences are none the wiser. LSI

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

Cover Story



Chvrches (Lauren Mayberry, Iain Cook and Martin Doherty) performing on the neat, sparsely appointed stage on the recent tour.

WORKING IN CONCERT

The audio tech effort behind the unique sound of Chvrches. ***by Kevin Young, photos by Steve Jennings***

FOR A THREE-PIECE band, Chvrches (pronounced as "churches") makes a lot of noise. Rather than depend primarily on playback to present its heavily layered synthpop-style music, the Glasgow, Scotland-based band – Lauren Mayberry (lead vocals, synthesizers, samplers), Iain Cook (synths, guitar, bass, vocals) and Martin Doherty (synths, samplers, vocals) – focuses on generating as much of it live as possible.

In fact, both on stage and in studio, Chvrches is all about that "live" feel, which loosely defined is the rough edges that make for signature performances. "That's important to all of them," notes Gareth Russell, the band's stage manager and backline tech, "and it's a very smart system they've come up with."

Hiring instruments and a stage monitoring system wasn't an option on the recent tour of North America, he explains, because everything is so specialized. Take the monitor rig, for example, a compact, integrated affair that relies on computers and a Behringer X32 rack-mount digital mixer. It's at the heart of an audio approach that's self-contained. "We carry a front of house console, have everything down to the media server, and are covered for audio and backline," he says. "For fly gigs, we hire risers, lighting and video, but all we need (on stage) is real estate and power, which is great for festivals."

SUBTLE REFINEMENTS

Russell, who worked Chvrches' first live shows in 2012, has teamed up with Ableton/MIDI/playback tech David Simpson to tweak the live sound approach since becoming involved again in 2016. Now that effort is about subtle refinements, he says, making set-up and tear-down faster, and the entire rig more streamlined and orderly.

Simpson is relatively new to touring, having previously worked as a composer/musician/producer in Glasgow on scores for film and television in the UK. (He also shared a studio with band member Cook.) "I didn't realize how unusual the monitoring system was until we start playing festivals," Simpson says. "It's plug and play as far as possible and it's about comfort and consistency for the musicians. Nobody in the band is particularly demanding about monitor mixes. For them it's more important that they get what they expect without me getting too involved in it."

Generally speaking, he adds, his approach is to provide spare mixes with very little processing, raw and unforgiving, but at the same time serving as a good way for the band to judge their performances and up their games, musically speaking.

Originally, all of the monitoring went through Abelton Live software, but it put a huge load on the computers and created issues with latency. "One of the first things I addressed was to take a lot of that out of Abelton and use the X32," he explains, adding, “It’s still very much a hybrid system. We do a lot of sub-mixing in Abelton because the system has evolved from what the band members set up themselves.”

There’s not much tweaking during shows on monitor mixes. Simpson’s mixing platform is a MacBook Pro, which is used to change scenes between songs, but most mixing is simply done with the X32’s editing software.

HERE TO THERE

While synthpop in style, the music of Chvrches also invokes indietronica, indie pop and electronic dance. Making it all work for the musicians and techs is a combination of instruments, audio gear, and an involved signal flow to get things where they need to be.

At stage right, Cook’s synths – a DSI Prophet 8, Moog Voyager and Roland Juno 106 – feed a Radial 8-channel direct box (DI) and then into the band’s playback racks. From there the signal flows through Focusrite OctoPre MK II pre-amps linked an RME M-32 A/D converter. Signal then splits into an RME Madiface XT interface, is processed and sub-mixed through Abelton, and then comes out as part of a set of audio outputs, via two RME Madiface XTs, into a Direct Out Exbox switcher.

“That’s our standard point of redundancy,” Simpson says. “Everything is duplicated by an A and B system plugged into the two Madiface XTs. From the Exbox that feeds the RME M-32 D/A converter, which we use as a split with a set of tails – 15 channels to front of house on D-Sub to XLR. Also from the D/A, there’s another MADI output that goes to the X32 rack. From there we connect to Sennheiser EW300 stereo transmitters on XLR which distributes the mixes to the band’s in-ear monitors.” (All players are on JH Audio JH16 IEMs.)

Cook also uses a Mike Hill A/B mute box to switch between bass and guitar, which connects to a Fractal Audio Axe FX unit on stage right, which then outputs to a Klark Teknik Square One splitter to FOH and the X32. At stage left, in Doherty’s world, his DSI Prophet 12’s output follows a similar path: into a Radial 8-channel DI that runs along D-Sub to an OctoPre, through Abelton, and then out to the X32 and FOH.

More centrally on stage there’s also a Roland SPD-SX drum sampler for Mayberry, positioned on a skid so that it can come on/off the stage as needed. It’s output on a stereo pair into the Radial 8-channel DI on Doherty’s stage left riser. “That then comes along to the playback racks, but isn’t processed through the Focusrite and Abelton path because of latency,” Simpson adds, “so instead goes directly to the Klark Teknik split to FOH and the X32.

“I’ll go through the playback racks in order,” he continues. “At the top we’ve got a Black Lion Audio Micro Clock MKIII feeding wordclock to the other units. Then there are two Focusrite OctoPres, which handle the synths, then the RME A/D M32, and then the two RME Madiface XTs. Also I’ve got some MIDI hardware in there – M-Audio Midisport 2x2 interfaces and a Kenton Merge box. That’s all run from a MacBook Pro connected via Thunderbolt using a Matrox MX02 Thunderbolt adapter, which converts Thunderbolt to EPCIE connections and connects to the RME Madiface boxes.”

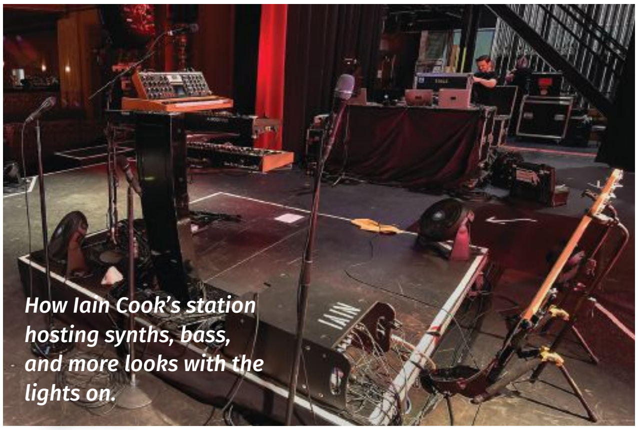
Top, the multitalented Mayberry contributing percussion, with a Sennheiser e965 condenser mic on hand for her lead vocals. Below, multifaceted tech David Simpson in his world of computers, interfaces and more.

FAVORING THE ANALOG

Playback is done via Abelton, but not the most recent version, Simpson notes. “Upgrading is a mine field, in my opinion. Until we have a significant amount of rehearsal time I’m not upgrading my OS or Abelton. Before this I worked in IT and I know that you don’t upgrade until you have to. Upgrades, I feel, are some kind of evil temptation.”

Both Cook and Doherty also play Roland A-800 controllers. “The main MIDI run is for the two A-800s, and the AKAI MPD 226 come into Abelton and run into MOTU MIDI XTs on each riser,” Simpson details. “The output from the A-800s goes into that and comes along another MIDI XT in playback world, splitting the signal to each instance of Abelton and the A and B rigs to maintain redundancy. That’s coming in through the MIDI connection on the XT, which cuts down on the amount of hardware that we’re using.”

In previous iterations, the band relied on a variety of soft synths, but Simpson points out, “One of the things we were


How Iain Cook's station hosting synths, bass, and more looks with the lights on.

all keen to do was to remove soft synths from the equation. Chvrches' thing is using hardware synths. They favor analog sounds and so were eager to get rid of soft synths from an aesthetic and reliability point of view." That said, the band still uses an Abelton drum rack running Simpler and Sampler controlled by an Akai MPD 226.

From the stage, Cook and Doherty use iPads to control playback and change song scenes. "The iPads come down a MIDI out into the Kenton MIDI merge box and go to the M-Audio midisport interfaces, which connects to Abelton," Simpson says. "The great thing about the iPad is that the interface is customized in the way that Iain wants; with MIDI feedback coming over Sysex telling him the name of the song and the section of the song, so he can see an overview of where he is in a song."

There have been occasions when a switch from system A to backup system B was necessary, but Simpson adds that the switch was so seamless he didn't notice until realizing one of the synths was on the wrong patch. "Because of the way the MOTU XT is set up, it needed to be switched manually so the program change hadn't fired," he explains, also noting that there's a also a backup system C, completely separate from the audio flow, for use only if something goes seriously wrong with the RME hardware.


It's a compact footprint for front of house engineer Paul Gallagher as well, centered upon a DiGiCo SD11i and Waves Impact Server.


Stage manager/backline tech Gareth Russell in pre-show prep mode.

MEANWHILE AT FOH

Front of house engineer Paul Gallagher notes that his part of the equation is more straightforward: "Keeping the band loud and getting vocals on top of everything." His approach to mixing Chvrches is keeping the sound as faithful as possible to the band's records while getting their live energy across.

He works with a compact DiGiCo SD11i digital console hosting Waves plugins that run on a Waves Impact Server. "When we chose the SD 11i, I had a few consoles in mind and looked at the smaller consoles offered by several companies," he says, "but the reason we went for the SD11i is the ability to run the Waves server and also have a remote stage box. Specifically, we chose the D2 rack on MADI BNC that allows a 100-meter run from the stage to front of house."

Gallagher also makes use of a Rupert Neve 5045 primary source enhancer, UAD Apollo Twin running Lexicon and Roland effects and an Eventide H9 Harmonizer multi-effect-processor. While most venues on the tour proved well-stocked in terms of house systems, at times he ordered extra subwoofers to attain the desired low-end impact.

Mayberry's lead vocal is captured with a Sennheiser e965 condenser mic that's hard-wired, and she switches to an e965 capsule on a Sennheiser 500 Series transmitter when she wants to move around. For Cook and Doherty on backing vocals, the choice is Sennheiser e935 dynamics. Doherty also utilizes an e965 on the occasions he takes lead vocal duties. All of the mics are split three ways via a Klark Teknik Square One splitter, going to FOH (D2 rack), the X32 for monitors, and broadcast when needed.

Overall it's a truly unique setup, one the band was at the core of in terms of its development. "They're extremely technical – the whole design, that's their vision and how they want the show to work," Russell concludes, adding that it helps everybody with their jobs. "It's brilliant. They are very hands on." LSI

*Based in Toronto, **Kevin Young** is a freelance music and tech writer, professional musician and composer.*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FIR-WARD THINKING, CONTINUED

Are longer filters and more taps better?

by Pat Brown

We live in an age of "bigger is better." During a recent SynAudCon Forum thread, I suggested that this philosophy is not necessarily true for the length of an FIR filter. As of this writing, the typical filter length supported by DSP is 1024 taps, with some as high as 4096 taps. Why would I not want or need longer FIR lengths? If a manufacturer introduced a 8192 tap FIR in its DSP, wouldn't I want to pass over competing units and buy it?

LOWER FREQUENCY, LONGER TIME

First, we need a file for practice and experimentation. A loudspeaker measurement could be used, but a simpler file will make some of the important points more obvious. As a reference, please see **Figure 1.** I created this series of boost filters (ISO octave band center frequencies) in FIRCapture software and saved it as a WAV file. The response could have just as easily been produced by any 9-band parametric EQ.

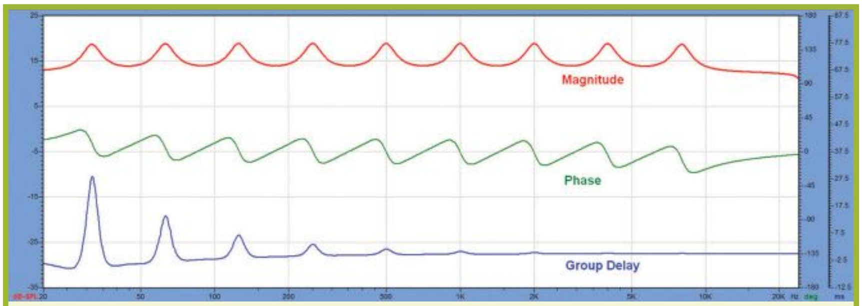

Figure 1: The transfer function of the author's test file.

COURTESY FIRCAPTURE

I selected a Q = 6 to minimize the overlap of the filters. This group of filters forms a filter in and of itself, and is not unlike the response of a loudspeaker, less the requisite high pass and low pass response at the frequency extremes. If this were a loudspeaker (let's pretend that it is), we might wish to equalize the bumps to produce an overall flat response. This file gives us a good way to examine the frequency resolution of FIR filters.

The frequency domain display shows the frequency response magnitude, phase, and group delay. Note that the phase and group delay are alternate ways of showing how the filter modifies the time behavior of the signal that passes through it. Since phase is relative, the phase behavior of each bump is the same.

From an earlier installment, the minimum phase response of a band pass filter starts positive, goes through zero at the filter's center frequency, and ends up negative. This phase shift produces frequency-dependent ringing for signals passing through the filter. The duration of this ringing is shown by the group delay plot. **Figure 2** is a joint Frequency/Time view of the filter. **Figure 3** shows the acoustic wavelengths for each ISO octave center. The relevance? Since the speed of sound is about 1 foot/millisecond (ft/ms), the wavelength approximately equals the time period for one cycle at each octave center. Wavelength serves as a visual reminder that sound waves span time and space, and so must the response of a filter.

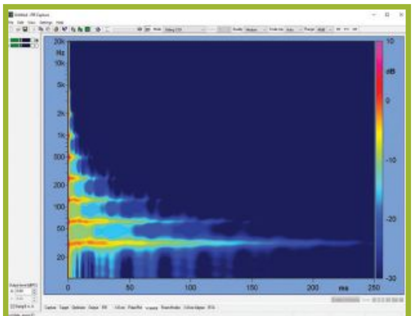

Figure 2: A frequency vs. time view of the same signal. Note the time domain ringing (not delay) of the boost filters. Note also that the ringing is better resolved in this plot (red color approximately equals group delay plot).

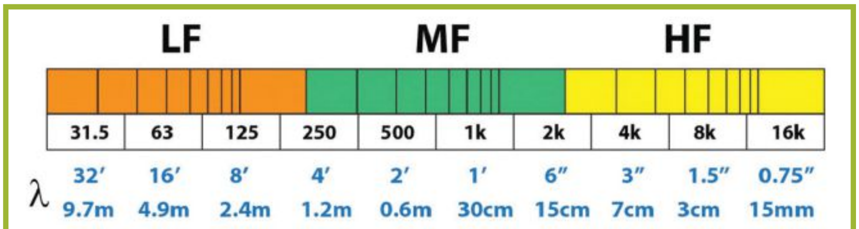

Figure 3: The approximate acoustic wavelengths for the octave centers. No matter what type of filters are used, the time/frequency/wavelength relationships ultimately determine the required resolution of an FIR filter.

COURTESY RANE HAL

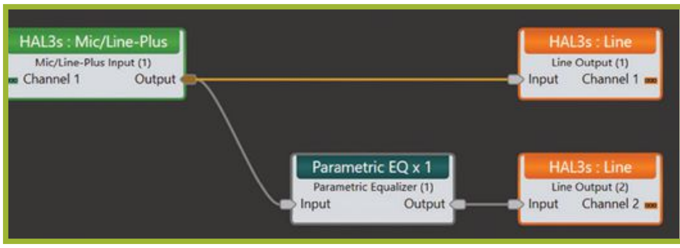

Figure 4: Signal processing setup on a Rane HAL DSP used to demonstrate filter ringing.

The group delay (GD) will be proportional to the wavelength of the filter frequency, increasing in length with decreasing frequency. While the name implies that the signal is delayed as it passes through the filter, this is a bit misleading. The storage properties of the filter cause ringing, where extra cycles are added to the signal. The GD shows the time span that the boost filter occupies. There is nothing strange happening to produce the very long ringing for the lower frequency bumps. It is inversely proportional to filter frequency (proportional to the wavelength), and a given amount of phase shift will produce a longer group delay with decreasing frequency.

To demonstrate, consider the signal processing chain shown in **Figure 4.** Let's pass a tone burst through it and see what happens. **Figure 5** shows the result as viewed on a 2-channel oscilloscope. The blue trace is a 1000 Hz 6.5 cycle wavelet. This is one of Don Keele's shaped tone burst signals. The red trace results from passing it through a 1000 Hz boost filter. Note that the original and filtered signals are in phase, and that the filtered signal has extra cycles. This is time domain ringing. The ringing of a cut filter would span the same amount of time.

IIR EQUALIZATION

What must an analog or digital IIR filter do to correct the response? Either must simply produce the opposite magnitude response and phase shift of the bump. Since IIR filters are recursive (utilize feedback) the filtering process happens "on the fly" based on the driving signal (think of a capacitor charging and discharging). As such, there are no low frequency limits on its operation (at least compared to an FIR filter) and it doesn't introduce a significant amount of processing delay. There will be an output signal so long as there is an input signal, and the filter's impulse response (theoretically) never decays to zero.

Since a positive group delay is the signature of a narrow band boost filter (Figure 1), a negative group delay is the signature of a cut filter. It means that the phase angle has decreased over the frequency span of the filter. **Figure 7** shows a group of IIR filters realized in an EQ block of a popular DSP. Compare the magnitude and phase of each "dip" with the magnitude and phase of each "bump" in Figure 1. The equalizer in Figure 7 would completely correct the response of my example file.

As a side note, for narrow band signals the GD is not a measure of signal delay, but it does show the time span occupied by the filter's ringing. Some terms in audio don't literally mean what their name implies. "Constant voltage" is another example, but that's a different article.

MINIMUM PHASE FIR EQ

In lieu of an IIR filter, let's consider a minimum phase FIR. Its response can be made identical to the IIR filters produced by the parametric equalizer. An important difference is that the FIR filter's response exists in totality independent of the signal that drives it. It is an impulse response of finite length, with a corresponding frequency resolution that is dependent on its length.

Since T = 1/F, the lower the frequency to be affected, the longer the filter length (time span, more taps) required. Correction of the LF bumps requires introduction of the conjugate of the magnitude and phase at each frequency, which also conjugates the GD. That's a long time span for the lower frequency bumps. Refer back to Figure 1 again for a good visual. Because the filter is minimum phase, the processing delay is very low, just like the IIR filter. The difference is that the FIR filter's bandwidth is determined by its length **(Figure 6)**, so a very long filter is needed to equalize the lower frequency bumps. This fact is independent of the processing algorithm used to create the filter.

FREQUENCY RESOLUTION

From Figure 6, a minimum phase FIR filter with 1024 taps and 48 kHz sample

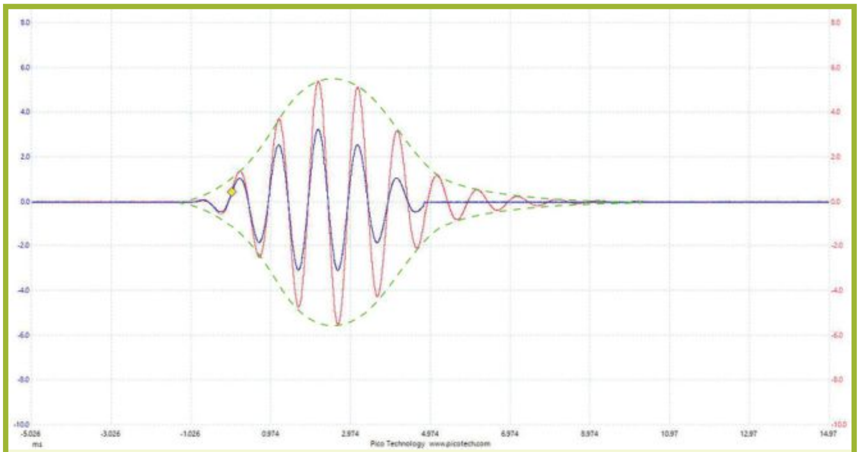

Figure 5: A 6.5 cycle (1 kHz) wavelet. The blue curve is the original signal. The red curve is the signal passed through a 1 kHz 6 dB boost filter (Q = 6). Note that the signal itself is not delayed by the filter block, but the signal envelope (green outline) spans more time.

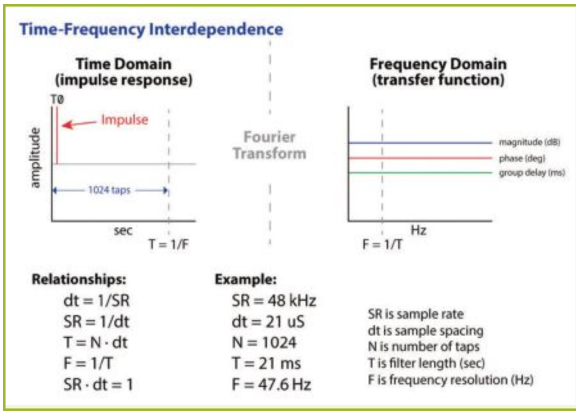

Figure 6: The interdependence of time and frequency resolution. The frequency domain plot is referenced to the exact arrival time of the pulse (TØ) and the time span extends to T. Since the arrival is a perfect impulse, this FIR acts like a pure delay whose bandwidth is determined by the number of taps.

rate will have a length of 21 ms. Since T = 1/F, this makes the frequency resolution of the filter about 47.6 Hz. What does this mean? It is the spacing of the frequency domain data points. It is also the lowest frequency that the filter can influence. In actuality, the realized resolution is a bit lower (higher in frequency) since a few periods of the waveform are required to determine frequency.

What would happen if we add more taps? Doubling the filter length will double the frequency resolution, making the frequency spacing of the data points 24 Hz. It will also reduce the LF limit of the filter by one-half, to about 24 Hz. This trend continues. Each doubling of the number of taps extends the filter's response downward by one octave. So, longer FIRs allow:

1) The filtering to extend to a lower frequency.
2) Higher detail in the frequency response of the filter, since the data points are closer together.

The minimum phase FIR is a case where more taps could be beneficial, since more taps means that the filter can extend lower in frequency. Since the filter is minimum phase, there is no additional processing delay over an IIR filter. About 4096 taps would be required to flatten my example file. This is far more taps than supported by any current DSP, and it fuels the argument that more taps are needed.

BUT WAIT, THERE'S MORE...

Here's the pushback. A minimum phase FIR behaves just like a minimum phase IIR filter, but requires a sufficient number of taps to affect the lowest frequency of interest. Low frequency equalization requires long filter lengths.

But why use a processor-hungry FIR filter block in your DSP to apply a min phase filter? A properly tuned parametric EQ block (IIR) will produce the same response with better frequency resolution using far fewer system resources (Figure 7). It's good audio practice to achieve the needed result using as few system resources as possible, and using FIRs for minimum phase equalization is not efficient.

Let's next consider a case where the response being corrected is non-minimum phase. I took my reference file and added a second order all pass filter (500 Hz) to the response **(Figure 8)**. This causes a phase shift over the bandwidth of the filter, with minimal effect on the magnitude response. In real life, this all pass behavior can result from the use of a crossover network. A minimum phase FIR cannot compensate for this additional phase shift.

LINEAR PHASE TO THE RESCUE!

A linear phase FIR has a two-sided impulse response, where the main signal arrival is centered in the IR **(Figure 9)**. Let's make the arrival peak relative time zero. The time span preceding the main arrival provides a place for the "negative time" arrivals necessary to conjugate energy arrivals after the main arrival peak. These "pre-delays" are causal in respect to real time, but acausal with respect to the main signal arrival. This allows the filter to compensate for reflections using an opposite "negative time" response. So, if your tap length is 1024 points, a linear phase FIR will place the main arrival at T/2, allowing one-half the filter length to provide the pre-arrivals necessary to conjugate the post arrivals produced in the loudspeaker or room. It also allows the introduction of the negative group delay necessary to compensate for the all pass response of the crossover.

By using one-half (1/2) of the filter length for "negative relative time" corrections, the frequency resolution of the filter is halved. For example, a 1024 tap min phase FIR has 47.6 Hz frequency resolution. The same tap length for a linear phase FIR has a frequency resolution of 95.2 Hz, since up to one-half of the filter length is reserved for phase equalization.

ARE MORE TAPS BETTER?

As with the minimum phase FIR, it would seem so. The pushback here is that one-half

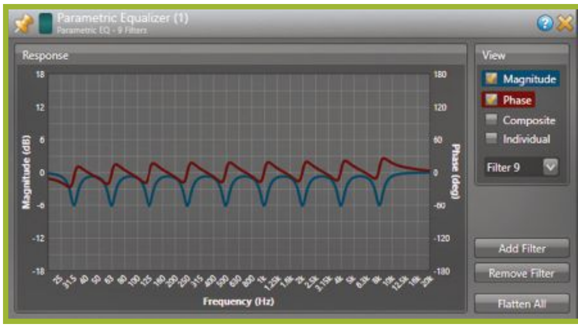

Figure 7: A parametric EQ block (IIR) can fully correct my example file. It has better frequency resolution than an FIR and requires far fewer DSP resources to deploy.

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of the filter length is preserved for non-minimum phase equalization. This translates into processing delay (often incorrectly referred to as latency). Unlike the minimum phase FIR, the delay of a linear phase FIR increases as it is lengthened to increase its frequency resolution. This is not due to the speed of the DSP. It results from TF = 1. There is just no escaping that relationship.

Let's bring up my reference scenario again, but this time for a series of cut filters at the same frequencies as the original boost filters **(Figure 10)**. From earlier, the group delay plot (blue trace) shows the time span of the filter ringing. Note that it appears to be negative, but that just means that the phase shift produced by the filter is increasing over the filter's bandwidth. It does not mean that a "time advance" has occurred or that the filter's response is acausal.

There is no getting around the fact that the time span that the filter must influence increases with decreasing frequency, and becomes very long (~40 ms) at the lowest octave center (31.5 Hz). For live sound and many installed sound applications, the long processing delay is unacceptable. A 1024 tap FIR (48 kHz sample rate) has a time span of 21 ms, which means that the processing delay is about 10.5 ms. Increasing the sample rate to 96 kHz does not reduce the processing delay since twice the number of samples must be processed. There's simply no getting around the fact that 21 ms of time only allows equalization down to about 48 Hz, with a realized frequency resolution of about 3 times 48 Hz or 150 Hz.

IT GETS WORSE

There is another drawback to linear phase FIR filters. Since "relative time zero" is centered in the filter's time span, the frequency resolution is one-half that of a minimum phase FIR. So, if a 1024 tap min phase FIR is effective to 150 Hz, a 1024 tap linear phase FIR will only be effective to 300 Hz. To get back down to 150 Hz would require a 2048 tap filter, which would mean a doubling of the processing delay from about 10.7 ms to 21 ms. For my example file, an 8196 tap FIR would be needed to flatten the entire pass band. The required processing delay is about 85 ms. Ouch!

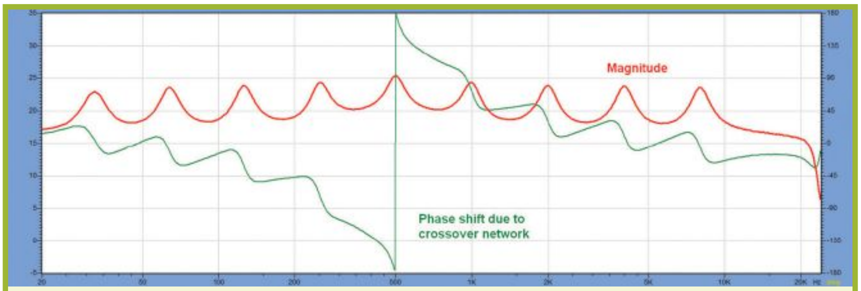


Figure 8: An all pass filter has been added to the filter set. This results in an overall response that is non-minimum phase. A linear phase FIR is required to correct this response.

At face value, the benefit of longer FIR filters appears to be an increase in frequency resolution, resulting in:

1) Operation to a lower frequency limit.
2) Finer filter detail across its bandwidth.

I've already shown that, at least for linear phase FIRs, operation to a lower frequency (more taps) increases the processing delay. This is not due to a lack processing power. It's due to the interdependence of time and frequency. Low frequencies last a long time and occupy a lot of space. Digital processing can't change that.

PROPORTIONAL BENEFITS

There are 10 octaves in the audible spectrum, so each one occupies about 10 percent of the whole (log scale). Doubling the filter length, which doubles the processing delay, only buys you the ability to filter about 10 percent more of the spectrum. Just like doubling the amplifier power to gain 3 dB of level, there is a point of diminishing returns where it's just not worth it. With amplifier power the price is paid in dollars. With linear phase FIRs the price is paid in milliseconds. It could be said that the currency of digital processing is time. You can have some amazing things – if you can wait long enough.

The lowest two octaves (subwoofer range) fall well below the Schroder Frequency for most rooms, where the room modes have a profound effect on the seat-to-seat frequency response. Even if we could make the response "perfect" at a point in space with a linear phase FIR, it will be different at all other seats, so where is the benefit?

Is finer filter detail needed? A 1024 tap filter has data point frequency spacing of about 48 Hz. If that is insufficient for correcting the frequency response, then you are probably trying to equalize something that you shouldn't, such as comb filtering caused by room reflections. The more "detail" there is in the frequency response, the more "position specific" is the time response, since the details are produced by sound reflecting from various surfaces.

Move the mic and the response changes dramatically. An FIR with very high detail will only be appropriate for "correcting" a single point in space. While this might be useful for acoustic echo cancellation for a specific microphone position in a conferencing system, it doesn't provide a benefit for an audience area. Long FIRs don't bring much to the picnic for live sound applications.

DO HIGHER SAMPLE RATES HELP?

That's the word on the street. Unfortunately it doesn't pan out. Referring back to Figure 6, the time spacing of the filter taps *dt* is 1/SR. Multiply this times the number of taps (N) to get the time span (T) of the filter. The frequency resolution (F) is 1/T. These are simple relationships, and they show that if you double the SR the frequency resolution of the filter is reduced by one-half. We made it worse! If you double the SR you now have 2x the number of samples to process, so you have to double the filter length to maintain the same frequency resolution.

Let's go the other way. Halving the SR will increase the frequency resolution of the filter. But this comes at the expense of high frequency response, which extends to about SR/2. Nyquist-Shannon will not be denied.

None of this changes the required processing delay for a linear phase FIR, which is up to one-half the time span that the filter must influence. Higher (or lower) sample rates don't change the time, frequency, or wavelength of the signal.

Time-Frequency Interdependence: Linear Phase FIR

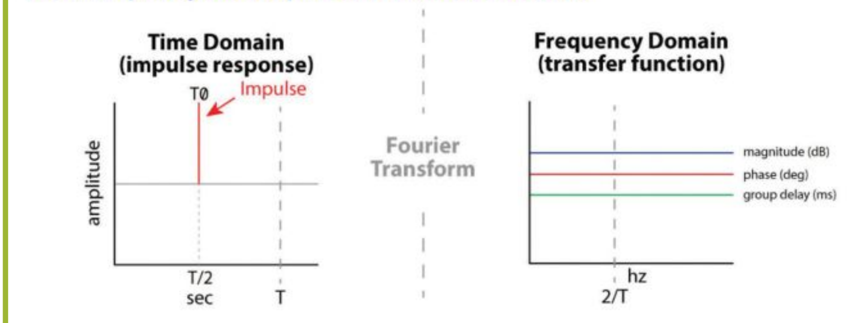


Figure 9: A linear phase filter reserves up to one-half of the taps for phase equalization/echo cancellation, and reduces the filter's bandwidth by one octave. The frequency domain plot shown is referenced to the exact arrival time of the pulse (TØ) and the time span extends to T. The example shown is a Dirac pulse, which behaves like a pure delay. Note that the frequency resolution (2/T) has been reduced by moving TØ to T/2.

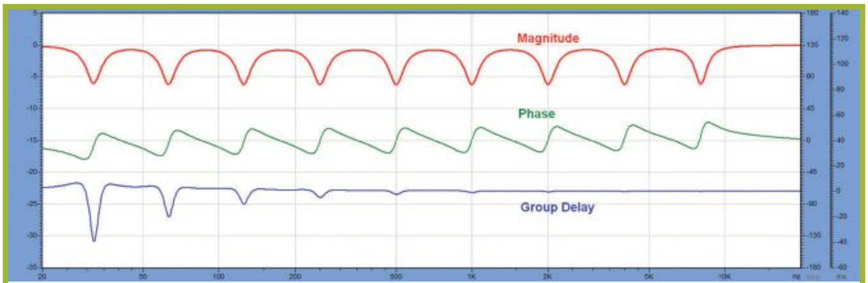


Figure 10: The series of cut filters required to equalize the reference file. Note the unavoidable very long group delay (right axis) for the lower few octaves.

CONCLUSION

There is no doubt that technology will gift us with longer digital filters. Chips keep improving, and so do the products that use them. Many of us can remember when 16 bit/44.1 kHz "CD quality" audio was barely possible. Today it's considered by many to be low resolution. FIRs will follow the same path.

But there are some road blocks to longer filters that have nothing to do with the technology. As I've shown, the major one is the processing delay, which is determined by time span that the filter must influence (see Figure 1). In live sound systems, we can only tolerate so much delay. It's a fuzzy limit, but most would agree that more than about 20 ms for filtering is a pretty long time to wait for a signal. This unavoidable processing delay adds to the latency of the other digital components in the chain, which can easily add another 10 ms or more. We simply have to give up on linear phase equalization at low frequencies due to the time premium, at least for live sound applications.

A combination of linear phase FIRs for HF equalization and mininum phase FIRs for LF equalization seems to be the way to go. These are called "mixed phase" filters, and in my opinion, they are the future. And while some are starting to consider IIR filters to be passé, there is no denying that they have the widest bandwidth, lowest processing delay, and use the least DSP horsepower of any digital filter. Please don't take away my parametric equalizer blocks!

Having a limitation on the number of taps also means that we must give some thought to what we are trying to correct and why. It's like only having one spoonful of peanut butter left in the jar. Use it wisely. How does my equalization affect the entire coverage area of the loudspeaker, not just a "sweet spot" where a measurement microphone is placed? The magic of linear phase FIR filters diminishes quickly when multiple seats are considered. They will likely continue to build auditoriums that way for the foreseeable future.

It's better to have a well-considered application of a 1024 tap FIR than a sloppy application of a much longer filter. Just as human perception places a practical ceiling on the required sample rate and bit depth for analog-to-digital conversion, so do time, frequency, wavelength, and delay place a limitation on the benefits of longer FIR filters. In audio, there is always "pushback" on making anything larger, and digital filters are no exception. It's important to let the pushback push back. This keeps things in balance and promotes proper thinking with regard to what really produces better sound versus what looks good on a spec sheet.

It's another example of "less can be more." LSI

Pat & Brenda Brown *lead SynAudCon, conducting audio seminars and workshops online and around the world. For more information go to www.prosoundtraining.com.*

BEYOND THE NORM

A wide range of problem-solving mic choices and approaches.

by Craig Leerman

Some of us may only have access to a limited microphone inventory in our day-to-day work, while others may have a wider range of choices. Regardless, most tend to stick with "go-to" favorites in terms of both mics and techniques. There's rarely if ever the luxury of testing different mics and positioning at sound check, so we opt for what we know works.

Amidst this routine, however, there are bound to be situations where a creative, unusual mic choice and/or technique is required save the gig. It's a part of adopting a mantra similar to units of the U.S. Marine Corps: Improvise, Adapt and Overcome. With that in mind, here are some mic selections/techniques I've used or seen others deploy over the years to solve a problem.

Let's start with the video director who told me he didn't want any mics or stands, save for the wireless vocal mic for the lead singer, visible in his camera shots. Specifically, he didn't want to see "messy mics around the drums" for the performance of a well-known band at a high-end corporate event in Vegas. We switched from mic stands to drum claws and clips in an effort to make the kit more "presentable," but he still hated the look of the mics.

So I asked him to show me the camera shots he was seeking and then devised a solution. For starters, overheads and a mic inside the kick drum worked because they were indeed not in the video shots, but we still needed to "invisibly" mike the snare, toms and hi-hat. So on hi-hat and snare, we stand-clamped a single mic underneath each and then reversed their polarity.

And since it was a corporate, we had on hand more than a dozen wireless packs with lavs, so some of these mics were simply taped to the lugs on the side of the toms (with the small packs also easy to conceal). It all produced a surprisingly good kit sound and made the director quite happy. By the way, we also hid the guitar combo amplifier mic by placing it at the rear of the open cabinet and reversing that polarity as well.

Speaking of moving, I was once asked to mike a piano that would be moved about the stage during a theatrical performance.

A Telefunken M80-SH with right-angle plug to reduce length.

TIGHT QUARTERS

Another non-standard drum approach that we employ regularly is using a standard podium mic on ride cymbal. It's clamped to the stand below the cymbal, with the mic head able to be positioned easily to where it sounds best by simply bending the gooseneck. Podium mics can also work well in percussion setups because their small size and long goosenecks afford placement options not available with standard-size models.

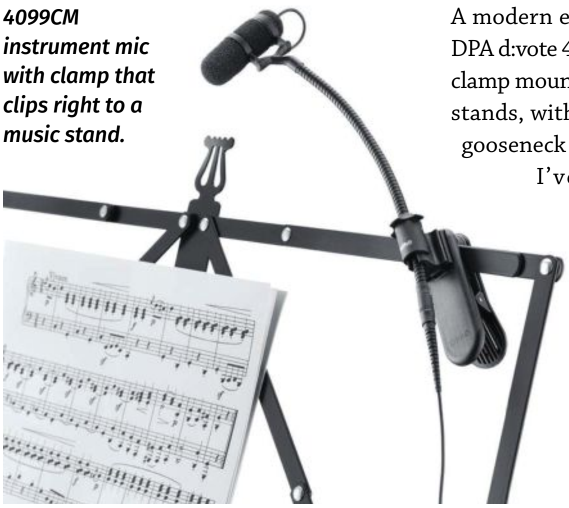
Sennheiser offers a very small and effective dynamic element mic, the e608, that is attached to a small gooseneck and clamp and affords positioning in the tightest spaces. Another solid option when space is at a premium is the Granelli G5790, a modified version of a Shure SM57. It has the same sonic signature as the original but includes a bent shape that makes for easier positioning around a crowded drum kit. The Electro-Voice N/D468 is another mic that

can get into tight spots, with a swivel head that allows pointing the element where it's needed.

To help in positioning, some manufacturers make shorter/more compact versions offering all of the features of their standard-sized models. Good examples include the Heil Sound PR 31BW, a shortened large-diaphragm dynamic PR 30, as well as the Telefunken M80-SH, a compact M80 that even ships with an XLR cable with right-angle plug to further reduce length.

An interesting way to mike percussion when there's no room for conventional units is to place a lavalier on the chest of the percussionist. Using an omnidirectional model that can handle high SPL, like a CO-8WL from Point Source Audio or the Shure MX150/O, allows the performer to play both hand and stick-struck drums, along with hand-held percussion instruments, without the mic getting in the way. Putting these on a wireless pack allows the player to also move about the stage during the show.

Speaking of moving, I was once asked to mike a piano that would be moved about the stage during a theatrical performance. The original thought of using wireless handheld transmitters wouldn't work because they were too large, so I turned to lavs with wireless packs. The lavs were taped to the piano lid in a makeshift attempt at turning them into boundary mics. It worked great and is something I've subsequently used when the need arises.



DPA d:vote 4099CM instrument mic with clamp that clips right to a music stand.

Flautist Bob Chadwick wearing his Countryman E6 Omni ear-set mic.

GOING SMALL

Another unusual approach that's worked out well is employing a head- or ear-worn mic with flute players. I find that if a flautist plays directly into a mic, the breath noise can be quite loud, so instead place a small head-worn model such as a Countryman E6 on the performer, adjusting the element so it's pointing at the instrument and not the player's mouth. This technique also works well with violin and viola.

Years ago, a colleague was working with an orchestra and he ran out of stands. His response was to attach the mics to the players' music stands with small drum clamps and gaff tape. It worked well, particularly for the brass instruments. A modern equivalent would be utilizing DPA d:vote 4099CM instrument mics with clamp mounts that clip right to the music stands, with the added bonus of a small gooseneck to help in positioning.

I've never clamped mics to music stands, but have used them to support boundary mics for recording. Specifically, this involves gaff-taping an Audio-Technica 871 (AT871R) on the desk (the part that holds the music), which can then be raised and tilted desk for positioning. This approach allows the room ambiance mics to blend in with the look of the chamber orchestra.

The same colleague also told me of a speech gig where the podium mic was broken so he gaffed a small lav to the broken gooseneck, using it as a stand. Over the years, I've expounded on his idea and now regularly tape small lavs on podium mics for use as backups. It's a better look than deploying a pair of goosenecks on the podium while also affording presenters a bit more space. The lavs can also serve as dedicated recording mics.

Lavs also provide a way to place a different pickup pattern at the podium, allowing a switch between the two if desired. When placing a lav on a podium mic, make sure it has a windscreen (or two) because many presenters will adjust the gooseneck to point toward their mouths and get right up on the mic. Many lavs aren't designed to be spoken directly into because they're usually worn on the chest, and thus the need for an additional windscreen.

In addition, lavs are commonly used on actors in theater, and my company has worked a lot of these gigs over the years. Normally we place wireless lavs on the actors with speaking/lead singing parts and suspend wired mics over the stage to pick up the chorus. Once we forgot to pack any small suspended "choir type" mics and so instead taped wireless lavs (and their packs) to a light bar above the stage. The audio quality was good, and we ended up using them for dress rehearsals and the three show nights. Plus the motorized light bar afforded an easy way to drop/raise it for battery changes.

BOUNDARIES & OLD SCHOOL

Small mics aren't the only models that can be suspended over stages. Over the years we've flown handhelds like Shure SM57s or Audix OM2s over stages when that's all we had at hand. If possible, select dark-colored mics so they blend into the background, and make sure the XLR connectors on the cables will lock to the mics so they won't fall when suspended.

Another approach is using handheld mics as floor mics for theater. While boundary options like Crown PCC 160s are typically used across the front of a stage for main or backup applications, many venues (schools in particular) don't have a budget for them.

When substituting handhelds in these applications, a key is getting the mics as low to the stage as possible without actually laying on it, because they pick up too much vibration and shoe noise. Put the mics on stands that are placed on the floor or that are located in the orchestra pit in front of the stage, with the front end of the mics just above the stage deck in an effort to get the diaphragms as close to the stage (boundary) as possible. Don't use "ball" mics if at all possible because they increase the distance between the diaphragm and the floor.

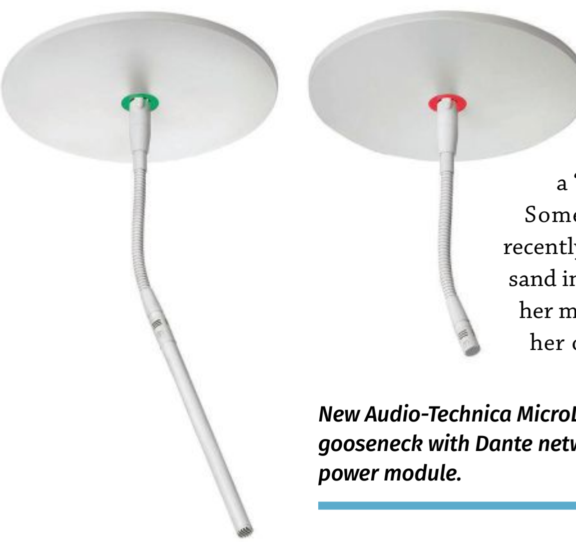
On ocassion vintage mics come in handy as well; for example, a theater setting calling for period-looking mics as props or working props, like the scene in Evita where the star performer stands behind several mics singing "Don't Cry For Me Argentina." If actual working vintage models aren't available or are cost-prohibitive, acquire non-working models and replace the "guts" with small lavs.

I've done this several times, attaching the lav inside, either clipping it on a support structure or stringing rubber bands inside and making an isolation shock mount. Then I can either wire up the lav to the original connector or page the lav's cable through a vent or grill hole and conceal it from the audience. In addition, many companies (i.e., Shure, CAD Audio, Heil Sound, MXL, Cascade, and others) now offer cool vintage-looking mics with "innards" that provide modern performance with a retro vibe.

Ear Trumpet Labs offers a range of "steam punk" retro-looking options.

I've also had inquiries from customers about distinctive mics, not necessarily vintage but ones that look a bit different from the typical models. Some distinctive choices include the copper-colored enCORE 200 from Blue Microphones, the Pearl Series from Violet Design, and eclectic models from Ear Trumpet Labs that have a "steam punk" vibe.

Sometimes color is a big issue. I recently worked with a Barbara Streisand impressionist who insisted that her mic had to be all white to match her outfit. Another client wanted the stage mics to match the company logo color, and some charity clients want emcees with mics in a color that identifies with the charitable cause, like pink for breast cancer fundraisers.



New Audio-Technica MicroLine gooseneck and cardioid gooseneck with Dante network output ceiling-mount power module.

Previously, options were limited to shipping the stripped mic body to a powder coating company or spray-painting the mic in-house. Now, however, several manufacturers offer custom finishes direct from the factory, so with some lead time almost any color can be acquired. There are also temporary slip-on covers offered in a variety of colors and patterns, and even made of beads or crystals.

NOW & THE FUTURE

Table mics, common at corporate events, can also present challenges. Because they're all usually "hot" during meetings, they can pick up distracting table noise. A simple trick is isolating their desk stands from tables with computer mouse pads. Place the smooth side down so presenters can easily slide mics toward them if they want to speak or push them away.

Also note that measurement mics can do more than measure. While many live mics have a bump or notch somewhere in their frequency range that gives them a characteristic sonic signature, measurement mics are made to be as flat as possible. So while they usually make poor live handheld vocal mics, I've found them to be excellent in recording applications and also in omnidirectional situations such as ambiance miking for in-ear monitors.

Finally, the future is networking. Dante has taken pro audio by storm and manufacturers are starting to design mics that interface directly into networks. Audio-Technica is leading the charge, with the ATND971 boundary wired mic that transmits audio and control data together over Dante, as well as the ATND8677 that allows Dante to be used with any phantom-powered condenser gooseneck mic with a 3-pin XLRM-type output. LSI

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

In Profile

TRUE TO THE MUSIC

The world of noted front of house engineer Kyle Hamilton.

by Kevin Young

THROUGHOUT HIS career, Kyle Hamilton has served as front of house engineer for some of the biggest names in music, past and present. Along the way, he's seen the world and also earned substantial recognition as a recording engineer, notably for mixing, among others, Grammy-winning records by Gladys Knight and Pharrell Williams.

Although Hamilton prefers the immediacy of live shows, he began his education in audio with recording. "I was always into music and I was always a technical kind of person," states the 41-year-old, Murrieta, CA-based engineer. "My mom was a hair stylist who styled for the stars, per se, in their homes and studios."

Often, he tagged along. "The consoles captivated me," he says. "I was always like, 'what do all those buttons do? What's this? What's that?' One of my mom's clients was Lionel Richie and he had this massive home studio and I'd be there seeing how records were made. It was magical."

THE RIGHT BALANCE

From that impetus, Hamilton's gone on to become a much sought-after live mix engineer, working with artists including Rihanna, The Isley Brothers and Mary J. Blige, and developing an approach to live sound he feels is integral to his success. "I think what sets me apart is that I want to bring you the record with a live feel. I stay true to the music."

From artist to artist, even from song to song, his method of doing so varies. "You control the atmosphere and mood of the show," he notes. "Every song doesn't beat you up. Every song isn't keyboard heavy. I have to know these records and mix applicably to the record."

He emphasizes finding the "sweet spot" for any backing tracks and treating them as another musician on stage, not too loud or soft. "We run sequencing because there's so much information on these record and if you had (all those instruments) on stage you'd have a 30-piece band. We strip it down, but make it a part of the show. If those nuances are missing, it's like making a cake without eggs; you can't see the eggs, but they add consistency. You can't leave out ingredients and look for the same results."

Hamilton also stresses his interaction with artists – balancing their vision, their musical director's needs, and even management requests, albeit diplomatically. "Management can be standing right next to you saying, 'I want more of this.' Then you think, well, the artist told me in rehearsals that they want it like that," he says, laughing. "So it's a tightrope act."

The surroundings and audience's mood plays into that, Hamilton adds. "You could have a dead audience one night, then do the same, exact thing the next night and the audience is rocking out." His favorite audiences are focused but still partying: "Not scrutinizing the music as opposed to enjoying it or watching the whole show through their phone."

BEING PREPARED

While his level of finesse on the console and ability to read the artists' and audience's moods have helped get him gigs, Hamilton's preparedness for any challenge and his tenacity – qualities instilled in him as a child – have also factored into his success.

Born in Los Angeles and raised in Pasadena by his mother in a single parent family, he and his brother were always encouraged to excel. "She always told us, 'You have to work hard, harder and hardest for what you want.' Life is what you make it – easy or difficult. It's about how much work you put in." His ethic? "Always stay ready so you don't have to get ready."

Growing up, Hamilton took every opportunity presented and felt there was nothing he couldn't accomplish. If he didn't know the answer to a question someone posed, or how to do something, he'd find out. "I would never say, 'I can't do it.' That's just another person telling you what you can't do."

Although he loved music, played drums for "a hot second in grade school" and DJ'd during high school, "My real instrument is my ears," he says. His early education in audio was a matter of being in the right place at the right time, shadowing family friends and his mother's clients and watching them work. "Seeing the records put together, I was like, well, I can't play on them, but there's another avenue in the industry behind the scenes." His fascination with audio gear, he adds, was an extension of his love of gadgets of all kinds and radio-controlled cars.

After graduating high school, Hamilton attended LA's Trebas Institute briefly in 1993, but found the program overly broad and ultimately switched to the LA Recording Workshop, where he finished the two-year program in nine months. "It was everything I wanted, literally, non-stop engineering. I went six days a week, 14 hours a day. The days were long was because in a studio those are typical days and they treated it as such."

CAREER JOURNEY

Recording was what most interested him at that time, but after graduating

Hamilton at the helm of a DiGiCo SD7, plying his trade for a top artist.

Hamilton took a roadie gig with Lalah Hathaway and Gerald Albright, watched, learned and soaked up every bit of information he could from whoever was mixing from the ground up.

Shortly thereafter, when a family friend who produced plays offered him a FOH mixing gig at one of his shows, Hamilton took it immediately. "He called the sound company, Sutter Audio of Tallahassee, FL, and didn't ask but rather told them point blank that I would be mixing. With plays, you're in the venue for two or three weeks so it was a great stepping stone because prior to that I had only been in the studio."

After four years of plays, in 2001, Sutter Audio owner Tom Hair offered him a shed tour with R&B duo KC and Jojo, to serve as a system tech. He'd never flown a PA before, he says, adding, "But every company did it their own way, so I said, 'As long as you send your crew chief out to show me the way you want it done, it'll be smooth sailing from there on out.'"

He continued mixing plays as well as working as a system tech for Sutter Audio along with mixing opening acts on various tours. "I guess people were listening because later I got a call from the Isley Brothers tour manager, the late Woody Johnston, asking if I wanted to mix FOH. At first, Hamilton was uncertain he'd heard right. "I said, 'Wait a minute, you're calling me to mix, not be system tech?' And he said, 'Yes, do you want to mix? Yes or no?' And I was like, 'Hell yeah.'"

He went on to mix the Isley Brothers for nearly a decade. In fact, most of his clients are long term, and although his schedule doesn't always permit him to go stick with every artist exclusively, he notes, "When I come back it's like we never skipped a beat. Job security, for me, is my mix."

KEEPING IT SIMPLE

Over time he's worked with top artists such as Janet Jackson and Nicki Minaj, legends including Lionel Richie and Stevie Wonder and, in 2012, with Prince. "When you go to what I call 'The Purple University' you had to pretty much throw away everything you learned about music and engineering and do it Prince's way." Regardless of what he wanted, Hamilton continues, "You just had to make it happen."

His ability to do so, regardless of the demands of the gig, is a function of staying ready. "Everybody I go out with, I truly know their records," Hamilton says. "At rehearsals I always let the artist hear what I'm doing. I say, 'This is what the room is doing. This is what your show sounds like.' I'm quick to give them a board mix any day because my mix sounds like their record, with the arrangements per their MD's desires."

While he has preferences regarding gear, having grown up in the business when analog and 2-inch tape were the norm, he's ready to mix on anything. His philosophy is to keep it simple, and although he owns a substantial amount of outboard equipment, he prefers to leave most of it at home.

"Adamson Energia is my PA of preference and the DiGiCo SD7 is my console of preference," he states, "but there's no console I can't use. I pride myself on learning all the desks because I'm not going to say I've never seen this before. That's never going to come out of my mouth."

Hamilton first used a DiGiCo D5 desk in mixing artists such as Mary J Blige and Lionel Richie, then transitioned to the SD7, which he praises for its "transparency, analog feel, and warmth." It's the console he used exclusively on Rihanna's Diamonds World Tour in 2013, which he singles out as providing one of his most memorable gigs: "Mixing Rihanna at the Stade de France was magical because she was the youngest artist to first sell out that place and, for me, as a young, black engineer, to rock that stadium, that was huge."

ADDITIONAL PURSUITS

Increasingly, however, what's more of a focus is his family. "I love traveling, but as the father of a 6-year-old, my son Austin, I don't want to miss out on his world." Hamilton also recently got engaged and, while he has no intention of getting out of the world music and audio, he is looking for other things to work on that will let him spend more time at home.

Typically, about 90-percent of his work is on the road, and although he still records – most recently mixing Mary J. Blige sideman Shon Hinton's solo project –Hamilton's interests extend beyond music to other entrepreneurial pursuits. And recently, owing to his life-long love of sports, he's begun marketing a variety of new training/medical devices developed by chiropractor Craig Dossman, including the Doc N Roll, a stand-up foam roller, the first of it's kind, that offers more flexibility than that of floor-based devices.

"It's not that I'm going to transition out of music," he states. "I just want to have different irons in the fire that so I can see my son grow up and be completely in his world because I didn't have a father."

He remains fascinated with gadgets, a passion he shares with his son: flying drones, radio-controlled helicopters and working on radio-controlled cars. "Flying drones is something that's real relaxing for me. It's an expensive hobby, but fun. And when you break something, rebuilding it is the most fun. Working on and fixing things make you appreciate them more."

That remains key to his career and is also a part of his ethic of staying ready. "It's what I live by," Hamilton concludes. "There's nothing that I'm not ready for. If I get asked today, 'Can you be at the airport in four hours?' I sure can. You have to stay ready because at a moment's notice everything changes. Life is never as promised. Nothing is guaranteed. You have to keep your edge." LSI

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

FACTORS OF A GOOD SYSTEM

A think tank on getting to the bottom of the problems. *by Bob Thurmond*

EDITOR'S NOTE: This fine article was featured in the March 2004 issue. We reprint it here in celebration of our 25th anniversary.

How many sound systems have been built and are in use? Many millions, for sure, and they're found in all types of venues and for all kinds of programs. So one would think we'd know exactly how to do it by now. But there seems to be plenty of examples to prove that we don't.

Why should this be? What is it we don't yet understand? Do we even know enough to know what we don't know? Perhaps we should start by trying to define the characteristics of a good system. Not just "it sounds good" but – exactly – what makes the difference between "good" sound and not so good. Then we might be able to quantify how good each characteristic needs to be and how to judge whether it's good enough or not.

After nearly 40 years spent designing and testing sound systems, I've finally come up with a list of the primary factors that I feel make up what we could call quality in a system, and why. In this installment, I'm going to confine my list and discussion to systems for speech reinforcement only.

1. RELIABILITY

The most important quality factor has to be reliability. No matter how good the performance of a system may be, if it fails to work, it is useless. Reliability is largely an engineering matter, involving component selection, configuration design, and assembly and installation correctness, for example, but any system can be abused to the point of failure.

Significantly, failure may not be abrupt and catastrophic, but instead may take the form of performance decline due to damage. One particular, and common, example of damage-induced deterioration can be found commonly-used transducer for higher audio frequencies, the horn and compression driver combination. Drivers have a severe amplitude limit; if overdriven, the driver diaphragm will impact the phasing plug, an essential part of the structure. If the diaphragm material is metallic, it can fracture and fail.

Some diaphragms, however, are made of a resin-impregnated fabric, which is much less brittle and, therefore, more able to survive a collision with the phasing plug. Repeated collisions, however, still cause progressive deformation (or warping) of the diaphragm, resulting in eventual failure and therefore, progressive decline of the driver's performance characteristics.

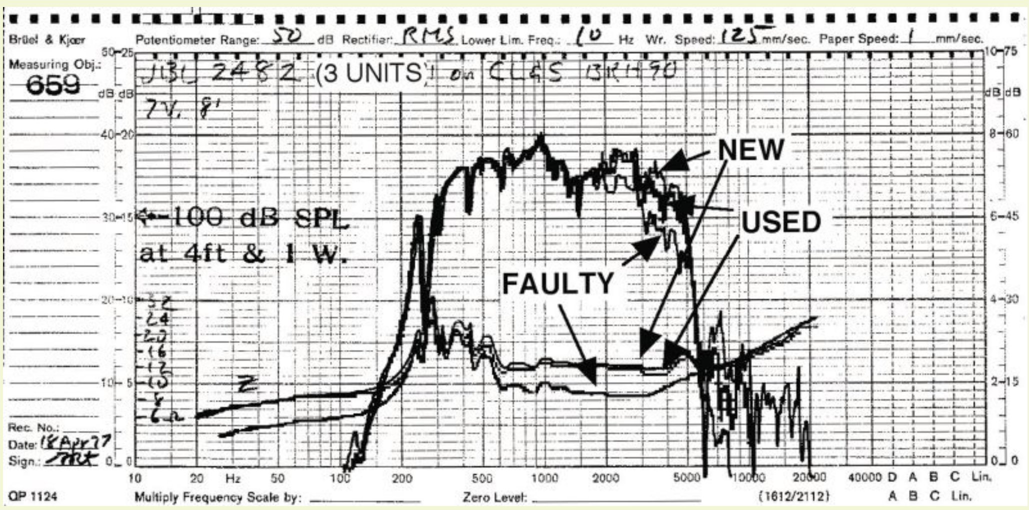


Figure 1: Measurement of output acoustical and input electrical impedance characteristics of three high-frequency horn drivers of identical model but different usage histories.

Predicting and detecting this impending failure, however, is not easy to do. The audible change in performance is fairly subtle and can be detected reliably only by careful comparison of the sound of a single questionable driver with that of a known good one. In the field, such a comparison is usually impractical. Further, a driver that has been used heavily for some time will also exhibit some performance deterioration, even though it has never been over driven into diaphragm collision.

Figure 1 illustrates these performance differences. The frequency response (amplitude versus frequency) of three drivers of the same model (with an impregnated-fabric diaphragm), one new, one well used but apparently undamaged, and one with observable damage. It can be seen that the response at higher frequencies changes with use or abuse. The differences between the upper two measurements are slight, while the third one is significantly different.

There seems to be a good relationship between the measured and (subjectively) observed performances in cases like these, but no real study of this relationship has been performed. So it would seem that a response measurement could be a valid substitute for a listening test. In fact, such a relationship has been established under certain circumstances, but not definitively in a sound reinforcement context. An investigation of this relationship would certainly be worthwhile.

However, there is another measurement that is easy to make, even though it's seldom done. The bottom three curves on Figure 1 represent the measured electrical impedance at the input terminals of each of the three drivers. Such a measurement is usually quite easy to make, even on a driver installed in a system.

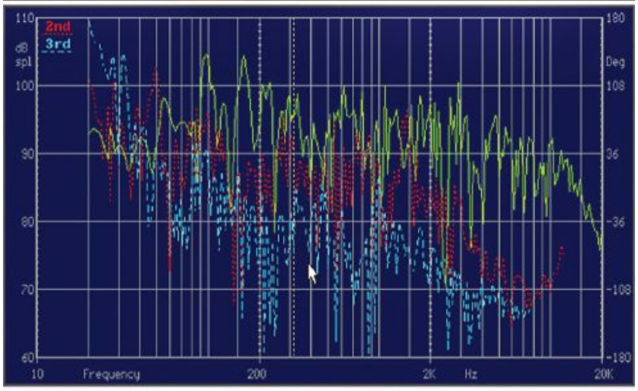
It's apparent that these curves separate the characteristics of the three drivers as well as any other common measurement does, especially in the case of the damaged unit, and much more easily. In fact, automated tests of this type have been designed into integrated systems as performance and reliability checks, with good results.

Thus it appears that different types of tests on the same items can yield corresponding results. Experience has shown that such relationships hold in some cases but not in others, and that it may be difficult to predict which is which. And in many cases, no acceptable substitute for a listening test has yet been found. Worse, some widely accepted tests might prove inadequate.

2. LOUDNESS

It's obvious that any sound system must provide enough sound level at the audience locations to ensure a satisfactory listening experience. Defining what this level actually should be is less obvious, and use of a valid measurement technique is not obvious at all. Subjective opinions on appropriate sound levels often vary widely as well, depending on a host of factors. (Investigating this matter alone could become a major research project!)

In fact, the correct sound level may not be just a matter of loudness. How well speech is understood (intelligibility) is often the overriding concern, and this is the result of many factors



Some desire the added distortion that can result from tube-based components. But is this a bad thing?

other than just loudness. In some cases, the loudness may need to be set other than as would normally be expected, because of adverse acoustical or system functional characteristics. It may also be found that the audience prefers a sound level different from that which exists near the performer.

Other acoustical factors may also be highly significant. The level of the reinforced sound must be sufficiently higher than that of any background noise so that speech intelligibility or program enjoyment is maintained. Some guidelines in this regard have been established empirically, and they may be adequate for most situations.

A common and complicating factor is that background noise level may vary significantly, rapidly and unpredictably. Further, since adequate performance in this area may be a matter of life safety, accuracy can be quite important. It's often the case that the desired sound level is greater than that which the system is capable of producing without difficulty. This difficulty is the result of one or more components overloading, which results in an audible distortion of the sound.

Distortion may take various forms, depending on the type of component that is overloaded, the magnitude of the overload, and the nature of the program material, among other factors. Therefore, the audibility of the distortion may vary greatly with the situation, and each type of distortion must be evaluated individually. Many listeners even believe that certain types of distortion are desirable, such as that typically produced by vacuum tube amplifiers. This usually applies to music playback systems in small rooms, however, so it's unclear if such an effect is valid in a larger sound reinforcement situation.

Some devices are available that deliberately introduce controlled distortion, specifically for pro audio applications. Many have noticed that a limited amount of distortion adds to the apparent loudness of amplified sound, and without being objectionable. If anyone has actually studied this effect, the results remain obscure.

3. TIMBRE

The overall timbre, or tonal balance, of a sound system undoubtedly has the strongest influence on the overall perceived quality. This characteristic is easy to measure, both subjectively and objectively, and there is a very good correlation between the two in a small-room configuration.

In a large-room sound reinforcement situation, however, this correlation does not hold. If the system has an overall response that is measurably flat (has nearly the same input-to-output level ratio at all frequencies), it will sound too bright, with the high frequencies being too loud. A system which sounds subjectively flat, so that the reproduced sound is perceived as being a close duplicate of the source, will have a measured response which rolls down at high frequencies.

Should the analysis be done with a swept filter, which yields more information, or is a stepped filter technique acceptable? What amplitude smoothing or averaging is appropriate? If measurements are taken at single, discrete frequencies, as are commonly done with contemporary techniques, how many measurement points are needed and at what spacing? This could be a major source of misleading data, especially at lower frequencies.

Despite countless practical field experiments in this area, beginning at least 65 years ago, little critical research has been carried out.

Whatever the technique, how many measurement locations should be taken, and where should they be located? And exactly how should the individual measurements be averaged to yield the overall system response? Also, how much variation between individual measurements is acceptable, and what should be done if the variation exceeds this tolerance? Schulein documented this discrepancy in 1975 in an elegant experiment and offered a plausible explanation. He noted that in all rooms, the listener receives sound directly from the source and also reflected from the room surfaces. *(See Sidebar)*

In a small room, the level of the direct sound is almost always higher than that of the reflected sound and, therefore, dominates in the perception process. Because of directional characteristics of human hearing at high frequencies, largely due to head shadowing effects, less total sound energy enters the ears at high frequencies than at lower. This imbalance is perceived as normal.

In a large room with typical acoustics, however, the opposite is true; the level of the reflected, or reverberant, sound is significantly higher than that of the direct at most listener locations. Since this reverberant sound arrives at the listener from all directions rather than just one, more of it enters the ears at high frequencies. Thus the highs are perceived as being louder.

A simple experiment tends to confirm this theory. A loudspeaker is located at head level in a relatively non-reverberant environment and fed with broadband noise. A listener stands one to two meters (about three to six feet) in front of the loudspeaker and slowly turns around while listening to the tonal character of the noise. Typically, the overall tonal balance will change little, if at all, with head direction.

However, if two identical loudspeakers are placed two or three meters apart facing each other and both are fed the same broadband noise, a listener between them, turning around as before, will hear the high frequencies more loudly when his ears are toward the loudspeakers than when he is facing one or the other loudspeaker.

The measured response (and perceived timbre) of a loudspeaker in a room deviates significantly from its performance in an anechoic environment, in ways that are complex and quite difficult to predict. Also, these deviations are different at each location in the room. Therefore, the only practical solution is to measure the actual response of the completed system and correct it as needed with additional circuitry.

This turns out to be a bit trickier than one might expect, however. If a pure tone, slowly swept in frequency, is fed over a sound system and the resulting level is measured at a point in the audience area, it will be found to consist of strong peaks and valleys, tens of decibels in amplitude, and spaced at intervals of about 1 Hz, caused by room resonances.

It's almost impossible to get meaningful information from

Schulein's Experiment

Two identical loudspeakers are arranged in a room so that the listener receives the direct sound from both, one close and one distant. The bandpass filter is first set at 1,000 Hz and the attenuator adjusted so that the sound level appears to be the same when the noise is switched from one loudspeaker to the other.

Then the filter is changed to another band and the corresponding band on the equalizer is adjusted until the apparent loudness is the same from each loudspeaker. This is continued through all bands, and repeated as necessary until no further adjustments are necessary.

Finally, the overall frequency response of the equalizer is measured, this being the correction necessary to make the distant loudspeaker sound like the near one in this room.

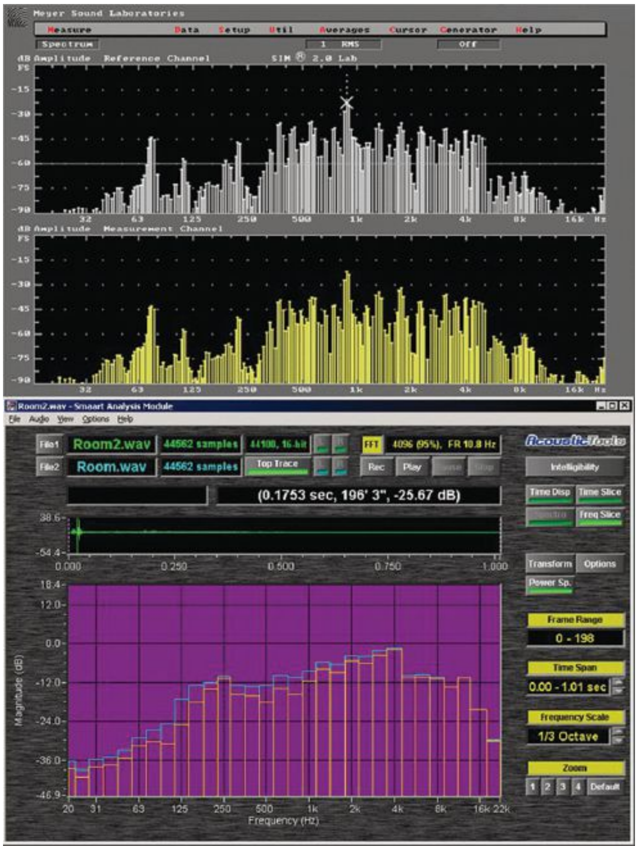
such readings. Besides, we don't perceive these variations because they are averaged by our hearing process in ways that are only partly understood. The measurements must incorporate averaging which simulates the hearing process.

However, this presents us with a shopping list of unanswered questions pertaining to the measurement techniques. What frequency resolution (bandwidth) is needed? A first assumption might be to use a bandwidth similar to that of the auditory (critical bandwidth) filters, but system measurements are typically done with third-octave filters, which are considerably wider than critical over much of the spectrum.

Should the analysis be done with a swept filter, which yields more information, or is a stepped filter technique acceptable? What amplitude smoothing or averaging is appropriate? How many measurement locations should be taken, and where should they be located? And exactly how should the individual measurements be averaged to yield the overall system response?

Despite countless practical field experiments in this area, beginning at least 65 years ago, little critical research has been carried out. As a result, there exist only a few *de facto* standards, and the actual results of these procedures vary considerably in quality.

In addition to the these considerations, it might be expected that nonlinear distortion in any of the system's components, especially the loudspeakers, would significantly affect its timbre, but such does not seem to be the case. The distortion levels of modern components, properly used, are low enough to be unnoticeable in a reinforcement situation.



Tools like Meyer Sound SIM and (now Rational Acoustics) Smaart can be of great help along with an entire understanding of what's happening with a system and room.

4. INTELLIGIBILITY

As the name suggests, intelligibility is the measure of how easy or difficult it is to understand speech over a system. It's ultimately measured subjectively and directly, typically using rhyming words as the test signal. The execution of this test is tedious and time-consuming with only one test subject, which is quite inadequate. Different subjects will render somewhat different results even under apparently identical conditions, and conditions vary significantly with location, program sound levels, room noise, hearing acuity, and many other factors.

The typically broad variance of test results makes it difficult to determine whether a system is actually performing acceptably or not. It hardly seems worth the rather considerable effort required to execute such a test, but there may be little choice. Because of these difficulties, a lot of effort has gone into devising an objective test regime, with several products resulting. All these involve dedicated gear and techniques, which, while not simple, are quite preferable to subjective tests.

These objective tests have been demonstrated to produce results comparable to those obtained subjectively in some, but not all, conditions. Unfortunately, the worst correlations tend to occur in conditions that produce low scores, exactly where accurate results are most desired. And it gets worse. Low intelligibility scores, which indicate serious problems, usually provide little or no information on the nature of these problems. Sometimes one or more physical problems are apparent in such cases, but are these really the causes of the poor performance?

Often, the only way to be sure is to correct the problems and see if that improves the scores. Of course, this may be completely impractical, and in fact, there may be multiple problems, some masking others, so that correcting the most obvious might accomplish nothing useful.

A much more practical approach might be to identify exactly which physical factors adversely affect speech intelligibility, and how, and calibrate physical measurements to subjective effects. If this were accomplished, then not only would meaningful test methods be available, but effective design criteria could be established to predict results and avoid problems in the design stage. Some significant work has already been done in this area, with results pointing to the ratio of direct to reflected (or reverberant) sound being the most important factor. LSI

At the time this article was published in LSI, ***Bob Thurmond*** *served as principal consultant with G. R. Thurmond and Associates of Austin, TX. Also note that it was originally presented as a paper at the 146th Meeting of the Acoustical Society of America.*

FULL DEPLOYMENT

Attention on recent sound reinforcement applications of note. ***by Live Sound Staff***

New Renkus-Heinz VL3 line arrays and more at Town Hall in Manhattan.

REFRESHED AUDIO AT A VENERABLE NYC VENUE

Town Hall in midtown Manhattan, one of the most venerated concert halls in the U.S. since it opened its doors in 1921, recently implemented a new main sound system specified by Audio Incorporated (Roselle Park, NJ) that's headed by Renkus-Heinz VL3 self-powered, 3-way line arrays.

"The VL3 line arrays deliver a clear, transparent sound," explains Audio Incorporated president Stephen Tolve. "Whether the show is classical or jazz, rock artists like David Crosby and Jackson Browne, or A Prairie Home Companion, you want it to sound as if you were listening to the performance without needing a sound system-especially in a venue that's famed for its acoustics."

The 1,500-seat Town Hall is a very wide room, with a short downstairs area, so there aren't many rows in the audience down on the floor. "But the balcony is quite deep," Tolve observes, "so coverage there was an issue in the past. Not with the Renkus-Heinz system, though."

Tolve's team chose three VL3 self-powered line arrays and one Renkus-Heinz DR18-2R dual 18-inch subwoofer per side for the main system. The system sits on carts that can be rolled out on either side of the stage. Coverage to the venue's deep balcony was solved with two Renkus-Heinz CF101LA modular point source arrays per side.

NEW MIX CAPABILITIES AT A MULTIFACETED PERFORMANCE CENTER

The Krannert Center for the Performing Arts at the University of Illinois Urbana-Champaign has a new Avid VENUE | S6L mix system to serve its diverse needs. Taking up a full city block, the venue houses four indoor theaters, five indoor stages, and an outdoor amphitheater.

After renting an Avid VENUE | Profile System for several years,

Left to right, Keith Norton (AV specialist), Rick Scholwin (audio director) and Alec LaBau (asst. audio director) with the new Avid VENUE | S6L at the Krannert Center.

the center decided to invest in a VENUE | S6L system acquired through Chicago-based audio equipment provider TC Furlong. "VENUE | S6L gives us the best of both worlds," states Rick Scholwin, audio director at Krannert Center. "There are many great consoles out there, but keeping in mind the educational aspect, it was a no-brainer for us to go with Avid."

The VENUE | S6L is primarily used in the 979-seat Tryon Festival Theatre, where musicals as well as rock bands and jazz artists like Brad Mehldau and Joshua Redman take to the stage. It will also be used in the 2,066-seat Foellinger Great Hall, a large symphonic space that hosts everything from the Chicago Symphony Orchestra to opera stars like Renée Fleming, as well as in the 678-seat Cowell Playhouse.

MONITORING IN THE ROUND FOR A DYNAMIC ARTIST

Eight EAW Redline RL12s forming a monitoring ring on the NYCB Theater stage for Patti LaBelle.

Patti LaBelle recently performed at the historic NYCB Theater in Jericho, NY (near Westbury) with Eastern Acoustic Works (EAW) Redline loudspeakers serving as stage monitors. Since 1966, the 3,000-seat theater-in-the-round has presented many top names in contemporary music and comedy.

Ronkonkoma, NY-based Audio Design Group supplied the eight Redline RL12 loudspeakers used in monitor mode, which were placed around the lip of the venue's turntable stage. Reuben James, LaBelle's monitor engineer since 2010, had specified bi-amped wedges for the downstage mix. "The Redline RL12s were more than up to the task," he states. "I was very impressed by the consistent coverage of the spectrum from 98 dB to the 125 dB climax of the show."

The RL12 includes an onboard 1,250-watt (bi-amplified) electronics package, with EAW Focusing processing refining impulse response and proprietary DynO algorithms optimizing headroom and sonic at maximum output. The enclosure incorporates symmetrical monitor angles to form left-right wedge pairs.

Phil Christie, president of Audio Design Group, concludes, "It was a very successful event with the RL12s playing a key role in the production. Everyone was very happy with the end result."

KEEPING LEVELS UNDER CONTROL DOWN SOUTH

d&b audiotechnik J-Series arrays deployed for the CMA Festival at Ascent Amphitheater in Nashville.

At the annual Country Music Association (CMA) Music Festival in Nashville in June at the Ascend Amphitheater, Morris (Nashville) utilized its new d&b audiotechnik J-Series system along with NoizCalc far-field immission modeling software to achieve as much noise reduction in the far field as possible while maintaining the concert sound experience.

"This was the first year Ascend Amphitheater was included as a venue for free outdoor concerts during CMA week, so we knew going in that we would be relying on our new J rig from d&b audiotechnik as well as NoizCalc, which is the best noise remediation solution on the market," states Danny Rosenbalm, Morris CEO. "With the help of NoizCalc and Array Processing, we were successfully able to keep sound levels within Nashville city government limits to avoid any fines while also guaranteeing crowds a memorable and enjoyable concert experience."

Ascend Amphitheater maintains a 102 dB(a) limit at front of house. With NoizCalc, Morris achieved a 12 dB(a) to 14 dB(a) drop at the end of the listening field, allowing the sound to remain impactful to the crowd while also staying well within the regulated limits.

"It was an exciting project for us to partner with Morris on because this was the first time I used NoizCalc in a situation with so much concern and pressure around noise mitigation," adds Nick Malgieri, d&b audiotechnik application support. "NoizCalc accounts for elevation, buildings and other structures in the far field, which gave us a clear understanding of areas that needed to be closely watched and ultimately resulted in Morris being able to make the necessary changes to reduce noise emission."

ENHANCED INTELIGIBILITY IN THE BAY AREA

Oakland's Fox Theater, a 2,800-seat Moorish Deco movie palace whose reincarnation as a multi-use performance space now hosts a wide range of live entertainment, is now supported by new Meyer Sound LYON linear sound reinforcement loudspeakers, selected by promoter Another Planet Entertainment (APE),

The Fox Theater in Oakland outfitted with new Meyer Sound LYON arrays and other components.

based in northern California.

"Intelligibility is the main thing," says Tony Leong, production manager for APE. "With the LYON rig, we can do spoken word, heavy metal, or comedy, and the sound hits you right in your face no matter where you are. Whether you're by the back bar or in the last row of the balcony, it sounds as clear and present as it does in the front row of the orchestra. It also sounds as good at a low volume at it does at high volume."

The new system comprises 14 flown LYON line array loudspeakers and six groundstacked 1100-LFC low-frequency control elements per side. Three 700-HP subwoofers are flown above center stage, and front fill comprises four MSL-4 and three M'elodie line array loudspeakers. Galileo Callisto loudspeaker management provides system drive and alignment.

CARRYING A LIGHT LOAD WITH THE FRONTMEN

Production manager and front of house engineer Eugene "Geno" Mulcahy was carrying his mixing console in his backpack when he and country group The Frontmen of Country Music flew onto an aircraft carrier in the Persian Gulf on the band's tour earlier this year. Specifically, Mulcahy was packing a QSC TouchMix compact digital mixer and a selection of microphones to provide live sound and also archive every show during the tour, which visited U.S. Navy vessels and military installations in the Middle East and Europe.

Production manager/engineer front of house engineer Geno Mulcahy touring with his QSC TouchMix.

"Space is extremely limited on military aircraft, so I had the monumental task of having to travel lightly but still have powerful audio tools to support my clients," says Mulcahy, who has worked with The Frontmen since 2009. "I need the ability to archive every show and also produce a consistent sound from venue to venue and night after night. With space at a premium, carrying a regular console was out of the question. So I reached for my TouchMix, my microphone package and my backpack, and away I went."

Using TouchMix, Mulcahy states that he was able to achieve a consistent front of house sound throughout the tour, from Bahrain to Djibouti to Italy to Spain, in clubs, outdoor courtyards, and ship hangar bays, and using a variety of loudspeaker systems. "My sound check on day one was in the desert in Bahrain, hitting presets and renaming them. The presets are the biggest thing with this desk. People ask me, 'Why does the desk sound good?' I say, just use the presets; they really work," Mulcahy concludes.

NEXO GEO T arrays flying for Dillo Day at Northwestern University.

MEETING THE CHALLENGES OF A STUDENT-RUN FESTIVAL

Armadillo Day began in 1972 when Texans attending Northwestern University held a small celebration in honor of the official mammal of their home state. More than 40 years later, Dillo Day is the largest student-run music festival in the nation, hosted at the Northwestern Lakefill on the shore of Lake Michigan in Evanston, IL.

This year's entertainment for the approximately 10,000 attendees included a range of musical acts, with Sound Works Productions (Mokena, IL) providing NEXO line array systems for the two stages. The main stage system consisted of 32 GEO T boxes, 16 CD18s, and four PS10 for fills, all driven by NX 4x4 amplifiers, while the second stage system incorporated 12 GEO S12 enclosures, four RS18 Ray Subs, and two PS10s for fills, with power again supplied by NX 4x4 amps.

"Dillo Day is a challenge because we have national acts and an audience that demand a concert level SPL," states Matt Wilson, account manager at Sound Works. "But we need to keep the noise down in the town that is located very close to the event site, so with the GEO T and CD18s directionality, we were easily able to give the audience exactly what they wanted without creating noise complaints from the neighboring community." LSI

REMARKABLE JOURNEY

Doug Jones returns to his roots to spread the gospel of science. ***by Live Sound Staff***

Currently serving as the director of Danley (Sound Labs) University, Doug Jones grew up in northeastern Congo, the child of missionary parents, and his feelings about his childhood experience are complex: the benefits — difficult to articulate — came at considerable costs. "It's hard to know how to share my story with anyone who wasn't a missionary kid," he says. "And yet it is my story; it shaped who I am today."

Jones has long mused about the radical discord between his childhood in rural Congo, steeped in a Christian institution, and his life's work in professional audio, 30 years of it in the urban bustle of downtown Chicago at the secular institution of Columbia College. "When I went back to visit the village I grew up in, I didn't even have words to describe my professional field," he notes. "'How can you work with sound? That's absurd!' everyone said."

But that discord recently resolved to harmony with the invitation from childhood friend and fellow "missionary-kid" Dr. Paul Robinson to teach two weeks of intensive classes on technology and electrical engineering at the Bilingual Christian University of Congo. "It's precisely because I'm a missionary kid from the Congo that I had a concrete vision for how to help Congolese students spark their own creativity," explains Jones. "And I wouldn't have had the tools to do that without the years I spent at Columbia, honing my craft as a teacher in the fields of technology and audio."

MAKING THE LEAP

The Bilingual Christian University of Congo (more commonly abbreviated to UCBC in French, the official language of the former Belgian colony) was founded nearly a decade ago by Drs. David and Kaswera Kasali (both of whom obtained their PhDs in the U.S.). It's a liberal arts college founded on Christian principles. The U.S.-based Congo Initiative raises money to help fund the school, as the vast majority of students do not have the means to pay their full ride.

Six years ago, Robinson, a senior advisor to Congo Initiative — asked Jones to help UCBC install a radio station and to develop curriculum in radio, which Jones happily did. More recently, Robinson secured a grant to pay distinguished U.S. professors to teach intensive courses at UCBC. Given their common history and Jones' cultural and professional expertise, Robinson asked Jones to be the first participant in the fledgling program.

Danley Sound Labs president/CEO Mike Hedden endorsed the leave that would be required for Jones to develop two intensive courses and to deliver them at UCBC. Logistically, he would instruct the same group of 40 students for both classes, which would occupy eight hours a day combined (Too long!" laughs Jones, who diversified their activities to usefully fill their days).

The first class focused on the history of electromagnetism: the science, sure, but with most of the emphasis on the lives of the scientists who discovered and articulated the science, such as Georg Ohm, James Watt, Alessandro Volta, André-Marie Ampère, Michael Faraday, et al. "These were real people who lived real lives and faced real challenges and uncertainties," Jones states. "For example, Ampère grew up in France during the revolution, and when he was 16, his father was guillotined for being on the wrong side of the political divide — a revolutionary at a

Doug Jones in the Congo surrounded by his students.

moment when anti-revolutionary forces regained control. I thought that would resonate among the Congolese, who have a long, sad history of oppression, first by the Belgians, then by a string of nasty dictators, and now by indigenous war. They've experienced horrible things that most Westerners could scarcely wrap their brains around."

He continues, "I though that by sharing the stories of these scientists, which in most cases amounted to a lifetime of struggle with zero guarantee of success, the students' own struggles might be given a new perspective. If these scientists could succeed, then, by implication, so could these students."

Jones quickly discovered that the Congolese education system is based on rote memorization. In fact, they had all memorized the names of the scientists to whom Jones had hoped to introduce them.

"They had all memorized massive numbers of facts — way more than most Westerners have — but they were never encouraged to synthesize or explore the connections or implications of those facts," he explains. "We talked about Newton, and I emphasized that in addition to his creativity and mathematical genius, Newton's greatest asset was his curiosity. He asked, 'Why do things fall?'" Jones started out every day by writing in big letters on the chalkboard, "BE CURIOUS!"

CONNECTING THE DOTS

The second course explored electromagnetism in a hands-on lab. Jones brought battery packs, light bulbs, resistors, nails, paper clips, and so on that would allow students to connect their strictly-academic understanding of electromagnetism with real experience.

"It was less about teaching and more about letting the students explore," he adds. "These were applied engineering students in their sophomore and junior years who were on their way to becoming electro-mechanical engineers. They could recite Ohm's Law and could do as much math as I could, but they could not connect that information with reality. Many of them couldn't get the light bulb to light up. They knew what circuits were in an academic sense, but not in a real sense."

One challenge proved especially illuminating: he asked the students to use a nail and lengths of wire to empirically determine the relationship between the number of wire turns around the nail and the strength of the resulting electromagnet, as measured by how many paper clips it could lift.

Jones leading a class focused on science and a whole lot more.

"I asked them to present me with a graph of the results," he details. "Every one of them flipped the x- and y-axes, the independent and dependent variables. We all came to the realization that although they had all graphed functions in math class, it had never even occurred to them to graph data! That was important and eye-opening for everyone."

Toward the end of tenure at UCBC, Jones delivered a sermon during one of the chapel assemblies that summarized his grand hopes for the Congolese and that placed their challenges in a unique light. His four-point message emphasized the vital need for 1) a vision for the future of the nation, 2) a curiosity about the world, 3) self-confidence that the Congolese are a capable and worthy people, and 4) hope that life in Congo will continue to improve.

Although he relished the classroom experience, Jones concludes that he came away from the experience believing that future professorial visits should focus more on improving university-wide instruction with teacher in-services. LSI

DIGICO S21

The overview on a compact digital console platform.

by Mark Frink

The DiGiCo S21 compact digital console.

Every few years a new live mixing console moves the bar and becomes an industry standard, including the DiGiCo SD7, which debuted in 2007 when the company launched its Stealth Digital (SD) consoles that employ a single large-scale FPGA chip. Now the DiGiCo S21, released a year ago and shipping since November, is transforming the sub-$10K console market.

With any new technology, longer service life and introductory pricing offset the risk taken by early adopters. My friend George Relles bought America's first Heritage 3000 and benefitted from a decade of use that resulted in a per-show cost of $100. The sooner you get it, the longer it's in service. DiGiCo is currently breathing new life into its SD consoles with Core 2 upgrades that increase channel, mix bus and FX counts, plus providing dynamic EQ, multiband compressors and DiGiTubes on every mix path.

The S21 tested in this review was used this spring for a variety of front of house and monitor applications at the Jacksonville (Florida) Symphony's Jacoby Hall. The D Rack Touring Package provided ($13,300 list) consists of the S21 in a "nose cone" flight case, plus a 32 x 8 D-Rack and a 246-foot (75-meter) Cat-5 cable. MSRP for the S21 alone is $6,995. With 40 channels, the S21 isn't up to the scale of the SD9 regularly provided by symphony audio vendor and DiGiCo dealer Everyman Sound of Gainesville, FL, but its pristine 96 kHz sonic quality proved identical.

GENERAL OVERVIEW

Named for its 21 faders, the S21 is both compact (30 inches wide x 23 inches deep) and light in weight (42 pounds), immediately recalling other brands that have gotten progressively smaller and lighter. The compact, sturdy design really feels like a DiGiCo and its polycarbonate surface has a sleek, rugged finish.

Faders and buttons are borrowed from the SD line, providing a familiar touch to experienced DiGiCo operators, though breaking from SD's 12-fader bank tradition by using two 10-fader banks below a pair of smaller multi-touch screens. Underneath each screen is a row of 10 encoders, and to the right of the right-hand screen are six assignable master encoders, all of which have multi-colored LED rings.

The S21 borrows the SD7's powerful scheme of colored LED backlighting around each encoder to match selected onscreen graphics, indicating the knob's function and leveraging the use of color in the graphic user interface. Like the SD7, the S21 is un-illuminated or "Hidden Till Lit" (HTL) when not in use, helping the operator focus on active controls.

Many experienced SD operators focus on a single touch screen surrounded by a plethora of encoders and buttons while employing two or three fader banks, as found on the SD8, SD9 or SD10. Thanks to a workflow that relies more on its touch screens, the S21 leverages its two screens with fewer knobs and switches for faster operation after adjusting to its new graphics and encoder workflow.

The ability to easily customize channel layout for four layers, as well as its Spill Set that can overlay the right bank of any layer, allows each operator to find a comfortable mode of operation. The Spill Set can contain up to 10 of any combination of processing channels: inputs, outputs, control or matrix. It fills the right screen from the right side, and less than 10 members leave one or more of the left-most display channels in any right screen layer uncovered.

Some engineers might prefer S21's 40 input channels arranged in two layers of 20 channels, its default configuration, and then perhaps call up control groups by assigning them to the Spill Set, available or hidden at the press of a button. As a monitor engineer, I liked assigning all inputs to all four layers of the left bank, with each dedicated vocal effect return nested beside its associated input, and aux masters available on the right screen via the spill set. Your mileage may vary.

Opening channel processing windows covers the right screen's graphics, but channels on any layer of the left bank remain unhidden. Channel layout is easily edited on the Overview page by dragging and dropping, similar to the way apps are moved on an iPhone. The S21 is intuitive and straightforward.

NAVIGATION

To the right of the two fader banks are two square buttons: one to access the console's overview screen, where all four layers are displayed and another above it for quickly displaying or hiding channels assigned to the Spill Set on the right bank, regardless of chosen layer. Beside them are a pair of triangular buttons for moving up and down the S21's four layers.

Above these, "forward" and "back" triangular buttons navigate snapshots. The far right is simply master fader, headphone control, twin 24-segment multi-colored master/solo meters and both 1/4- and 1/8-inch high-power headphone jacks under the armrest.

SOFTWARE

The S21's new "flat" graphic user interface (GUI) is a departure for DiGiCo and reminds me of an iPhone. Seasoned DiGiCo users will discover they spend more time touching S21's screens, which takes getting used to; however, after a week I realized that many buttons and knobs on my SD consoles were relatively unused and I enjoyed the new S21 workflow more. I eventually found myself eager for the S21's new design aesthetic to slowly be incorporated into SD console revisions.

DiGiCo regulars are familiar with the company's tradition of semi-annual software revisions that gradually adds features across the entire SD range. Even in its third revision, the S21 still has a way to go, especially compared with the SD range's mature feature set. It's easy to imagine many of the SD's (often unique) features making their way to the S21 over time, but the company may be in no hurry to have it compete strongly with the SD range.

That said, any console is made of three elements: parts, algorithms and engineering. Clearly DiGiCo carefully selected some of the best parts – though fewer of them – found a more affordable FPGA chip, and has many of the same engineers and software processes employed on SD. The result is DiGiCo quality at a new price point.

FUNCTIONALITY

The S21 operates at 48 or 96 kHz without penalty, but SD9 users are familiar with D-Rack's use of 28 channels instead of 32 at the higher sampling rate. The console's rear is generously appointed with 24 local inputs and 12 outputs, allowing us to deploy it in Jacoby Hall's control booth immediately without the D-Rack, simply replacing the hall's ancient Crest console. When the D-Rack was added weeks later, everyone immediately noticed the improvement in sound quality provided by locating preamps on stage.

The S21 has a fairly fixed arrangement, though both its inputs and "group/auxes" are "flexi" channels that can be stereo or mono. In addition to the main stereo mix, it has 40 inputs as well as 16 mix buses that can be either variable auxiliary buses or fixed gain groups. Any or all can be mono or stereo, so far more than 40 inputs can be handled, depending on how many sources can be used as stereo.

Opening any channel's Setup View displays its signal path on the right-hand screen: its preamp source, high-pass/low-pass filters and delay, EQ and dual dynamics, and both pre- and post channel processing insert points. These processing blocks can be further opened and adjusted on the right screen.

The rear panel of the S21, with Dante expansion card highlighted.

Each channel and bus has 4-band parametric EQ plus high-pass and low-pass filters. For those not accustomed to low-pass, its benefit is a cornerstone of "big boy" mixing, even though many consoles force users to "steal" it from the parametric EQ high band. The S21's EQ also opens onto its right-hand screen, where encoders to its right are colored to match the selected band. The multi-touch screen allows users to use "pinch" and "spread" gestures to narrow or widen EQ filters intuitively.

Gate and compressor also open on the right screen. Dynamics 2 is a keyed, filtered gate that shows signal spectrum on the left to aid tuning its HPF and LPF. It can alternately be configured as a compressor with side-chain or as a ducker. Dynamics 1 can be selected on up to four channels as a multiband compressor that splits the familiar "bent knee" graphic interface in three, with signal spectrum shown across the crossover point display below the three knees in a powerful interface that makes it easy to get desired results.

There are also 16 freely assignable graphic EQs whose 32 filters don't all fit on 20 faders, but easily "slide" left or right with a swipe motion to move frequency control up or down on the faders where it's needed. Fun.

The S21 also includes four DigiTubes and eight effects. I'm not a fan of distortion, but some will enjoy DiGiCo's tube emulation on particular channels for effect. Each of the console's eight effects engines also open on the right screen from a familiar rack-like graphic interface and have a modest number of adjustable parameters and decent variety of algorithms, including tap delay.

I/O OPTIONS

In addition to the generous local 24 x 12 analog XLR I/O on the rear, the S21 has two DiGiCo Multichannel Interface (DMI) expansion card slots as used in the company's Orange Box for expanding SD desks. There are 10 choices of DMI cards: 64 x 64 Dante, co-axial MADI, Cat-5 MADI (used with D-Rack), DiGiCo Optocore, Aviom A-Net, 16-channel analog outputs, 16-channel analog mic/line inputs, 16-channel AES, Waves Soundgrid interface, and Calrec Hydra2 connectivity.

There's also a built-in UB-MADI, a Type-B USB connector that provides a built-in 48-channel, 48 kHz recording interface. Running at 96 kHz, the USB output down-samples to 48 kHz to not sacrifice recording channel count, and 48 kHz audio played back to the console for virtual sound check is up-sampled to 96 kHz, making multi-track recording and virtual sound check as easy as putting your favorite DAW on a laptop. It also makes the S21 an outstanding $7K recording interface, especially when considering the various DMI options for bridging to other consoles. It also makes a great desk for support acts touring with a DiGiCo (or Dante) ecosystem.

CONCLUSION

Having mixed on an SD10 with the Zac Brown Band on tour, an SD9 with the Jacksonville Symphony, and an SD7 with Dr. John on tour, I'm very aware of the "DiGiCo sound." At one pops show with the symphony we deployed an SD9 at FOH and the S21 on stage for monitors, and they really sound identical. Though the S21 lacks a few features found on the SD9, it sounds just as good. Over time I'm sure software updates will provide many features found on the more mature SD platforms.

The S21 carefully brings DiGiCo quality down-market in a package that combines flexibility and quick touch screen operation with outstanding sound in a powerful, compact and affordable system. Further, customer support is excellent, including the services of U.S. distributor Group One Ltd., headed by Jack Kelly.

U.S. MSRP: S21 – $6,995; S21 with 32 x 8 D-Rack – $11,995; D-Rack Touring Package (S21 with flight case, 75-meter Cat-5 and 32 x 8 D-Rack) – $13,300 LSI

***Mark Frink** (livesound@markfrink.com) is serving as production manager at the South Shore Music Circus in Cohasset, MA this summer and would like to thank James Pitts, Shamus McConney and Ray Klasse at the Jacksonville Symphony for their assistance and support with this review.*

WAVES AUDIO X-FDBK

Hands on with a new feedback elimination plugin. *by Phil Hagood*

The new Waves X-FDBK Feedback Eliminator plugin is a tool for quickly and easily ringing out feedback from loudspeaker systems. This plugin builds on technology of feedback eliminating hardware devices while also providing engineers with a helpful and powerful visual interface.

As I began my evaluation of X-FDBK, I went to the Waves website and checked out the tutorial video. These videos are important to build a basic understanding of the features in new plugins. In the live sound environment, having knowledge and experience with the gear can save valuable time and prevent embarrassing issues or mistakes that cost you a gig. After quickly viewing the video, the installation and authorization process took mere minutes in Waves Central.

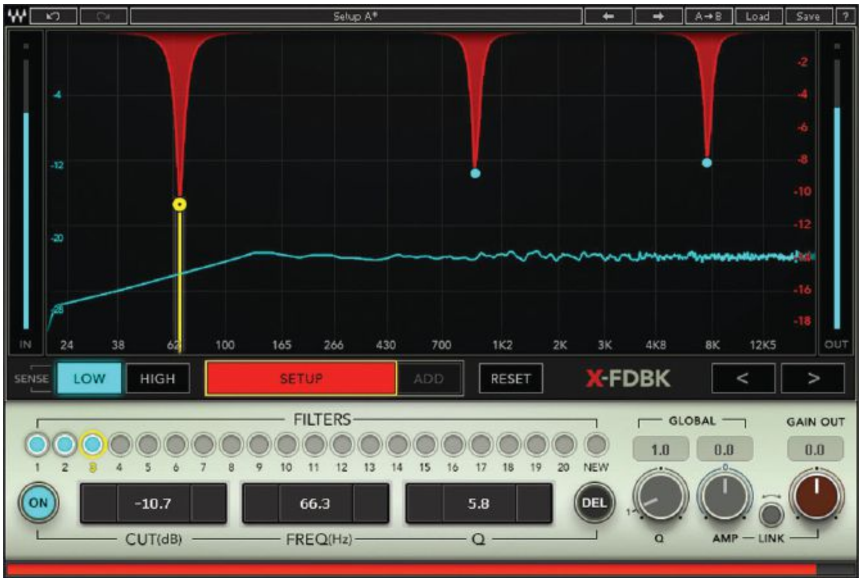
After purchasing or demoing the product, it immediately shows up in Waves Central and allows you to authorize the plugin on a USB drive or computer. When using plugins in a live scenario, having an easily transferred license can be crucial when dealing with venue-owned consoles.

When I first opened the plugin within my Waves Multirack rig, it loaded quickly and was clearly designed to be straightforward and clean. There are a couple basic controls to get started: sensitivity (SENSE) to feedback and the detection mode where it makes cuts to the signal (SETUP).

DIGGING IN & TESTING

To evaluate this plugin from a stage monitoring perspective, an SSL L300 Live console and a Waves MGO + Server One was set up to drive a d&b audiotechnik M4 stage monitor powered by a d&b D80 amplifier. I find the best way to evaluate any new tool or component in the monitor mixing world is a Shure SM58 microphone with a high-pass of 120 Hz and flat EQ. I connected the mic and pushed it into the monitor.

When X-FDBK cuts feedback, it does so at the exact frequency while also endeavoring to adopt the best possible Q and



The interface for the Waves X-FDBK feedback eliminator plugin.

amplitude to eliminate the feedback. The plugin also provides an extremely handy RTA feature so you can not only hear the feedback elimination, you can also see it.

As I pushed the mic harder into the monitor, it started to feed back at about 900 Hz. I pulled the level back down and activated the SETUP mode where the plugin begins to "listen" for feedback and pushed the mic back to the volume where feedback began.

Ordinarily as feedback starts I would cut that frequency and slowly bring it back into the monitor until it sounds natural. Using X-FDBK, I was able to push the volume in the monitor to the SPL I needed and then let the plugin balance between too much cut (which would hurt tonality) and not cutting enough to actually fix the feedback. Once I reached the needed SPL and made sure it was stable around the monitor, I took it out of SETUP mode and my EQ cuts were made.

After this process there were several things that I noticed. First, the SETUP mode has a countdown feature to prevent the plugin from cutting frequencies after the monitors have been rung out. This can be beneficial in a high-speed festival environment, where if you were to leave the SETUP mode active, a singer sustaining a note in the mic would result in a cut.

Second, the plugin takes approximately one second to react to ringing, so it will not fix the transient squeals and squeaks that can occur when a vocalist sticks the mic directly into the monitor. X-FDBK product manager Valery Gamarnik notes that he made it slower by design, adding "make it too fast and it reacts too drastically. So easy does it."

He also told me, "It's almost impossible for X-FDBK to differentiate during a show between guitar feedback and a vocal mic feedback during a show. So if you want to add an additional cut, use the ADD button, which activates the engine only while you're holding it, and adds one cut only without affecting the existing curve."

During the development process, Gamarnik explains that he compared X-FDBK to all current hardware units on the market for feedback elimination, and the plugin is faster and more precise in comparison. He cites average reaction time of two to three seconds for hardware units, in contrast to the previously noted X-FDBK reaction time of up to one second.

Finally, users can go back and add additional cuts either manually or automatically with the detection feature if needed. After the feedback was cut and the monitor was ready for my vocalist, I went back and listened to the musicality of the monitor. If I found that I cut too much, I used the controls to make all cuts shallower and effectively got more volume from the monitor. Or conversely, if I cut too little, I could pull all of the cuts down, as well as cut several more dB.

MAKING A DIFFERENCE

Technology in live sound continues to advance in making the job of engineers more efficient and effective. Waves X-FDBK does exactly that. This plugin can be invaluable to quickly and precisely cut feedback out of house and monitor systems. As with any new technology, there's no substitute for an engineer with experience and an ear that can hear and decipher frequencies. Knowing your vocalist, mics, loudspeakers and venue are required skills, but many times they are luxuries that busy monitor engineers do not have.

The bottom line is pretty clear: X-FDBK is a great tool that can be useful when there's the need to quickly fix feedback problems. There is no piece of gear that is a "fix-all," and X-FDBK is only as powerful as the engineer who drives it. That said, it can still be useful for novice engineers just learning to hear frequencies and fix feedback as well as for seasoned pros looking to get baseline cuts on a festival stage full of monitors.

Waves continues to develop innovative and original plugins that allow engineers to truly be able to focus on the music and creativity in mixing. X-FDBK is no exception. It definitively provides a way to quickly and successfully recalibrate systems for a better mix and a better show. LSI

***Phil Hagood** is director of operations for Morris Integration, based in Nashville.*

FLEXIBLE RESOURCES

Checking out the latest compact 2-way loudspeakers.

by Gary Parks

The venerable 2-way loudspeaker, combining a cone driver with a high-frequency device through a directivity-shaping horn/waveguide, has benefitted from advances in driver technology, signal processing, and amplification. Typically offered by many of the same companies that build line arrays and larger-scale systems, these loudspeakers often have a similar sonic signature to their "big brothers" and lend themselves to front fill and other supplementary duties in larger venues in addition to serving as mains in smaller applications and also as stage monitors.

As you'll see in the roundup of recent models that follows, most now include onboard amplifiers with sophisticated DSP, although many are also offered in passive versions. "Automatic" features might include signal delay to align the acoustic centers of each driver, phase correction, equalization, and driver protection. Some have selectable presets for different applications such as main or monitor placement, vocal or music sources, and high-pass filtering when used with subwoofers.

Additional EQ control may be present, including parametric EQ in at least one case. Others provide rudimentary mixing capability, with level controls, mic/line switches, and EQ on XLR or combo (XLR, 1/4-inch) inputs. We're also seeing more models that will accept Dante digital audio input, along with an analog audio signal, and it's likely this trend continues.

The transducer innovations have in many cases trickled down from touring products. Many have lightweight magnetic structures using neodymium magnets, rugged cone and spider materials, and robust voice coil designs. High-frequency horns are often rotatable, and with improved directivity.

Finally, cabinetry is roadworthy, using multi-ply wood or high-impact molded materials, durable coatings, and reinforced joints. Mounting options are built-in, with most having suspension points or plates on top, bottom, and sides, plus pole-mount hardware. For ease of comparison, in this roundup we're looking at (mostly) 12-inch models, with several options along with companion subs also noted. LSI

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

Adamson Point Series P-12
www.adamsonsystems.com

Dispersion (h x v): 60 x 40 or 40 x 20 degrees, both rotatable
HF: 3-inch diaphragm compression driver
Frequency Response: 60 Hz – 18 kHz (+/-3 dB)
Maximum SPL (peak): 134.4 dB
Power Handling: LF: 500 watts; HF: 110 watts (AES)
Connectors: Dual Neutrik Speakon
Positioning/Rigging: SLR sockets; pole-mount
Dimensions (h x w x d): 25.5 x 13.9 x 12.1 inches
Weight: 45.5 pounds
Notes: Active bi-amped and passive versions; Lake and XTA processing; 8- and 15-inch versions also available, plus single and dual 15-inch subs

Alcons Audio VR12
www.alconsaudio.com

Dispersion: 90 x 40 degrees, "revolvable"
HF: RBN601 pro-ribbon driver
Frequency Response: 60 Hz – 20 kHz (+/-3 dB)
Maximum SPL (peak): 132 dB
Power Handling: 1,500 watts (peak)
Connectors: Neutrik Speakon
Positioning/Rigging: Flytrack, mounting points, pole-mount, swivel bracket option
Dimensions (h x v x d): 26.5 x 13.8 x 13.6 inches
Weight: 39.6 pounds
Notes: Use with company's ALC amplifier/controllers recommended; 60-degree (h) version also available and it too is revolvable; 7-inch version offered as well

Cerwin-Vega P1500X
www.cerwinvega.com

Dispersion (h x v): 90 x 65 degrees
HF: 1.75-inch compression driver (15-inch LF)
Frequency Response: 50 Hz – 23 kHz (+/-10 dB)
Maximum SPL (peak): 134 dB
Amplification: Onboard 2-channel amp
Connectors: XLR/TRS inputs (ch 1-2), dual 1/4-inch TS (ch 3), XLR (thru and mix)
Positioning/Rigging: Suspension points, dual-angle pole-mount
Size (h x w x d): 27.5 x 17 x 13.5 inches
Weight: 53 pounds
Notes: 10-inch version also available plus 18-inch sub; built-in mixer with I/O connections provides EQ, Vega-Bass boost, and high-pass filters

NEXO PS Series
www.yamahaca.com

PS Series high-output loudspeakers provide more power, greater versatility and better value in both installed sound and touring applications. The asymmetrical horns used in all models are engineered so the vertical coverage is narrower above horn axis than below.

When used as the main PA, PS systems significantly reduce the amount of ambient, reverberant energy caused when loudspeakers misdirect their output towards walls and ceilings. And, when used in monitor mode, more of their output can be focused on the performer.

The 2-way passive design (8-ohm) uses a single amplifier channel for simpler installation and lower cost, while sophisticated control electronics (digital TD Controller or NXAMP) ensure reliable, linear operation.

TECHNOLOGY FOCUS: *A proprietary horn design better focuses coverage, with unique progressive horizontal (50- to 100-degree) and vertical (55-degree) dispersion providing selection of the most suitable pattern for vertical or horizontal PA usage, or wedge monitoring.*

OF NOTE: *Servo-controlled processing driven by amplifier output monitoring is a central element. VCAs and VCEQs instantaneously respond to voltage and current via algorithms that model electroacoustic complexities, including voice coil temperature and power compression. Series includes 8- and 15-inch versions and several sub options.*

KEY SPECIFICATIONS:

Model: PS10-R2
Dispersion (h x v): 50 to 100 x 55 degrees, rotatable
HF: 1-inch throat compression driver
Frequency Response: 65 Hz – 20 kHz (+/-3 dB)
Maximum SPL (peak): 132 dB
Power Handling: 500 watts to 1,250 watts recommended (8 ohms, 1 cabinet)
Connectors: Neutrik NL4 x 2 (input, thru)
Positioning/Rigging: Suspension points, pole-mount
Dimensions (h x w x d): 20.3 x 12.4 x 10.9 inches
Weight: 31 pounds

d&b audiotechnik E12
www.dbaudio.com

Dispersion: 80 x 50 or 110 x 50 (E12-D) degrees, both rotatable
HF: 3-inch voice coil compression driver
Frequency Response: 50 Hz – 18 kHz (-5 dB)
Maximum SPL (peak): 134 dB (w/several D Series amp models)
Power Handling: 1,600 watts (peak), D Series amplification recommended
Connectors: Dual NLT4, optional dual EP5 or NL4
Positioning/Rigging: Several flying/mounting options, pole-mount
Dimensions (h x v x d): 22.8 x 13.8 x 13.2 inches
Weight: 35 pounds
Notes: 15-inch version also available plus 12- and 15-inch subs; employs integrated coaxial driver; enclosure shape allows use either in upright or horizontal orientation

D.A.S. Audio Vantec 12A
www.dasaudio.com

Dispersion: 90 x 50 degrees
HF: 1-inch-exit compression driver
Frequency Response: 60 Hz – 20 kHz
Maximum SPL (peak): 135 dB
Amplification: Onboard class D amp, 1,500 watts (peak), biamplified (passive version also available)
Connectors: XLR (input and loop thru), 1/8-inch (aux in)
Positioning/Rigging: Suspension points, dual-angle pole-mount
Dimensions (h x w x d): 24 x 15 x 14.8 inches
Weight: 42 pounds
Notes: Single and dual 15-inch versions available plus 18-inch sub; onboard DSP includes FIR filters, several presets, and stackable 3-band EQUltra

dBTechnologies DVX D-12
www.dbtechnologies.com
www.americanmusicandsound.com

Dispersion (h x v): 60 x 40 degrees, rotatable
HF: 3-inch diaphragm compression driver
Frequency Response: 68 Hz – 19 kHz (+/-3 dB)
Maximum SPL (peak): 131 dB
Amplification: Onboard class D amp, 700 watts (RMS), biamped
Connectors: Neutrik Speakon (output and link), XLR (input and link), 1/4-inch TRS (link)
Positioning/Rigging: Suspension points; top/bottom flytracks; pole-mount
Dimensions (h x w x d): 24.6 x 14.6 x 15.6 inches
Weight: 60.6 pounds
Notes: 8-, 10- and 15-inch versions also available, plus three sizes of floor monitors with 12-, 15-, and dual 8-inch woofers.

RWG Spotlight Listing

FBT VENTIS Series
www.fbtusa.com

For the *new* VENTIS family of products, FBT has designed a series of products to meet the highest standards in acoustic performance and aesthetic appeal. Based on a powerful combination of B&C compression drivers with FBT's own custom, long-excursion woofers coupled with the company's renowned 15 mm birch plywood designs, VENTIS represents the pinnacle of audio performance.

Its one of the first products to utilize FBT's newest technologies in amp designs, with a 700-watt (RMS, full bridge) class D amp on the LF section and a 200-watt (RMS) meticulously crafted class H/AB amp design on the HF section. New and improved DSP allows users to choose between six factory presets or users can develop their own and save them into two user slots.

The top-quality cabinet features rubberized aluminum handles, several M10 points, a 45-degree sloped profile cut, and a full grille backed with treated acoustical cloth.

TECHNOLOGY FOCUS: *Located on the rear of each enclosure, a menu-driven DSP system is navigated via a single rotary control with push-to-select functionality. The two preset slots for users offer five parametric equalizers per slot. There's also a choice between two limiter modes (MAX-SPL or MAX-QUALITY) to best suit the specific needs of a performance.*

OF NOTE: *Also available is a high-pass filter, a mic/line selector, low, mid and high tone settings, and an optional delay of between 0 and 3.5 meters. Series also includes dual 6-inch as well as single 10- and 15-inch versions, plus several sub options.*

KEY SPECIFICATIONS:

Model: VENTIS 112A
Dispersion (h x v): 80 x 50 degrees, rotatable
HF: 1.4-inch voice coil compression driver
Frequency Response: 48 Hz – 20 kHz
Maximum SPL (peak): 133 dB
Amplification: Onboard 2-channel amp (class D LF, class H/AB HF) 1,400/400 watts peak
Connectors: 2-channel mixer offers XLR combo, stereo RCA
Positioning/Rigging: Suspension points, pole-mount
Dimensions (h x w x d): 26 x 12.2 x 15 inches
Weight: 23 pounds

EAW Redline RL-12
www.eaw.com

Dispersion (h x v): 90 x 60 degrees, rotatable
HF: 1.75-inch voice coil compression driver
Frequency Response: 55 Hz – 19 kHz
Maximum SPL (peak): 135 dB
Amplification: Onboard class D amp, 1,250 watts "maximum output," biamped
Connectors: XLR input
Positioning/Rigging: Suspension points; pole-mount
Dimensions (h x w x d): 22.7 x 14.6 x 17.3 inches
Weight: 42 pounds
Notes: 15-inch version and an 18-inch sub are also available; amplifier DSP includes three pre-defined voicing options

Electro-Voice ETX-12P
www.electrovoice.com

Dispersion (h x v): 90 x 60 degrees
HF: 1.25-inch titanium dome compression driver
Frequency Response: 55 Hz – 20 kHz (-3 dB)
Maximum SPL (peak): 135 dB
Amplification: Onboard class D amp, 2,000 watts (maximum), biamped
Connectors: Two XLR/TRS combo jacks plus one XLR link output
Positioning/Rigging: Suspension points; straight and tilted pole-mount
Dimensions (h x w x d): 24 x 15 x 16 inches
Weight: 52 pounds
Notes: 10- and 15-inch versions are also available, as well as a 15-inch 3-way model plus 15- and 18-inch subs; internal DSP with multiple optimization settings

Fulcrum Acoustic FA12ac
www.fulcrum-acoustic.com

Dispersion: 90 x 45 degrees, rotatable
HF: 3-inch titanium diaphragm compression driver (coaxial)
Frequency Response: 46 Hz – 20 kHz
Maximum SPL (peak): 127 dB
Amplification: Onboard class D, 2 x 1,050 watts (peak), biamplified
Connectors: XLR (input and output), AES3 (input), dual RJ45 (Ethernet)
Positioning/Rigging: Suspension points, pole-mount, several other options
Dimensions (h x v x d): 17.9 x 21.9 x 14.8 inches
Weight: 46.5 pounds
Notes: Dual 8- and 12-inch versions available, as well as single 15-inch plus several sub options; includes proprietary TQ processing; four selectable presets; input filters, delay and more can be accessed via Ethernet using Powersoft Armonía

Yamaha DXR Series
http://usa.yamaha.com

DXR active loudspeakers are the fruit of collaboration with sister company NEXO. Their detailed analysis of the transducers and cabinet designs allowed the engineering team to take output performance farther than previous designs.

By employing proprietary 48-bit signal processing we are able to produce accurate sound while carefully controlling the amplifiers to maximize SPL. FIR filtering provides remarkable phase accuracy, allowing better clarity and imaging. DXR loudspeakers offer settings for both FOH and monitor use. Not just EQ circuitry, this intelligent dynamic control adjusts as output increases.

Incorporating separate 24-bit A/D and D/A convertors allows higher signal-to-noise ratio and reduces self-noise. Comprehensive protection circuitry monitors all aspects of the power supply, amplifiers and transducers to ensure reliable operation.

TECHNOLOGY FOCUS: *All full range models offer proprietary FIR-X tuning utilizing linear phase FIR filters for the crossover network. FIR-X tuning simultaneously optimizes frequency and phase response while adjusting the time alignment between the HF and LF transducers. This creates a very smooth response around the crossover point, providing much better clarity and imaging than is possible with typical crossovers.*

OF NOTE: *DXR loudspeakers employ many of the same protection functions used in Yamaha's top-class TXn amplifiers. Series also includes 8-, 10- and 15-inch versions and sub options.*

KEY SPECIFICATIONS:

Type: DXR12
Dispersion (h x v): 90 x 60 degrees
HF: 1.4-inch diaphragm compression driver
Frequency Response: 52 Hz – 20 kHz (-10 dB)
Maximum SPL (peak): 133 dB
Amplification: Onboard 2-channel amp, 1,100 watts, biamplified
Connectors: Input: XLR; Input 2: Phone x 2; Input 3: RCA x 2; Thru: XLR (parallel with In 1); Link Out: XLR
Positioning/Rigging: Suspension points, dual-angle pole-mount
Dimensions (h x w x d): 23.7 x 14.2 x 13.8 inches
Weight: 42.5 pounds

Grund Audio GQ-12A
www.grundaudio.com

Dispersion: 90 x 50 degrees, rotatable
HF: 1-inch compression driver
Frequency Response: 48 Hz – 20 kHz (-10 dB)
Maximum SPL (peak): N/A
Amplification: Onboard amp rated at 400/800/1,600 watts (RMS/program/peak)
Connectors: XLR (input and output), 1/4-inch (line input), RCA (inputs)
Positioning/Rigging: Pole-mount
Dimensions (h x v x d): 24.5 x 15.2 x 15.4 inches
Weight: N/A
Notes: 10- and 15-inch versions, along with 3-way model, also available plus 18-inch sub; any two sources can be mixed internally

JBL Professional VTX F12
www.jblpro.com

Dispersion (h x v): 90 x 50 degrees
HF: 3-inch Dual Diaphragm, Dual Voice Coil compression driver
Frequency Response: 69 Hz – 20 kHz (+/-3 dB)
Maximum SPL (peak): LF: 132 dB; HF: 137 dB (free-field)
Power Handling: LF: 1,000 watts; HF: 200 watts (AES)
Connectors: Neutrik Speakon
Positioning/Rigging: Suspension points; pole-mount
Dimensions (h x w x d): 21.6 x 15.6 x 12 inches
Weight: 42 pounds
Notes: 15-inch version and an 18-inch sub are also available; Crown I-Tech amplification recommended, including model-specific presets; also integrates with JBL HiQnet Performance Manager

L-Acoustics X12
www.l-acoustics.com

Dispersion (h x v): 90 x 60 degrees
HF: 3-inch diaphragm compression driver (coaxial design)
Frequency Response: 59 Hz – 20 kHz
Maximum SPL (peak): 134 dB
Amplification: External LA4X or LA8 amplified controller to internal passive crossover
Connectors: Neutrik Speakon NL-4
Positioning/Rigging: Suspension points; dual pole-mount
Dimensions (h x w x d): 16.9 x 10.5 x 14.8 inches
Weight: 44.1 pounds
Notes: 8- and 15-inch versions (both coaxial as well) also available, plus 15- and 18-inch subs; variety of bracket and other mounting accessories

RWG Spotlight Listing

Clair Brothers kiT Series
www.clairbrothers.com

The kiT Series are the first self-powered loudspeakers in the Clair Brothers line with three full-range boxes and two subwoofers (kiTCURVE12+, kiT12+, kiT15+, kiT-Sub+, and kiT-Sub-mini+). The series features traditional Clair craftsmanship in wood enclosures and CNC machined wood waveguides, offering the same quality found in high-end, tour-grade products to users in smaller applications. The series delivers exceptional intelligibility, high-output performance and durability, making them perfect choices for virtually any indoor or outdoor application.

The kiTCURVE12+ has its roots in the Clair Brothers CAT series, the origin of Curved Array Technology waveguide. Tremendous versatility marks one of the benefits of the entire kiT Series line. Each loudspeaker comes equipped with multifunctional kiT Series rigging corners and features common dimensions allowing for virtually limitless rigging and stacking possibilities.

TECHNOLOGY FOCUS: *All kiT+ loudspeakers are powered by an integrated 2-channel, 1,600-watt each (3,200 watts total) power amplifier module that includes internal DSP loudspeaker processing: Xover (IIR), EQ (FIR) and limiter. The processing is factory programmed and provides the user with the choice between four presets. Networked DSP upgrade option available.*

OF NOTE: *All full-range kiT Series models sport wood horns/waveguides available in multiple dispersions. Standard pole-mount sockets offer multiple configurations. Series includes 15-inch version and sub options. Passive versions are also offered.*

KEY SPECIFICATIONS:

Model: kiTCURVE12+
Frequency Response: 51 Hz – 20 kHz (+/- 2 dB)
Dispersion (h x v): 90 x 15 degrees
HF: 2 x 1.4-inch voice coil compression drivers
Max SPL (peak): 141 dB
Amplification: Onboard 2-channel amp, 3,200 watts (maximum), biamped
Connectors: Dual XLR (input and thru)
Positioning/Rigging: kiT rigging corners, dual-angle pole-mount
Size (h x w x d): 14.8 x 28.1 x 15.2 inches
Weight: 66 pounds

Mackie SRM450
www.mackie.com

Dispersion: 90 x 45 degrees
HF: 1.4-inch titanium dome compression driver
Frequency Response: 47 Hz – 20 kHz (-3 dB)
Maximum SPL (peak): 128 dB
Amplification: Onboard class D amp, 1,000 watts peak, biamplified
Connectors: Proprietary Wide-Z inputs, stereo RCA inputs, XLR (thru)
Positioning/Rigging: Suspension points, pole-mount
Dimensions (h x v x d): 26.1 x 16 x 14.8 inches
Weight: 37 pounds
Notes: 10-inch version also available plus 15-inch sub; 1-buttom Speaker Mode selection; integrated 2-channel mixer; "Smart Protect" DSP; patented Acoustic Correction

Martin Audio CDD-LIVE! 12
www.martin-audio.com

Dispersion (h x v): 60 x 60 degrees mid-field, 110 degrees horizontal near-field
Components: 1.7-inch polyimide dome compression driver (coaxial design)
Frequency Response: 62 Hz – 20 kHz (+/-3 dB)
Maximum SPL (peak): 128 dB
Amplification: Onboard class D amp, 1,250 watts (continuous), biamped
Connectors: XLR input and output; dual Dante-enabled Neutrik Ethercon
Positioning/Rigging: Suspension points; pole-mount
Dimensions (h x w x d): 22.8 x 14.1 x 14.7 inches
Weight: 61.6 pounds
Notes: 8- and 15-inch versions available (both coaxial as well), plus single and dual 18-inch subs; internal amplification with a variety of presets; Dante enabled

Meyer Sound UPA-1P
www.meyersound.com

Dispersion (h x v): 100 x 40 degrees
HF: 3-inch diaphragm compression driver
Frequency Response: 80 Hz – 17 kHz (+/-4 dB)
Maximum SPL (peak): 133 dB
Amplification: Onboard 2-channel MOSFET amp, 550 watts, biamped
Connectors: XLR input/output
Positioning/Rigging: Ring and stud pan fittings; pole-mount adapters & yoke-mounts
Dimensions (h x w x d): 22.4 x 14.5 x 14.3 inches
Weight: 77 pounds
Notes: Internal DSP includes EQ, phase correction, and protection; optional 3/8-inch or M10 nut plates available

RWG Spotlight Listing

KV2 Audio ESD15

www.kv2audio.com

The ESD 15 is a three-way, full-range passive top loudspeaker that incorporates a unique coaxial 15-inch transducer with a 1.75-inch nitride titanium neodymium compression driver on a wide dispersion 80- x 60-degree horn.

While most coaxial loudspeakers will try to cover the full-frequency response, the ESD15's coaxial driver only covers bass and highs, leaving the all important mid-range to a 6-inch neodymium cone driver mounted on a large 80- x 60-degree horn. This resolves the issue of the 15-inch driver trying and failing to reproduce mid-range, a common problem with conventional coaxial designs.

The ESD15's unique design provides extremely flat, full range reproduction from a compact yet powerful package that out performs models twice its size, weight and price.

TECHNOLOGY FOCUS: *KV2's entire ESD range is designed for use with its ESP amplifiers that greatly improve impulse response time, increasing resolution and reducing distortion. However, third-party amplifiers are also supported via NL4 connectors. RMS and peak (HF) limiting are built in.*

OF NOTE: *KV2 is releasing the EX15 this fall, an active cabinet with the same design as the ESD15. Also currently available are ESD loudspeakers in 5-, 6-, 10- and 12-inch versions plus several sub options.*

KEY SPECIFICATIONS:

Model: ESD15
Frequency Response: 48 Hz – 18 kHz (- 3 dB)
Dispersion (h x v): 80 x 60 degrees
HF: 1.75-inch diaphragm compression driver
Max SPL (peak): 129 dB
Power Handling: 500 watts (4 ohms), ESP amplifiers recommended
Connectors: Dual NL4, terminal blocks
Positioning/Rigging: Suspension points, pole-mount
Size (h x w x d): 27.5 x 17.7 x 17.7 inches
Weight: 77.2 pounds

PreSonus ULT12

www.presonus.com

Dispersion (h x v): 110 x 50 degrees (rotatable)
HF: 1.75-inch voice coil compression driver
Frequency Response: 55 Hz – 18 kHz (+/-3 dB)
Maximum SPL (peak): 135 dB
Amplification: Onboard class D amplifier, 1,300 watts (peak), biamped
Connectors: XLR outputs, combo jack inputs
Positioning/Rigging: Suspension points, dual-position pole-mount
Dimensions (h x w x d): 25.2 x 14 x 13.9 inches
Weight: 52 pounds
Notes: Mic/line input includes company's XMAX mic preamp and a line-level-only input; inputs have independent level control; 15-inch version also available plus 18-inch sub

QSC E12

www.qsc.com

Dispersion (h x v): 85 degrees conical
HF: 1.75-inch voice coil compression driver
Frequency Response: 64 Hz – 20 kHz (-6 dB)
Maximum SPL (peak): 128 dB
Power Handling: 400 watts continuous, passive crossover
Connectors: Neutrik Speakon, barrier strip
Positioning/Rigging: Suspension points, dual-angle pole-mount
Dimensions (h x w x d): 24.2 x 14.2 x 14.6 inches
Weight: 51 pounds
Notes: 10- and 15-inch versions also available, plus 18-inch sub; QSC PLD or GXD amplifiers or TouchMix digital mixer recommended for proprietary DSP optimization

Ramsdell Pro Audio 12-X-SS

ramsdellproaudio.com

Dispersion: 60 x 40 degrees
HF: 2-inch compression driver
Frequency Response: 70 Hz – 20 kHz (+/-3 dB)
Maximum SPL (peak): 126 dB
Power Handling: 800 watts program at 8 ohms, biamp or passive modes
Connectors: Dual XLR (input, paralleled), barrier strip (install version)
Positioning/Rigging: Several suspension options, pole-mount
Dimensions (h x v x d): 28 x 14 x 15 inches
Weight: 80 pounds
Notes: 15-inch version also available plus several sub options; replaceable baskets provide fast, easy voice coil replacement

RCF TT2-A
www.rcf-usa.com

Dispersion (h x v): 90 x 50 degrees
HF: 3-inch voice coil compression driver
Frequency Response: 50 Hz – 20 kHz
Maximum SPL (peak): 134 dB
Amplification: Onboard class D 2-channel amp, 1,600 watts (total), biamped
Connectors: XLR in/out, RDNet in/out
Positioning/Rigging: Rigging plates top and bottom, suspension points, pole-mount
Dimensions (h x w x d): 23.5 x 14.6 x 18.5 inches
Weight: 75 pounds
Notes: 10- and 15-inch versions available; rear-panel control of functions including gain reduction, delay, and presets; also controllable remotely via RDNet

Renkus-Heinz PN121
www.renkus-heinz.com

Dispersion (h x v): 60 x 40 or 90 x 40 degrees
HF: 1-inch compression driver on Complex Conic horn
Frequency Response: 55 Hz – 18 kHz
Maximum SPL (peak): 129 dB
Amplification: Onboard 300-watt (total, RMS) amp
Connectors: Neutrik 4-pin Speakon style, screw terminals
Positioning/Rigging: Suspension points, U-bracket, pole-mount, Aeroquip flytrack
Dimensions (h x w x d): 26.5 x 15.5 x 13.7 inches
Weight: 67 pounds
Notes: 6-, 8-, dual 8- and 15-inch versions offered, plus single and dual 12-inch subs; also available in non-powered version; network-controllable amp modules offered

Tannoy VXP 12
www.tannoy.com

Dispersion: 90 degrees conical
HF: Proprietary Dual Concentric drive unit
Frequency Response: 70 Hz – 25 kHz (-3 dB)
Maximum SPL (peak): 127 dB
Amplification: Onboard Lab.gruppen IDEEA class D power module, single channel
Connectors: XLR (input and output)
Positioning/Rigging: Suspension points, yoke bracket inserts, pole-mount option
Dimensions (h x w x d): 19.3 x 14.6 x 15 inches
Weight: 37 pounds
Notes: Single/dual 8-inch as well as 15-inch versions available, plus several subs; options for more specific vertical directivity control offered as well; switchable 90 Hz high-pass filter tailors response for use with a sub

Turbosound Milan M12
www.turbosound.com

Dispersion (h x v): 90 x 60 degrees
HF: 1-inch compression driver on elliptical waveguide
Frequency Response: 50 Hz – 18 kHz (+/-3 dB)
Maximum SPL (peak): 128 dB
Amplification: Onboard class D amp, 1,100 watts (total), biamped
Connectors: Combo jacks for input/output
Positioning/Rigging: Suspension points, dual-angle pole-mount
Dimensions (h x w x d): 24.4 x 15.5 x 13 inches
Weight: 45.1 pounds
Notes: Two independent input channels with mic/line switch and integrated two-channel mixer; switchable high-pass filter; 10- and 15-versions also available, plus 18-inch sub

VUE audiotechnik h-12
www.vueaudio.com

Dispersion: 100 x 50 (W version) or 60 x 40 (N version) degrees
HF: 4-inch voice coil (Truextent beryllium diaphragm) compression driver
Frequency Response: 55 Hz – 21 kHz (-3 dB)
Maximum SPL (peak): 135 dB
Amplification: Onboard class D 2-channel amp, 2,400 watts peak, biamplified
Connectors: XLR (input and loop output) for both analog input and AES EBU digital input
Positioning/Rigging: Suspension points, pole-mount
Dimensions (h x v x d): 24 x 14.5 x 15.2 inches
Weight: 78 pounds
Notes: Dual 5-inch as well as single 8-, and 15-inch versions also available, plus several sub options; integrated DSP with System-VUE network control and monitoring

WorxAudio WaveSeries 12A
www.worxaudio.com

Dispersion: 75 x 75 degrees
HF: 1.75-inch voice coil compression driver
Frequency Response: 54 Hz – 17 kHz (-3 dB)
Maximum SPL (peak): 120 dB (biamped)
Power Handling: LF – 200 watts (800 watts peak); HF – 60 watts (180 watts peak)
Connectors: Dual parallel-wired Neutrik Speakon
Positioning/Rigging: Portable version has pole-mount, install version equipped with flyware
Dimensions (h x v x d): 23.5 x 18.2 x 17.5 inches
Weight: 54 pounds
Notes: Dual 6-inch version as well as 8- and 15-inch versions available plus several subs; passive network provides EQ points for a flat frequency response

PHASE SHIFTS

The harmonics lead the fundamental.

by Merlijn van Veen

"It turns out that, within very generous tolerances, humans are insensitive to phase shifts. Under carefully contrived circumstances, special signals auditioned in anechoic conditions, or through headphones, people have heard slight differences. However, even these limited results have failed to provide clear evidence of a "preference" for a lack of phase shift. When auditioned in real rooms, these differences disappear..." ***—Dr. Floyd Toole***

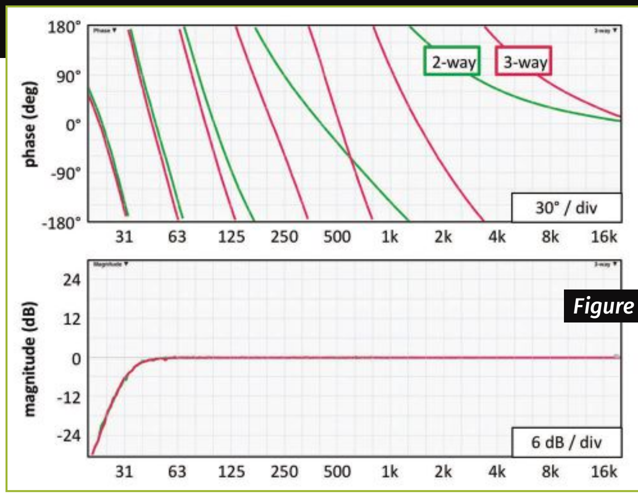


Figure 1

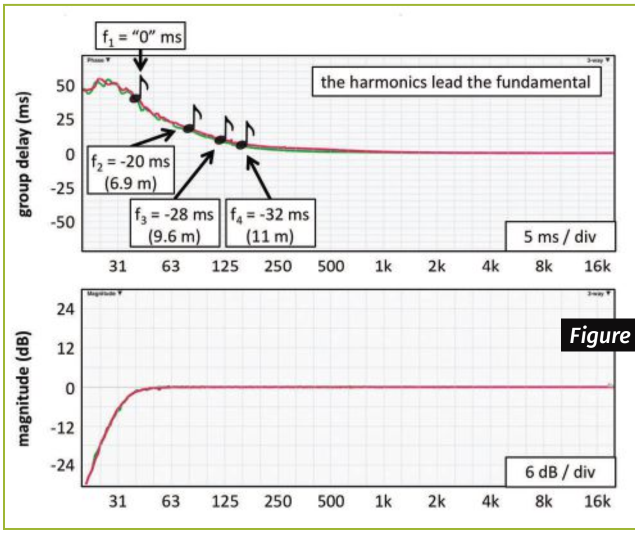


Figure 2

As Dr. Toole states, a lot of people, including myself, have a hard time detecting phase shift (not to be mistaken with phase offset). In experimenting with this issue and documenting it on a video that can be viewed on my website as well as ProSoundWeb, I applied gratuitous amounts of phase shift to a music track by means of all-pass filters.

After demonstrating that the plugins were actually doing something, it was difficult to hear any differences but for the occasional pop and dropout of my computer trying to cope. At the end of the experiment, when I switched over from phase to group delay, as much as 15 milliseconds (ms) of frequency-dependent "time smearing" was introduced. And that's on top of the inherent phase shift of the loudspeakers or headphones used to listen to the sound of the video.

A couple of people I hold in high regard, Mauricio "Magu" Ramírez of Meyer Sound and François "Frankie" Desjardins of Solotech (among others), have heard actual loudspeakers with linear "flat" phase behavior from virtually DC to light. Doing blind A/B testing with their favorite tracks, they initially detected no differences between linear phase and minimum phase (typical behavior for real-world loudspeakers) responses. But when a track contained substantial low-frequency information (e.g., bass guitar, drums, or deep vocals), they immediately noticed a difference that can be best described, from what I came to understand, as presence. Where these instruments or voices previously sounded ambiguous, now they suddenly were up close and personal, with focus and impact.

This raised the question as to what causes this sensation. The only explanation I can think of, and it probably isn't novel, is that in a typical sound system, for low-frequency sources, the harmonics lead the fundamental. The fundamental allows us to detect pitch and the harmonics (multiples of the fundamental frequency) enable us to distinguish violin from clarinet or Willie Nelson from Leonard Cohen. Without harmonics, all sources would be producing sine waves exclusively and thus be indistinguishable from each other.

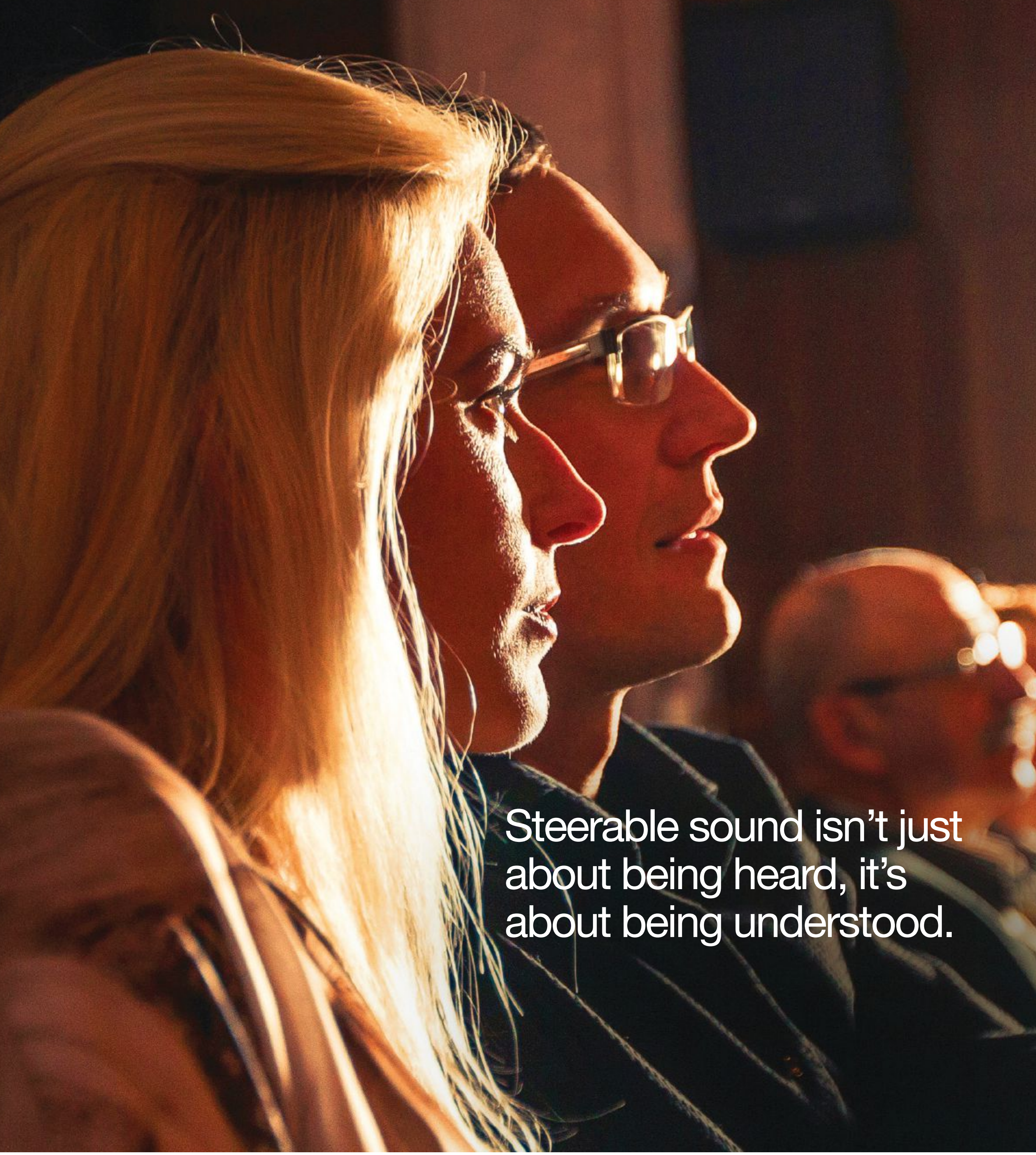
Figure 1 shows the typical phase shift for a regular (no FIR or all-pass filters) 2-way or 3-way loudspeaker, complemented with a non-overlapping, phase-aligned (ported) subwoofer. All crossovers consist of 4th-order Linkwitz-Riley filters at unity gain. **Figure 2** tells the same story but now instead of phase, group delay is shown.

In the case of the lowest open E-string on a typical bass guitar with a fundamental frequency of 40 Hz, the 2nd, 3rd and 4th harmonics lead by 20, 28 and 32 ms, respectively. That's 7 to 11 meters! If we nullify this phase shift, I would expect to hear a difference. Having the harmonics delivered simultaneously with the fundamental instead of in series produces steeper, more transient edges in the waveform that tightens and cleans up the sound.

Flattening the phase can be done, e.g., with FIR filters, in exchange for latency. The duration of the Finite Impulse Response filter needs to be at least twice the period duration of the lowest frequency it's describing. That's 100 ms at 20 Hz! Unacceptable for live sound reinforcement but not for playback situations.

I, for one, am hoping to experience it myself one day. LSI

Based in The Netherlands, ***Merlijn van Veen*** *(https://www.merlijnvanveen.nl) is a consultant specializing in sound system design and optimization, and he's also a noted audio educator.*


Steerable sound isn't just about being heard, it's about being understood.