THE JOURNAL FOR LIVE EVENT TECHNOLOGY PROFESSIONALS
JANUARY 2017 | PROSOUNDWEB.COM

LIVE SOUND INTERNATIONAL

INSIDE

SEARCHING FOR SIMPLICITY

Inside the concert sound approach for Switchfoot on tour.

A FRESH LOOK AT DRUM MICROPHONE TECHNIQUES

COMPUTER WORLD: FIELD APPLICATIONS & SOFTWARE

STEP-BY-STEP IN TRACKING DOWN RF PROBLEMS

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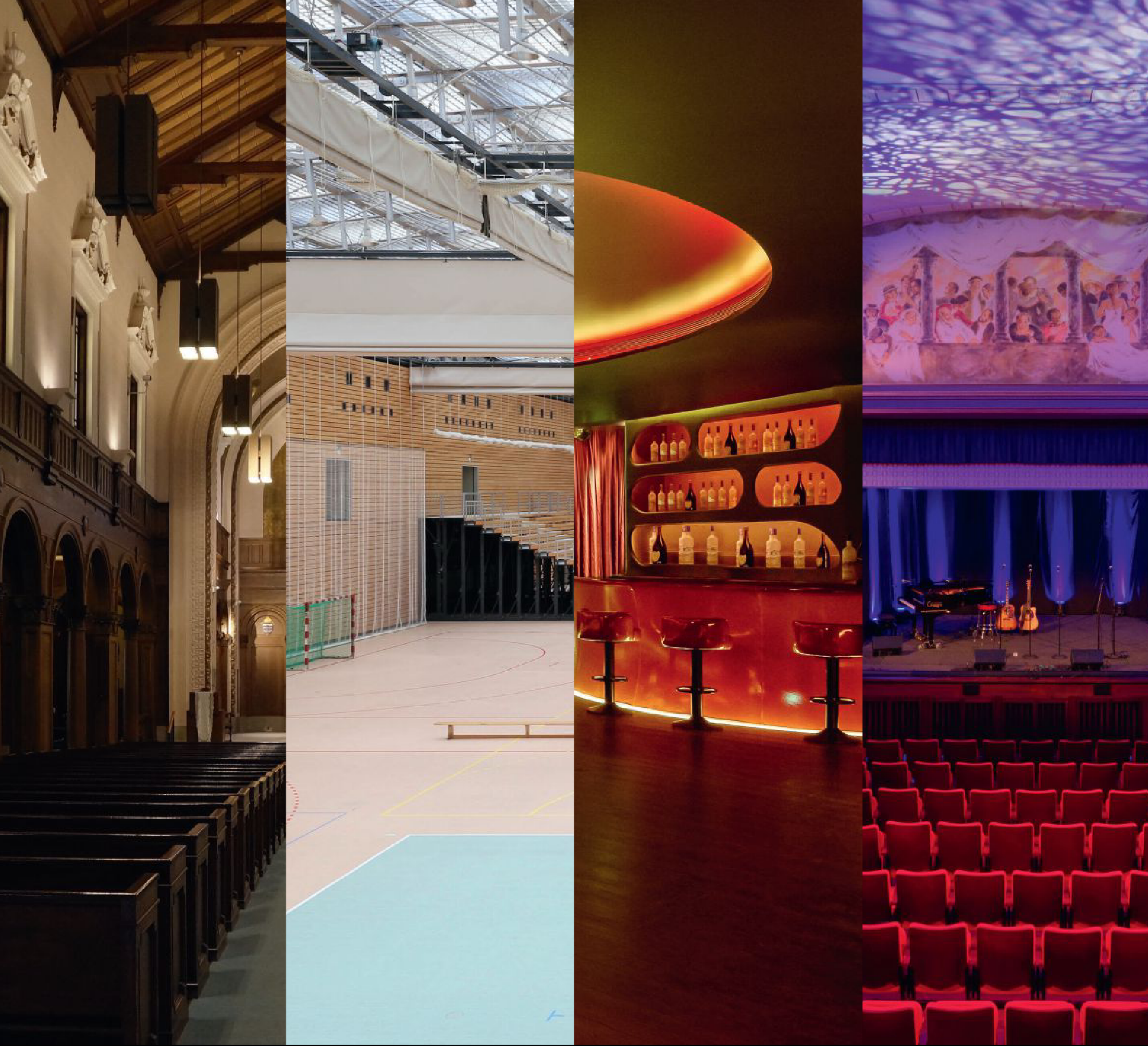

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CREDIT: STEVE JENNINGS

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From the Editor's Desk

WE BEGIN THE NEW YEAR with numerous contributors all bringing their "A game" to this issue in addressing a diverse group of topics, ranging from the straightforward to the more challenging.

For example, Ike Zimbel presents procedures for tracking down common wireless system problems in what's sure to be a reference article. His advice, cultivated from decades in the field, is clear-cut and largely uncomplicated while succinctly addressing key issues and practices faced by so many audio professionals working in what's currently a challenging RF landscape.

On the other hand, Jonah Altrove's take on drum miking moves outside the "typical" realm of discussion. It's a different consideration of the topic, one that may prove to be thought-provoking to many readers.

We're thrilled to have Bob McCarthy return with part four of his series on system optimization. He continues to fill in the key pieces of the puzzle making up the picture of where we are today, and doing so in entertaining fashion.

Meanwhile, Andy Coules focuses on another key component for optimization: our ears. He offers an interesting look at how these devices actually work, along with providing some relatively simple yet absolutely necessary aspects in maintaining hearing health and wellness.

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

Keith Clark
Editor In Chief, Live Sound International/ProSoundWeb
kclark@livesoundint.com

ON THE COVER: Switchfoot frontman Jon Foreman goes airborne (with Shure mic in hand) on the Looking for America Tour. *(Photo by Steve Jennings)*

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Radio Active Designs TX-8/900M

A wireless antenna combiner that can be used with any UHF in-ear monitoring (IEM) or IFB system operating in the 900 MHz band, specifically 941 to 960 MHz. The unit allows the connection of up to eight UHF transmitters to one antenna. An active-linear design is intended to reduce intermodulation products to low levels to minimize interference while providing compatibility with all forms of modulation, including FM, AM, SSB, ENB, as well as a variety of digital forms, including FSK and MIMO. Individual channel indicators on the front panel show RF activity and amplifier faults. Output power is selectable to 50, 100 or 250 mW, independent of input power level. ***radioactiverf.com***

MIPRO MTG-100 & MTS-100

The MTG-100 is a digital wireless tour guide system and the MTS-100 is a digital wireless language interpretation system, both operating in the 900 MHz band. They offer a choice of 16 preset channels, facilitating up to four simultaneous tours or the ability to communicate in four different languages on the same tour. The systems offer switchable RF output, with settings for low, medium, and high, with an operating range specified as 330 feet (line of sight). Options include both stationary and portable digital receivers and transmitters. Microphone choices are gooseneck, headset, and lavalier in addition to the availability of an adapter that enables the use of condenser mics. Monitoring choices include both stereo and single earphones as well as stereo earbuds. Components can be powered by Lithium rechargeable and AA batteries. Optional charging bays are available in 4-slot configurations that can be mounted in a charging rack. ***avlex.com, mipro.com.tw***

Eastern Acoustic Works Radius RSX12M

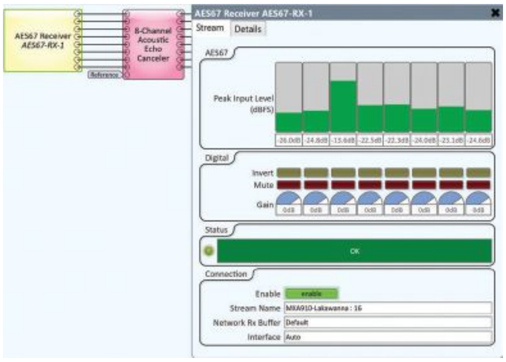
A stage monitor joining the company's Radius line that's equipped with onboard bi-amplified electronics (500 watts per channel), proprietary technologies such as Focusing and DynO, and Dante networking (with loop-thru). It's loaded with a 12-inch vented cone driver (2.5-inch voice coil), joined by a 1-inch exit (1.77-inch voice coil) compression driver oriented in a coaxial configuration. Dispersion is listed as 105 by 105 degrees, with latency stated as 2.6 milliseconds. The free (available via iTunes) EAWMosaic app provides system prediction, control and monitoring of all Radius models, including the RSX12M. Four pre-defined voicings are also included. ***eaw.com***

Behringer KM1700 & KM750

Dual-channel power amplifiers equipped with the company's proprietary ATR (Accelerated Transient Response) technology that's designed to improve punch and clarity while reducing overall weight. Stated power ratings for the KM1700, which weighs 27.5 pounds, are 800 watts/channel at 4 ohms, 500 watts/channel at 8 ohms, and 1,700 watts bridged. For the KM750, which weighs 18.7 pounds, the spec is 400 watts/channel at 4 ohms, 200 watts/channel at 8 ohms, and 750 watts bridged. ***behringer.com***

Radial Engineering 4-Play

A direct (DI) box to handle multiple instruments on stage, outfitted with a standard 1/4-inch input for acoustic guitar, mandolin, fiddle, bass and other instruments. A mute footswitch can be engaged when connecting or disconnecting an instrument. The signal is then routed to a selector footswitch that sequentially activates up to four outputs, allowing each instrument to have its own dedicated channel. LED indicators provide clear visual feedback as to which output is active. 4-Play also includes four balanced XLR outputs, each equipped with a ground lift switch. All connectors are made from glass-filled nylon with nickel-silver contacts. Two "set & forget" switches allow the user to increase the range of the selector footswitch to 2, 3 or 4 outputs. There is also a dedicated tuner out that can be assigned to be always on or function in tandem with the mute switch. ***radialeng.com***



QSC Q-SYS Support Of AES67

An addition to the Q-SYS platform that's part of the upcoming Designer v5.3 release with software-based audio/video/control. AES67 allows performance audio streaming between Q-SYS and third-party products supporting different native networked audio technology such as Dante, RAVENNA, and Livewire without requiring any additional hardware or license costs. It's a standard for audio-over-IP interoperability and was published by the Audio Engineering Society (AES) in 2013, and more specifically, it's a Layer-3 protocol suite designed to allow audio interoperability between any networked audio solution based on Layer-3 technology. ***qsc.com***

Linea Research C Series

A range of power amplifiers comprising three 4-channel and three 8-channel models, all with DSP control and an optional Dante interface in a 2RU chassis with a tamperproof front panel. The 4-channel units are available with power ratings of 5,000, 2,500 and 1,500 watts per channel. The 8-channel units are available with 1,250, 750 and 400 watts per channel (all ratings at 2 ohms). They're specified as capable of delivering full rated power into 2, 4 or 8 ohm loads, or into 70/100v systems, selectable on a channel-by-channel basis. All can operate on any supply voltage from 100 to 240 volts. Setup is achieved with the company's System Engineer software. Additional third-party control can be done via the built-in contact closure, fault relay, RS232, RS485, and TCP/IP interfaces. ***linea-research.com, alliedsmd.com***

Lectrosonics SRAES3

A bottom plate adapter for all SR Series dual-channel slot receivers, including the SRc, SRb, SRa and SR, and the 5P variants of these units, providing audio in the following formats: AES3 digital audio at 48 kHz sampling with an internal clock; AES3 digital audio with an external word clock at 44.1, 48 or 96 kHz sampling; and balanced analog audio from -50 dBu to +5 dBu in 1 dB steps. One or two channels of digital audio are delivered from the CH1 jack, while the CH2 jack delivers analog audio at all times. External word clock can be fed to the SRAES3 via the BNC connector. A 75-ohm termination of the word clock connection is applied by a field changeable jumper. ***lectrosonics.com***

Waves Audio Primary Source Expander (PSE)

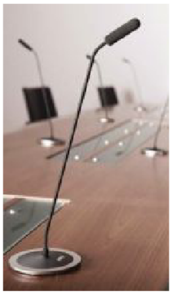
A plugin designed to help reduce stage bleed and sensitivity to feedback when a microphone is idle by automatically lowering mic levels between musical phrases. It includes a precision expander tailored for melodic sources such as vocals, guitars, strings, brass, woodwinds and more. PSE works like a fader that attenuates the channel's level when the source goes below the threshold that's been set. The user determines both the threshold and the amount of attenuation. The plugin also provides side-chain and ducking controls to enhance consistency. ***waves.com***

Shure ULX-D & QLX-D VHF

ULX-D digital wireless systems and related accessories are now available in the VHF frequency band, with QLX-D digital wireless systems following in the near future. The VHF options both provide 42 MHz of tuning bandwidth. ULX-D systems offer networked control, Dante capability (on ULXD4D and ULXD4Q dual and quad receivers), and AES-256 encryption, while QLX-D systems have 24-bit digital audio. VHF-ready accessories include the UA844+V antenna distribution system, UA834V in-line amplifier, UA874V active directional antenna, and UA860V passive ground-referenced omnidirectional antenna. The UA845UWB antenna distribution system is switchable for UHF/VHF. ***shure.com***

PK Sound Gravity 30

A subwoofer incorporating a 15,000-watt class D amplifier with differential pressure loop technology driving a 30-inch moving magnet linear motor M-Force transducer. The dual-reflex bandpass enclosure provides high, symmetric loading to the cone, designed to increase efficiency, improve transient response, reduce distortion and maximize LF extension. A steel alloy motor mount provides a rigid connecting assembly that acts as a large heatsink for the transducer and as a grill for the HF chamber. The Gravity 30 is cardioid arrayable and available in both touring and install versions. Both integrate with PK's Kontrol software, providing access to key DSP features and real-time monitoring. The touring model includes an integrated rigging assembly that allows up to 12 cabinets to be suspended. ***pksound.ca***



DPA d:screet Microphone Base

A stand for the company's d:screet SC4098 podium microphone that's available in several versions. Designed to be placed on a table or podium, or attached to a ceiling or wall, the stand is available in black and white, and comes with either a MicroDot connector, an XLR connector, or un-terminated leads for connections to Phoenix blocks. The d:screet capsule is mounted on a boom that provides flexible positioning. ***dpamicrophones.com***

Symetrix Prism 0x0

The latest in the company's series of Dante-enabled DSPs offering expansion via 64 channels of bidirectional networking. Used as the DSP core of a Dante network, or as a DSP co-processor, Prism 0x0 is suited for applications requiring advanced signal processing coupled with a network audio interface. It has no analog inputs or outputs and no external control inputs or logic outputs. It's housed in a half-rack x 1U rack-mount chassis. Mounting options are offered as well as power over Ethernet (PoE+). ***www.symetrix.com***

Lawo mc² v5.8 & Nova73

The latest software upgrade for the company's mc² Series consoles includes the new Module ISO, which enables the prevention of certain settings from being loaded from snapshots, and it also provides RAVENNA AoIP integration for Neumann DMI-8 digital microphone interfaces. In addition, a new 981/65 interface card for the Nova73 processing and routing core supplies native Dante support for mc² consoles. It offers four redundant Dante ports with 64 channels each, resulting in a total capacity of 256 audio channels per card. ***lawo.com***

Biamp Systems AMP-A460H

A 4-channel power amplifier that can be customized using the company's TesiraFORTÉ processing platform. A balanced analog line in fosters simple integration with any of the eight TesiraFORTÉ models. Each channel is rated to deliver 60 watts per channel at 8 ohms, and bridged in pairs, are specified to provide 120 watts at 8 ohms or with constant voltage (70-/100-volt) direct drive. The half-rack design includes a kit for mounting into any standard rack. ***www.biamp.com***

Galaxy Audio EDX

A frequency-agile, dual-channel wireless microphone system with a stated range of 200 feet. Each receiver includes 16 selectable UHF channels and two internal antennas. There's also IR Sync from transmitters to the receiver. Receivers have a 1/4-inch output and two discrete XLRM outputs on the back. Output is controlled by a volume control for each receiver, located on the front of the unit, along with AF/RF and channel indicators. Transmitter options include the HH38 handheld microphone and the MBP38 bodypack with choice of headset or lavalier mic. ***galaxyaudio.com*** LSI

Spotlight

A USER'S GUIDE

The design and care of an audio professional's most valuable tools.

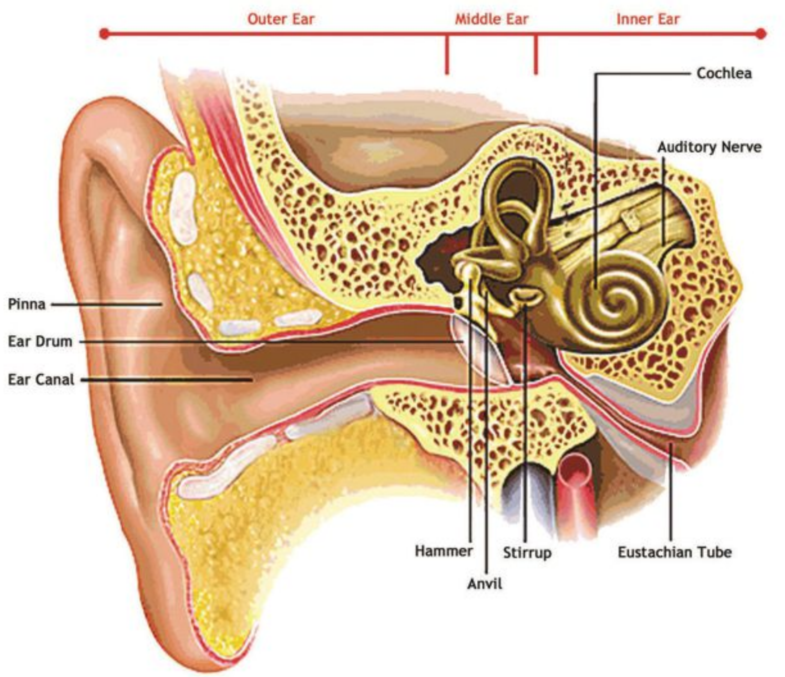
by Andy Coules

Of all the tools at the disposal of the sound engineers, our ears are probably the most important yet also the most over looked. Much like any other tool it helps to fully understand how they work to get the most out of them and ensure a productive working life — because if they break, that's it, we can't just go out and buy a new pair.

Ears are transducers, devices that receive signals in one form of energy and convert them to another. In this case, vibrations are converted into electrical signals which are then interpreted by the brain. Microphones perform a similar job; however, no mic exists that can come close to the frequency response and dynamic sensitivity of the human ear, which explains why we use so many different mics to capture a wide array of sound sources.

HOW THEY'RE BUILT

The thing we call the ear is just the external component, a specific arrangement of skin and cartilage called the pinna. It acts as a parabolic reflector, amplifying and focusing sound waves into the ear canal, and it also serves as a filter that affects the frequency content of the sound depending on which direction it arrives from, thus aiding in the localization of sources. The other thing that aids in localization is that ears come in pairs. We can detect the most minute differences in the arrival time of sounds to each ear, which, together with the filtering provided by the pinna (and the head itself), enables us to pinpoint sound sources with startling accuracy.



Once sound is amplified and focused by the pinna, it enters the ear canal, a slightly bent tube about 1 inch in length and 0.3 inch in diameter, at the end of which is the ear drum. The deep embedding of the ear drum keeps this sensitive mechanism safely protected from external interference — imagine what would happen to your favorite mic if the diaphragm were openly exposed. The ear canal secretes a yellowish substance called cerumen (also known as ear wax), which cleans and lubricates the canal as well as providing protection from bacteria, fungi and insects. The ear drum itself is a thin cone shaped membrane about 0.4 inches in diameter that moves in response to external vibrations and transmits these vibrations to the mechanism of the middle ear.

The middle ear is an air-filled cavity inside the skull containing three tiny bones called ossicles that directly couple sound energy from the ear drum to the oval window of the cochlea (more about this a bit later.) These bones are known as the hammer, anvil and stirrup, so named for their shape. (By the way, the stirrup is the smallest bone in the human body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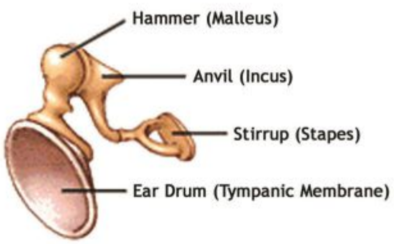
at about 1 inch long.)

The relative sizes of the ear drum and oval window, in combination with the mechanics of the ossicles, means that signals travelling through the middle ear are amplified by a factor of almost 20. Such amplification is necessary because the cochlea is liquid filled and thus more energy is required for sound waves to travel through such a medium — this gain is actually a natural form of impedance matching.

However, due to the nature of the mechanism of the ossicles, the amount of amplification is not equal at all frequencies, peak efficiency is achieved at 1 kHz, which in combination with the transfer function of the middle ear, gives our ears peak sensitivity between 1 kHz to 3 kHz. It's no coincidence that these are key frequencies for the intelligibility of human speech.



At the base of the middle ear is the Eustachian tube, a narrow passage that links the cavity to the back of the throat. In its natural state this tube is collapsed, but it opens in response to both swallowing and positive pressure. Its purpose is to ensure that the air pressure behind the ear drum is the same as the external pressure in the canal, because if the pressure differs, the drum distorts, resulting in mild discomfort and hearing impairment.

Anyone who's flown in an airplane has experienced this phenomenon — as the plane gains altitude, the air in the middle ear expands and pushes the ear drum outwards, while as it descends, the air in the cavity shrinks, pulling the ear drum inwards. The normal mechanism of the Eustachian tube can't keep up with these rapid changes in pressure so active opening is often required to equalize the pressure (i.e., closing the mouth/nose and blowing).

The final part of the ear mechanism is the inner ear, which contains a bony labyrinth that houses two key mechanisms: the vestibular system responsible for our sense of balance and the cochlea that converts sound vibrations into nerve impulses. The cochlea comprises three liquid filled ducts (tympanic, cochlear and vestibular) separated by two membranes (Resissners and basilar) that span the entire length of the curved structure of the cochlea.

It really is a truly remarkable mechanism, but any mechanism this complex is bound to be prone to faults. So what can go wrong?

Sound waves enter via the oval window and cause vibrations in the liquid, which displace the basilar membrane in a pattern that peaks a distance from the oval window depending on the frequency of the sound wave. Thus high frequencies are detected at the base (nearest the oval window) and low frequencies at the apex. The organ of Corti (which spans the length of the basilar membrane) detects these specific displacements and sends an electrical signal along the auditory nerve to the auditory cortex of the brain as a neural message.

And this is where the actually hearing happens as our brains make sense of the nerve impulses arriving from both ears in real time. This happens 24 hours a day, seven days a week — our ears never really turn off, they're always on, ready to alert us to any potential threats. It really is a truly remarkable mechanism, but any mechanism this complex is bound to be prone to faults. So what can go wrong?

HOW THEY'RE DAMAGED

By far the biggest issue is Noise Induced Hearing Loss (NIHL), which, as it's name suggests, is damage to the mechanism of hearing as a result of exposure to loud sound. This can manifest in various ways, such as a general impairment of hearing perception, a loss of perception of a specific band of frequencies, or the hearing of a persistent sound when no sound is present (most commonly a ringing sound, called tinnitus, but it can be a clicking, hiss or roaring sound). This can happen gradually, as a result of regular exposure to loud sound, or suddenly from a short high-intensity sound such as a gunshot or an air horn.

The key to preventing NIHL is simultaneously being aware of noise levels as well as exposure times, and thanks to wide ranging legislation on the subject, we have plenty of accurate data to draw on. A baseline figure agreed almost universally is that levels below 85 decibels (or dB) are considered safe.

The Center for Disease Control (CDC) and the National Institute for Occupational Safety and Health (NIOSH) have published figures stating that exposure time at 85 dB should be no more than eight hours. Further, exposure time is halved for each doubling of sound energy (i.e., an increase of 3 dB), which gives us **Figure 1** (on next page).

I've highlighted 94 dB because it's a handy figure to remember (as the recommended exposure time is one hour), but how can we tell how loud things are?

Decibels is a measurement of a power ratio relative to a specific reference value; in this context, the figures are all dB SPL (or Sound Pressure Level) which defines 0 dB as 20 μPa. Just in case you have no idea what 20 micro Pascals are, it just so happens to be the quietest thing a healthy human ear can hear. In other words it represents the threshold of hearing and goes all the way up to 140 dB SPL (or 200 Pa), which is the threshold of pain (i.e., where sound is so loud that it becomes painful).

The best way to learn when levels become potentially harmful is to get a sound level meter and monitor accordingly. It's important to keep in mind that phone apps providing this information are only accurate if they've been properly calibrated with the built-in microphone. There are even key rings with simple LED displays that can alert us to dangerous levels.

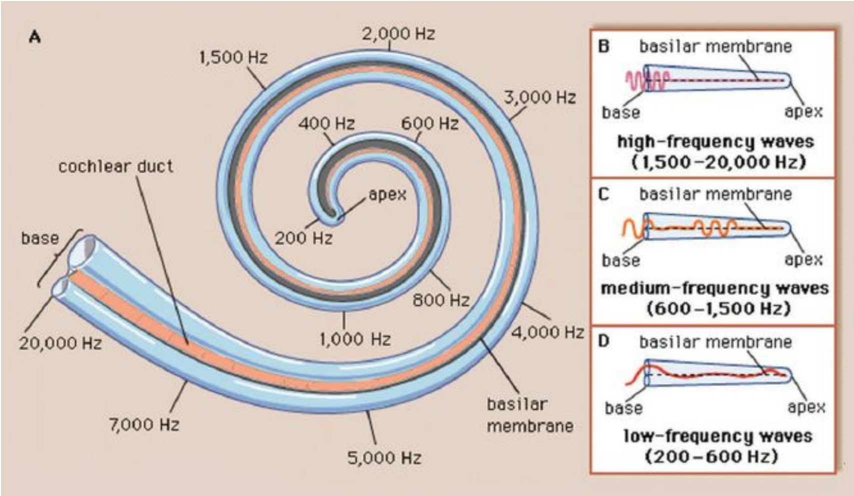
As a rough guide, an unamplified rock band can routinely generate 90 to 100 dB without breaking a sweat, and once the PA is cranked up they'll sit quite comfortably in the 110 dB region. As a result, some form of protection is required, particularly if you want a long career.

There are a variety of ear plugs available that provide a range of attenuation (which is rarely equal at all frequencies). Expensive molded plugs generally provide a relatively flat frequency response while less expensive ("cheap") ones made of memory foam tend to dramatically attenuate high frequencies.

If I'm not mixing or involved in a critical listening role, I'll pop in generic foam plugs for maximum protection; otherwise I use different ear plugs at different times to limit my exposure. Don't forget that risks exist outside of the live concert environment — just because you're off duty doesn't mean your ears are. I've worn ear plugs in cinemas, night clubs, karaoke bars, and even on loud subway trains.

Fig. 1

Continuous dB	Permissible Exposure Times
85 dB	8 hours
88 dB	4 hours
91 dB	2 hours
94 dB	**1 hour**
97 dB	30 minutes
100 dB	15 minutes
103 dB	7.5 minutes
106 dB	4 minutes (approx)
109 dB	2 minutes (approx)
112 dB	1 minute (approx)
115 dB	30 seconds



HOW THEY'RE MAINTAINED

In addition to environmental factors that can affect the mechanism of hearing, there are also a host of medical complications. Like any part of the body, ears are prone to infection. Otitis Externa and Otitis Media are infections of the outer and middle ear, respectively, which are easily treated with antibiotics and in some cases a spray. Otitis Externa (also known as swimmer's ear) can also be caused by the use of dirty ear plugs so it's important to keep them clean (or if they're disposable change them often).

Also never insert things (i.e., thin cotton swabs a.k.a. Q-tips) to clean ears. Aside from the risk of damaging the ear drum, the removal of wax actually hampers the ear's natural cleaning mechanism and you can push the wax towards the ear drum and cause troublesome build-ups. If in doubt about this or any other issues, always seek professional medical advice because if your ears are out of action for a period of time, then so are you.

Annual check-ups should also be an important part of your maintenance regime. They can prove invaluable in identifying issues and potential complications in advance of them becoming permanent, life-changing problems. When in doubt see an audiologist.

Don't forget that risks exist outside of the live concert environment — just because you're off duty doesn't mean your ears are.

The bottom line is just how important hearing is to a long and prosperous career as a sound engineer and/or technician. Thankfully, ears are incredibly resilient and with proper care and attention, they're usually capable of providing a lifetime of reliable service. LSI

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

BANG ON THE DRUM ALL DAY

A fresh look at drum microphone techniques.

by Jonah Altrove

My father used to say, "Son, opinions are like drum miking techniques. Everyone has one." Or something like that...

Truthfully, there's an overwhelming amount of information already out there on this topic, so rather than rehash it here, let's explore a less conventional approach. I'll admit that it's a unique method, as it's a hybrid of studio techniques and some ideas "begged, borrowed and stolen" from engineers I admire.

Picture a snare drum. Let's mike the top and bottom heads. Remember that in a properly wired system, a positive pressure on the mic's diaphragm creates a positive voltage on XLR pin 2, which pushes the loudspeaker cone forward. OK, now hit the drum. Bang. The stick pushes down on the drum head, which moves away from the top mic (negative pressure) and towards the bottom mic (positive pressure). This means our bottom mic's initial transient is a positive voltage, while the top mic's is negative. (If you doubt this, make a recording and use DAW software to zoom way in on the waveform.)

We send both signals to the PA and create a tug of war at the loudspeaker cone, which is being told to move in and out at the same time. This creates a partial cancellation that's very audible, typically as a loss of low-end "body" in the snare sound. Luckily we have a polarity inversion switch on the input channel, which "flips over" the waveform (by swapping XLR pins 2 and 3 at the preamp), so both transients are headed the same direction.

STAYING POSITIVE

I don't intend to enter the fray regarding whether absolute polarity is audible (most research suggests that it is not). Rather, let's focus on keeping as many of our drum inputs as possible in polarity with *each other*. Convention and AES Standard both say positive pressure = positive voltage = outward (toward the audience) loudspeaker movement, so let's go with that.

Start with Input 1, the kick drum mic. The beater (the part of the pedal that strikes the drum) will push the drum head forward (away from the drummer), which is toward the kick mic. It's pretty common to mike both inside and outside kick, but they'll both generate a positive transient.

It's conceivable that the kick and snare would be played at the same time, so shouldn't they be in polarity with each other. If we follow standard practice of inverting polarity on the snare bottom, our kick transient is positive-going, while both snare channels are negative-going. For this reason, I advocate flipping polarity on snare top instead – now kick and snare are all positive transients coming through the PA.

Basically, any part of the kit miked on the same side it's hit is going to generate a negative initial transient, so let's flip the polarity on all of the tom mics. I occasionally use a bottom mic on floor tom, so that can stay positive.

And now we come to the cymbals. I have a confession to make: I have overhead drum mics. In the studio, they're great – studios spend a lot of money creating a pleasing acoustic environment, so overheads, room mics, and ambient mics can yield good results.

However, there isn't a huge system blasting that room sound back into the room. I've always thought it a bit odd that we strive for maximum isolation on every other input, then hang super-sensitive condenser mics several feet above the stage. Solo overheads in your headphones some time and listen to how much "room sound" they're leaking into your mix.

So instead, I utilize underheads. There

are a variety of "clips and claws" that can be used to mount mics straight to the stands underneath the cymbals. This does a number of good things:

- No additional mic stands needed
- It's low-profile, in nobody's way, and there aren't mics bouncing around above the drummer's head, which I always find distracting
- Isolation is much improved; there's less bleed from the other parts of the kit (hold that thought) because the mics are pointed away from the drums. Further, this rejection can be fine-tuned with hyper or supercardioid mics to reject the snare, which is the worst offender
- Set up is really fast
- To me, it sounds better – tighter and cleaner, which to my ears has much more clarity
- It preserves the polarity of the drum kit

PROS & CONS

Like anything in audio, using underhead cymbal mics has its downsides (zing!). For

A Pultec EQP-1A with pots for both boost and cut.

starters, there's likely the need for more inputs, as close-miking necessitates a mic for each cymbal or small group of cymbals. If the drummer has a gargantuan kit, this might be a concern. There will also be more open mics, but gain before feedback actually seems to improve with this approach. Here's why:

Let's say a standard overhead mic is 3 feet from the cymbal, and our underhead mic is 1 foot away. That 3:1 advantage in SPL translates to 10 dB less gain needed at the preamp to get the same level in the mix. We've effectively attenuated room noise by 10 dB, increasing the signal-to-noise ratio (SNR). This is the mechanism that drives the close-mike approach to isolation, which dominates our live sound strategy.

Also be aware that under-miked kits can be harder to mix. Since an overhead picks up everything, it naturally preserves the balance of the drummer's performance. With close mics, a lot more attention needs to be paid to making sure the elements of the kit are balanced in the mix. This can be a lot to juggle.

Finally, be aware that the "close-miked sound" is not appropriate for all styles of music. For example, often with jazz a single overhead is the only drum mic, relying on the player to balance the sound.

TNT, I'M DYNAMITE

Speaking of which, let's talk about kick drum EQ for a minute. Imagine you're at a rock concert. What does the kick

DRUM MIC OPTIONS

Audio-Technica ATM230

A dynamic mic with a hypercardioid polar pattern that reduces pickup of sounds from the sides and rear, improving isolation. The included AT8665 mount screws into the stand clamp for mounting to drum rims. Frequency response is stated as 30 Hz to 12 kHz. Primary drum kit applications: snare and toms. ***audio-technica.com***

Audix D6

A dynamic cardioid mic equipped with the company's proprietary VLM (Very Low Mass) diaphragm, designed to provide natural, accurate reproduction. Frequency response is stated as 30 Hz to 15 kHz. Primary drum kit applications: kick and large toms. ***audixusa.com***

Shure SM81

A unidirectional condenser cardioid mic that offers selectable low-frequency response: flat, 6 or 18 dB/octave roll-off. Also included is a 0/10 dB lockable attenuator switch as well as a swivel adapter. Frequency response is stated as 20 Hz to 20 kHz. Primary drum kit application: cymbals. ***shure.com***

Sennheiser E 604

A dynamic cardioid mic in a designed based upon the company's popular MD 421 that's supplied with an integrated stand mount and universal clip to attach to the rims of toms and snares. Frequency response is stated as 40 Hz to 18 kHz. Primary drum kit applications: snare and toms. ***sennheiserusa.com*** drum sound like? OK, now imagine how the kick drum might sound at a hip-hop show. Then picture a jazz band. You can see that any across-the-board advice on how to EQ a kick drum should be looked upon with suspicion.

There must be a consideration of the genre of music, as well as the expectations of the audience. Fans at a Chick Corea concert would be extremely upset if the kick drum sounded like some sort of explosive device detonating. Fans at a Five Finger Death Punch show would be extremely upset if it *didn't*.

Having just advised against any blanket statements regarding kick drum EQ, I'm now going to make a blanket statement regarding kick drum EQ. Here's a little trick that I have been using for yours before I discovered why it worked. So let's start there.

Although you'd be excused for not having seen it in a live setting, the Pultec EQP-1A is a mainstay in recording studios across the globe. Examining the front panel reveals a curiosity: each EQ band has two gain pots: one for boost and one for cut, a head-scratcher for sure.

What's really going on is that the attenuation frequency is intentionally offset from the boost frequency by about 100 Hz. If you set the EQ to boost kick drum resonance at 60 Hz, the attenuator pot would add a cut around 160 Hz. This creates a gentle yin-yang shaped curve that makes the boost sound bigger that it actually is, which really "tightens up" the sound and makes some room for the bass guitar. So if you like to use LF boost at the kick's fundamental frequency (usually 50 to 60 Hz), you can emulate the Pultec curve by adding a gentle cut a little further up.

A quick chat with the drummer can streamline the process and prevent having to make changes later.

ADDITIONAL MATTERS

By the way, that 18-inch sub behind the drummer? Yeah, the one that he keeps asking you to turn up? Kick drum transients push the drum sub loudspeaker cone outward – towards the kick drum head – which can sympathetically resonate the drum head, like pushing a kid on a swing. Flip the polarity on the drum sub and its excursions are now opposite those of the drum head, which can actually damp the head a bit and buy you a few extra dB of gain before feedback. You know, so you can turn it up more.

Finally, it may be worth having a conversation with the drummer about his mic preferences. I asked a drummer buddy about this, and he said, "I don't care as long as nothing's in my way." He also told me that he's particularly sensitive to "too much going on" in the space between the hi-hat and snare drum. Good things to know when I go to mike up his kit! A quick chat with the drummer can streamline the process and prevent having to make changes later.

Ask 10 drummers – or 10 sound engineers – and you'll get 10 different suggestions for how to mike a kit. And that's fine. Ultimately, let you ears be the judge. If it sounds good, go with it! LSI

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

DRUM MIC OPTIONS

Telefunken M82 Dynamic
A large-diaphragm mic that's equipped with two separate EQ switches: Kick EQ (reduces lower-mid frequencies) and High Boost (increase of upper frequencies). They function independently to provide four unique settings. Frequency response is stated as 25 Hz to 18 kHz. Primary drum kit application: kick. ***t-funk.com***

Earthworks DP30/C
A condenser cardioid mic (also called the "Drum Periscope") with a right angle head on a mini gooseneck designed to not move when drums are hit hard. It's mounted parallel to the drum shell with the RM1 RimMount. Frequency response is stated as 30 Hz to 30 kHz. Primary drum kit applications: snare and toms. ***earthworksaudio.com***

DPA d:vote 4099D
A condenser supercardioid mic with a flexible gooseneck. An optional gooseneck extension adds 50 percent extra length. The gooseneck attaches to a clip designed specifically for drum rims. Frequency response is stated as 20 Hz to 20 kHz. Primary drum kit applications: snare, toms and kick. ***dpamicrophones.com***

Cover Story

SEARCHING FOR SIMPLICITY

A stripped-down sonic landscape for alt-rockers Switchfoot on tour.

by Gregory A. DeTogne, photos by Steve Jennings

TRAVELING TO MORE than 50 cities last fall for live dates in support of its 10th studio album, after a brief hiatus Switchfoot has returned to the road this month for a second leg of the Looking for America Tour, with stops scheduled through mid-February that will wind east from Knoxville, TN to Rochester, NY.

The San Diego-based alt-rock band, comprised of Jon Foreman (lead vocals, guitar), Tim Foreman (bass guitar, backing vocals), Chad Butler (drums, percussion), Jerome Fontamillas (guitar, keyboards, backing vocals), and Drew Shirley (guitar, backing vocals), are comfortably cruising the theatre circuit, playing rooms seating 1,000 to 2,500.

As with the tour's first leg, Switchfoot's production crew will continue to utilize house PAs at most stops, pausing from this strategy only briefly to bring in new ShowMatch arrays from Bose Professional at a few select venues. With Clair Global providing a basic stage package and the band carrying its own input selections, the crew is traveling in the same bus as the band and hauling everything it needs for the show in a single truck.

Front of house engineer Ryan Nichols at his Avid VENUE SC48 on the current tour.

"Last year on the first leg," explains Ryan Nichols, the band's veteran front of house engineer, "I looked at the tour stops and saw that at least 80 percent had great PAs of their own – the House of Blues locations, Fillmores, and whatnot. So why would I have wanted to carry the extra gear and extra people? I guess you could say we were running with a skeleton crew then, just as now. The tour manager is also my monitor engineer, I'm FOH, but I'm also the production manager, and our stage/guitar tech Josh Phiffer even does merch too. Everyone here serves in multiple roles."

The Clair stage package is stripped-down to its core and basic, including mic stands, stage boxes, a drum snake, and a main snake, along with an Avid VENUE SC48 that Nichols uses to orchestrate his house mixes. As on the first leg, rock band Relient K is onboard for part two of the tour, a fact of life accommodated by old

school, copper, 48-channel snakes that are simply swapped out front to coincide with either Nichols' presence or that of Relient K FOH engineer Eddie Hudson.

"I've always strived to base my world around the notion that simpler is better," Nichols notes. "And I mean that in a sense like Albert Einstein maintained, that everything should be made as simple as possible, but not simpler. This is live rock 'n' roll, not rocket science. There is definitely some science too it, but let's not get too crazy. I don't think that's a bad idea; I embrace it."

Nichols came to this philosophical conclusion following a time in his career when, by his own admission, "I got carried away with plugins to the point where I felt that my mix wasn't getting any better. So I stripped down and went to nothing, then just added a couple things that I truly felt helped and made things stand out."

A straight-up approach suits Switchfoot and its sound crew on the road.

Guided by these same principles today, his Switchfoot plugins include Avid Smack!, which he uses as "a dynamic tool, not just a compressor," as well as Waves C6 compression, which is applied on lead vocals, acoustic guitar, and elsewhere very sparingly in a low-key fashion.

Snapshots are set up for the effects he feels are best for each song. "It provides me with the opportunity to put some really deep 'verbs in some songs and nice light ones in others," he explains, "and be able to make those kinds of changes instantly. My delay times are set for each song, and where all this comes in handy is when the band changes its set list, which it does constantly, even on the fly.

"Just because something is written down on a piece of paper doesn't mean anything here," he continues. "They'll

Capturing guitar amps has been the purview of Shure SM57s for a decade.

The mic outlay to capture drummer Chad Butler's kit.

start one song and switch to another, and do the same thing even in the middle of a song they're already playing. I have to be able to adapt to that. That's why I use snapshots. I can change my 'verbs, delay times, and panning instantly without changing my whole desk."

SCHOOLS OF THOUGHT

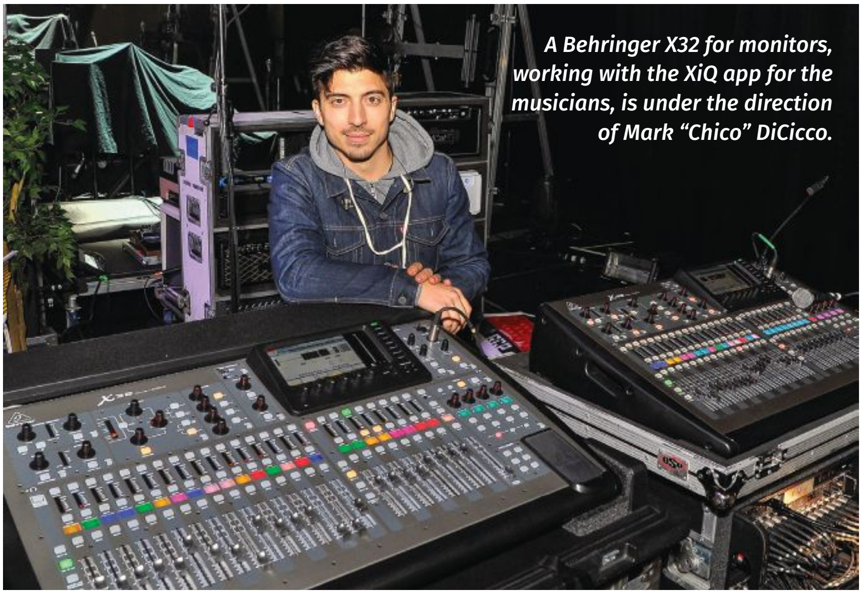
Nichols' job is very hands on and fader intensive. By contrast, the monitoring side of the equation has a more self-sufficient side facilitated by a 32-channel Behringer X32 console. Falling under the guidance of Mark "Chico" DiCicco, who also serves as tour manager, the monitor mixes were essentially dialed in on the first leg for the long haul. As a result, there isn't a lot of on-the-fly mixing to do, so DiCicco – despite his own solid engineering chops gleaned from both the studio and stage – can play more of a "monitor of the monitors" role while focusing on his management responsibilities too.

Sennheiser G3 in-ear wireless systems are the choice for everyone on stage, along with UE11 earpieces from Ultimate Ears. Each band member additionally has XiQ, the monitor app for the X32, loaded on their phones to allow them to adjust and refine their mixes as needed, all within a realm of safety that keeps them from accidentally dropping the drums from another band member's mix while adjusting their own.

Switchfoot maintains a stage plot dominated by Shure components. With the simple-is-better algorithm serving as a backdrop once again, the backline starts with a BETA 91A microphone mounted inside at kick. Nichols previously practiced the traditional "91A inside BETA 52A outside" formula, but halted the habit after the umpteenth time he found himself wondering why his kick drum sounded lousy only to look up on stage to find that someone in the band had kicked the 52 loose and out onto the floor.

"I adapted and discovered that I could get a great kick drum sound just with the 91 alone," he says with a hearty laugh. "So again, why make things more complicated than they have to be, right?"

For snare top, Nichols contends that the Heil Sound PR 20 is very similar sounding to a Shure SM57, but has "a little bump that just makes it talk a little more." Still relying upon the venerable SM57 for snare bottom, however, these mics are complemented on the kit by KSM137s on hi-hat and under-ride cymbal, while Sennheiser 604s stand in for toms and KSM32s fly overhead. SM57s are also Nichols' choice across



A Behringer X32 for monitors, working with the XiQ app for the musicians, is under the direction of Mark "Chico" DiCicco.

the board for guitars; currently these inexhaustible workhorses are the same ones he's been using in this application for the past decade.

AS AN ENSEMBLE

Vocal mics are SM58s, all hard-wired except for frontman Jon Foreman, who cuts the cord with a UR2 wireless system. Instrument wireless is all Shure UR2 as well, except for bass player Tim Foreman, who uses a Shure digital system.

"For me, the whole 58 thing revolves around the mic's ability to reject some stage noise while allowing some stage noise through that's fairly easy to make musical," Nichols says. "Some people look at their stage and decide that everything should be isolated. They take their guitar amps and face them sideways, or put them underneath the stage. They start erecting plexiglas screens. I subscribe to the notion that to play as a rock band means that you play as an ensemble. That's a concept some don't understand or might not even have heard of before.

"Even though I've been with Switchfoot for 14 years, I have an orchestral background. That's a world where mics don't exist, and instruments are played onstage in a way that they work together to sound bigger as they travel out to listeners' ears in the room. I've expounded on that idea here in the world of rock, and that's how I got to where I am today. I put mics out in the open where I have to, and make everything work. It's a challenge, but when you hit the note the rewards are endlessly satisfying."

OUT OF SIGHT

All that said, one might conclude that Nichols might be the kind to rely heavily on gates. True, he gates snare bottom and toms, but other than that everything else is open. A fair bit of compression is put to work, but with the mantra that nothing is over-compressed.

"Bottom line, I guess you could say it's my overall goal to make this band sound the best I can in whatever room I'm in, and on whatever PA," he concludes. "It's my task to insure that the audience enjoys an incredible rock show every time, with special focus on the intensity of the drums and bass, a nice fit in my mix for the guitars, and the vocals sitting in that perfect spot on top where everyone can hear and understand every word that's sung.

"I've always approached mixing from the standpoint that if someone in the crowd realizes I'm there, chances are I'm doing something wrong. I don't want to make a mark, I want to remain invisible. Everyone should walk away shaking their heads and saying to themselves, 'Man, that sounded incredible!' I never want them to be asking themselves why it sounded that way. That's not the point, and it's certainly not rock 'n' roll." LSI

***Gregory A. DeTogne** is a writer and editor who has served the pro audio industry for the past 32 years.*

MODERN ANALYZER DEVELOPMENT

The evolution of large-scale sound system optimization, part 4.

by Bob McCarthy

A Meyer Sound SIM 1 System circa 1987. From top to bottom: Hewlett Packard Integral Personal Computer, the HP 3582A dual-channel FFT analyzer, the delay controller with a customized Audio-Digital delay line, the 8 channel mic switcher, and two 8-channel line switchers (enough for 8 EQ inputs and 8 EQ outputs). All of the devices were under the remote control of the computer, which could switch between any mic or EQ in the system and manage the data library.

The analyzer is at the core of a measurement system. The manufacturers of laboratory test equipment design their products to be adaptable to a wide variety of scientific applications. Their working assumption is that individual users will add the peripheral devices required for their application.

The laboratory analyzer needed a lot of help to be ready for the down and dirty of touring with a sound reinforcement system. The process began slowly, but eventually became standardized since the basic structure of sound systems is the same throughout the industry.

A key challenge was to get the maximum practical use from the scientific power of the fast Fourier transform (FFT). It had to easily patch into the sound system and get answers quickly in a way that engineers could understand. It had to be small, portable, durable and quiet enough to sit at front of house. Most importantly, it could never stop the show.

There were quite a few customizations required to begin this transition in 1984. First we needed an external microphone preamp since the Hewlett-Packard 3582 spectrum analyzer had unbalanced line level inputs (banana plugs!). We also needed an external delay line so we could synchronize the measured mix console output to the sound arriving at the mic.

Digital delays were barely extant at that time and our initial work was done with an analog delay with a limit of 100 milliseconds (ms). One of the early jobs had a mix position at 105 ms so we had to set a fence around the mic a few feet ahead of the mixer.

At the earliest gigs we used a "Y" cord with bare wires and gator clips to split the mix console output feed for us and then wired the other end to go unbalanced into the HP analyzer. (Kids, don't try this at home!)

We won't be allowed to measure anybody's system if we cause it to hum, so one of the first steps was to create transformer isolated access points that see the signal at the input and output of the system EQ. We could now see the signal flowing through the system without risking an interruption or signal degradation.

MEMORIES

It's easy to forget how recently computers joined our workplace. It was unlikely to see a computer at a gig in the early 1980s when the real-time analyzer (RTA) was the primary measurement tool. If we did see one, it was certainly not being used for system measurement. Even the best measurement tools of that era had very little memory, which greatly limited what we could learn from them.

A handheld RTA like the Ivie IE30 could hold one or two stored traces. You could toggle between the live trace and a stored one to make a "before vs. after" or "here vs. there" kind of comparison. In order to store another trace we had to overwrite the previous one. Once the device was powered down, the data was lost for good. This lack of memory made it very difficult to learn about the behavior of sound systems and how to optimize them. Essentially we had two responses to work with: right now and one from memory. We could compare two locations out of 10,000 seats in an arena. There was very little we could act on with such a minimal amount of data.

We'd made a huge step forward when we moved from the single-channel RTA (1/3rd-octave, amplitude only) to the dual-channel FFT analyzer (24th-octave amplitude, phase and coherence). But without computer memory there was an extremely clear picture of only one location.

Moving from low-resolution, one-dimensional analysis at the mix position to high-resolution, complex signal analysis at the mix position was the extent of our progress. We don't need memory to turn the EQ knobs at the mix position. In fact, we don't need a high-resolution analyzer to do this. We don't need an analyzer at all because there is no objectively right or wrong EQ setting for the mix position. Whatever the mixer says is "right" is right. End of discussion.

But there are objective rights and wrongs for optimization, the process of creating uniform response over the space. Optimization requires us to measure at locations other than the mix position.

Memory is the mobilizing ingredient for optimization that moves us beyond the mix position. Multiple trace storage allows us to compare responses all over the room. We can make a change in aim, splay, level, EQ or delay and see how it affects the response in all of the relevant areas. Are the responses down on the floor and in the balcony more closely matched than before? If so, we're making forward progress toward optimization. If not, we've eliminated one wrong option and can now try another till we get it right.

We also need a good librarian. It won't do us any good to store data if we can't find it later. For this we need an orderly system of organization: a data library. We're measuring multiple loudspeaker systems with multiple mics in multiple locations. Our data library has to allow comparison of the mains upstairs with the down fill below, each position with the hall empty to the house full, and so on.

THREE TRANSFER FUNCTIONS

Transfer function measurement allows us to characterize the system as a whole or subdivided into parts. The standard signal flow is mix console (our electrical input reference) to signal processor to speaker system (our acoustical output measured by a mic).

It's easy to visualize the transfer function measurement between console and microphone. If we see a peak we can add a dip at the equalizer. But here's the twist. Once we have started turning knobs on the equalizer we no longer know who is responsible for a peak or a dip. Did the loudspeaker system in the room cause it, or is it something we did on the EQ?

Once the filters go in we lose the evidence for why we chose to put them there. This may seem like an obscure point but in my experience it is absolutely vital to know why every filter is selected. This is a major factor in minimizing the probability that we are doing stupid things — most notably over-equalization.

But how can we monitor the EQ itself? The key is to add a third measurement access point at the mid-point in the series connections: the signal processor output. This location is both an output (of the processor) and an input (of the loudspeaker system).

If we keep the console input reference the same and move the measurement probe upstream to the processor output, we will see just see the EQ alone. Conversely, we can isolate just the speaker system in the room (independent of the signal processor) by moving the input reference downstream to the mid-point and use the mic for the output.

These three measurement points have been integral to every generation of the Meyer Sound SIM platform. The three data points (EQ input, EQ output and mic) can be paired into three distinct transfer functions: Room (EQ out vs. mic), EQ (EQ in vs. EQ out) and Result (EQ in vs. mic). The Room transfer function shows the raw, unequalized response (even after it is equalized). The EQ response shows the EQ response (duh!), and the Result shows the combined effects of the speaker in the room and the applied EQ. Room shows us the problem. EQ shows the treatment, and Result shows whether we've achieved our goal.

It's extremely helpful in practice to have this complete characterization. Of course in the old days it helped because it was the only way we could see the EQ response. Now we can just look at the

The author and a SIM system at the 30th Grammy Awards at Radio City Music Hall in NYC. Multiple mics monitored the response in the hall during the broadcast.

control software.

But there's more. The modern EQ is a full suite DSP, and can be doing all sorts of other things you might not see on the user interface. I like to see the response and not be surprised. The data shown in the room response contains only the loudspeaker in the room, and therefore complies with the laws of physics.

If I see a propagation time of 100 ms, I want to know if the loudspeaker is going through 100 ms of air or if it's 90 ms of digital delay and 10 ms of air. This makes a big difference, because I expect a loudspeaker response to have some damage after 100 ms of air travel, but not after 10 ms. That is just a single example, but simply put, knowing the processor from the room helps keep us wise as to the physical reality of the room. DSP + acoustics = magic. Be careful of magic.

SIM in action for Les Miserables in Boston (1988 or 89). Notice that the analysis system is larger than the signal processing (EQs and delays) that it's monitoring.

PRIMORDIAL SIM

The experimental phase of SIM ran from 1984 to 1987. During this period our hardware was custom made and constantly evolving as we learned more. The gear consisted of the HP 3582, a custom two-channel interface to the sound system, a digital delay, and a measurement mic and preamp. Our "computer" was the HP 87, which ran a language called Basic and had a whopping 256 kilobytes (kB) of memory. How much data could we store in it: two entire responses (remember that each response took a whole series of 63 traces from the HP). The computer had only one function: store and retrieve data so we can make comparisons.

However, in 1987 we broke the multichannel barrier with the official introduction of SIM. We now call it SIM 1, but at the time we didn't know if there would be a sequel. The analyzer was still the HP 3582 but we were set up to provide it with a lot more to measure. SIM 1 was fully multichannel. The standard system had an interface to measure 8 equalizer inputs, 8 equalizer outputs and 8 microphones. It could be expanded to 16 or even 24 though not many people did so (it was already 20 rack units).

The system also had a digital delay and a computer with an expanded role. The computer, the HP IPC was the master controller of the whole system. The analyzer, the delay line and the switcher interfaces were all driven by the IPC.

The capabilities of the system were unprecedented at the time: we could patch into the system and switch between multiple equalizers and measurement mics, store the data at 8 locations and be ready to compare them to another 8 sets of incoming live data. All told we had enough memory to store 16 sets of traces. The 1 megabyte (MB) memory card cost $1,200 at the time!

The multiple mic capability changed everything because now we could start to really see how a change in one processor setting or splay angle affected multiple locations.

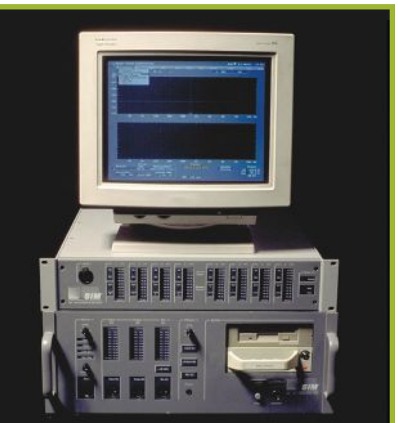
The SIM 1 platform gave us all the information we needed to set delays, levels, EQ and aim loudspeakers. The data could be stored in memory and post-processed. In short, it could do almost everything a modern analyzer software program can do now. But because it was built around the "some assembly required" HP3582A, it required 63 steps, not to mention taking up 24 rack units, weighing 600 pounds and costing $80,000. We did a lot of shows, learned a lot of lessons, and were ready to move forward to the next generation.

Our initial strategy was to get one of the companies that manufactured analyzers to undertake the project. They were the big players in the measurement market and we were a relatively small loudspeaker manufacturer. We laid out the plan but there were no takers for a market so small.

Hewlett-Packard did come up with a proposal to network 21 HP3582's together with a computer to display a composite response. This would have cost around $400,000, weighed almost a ton and required "only" 100 rack spaces. Very practical! They already had customers that were using this type of system: U.S Navy submarines. It was clear that we had to either get Pentagon funding or do it ourselves.

BRINGING IT TOGETHER

SIM II, introduced in 1991, was the first analyzer to put all the linear spans together into a single high-resolution display. This was accomplished by taking 8 separate time



SIM System II (1991) brought the analyzer and computer control into a single chassis. SIM II calculated 3 transfer functions (Room/EQ/Result) at 24 points/octave and displayed amplitude, phase coherence and the impulse response. The switcher accessed 8 microphones and 8 equalizers, and the system was 21 times faster than the original at one-third the size (and also possible for a human to comprehend).

SIM 3 (2003) represented an incremental expansion of capability, ease of operation and reduction of size, helping the optimization process become more mainstream.

records and using only the top octave (its highest resolution) and throwing out the rest. The linear pieces were spliced together as a log series to create a single trace of 24 points/octave data. It was not technically 1/24th octave data, since each octave contained 24 linearly spaced data points, which is why it's termed "quasi-log."

Meyer Sound used the term "Constant Q transform" to describe this groundbreaking implementation of the Parks-McClellan algorithm. All of the modern analysis systems in large-scale use in sound reinforcement have implemented an adaptation of the algorithm. Each octave (going down) is derived from a time record twice as long as the one above, while maintaining the same number of data points. The result is constant, high resolution.

The introduction of SIM II represented the first time that people had seen A) full-range high-resolution data, B) full-range log phase response, and C) coherence/signal to noise over frequency. This development created a permanent place for the FFT analyzer at front of house because the data was now displayed in a way that the average engineer could understand. This did not mean that the work of optimization was suddenly easy, but it was, at last understandable and teachable. SIM 3, the next generation, was very much in the mold of its predecessor but with faster processing joined by a smaller size and price point.

FFT GOES MAINSTREAM

The SMAART platform was introduced by SIA in 1995 under the leadership of Sam Berkow. The key breakthrough with SMAART was that it could run the FFTs on a standard PC without the need for a dedicated DSP engine. SMAART was a program whereas SIM III was a 4-rack-space piece of hardware with internal A/D, DSP, delay line, etc. (and software).

SMAART required each user to provide all the peripherals, which was a challenge at the time, while SIM was complete but also very expensive (especially for something that people were unsure about what it was/did). With SMAART, there was "some assembly required" but it had implemented the quasi-log FFT and its price point was so low that sales expanded quickly. The FFT had moved into the mainstream, but the majority of early SMAART users were basically using the tool as a high-resolution RTA.

We now had high resolution at FOH, but still, the mentality was equalization rather than optimization. SMAART (now stylized as Smaart) has evolved greatly under the guidance of Jamie Anderson and the team at Rational Acoustics, moving FFT optimization squarely into the mainstream. They've taken advantage of the ever-increasing processing power of the personal computer to expand its capabilities with each of its 8 generations.

I can still remember arriving at tunings 30 years ago and having people laughing at what a stupid idea it was to bring all this laboratory science into their world. I'd be lectured about how I lived in "theory world" and they lived in the "real world." I don't hear that any more. The FFT is now the new normal. It just took a long time to get there. LSI

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

In Profile

PASSION FOR THE CRAFT

The world of touring front of house engineer Michelle Sabolchick Pettinato.

by Kevin Young


CREDIT: JASON POWELL

OVER MICHELLE Sabolchick Pettinato's 25 years as a front of house engineer, she's worked with a wide variety of artists, among them: pop icons Christina Aguilera and Gwen Stefani, 90s rock outfits Collective Soul and Goo Goo Dolls, and currently, she's on tour with one of her favorite bands from her teenage years, Styx.

Sabolchick recalls working with a colleague, Jeff Heintz – who's since served as a co-production manager and keyboard tech for Styx – on Stefani's first solo tour in 2005: "On that tour, we talked about our favorite bands growing up; Styx was mine, and Journey was his. We always said, 'You know, one day, I really want to work with them.' Within a month of wrapping up that tour, Jeff called me and said, 'You're never going to believe who I'm working for' – it was Styx."

STEP BY STEP

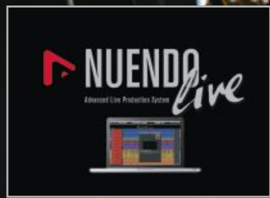
Born and raised in the small town of Ashland, PA, in the heart of the state's coal region, Sabolchick, a self-described science geek, spent a fair bit of time taking old recording and stereo gear apart in an effort to figure out how it worked. She also loved music and studied piano for many years, but never wanted to be on stage or perform in front of people. Consequently, early on, she felt that a career as a recording engineer was a good fit.

"Live sound hadn't occurred to me to that point. I'd gone to tons of concerts, but never even thought you could do it for a living," she notes. During high school, a friend introduced her to a local promoter who let her shadow him and learn the business. After graduating she applied for a university recording arts program but unfortunately, the tuition was prohibitive, so instead she opted to study at Penn State University with a major in music.

After a year, she realized the program wasn't going to help her achieve her goal of working in audio and instead enrolled in a Recording Engineering and Studio Maintenance program at The Recording Workshop in Chillicothe, OH. Completing that helped lead to a gig at a local radio station, WMGH, selling and recording commercials, but it was only a stepping stone.

Hungry for actual pro audio work, Sabolchick hopped a bus to Nashville, bunked with a cousin for a few weeks, and knocked on pretty much every door on Music Row. She soon realized, however, given her two-line resume, that she needed more training and experience, and enrolled at Full Sail University in Winter Park, FL, where it was the first time she'd felt that others truly supported her career choice.

While she entered the business at a time when far fewer women worked in audio, Sabolchick downplays the impact being female in a largely male-dominated industry had on her long-term goals. However, she adds, there were times when people misunderstood her role. As an example, she cites returning to Pennsylvania, where she interned at a small sound company near Philadel-



phia: "Me and another guy would go out and do bar bands, set up the gear, sound check, then leave and have dinner. When we came back for the show the bouncers would try to charge me cover, thinking I was the guy's girlfriend."

Throughout her career Sabolchick has been driven to go where the work was. Early on, she regularly traveled back and forth between Pennsylvania and Florida to make ends meet, working as a stagehand, running lights, mixing, doing whatever was necessary. "I worked at any job, anywhere, until I reached the point where I wasn't going to learn more or progress," she says.

A big break came about by being in the right place at the right time – on a day she'd intended to take a train to Florida to work, Amtrak went on strike. That afternoon, a friend called to ask if she'd take over his gig doing front of house for Spin Doctors, and roughly a month after she signed on, the band broke nationally.

During that first tour, Sabolchick also met her husband, Jim Pettinato, who had signed on as lighting designer and production manager. ("We grew up 90 miles apart, but never met until Spin Doctors, and we just clicked.") They were engaged for five years before getting married, she notes, laughing, "We just couldn't get our schedules together," adding that balancing real life with being on tour regularly isn't always easy: "But this is our life. We both get that and deal with it."

So then I'm thinking, 'Awesome, we're gonna die,' but it was an amazing show.

CHOOSING PLATFORMS

When Sabolchick and I recently caught up via phone, she'd just returned to her home in Scranton, PA, after touring, alternately, with Styx and Mr. Big (a band she's mixed since 2009). She's been full time with Styx since July 2015, but also worked with them briefly in 2013. That initial stint was to fill in for the band's long-time live and recording engineer – Gary Loizzo, owner of Pumpkin Studios – who passed away in early 2016 after a lengthy battle with pancreatic cancer.

Sabolchick in an analog FOH world on the road with Big Time Rush in 2012.

When Sabolchick took the Styx gig she was working on a Yamaha PM5D console, noting, "Gary had one for years and, because they do over a hundred shows a year – going out for several weeks, coming home for a week, then going out again – they don't rehearse, because they're constantly touring."

Ultimately, Sabolchick switched to a Midas Pro X digital board when the band was prepping for a gig in Las Vegas in 2015. Still, she describes herself as an "analog girl" who'd prefer that approach with Styx. "I'd actually still have an (Midas) XL4, but we can't fit it in the truck," she says. "I love analog, and with Styx it's a crazy mix intensive show.

"For instance," she continues, "at the top of 'Lady,' I've got four vocals, a little chime mic that's got to be cranked, and a keyboard part. When the song kicks in after the first verse I have to mute the chime mic, mix a four-part harmony, and I've also got the lead singer switching to an RF mic from a hardwired mic in the middle of a line and all that's happening in a split second. On a more straightforward show, where you don't have a lot of cues, it's one thing; you kind of set it and forget it. But with a lot of the bands I mix there are a lot of changes throughout the night and a lot of things to think about."

With Styx or any mix intensive show, she adds: "There's so much going on that, with digital, it can be difficult at times. I have to do a lot of thinking: 'How am I going to lay out the board?' It takes a lot of thought as far as workflow, efficiency and getting at what I need to as quickly as possible. With analog, there's no scrolling through pages. You just grab what you need when you need it. It's all right there in front of you."

Sabolchick's happy to roll with the times, and while she doesn't mince words when it comes to her preference the warm, fat sound of consoles from back in the day, she does admit digital consoles have become significantly better over time. "That's why I'm on the Midas now. I think it's one of the best sounding digital consoles out there, and if you start with the right tool, you shouldn't need $5,000 worth of plugins to make it sound good."

She also doesn't miss carrying extra racks, dealing with more patching, and the added potential for buzz and noise floor issues that analog gear entails. "There are definitely pros and cons," she states. "When I first started using digital I was carrying a rack of Empirical Lab Distressors, TC Electronics reverb and an Eventide H3000. That was my standard rack."

Ideally, she'd love to use that gear for Styx, but given the lack of truck space it's not feasible. "Besides," she adds, "the Pro X can handle all that and I've actually found, lately, that trying to integrate analogue outboard gear with digital can create more problems than it's worth – latency, noise issues – and the Midas compressors sound great, so I don't miss the outboard gear."

KEY ELEMENTS

Regardless of the project, Sabolchick's preparation begins with going through the artist's catalogue. "Some, like Melissa Etheridge, have a number of records, others just have a couple, but I start there and when we do rehearsals – if we do rehearsals – I have a conversation with them," she explains. "Some say, 'Just mix. Have fun.' But others have very distinct ideas about what they want. Even if there are no production rehearsals, I try to be in band rehearsals to see what everybody's got going on because, so often, it's completely different live than on record."

On Stefani's tours, Sabolchick worked closely with the band's MD/keyboard player, Kris Pooley. "He did incredible arrangements for the show and put it together in a cohesive fashion. With Gwen, the first shows were festivals, kind of throw and go," she says. "When we did the full tour, we had two or three rehearsals before we a chance to talk and I said, 'Do you have any suggestions or requests?' She said, 'Just make it sound good.' She hired people she trusted and let you do your job."

During her career, Sabolchick has emphasized persistence, the willingness

Mixing in the digital realm with the Goo Goo Dolls in 2014.

to continue to learn constantly, networking relentlessly, and the necessity of understanding the craft inside and out as the best way to succeed. Communication is also key: "It's all about figuring out what the artist's vision is."

Additionally, she stresses the fact that although there's a creative element to mixing, there's something very important to always keep in mind. "You work for the band. They don't work for you," she states. "You're presenting the artist's music to the audience. The artist has to trust you're going to do that the way they want."

In 2013, Sabolchick co-founded Soundgirls.org with long-time Pearl Jam monitor engineer Karrie Keyes after they both served on a panel at the 2012 AES show. Although she's since removed herself from an active role with Soundgirls, she remains an avid supporter and continues to be a contributor when time permits.

EVERYTHING FITS

Beyond satisfying her love of music, science, and technology, working as a touring engineer has fed an enduring passion for travel. "My family used to do a lot of traveling and camping every summer," she says. "I always knew there was a huge world out there and wanted to see it. So live sound was perfect. I could do what I love, travel, and get paid for it."

In seeing the world, going to places she never dreamed she'd visit, Sabolchick's found herself in some areas where, frankly, personal safety was a concern. Case in point was a tour with Mr. Big in India. "A couple of years ago we did three shows in 10 days there, and it was the hardest touring I've ever done, emotionally and physically draining," she says. "It's hard to fathom that there are still millions of people without basic necessities: clean water, shelter, electricity. I've been to a lot of countries where you see poverty, but in India there's just another layer to it.

"We were outdoors playing in these huge polo fields," she continues. "The third show was for 15,000 people, with another 7,000 outside that they couldn't let in because the venue was packed. Right before the band walked on stage, the system tech from our Indian sound company said to me, 'It's amazing you're here.' He'd been at all the shows, and I said, 'Yeah, It's amazing.' But he said, 'No. It's amazing because it's really dangerous. There are two tribes here that fight constantly. There are murders and kidnappings, but the government negotiated with them to stop fighting so we could have this concert.'

"So then I'm thinking, 'Awesome, we're gonna die,' but it was an amazing show. The band was exhausted, but they gave it 120 percent, and the audience was insane. It was incredible."

Ultimately, it's the challenging gigs you remember, she concludes, rather than those that go off without a hitch. "When you get into a tour, day in and day out, you do the same thing, and it's nice when it's easy and it all comes together, but then you have a day when you're outdoors and all of sudden there's a tornado coming in, and it's like, 'What are we going to do?' That's the day you remember, when everybody pulls together to make the show happen. That really makes it memorable and rewarding." LSI

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

Backstage Class

COMPUTER WORLD

Field applications of laptops, as well as a range of software.

by Craig Leerman

Working with RF Explorer Client for Windows.

Of all the tools my company brings to shows, computers (in laptop form) are certainly among the most important. They allow us to edit and play back music tracks, convert audio file formats, measure/analyze audio, and evaluate/monitor the RF spectrum and coordinate frequencies for wireless systems. We can also get equipment manuals and quick start guides, surf the web during downtime and even record the gig.

Whether it's a Mac or PC doesn't matter as long as it helps make the client happy. Our choice is PC, normally two on a show. Both contain the same software; one is tasked as the main audio player and the other usually runs measurement software while also serving as the backup audio player and fulfilling a range of distinct functions to that particular gig.

STREAMLINING THINGS

For years we utilized Instant Replay by 360 Systems for audio playback. These hardware machines are very common at corporate shows and in the broadcast world. Instant Replay provides playback at the push of a button, and with 50 buttons on its face and 20 banks, there are 1,000 tracks at your fingertips.

The downside is the necessity to load the files into the machine as analog files or AES — no plugging in a thumb drive directly to transfer files. It also only allows basic "heads and tails" editing, and if you have to fly to gigs, it's another large hardware device to lug around.

A few years ago, we switched to Sports Sounds Pro software. Already carrying at least two computers to gigs, it made sense. The software lets us load buttons on the computer and click on them with the mouse (or set up keys on the keyboard as hot buttons). With 10 banks of 10 pages, and 48 buttons per page, it supplies a whopping total of 8,640 user-configurable buttons.

Unlike hardware units, these buttons are labeled with the track name and run time so we don't need cheat sheets or board tape. Files are played from their folder(s) located anywhere on the computer as well as from a hard drive or thumb drive. Pages and buttons can be color-coded to keep things better organized.

In addition, a feature called Instant Play scans a file and listens for audio, then plays the file from the start of the audio, no editing the "head" of a track, which is perfect for when you're handed a file at the last minute. Clients hand these to us all of the time. We download the file(s) into a labeled folder that's usually located on the desktop so that it's easy to find.

Then we convert the file(s) to Wave (.wav) format using Switch Audio File Converter from NCH to convert the files. The program can convert between more than 40 audio file formats, and can also extract files from (almost) any media. NCH offers a free trial if you'd like to check it out (www.nch.com.au), and the company also offers a range of other software for editing, recording, mixing, ripping and burning CDs.

The label for the folder on the desktop includes the client and/or show name, as well as the date. Then if needed, the tracks can be quickly called up and edited (including implementing client requests). For this we use free editing software called Wavosaur (www.wavosaur.com) that handles .wav (and MP3) files. It also accommodates recording, so when there's the need to create announcements for a client, it's quite handy for one-stop record/edit. Typically the only editing that's needed on client files is trimming any dead space before the audio starts, but sometimes the track needs to start at a particular spot or be edited it to a particular length. The final thing to make the track(s) ready is to normalize it/them (via Wavosaur). Normalizing them all to the same level eliminates having to ride the faders for each cut.

Opening Sports Sounds Pro, all normalized tracks can be assigned to their own buttons. It's also a good idea if there's time to test each track through the PA. For a single track this process just takes just takes a few minutes for conversion, editing, normalizing and loading into the play, way less time that it took to load a track (let alone more than one track) into a separate hardware device.

MAKING CONNECTIONS

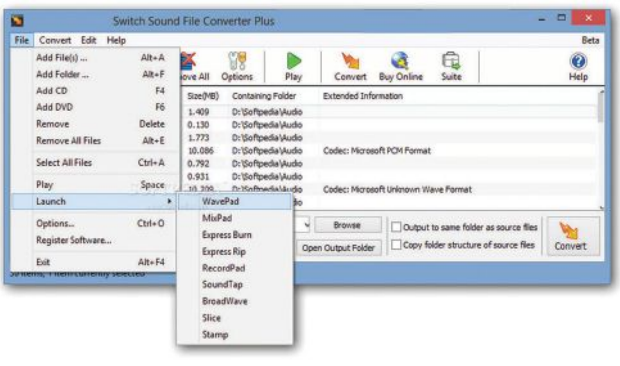
At shows, the main audio computer is positioned where it can be easily viewed, working with a wireless mouse for control. We have a few ways of routing audio out of the computer.

The approach for some time was to utilize a Focusrite Scarlett 8i6, an 8-input/6-output interface with two XLR inputs and two TRS balanced outputs. It required downloading Scarlett software on my computers and carrying around a set of TRS-to-XLR adapter cables. The cool thing is that the 8i6 offers a S/PIDF digital output, but the not-so-cool thing is that most modern consoles have omitted a S/PDIF input in favor of more standard digital connections like Dante and AES.

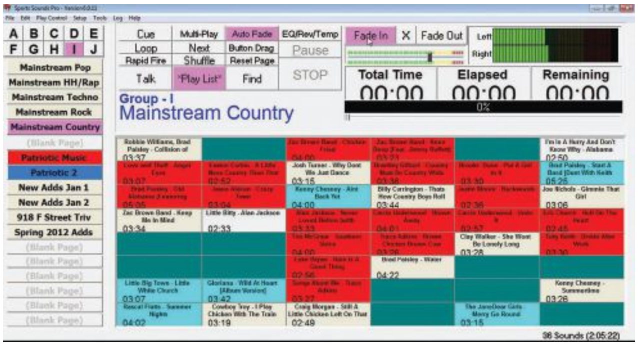
My new favorite analog interface is the USB-P from Peavey. It's a compact box that plugs into a USB port and shows up on any computer as stereo outputs. It also has a stereo to mono switch, so it can run a single channel if that's all that is available. The USB-P has two XLR outputs and is powered by the USB port.

The headphone jack on the playback computer also gets some use when the USB-P is needed for a client's computer or for connecting to graphics/video playback machines. We carry 1/8-inch TRS-to-dual-RCA cables with RCA-to-1/4-inch and RCA-to-XLR adapters to link the headphone jack to the console.

On a recent freelance gig, the A/V company deployed several Switchcraft 318 Mini AudioStix to serve as DIs on the podium. The 318 is a (very) compact unit with a 1/8-inch stereo jack on one end and a single XLR on the other, joined by a volume control and ground switch. It converts a stereo signal to balanced mono mic level output. I was so impressed with the quality and



A look at Switch Audio File Converter from NCH.



With Sports Sounds Pro, audio files can be played at the click of a mouse or via "hot buttons" set up on the keyboard.

small size that I added a few to our inventory, where they'll come in quite handy when we need to get audio from a laptop or mobile device into a single channel of a console.

There's also interfacing with Dante networks; we've been using Dante Virtual Soundcard for a few years now, carrying 5- and 10-foot Cat-5e cables as well as a compact 5-port Netgear gigabit switch for quick and easy interconnect. Both of our primary laptops at shows have Dante controller installed and can be linked to the network to patch inputs.

We also run the newer Dante Via, which provides access to any audio device on my computer, internal or external, into a Dante network. If desired, we can connect the computer's microphone jack into the network, but prefer using the computer's headphone jack to monitor Dante channels, particularly handy for troubleshooting and/or when it's a larger network.

MORE BANG FOR THE BUCK

On fly dates where a primary objective is keeping gear weight as low as possible, I carry an Icicle USB converter from Blue Microphones to feed analog audio to the laptop. A small, light, single-channel unit, it works well in addition to providing phantom power. Normally it's deployed with measurement mics.

Speaking of which, our computers are also loaded with both Rational Acoustics Smaart and AFMG SysTune platforms for measurement and analysis. I usually gravitate toward SysTune but many techs prefer Smaart, so our choice is both. Measurement mics are usually M30s from Earthworks, but when space is extra tight, the choice is the miniscule Apex220.

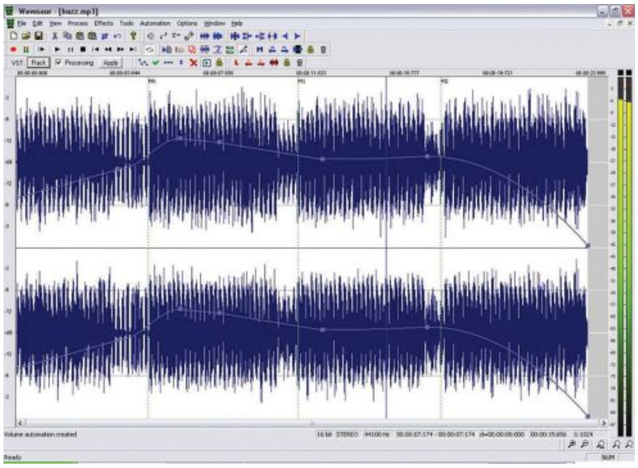
Yet another big computer application is managing wireless systems. While most music shows only use a handful of channels, corporate events can occupy a considerable number of frequencies throughout a venue when you factor in a general session, breakout rooms and production crew communications.

A lot of my company's work is in Las Vegas at casino/resort ballrooms, where there may be an in-house show right next door that has dozens to hundreds of frequencies already in use.

The Earthworks M30 (left) is the author's "go to" measurement mic, but when travel space is particularly tight, there's the option of the Apex220, which measures just 3.7 inches long.

Therefore, frequency coordination is especially crucial. To check the airwaves we deploy a compact standalone spectrum analyzer called RF Explorer, available from several sources. (Ours was acquired from NutsAboutNets.com.)

RF Explorer comes in many versions, including some that also account for Wi-Fi frequencies, handy if you're using wireless operating in the 2.4 GHz bands. While the screen on the unit is very informative, the scanner really shines when interfaced with a computer running data acquisition and analysis software. We employ two different software packages that basically do the same thing: RF Explorer Client for Windows and Touchstone. Both offer a large frequency



Wavosaur audio editor providing a representation of audio files without graphic aliasing.

spectrum display and a waterfall display (if that's desired). Using either, we can log data and/or check out what's happening in real time.

After seeing what's going on in the wireless spectrum, it's time to coordinate frequencies. Most of the time we utilize Intermodulation Analysis System (IAS) from Professional Wireless Systems. The pro version is a bit costly but allows

A Word About Audio Files...

AS NOTED IN THE ARTICLE, we use Wavosaur for basic voiceover and announcement recording, but for client recordings of events (most of them corporates) and multi-tracking, we've recently switched to Audacity (it's free and available at www.audacityteam.org). So far we've recorded up to 18 channels and some stereo mixes with it, without problems. Using a plugin that simply drops into the Audacity folder, we can also export the recorded files as MP3s that any client can use.

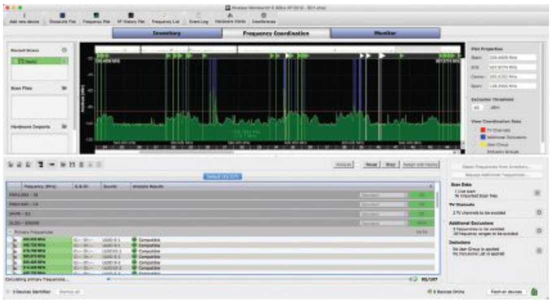
If the client has provided us with tracks for walk-in and/or walk-out music, we can set up a playlist in Sports Sound Pro. If not, we have a ton of walk-in/out playlists already set up in iTunes (about 10,000 songs total that can be played from iTunes or Sports Sounds Pro). While we're not DJs, corporate clients expect us to have walk-in/out music as well as to be able to play back tracks on cue.

Our generic "go-to" music in this application is usually instrumental smooth jazz, but we also offer various themes, including "Rat Pack" tracks as I'm from Vegas and work here a bunch. This playlist has standards from Frank Sinatra, Dean Martin, Sammy Davis, Jr., as well as other crooners like Tony Bennett and Michael Bublé.

We also carry a lot of "royalty free" instrumental music that can be used for backgrounds and walkups (where a person is introduced and music plays while they walk to the stage). Royalty-free tracks have been paid for and can be used over and over without paying additional royalties or license fees. However, be sure to check the contract and terms of the specific royalty-free service you're working with, as some place limitations on usage.

Normally the client or venue is responsible for paying any licensing fees for recorded or live music, but if we find that a license has not been obtained or the customer intends to broadcast and/or record the event even if a local license has been obtained, we offer to use our collection of royalty-free tracks.

Our track collection also includes royalty-free sound effects that have come in handy over the years both at corporate events and also for theatrical productions. And no audio track folder is complete without a few generic announcements. I have announcements using my own voice, as well as other male and female speakers, stating things like, "Ladies and gentlemen, please find your seats the program will begin soon" and "Please silence your mobile devices." These have come in handy numerous times over the years. LSI



Shure Wireless Workbench, a source to manage and monitor wireless systems as well as coordinate frequencies.

an unlimited number of coordinated frequencies. (The basic version is limited to 30 coordinated frequencies.) In addition, the pro version provides both pre-built wireless models and the ability to create user-defined models.

Another excellent (and free) program is Shure Wireless Workbench, which works in concert with many of the company's wireless systems, including Axient, ULX-D, QLX-D, UHF-R, and PSM 1000 personal monitor systems. It provides management and monitoring of connected wireless systems, and it can also coordinate frequencies for non-connected systems from both Shure and others. We also utilize free Sennheiser SIFM intermodulation calculation software, and it too works quite well.

TYPICAL & VALUABLE

Other applications of our computers aren't specifically audio related but nonetheless handy at shows. Most frequently used is Microsoft Office, specifically Word and PowerPoint, the latter having "saved the day" after a client's computer with a key PowerPoint presentation has gone south. Word is convenient for typing up show notes, making signs, and of course, writing articles for this publication during down times.

Yet another computer aspect has also helped/saved clients at a number of shows: a video/media player. We keep several on our computers, and I gravitate toward VLC media player (www.videolan.org). It's a free, open-source, cross-platform format that plays most multimedia files as well as DVDs, audio CDs, VCDs, and various streaming protocols.

Finally, our computers provide web access. Being able to download songs, check equipment manuals, download firmware updates and check email might seem mundane, but it's important nonetheless. Plus we can quickly check our upcoming business calendar, schedule the next show, and invoice clients (literally) right after the gig. And that's somewhat the point, right? LSI

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

DO YOU HEAR WHAT I HEAR?

Problems and solutions in the quest for quality mixes.

by Mike Sokol

Many live mixes are not only too loud, but also sonically unbalanced. I got to thinking about this (again) when recently re-reading "How Loud Does It Need To Be?" by Peter Janis of Radial Engineering on ProSoundWeb.

Yes, I'm saying it right here: far too many live mixes are just plain bad. That is, too much bass, too many highs, and dynamics out of control. Add this to SPL in excess of 110 to 115 dB and it's not only painful, it's usually a poor version of the intended mix of the song.

Why does this happen? Isn't there some meter or app that will tell us when the mix is dialed in? And don't those digital mixers mix the band for us automatically? No and No. The final music mix is solely decided by one person: the front of house engineer (and sometimes the producer standing beside him). That mix is filtered by their ears and listening experience. As Peter alludes to, these unbalanced mixes are caused not only by hearing damage from years of just plain too loud, but also by our brain's auto-balance control of the spectrum of the mix.

I'd like to offer one more possible reason for this problem, along with a solution. Just like you are what we eat, we mix like we listen. And if we listen badly, well, we're going to pay the price.

WANT FRIES WITH THAT?

When I was a young musician in the 1960s, I didn't own a decent hi-fi system, so it was difficult to find out what songs really sounded like in the studio. My inexpensive record player with a 3 x 5-inch mono loudspeaker was my initial schooling in what Jimi Hendrix, Led Zeppelin and Santana sounded like. Because it was "all mid range all the time" with little in the way of lows or highs (think whizzer cone on the loudspeaker), that's pretty much what our bands sounded like.

Further, my first PA, a Kustom 100, had four 10-inch cones per cabinet but really only carried vocals. But it sounded just fine to us (at least most of us) at the time since we'd never heard a real full-range system. What we were listening to set our opinion on what sounded acceptable.

I call this the cheap cheeseburger effect — if you only eat fast food, then fast food becomes acceptable. But eat a really great steak at a chophouse and all of the sudden that burger doesn't taste quite as good the next time.

MIX LIKE YOU LISTEN

Just as there's no "suck knob" on the console, there's also no "great mix knob." That control is located inside of our heads, not on the physical mixing console. And just as we train our taste buds as to what *tastes* great, so we train our ears as to what sounds great. Listen to a lot of unbalanced sound systems with badly mixed music and that's what we'll end up mixing. Listen to great mixes on a well-balanced system and we begin to really understand how the Grammy Award-winning folks do it.

This is not complicated. It begins by listening to and analyzing great mixes on great sound systems on a regular basis. (Yes, listening to great music on a great sound system is your homework assignment. You may thank me now...)

BAD APPLE

A few years ago I was teaching a live sound mixing seminar at a big church in Texas. One of the students informed me of how crappy my mixes sounded. Rather than get defensive, I asked him what he thought was wrong. First he added that nearly *all* mixes for *all* artists on *all* albums were pretty much terrible, and then went on to describe the sound system he was fortunate enough to have at his church. It included a lot of quality pro-level loudspeakers and amplifiers, as well as a ton of subwoofers. (Probably twice as many subs needed for even heavy bass-oriented music, but anyway...)

He noted that even when playing acknowledged top-quality mixes on this system, they sounded "wrong." He'd contacted tech support at the loudspeaker manufacturer who (I gathered, reading between the lines) politely informed him that he was nuts. Engineers visiting his church told him likewise, as did the congregation and band.

And I was getting ready to tell him he was nuts until I thought to ask one simple question: "What's your home stereo system?" That is, what's the reference system he listens to on a regular basis? He replied that he didn't have a home system at all, but that his car had a really nice one — including a pair of 18-inch subs in band-pass cabinets in the trunk driven by a few thousand watts of amplification.

Well now, I thought, there's your problem! This was his mix reference. He'd taught his own mind and ears that all music mixes were supposed to sound like one-note thumps that hurt. And that's what he was doing in his own church, tuning his praise team mixes so they sounded like an over-amped car stereo with band-passed woofers.

There's no "great mix knob" on the console — that control is located inside of our heads.

KIDS SAY THE DARNDEST THINGS

A few years later my own twin sons came to a realization that, while simple on its face, I thought was profound. I have a quality M&K surround system in my living room, an NHT Pro monitoring system in my home studio, a custom playback system in my vehicle with a 100-watt sub and Alpine 2-way loudspeakers in the dash, and Tannoy Reveal loudspeakers with a Polk subwoofer in the kids' gaming room.

One day, out of the blue, my sons came to me and said, "Dad, we just realized that that all of your sound systems sound the same. You did that on purpose, didn't you?" And they were right. I'd carefully balanced the sub crossover points and relative levels on all of these listening systems so that they sounded very similar.

Further, I don't add bass or highs via the various tone controls on a song-by-song basis. That is, no smiley face EQ, no thump at 30 Hz. Nope, I run pink noise through everything and get it as flat as reasonably possible, then stick with it. My reason is straightforward: I don't want to un-calibrate my hearing by listening to bass-heavy or high-heavy or whatever-heavy mixes. So when I'm mixing live I now have an internal reference of what things are supposed to sound like. This also means I can balance a live sound system on the fly without having to impose pink noise on a cocktail crowd or congregation. I've *learned* via consistent quality listening what it's supposed to sound like.

MAKING SPACE

Circling back to Peter's article, louder is not necessarily better. Once stage volume is under control, it's not brain surgery to mix at a reasonable 90 to 95 dB SPL (A-Slow) and hear every detail. There's no need to rip peoples' heads off with the vocals, because the back line is under control, and there's no need to hurt them with bass because the mids and highs are in balance.

Note that my practice is to utilize a lot of loudspeakers and amplifiers whenever possible (and appropriate to the gig) because I want the one thing that's missing in so many of the mixes: headroom. Yes, it's cool to get the impact of the kick, and the crash of a cymbal that's not hitting the limiters can generate much happiness. And warm and fuzzy bass can make everyone feel, well, warm and fuzzy.

I don't want to un-calibrate my hearing by listening to bass-heavy or high-heavy or whatever-heavy mixes.

So I encourage you try it. Get thee a decent hi-fi system and buy the CD box sets of your favorite bands of your youth (as well as the present). Make sure there's no EQ on the system, then listen to the genius of Hendrix and the musicality of Miles, the vocal control of Norah Jones and Adele (or whoever you like). If you prefer Nine Inch Nails, I'm not judging because I happen to *like* Nine Inch Nails and think that Trent Reznor is a pretty savvy producer. Pick what moves you and just listen to the balance of the mix.

After getting used to it, mixing at 95 dB in a room is a lot more musical and pleasant than 110 to 115 dB. Done correctly, the band will love it, the audience will love it, and you'll love it. Plus you won't go deaf. LSI

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STEP-BY-STEP RF

A logical approach for tracking down common wireless problems.

by Ike Zimbel

For many in pro audio, working with wireless systems is still thought to be a "nightmare" filled with peril, an accident waiting to happen, and so on. In actual fact wireless systems are meant to work and do work every day in thousands of concerts, theatrical productions, festivals, churches, broadcasts and the like, all around the world.

With that in mind, I've compiled this list of troubleshooting steps to take when encountering wireless problems in the field. Hopefully it will alleviate the apprehension many experience while also solving the problem.

Step 1. Develop a positive attitude. As noted, RF gear is supposed to work. Understanding and believing that — instead of believing that it's nothing but trouble — is critical to resolving issues when they arise.

Step 2. The first question to ask when encountering RF problems at an event: Is this system coordinated? It's especially true if there are problems with multiple systems at the same time. I've been doing frequency coordination since the early 1990s, and in all that time I've had barely enough problems to come up with material for this article. Frequency coordination is no longer an option — it's an essential.

Ideally, frequency coordination should be done before the equipment rolls in but in the event that it hasn't been done, here are some things to do to improve things:

- Someone must take charge and get all RF users on the same page.
- Compile a list of all wireless equipment and frequency ranges in use.
- Doing rudimentary band planning and doling out a chunk of spectrum to each user can help. For example, the two backline techs on a gig both have Sennheiser "B" (626 to 668 MHz) band systems. A quick scan with one of the receivers shows DTV channels 40, 41 and 44 are present, which takes up 626 to 638 MHz and 650 to 656 MHz. Solution: tech number 1 gets 639 to 650 MHz and tech number 2 gets 657 to 668 MHz. It's not frequency coordination, but it does help eliminate the "every man for himself" mentality and brings some order to a chaotic situation.

If you think a receiver is truly getting interference, the one sure way to tell is to turn off its associated transmitter.

Another approach, in the absence of a real coordination, is sequential scanning. It works as follows:

- Turn on "money channel" systems and scan for the best frequency available.
- Place these transmitters on stands (don't leave them in a pile on a road case) and leave them on.
- Then scan the next most important systems, place them where they'll be used and leave them on, and then repeat with each successive system until all are turned on. Again, this is a long way from a proper coordination, but it's better than nothing.

One last point, if there is frequency coordination, make sure that everyone is on the same coordination. I've noticed that now that coordination is becoming more common, a lot of rental houses are sending a coordination report out with RF rentals. This is great if they're providing the whole equipment package, but if, say, the mics and IEMs are from one company, the backline from another, and the RF intercom from a third, and each company provides a coordination report with its rental, there's a very good chance that all three coordinations will be null-and-void.

Step 3. A key question to ask yourself about a potential problem: Is this real or a red herring? For example, a vocal channel that has been rock-solid all day is suddenly taking hits a half hour before show time. The natural tendency is to look for another frequency, but before doing that, an important thing to determine is "where is the transmitter right now?"

If it's out on stage, on it's stand, there may be a real issue. However, if it's not in the stage area because it's been taken down the hall to the artist's dressing room, what you're probably seeing is the receiver trying really hard to keep receiving its transmitter. Modern RF gear is designed to be very selective about what signals will actually cause it to "open up" and pass an audio signal to its output, so

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it pays to remember that the one thing that will do that without fail is the transmitter that has been programmed to the same frequency.

Also keep in mind is that if you do change the frequency, the problem will *appear* to be fixed, but really all you're doing is setting the receiver to a frequency that it doesn't see its own carrier on. Then, when the transmitter is brought back to the stage to sync to the new frequency, the problem is "solved" because the transmitter is back within range of the receiver.

Step 4. If a problem crops up after all has been fine for some time, ask "what's changed?" At a recent festival gig, for example, I arrived on site and was told by

On two occasions I've had walkie talkie signals get into open receiver channels.

the monitor engineer that he was taking hits on one of the eight vocal mics… on day nine of a 12-day gig. Looking into the problem, I could see that the receiver was taking hits with the transmitter turned off. I checked that frequency on two spectrum analyzers but couldn't see anything that would be causing interference, and also evaluated several other things, including a tour around the site to see if any unauthorized news crews were roaming around.

Finding nothing obviously wrong, I turned my attention to the antennas, a pair of the new Shure UA874XA paddles with the gain/attenuation switch. As soon as I walked up to the first one I could see that the RF overload LED was on continuously, and a glance at the "B" antenna showed the same light flashing.

These had been set to 0 dB on set-up day, in consultation with the monitor engineer; however, it turned out that he'd switched in 6 dB of gain on both antennas the night before in an attempt to better pick up the host, who'd been running all over the site with his transmitter. Reverting to 0 dB of gain immediately corrected the overload issue, while the hits were in fact intermod products being generated by the overdriven amplifiers in the antennas. (Think of this situation as an "RF fuzz box"— the overdrive creates additional harmonics.)

Step 5. Listen! Spectrum analyzers are a useful tool, but I find a lot of time wasted in staring at them when our ears can tell us the real problem. Examples:

▷ On two occasions I've had walkie talkie signals get into open receiver channels. In both cases I was finally able to track down the source by listening to the affected receivers, noting a bit of what was said and then asking around "Who just said...?" In one case it was the walkie talkie-based intercom system that the TV crew was using (resolved by moving the Tx antenna off the arena floor into one of the voms), and in the other, the source was the 20-watt repeater from the arena emergency response system. In the latter instance, the signal was only getting into one channel that had its transmitter turned off during sound check, something that wouldn't be the case during the show.

▷ I once received a panic call during a changeover, with the issue being that one of the vocal mic channels sounded "gated." A quick listen at the headphone jack on the front of the receiver told me that the problem was not in the RF domain (i.e., not a frequency problem). Note that this didn't rule out an actual electronic fault in the output of the receiver (it ultimately turned out to be a stage box issue), but it did prove that changing the frequency would have been a waste of valuable troubleshooting time.

Step 6. Turn off transmitters. If you think a receiver is truly getting interference, the one sure way to tell is to turn off its associated transmitter. If the receiver goes dark (i.e., no RF level indication) and quiet with the Tx off, it's probably not straight up RF interference.

That said, the problem could still be other issues mentioned above, such as overloading the front end of either the antennas (if they're active) or the receiver. Similarly, if you turn off the IEM transmitters and the mic receivers clean up, the problem could be something along the lines of the Tx antenna being too close to one or both of the receiver antennas.

Step 7. Keep in mind that RF gear is gear, just like mixing consoles, power amplifiers, loudspeakers, etc. — and sometimes gear breaks down. In the past three weeks alone, I've had two different RF intercom systems that were sending intermittent crackling down the comm line. RF interference, right? No, it was loose wiring inside the base stations.

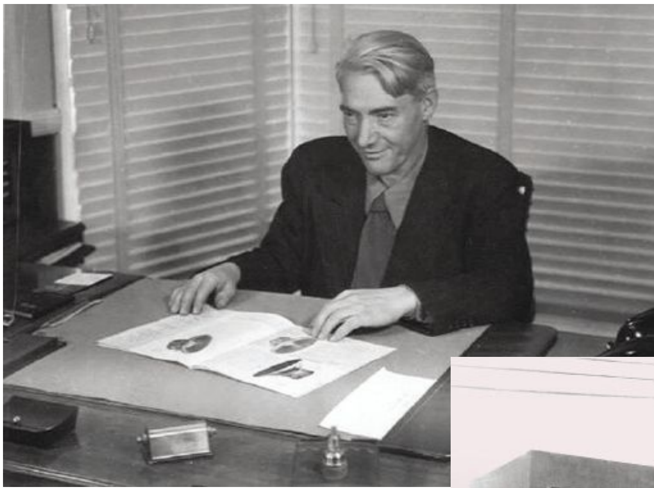
Finally, when someone complains of intermittent noise on IEMs, especially at a rehearsal or sound check, the first thing to ask them before going into full troubleshooting mode is: Where's your cell phone? It's amazing how often they just happened to be getting a flurry of text messages when the "interference" cropped up. LSI

***Ike Zimbel** is a wireless frequency coordinator and tech based in Toronto. You can find him on LinkedIn.*

JBL AT 70

Key product and technological milestones along the way.

by Live Sound Staff



James B. Lansing and early headquarters facilities of JBL in Southern California..

Ever heard of "James B. Lansing Sound, Incorporated"? No? Perhaps you know it by the more familiar name of JBL. What Mr. Lansing, a trailblazing loudspeaker engineer/designer, started in 1946 in what was then much more sparsely populated Southern California has led to quite a legacy in professional audio.

October 1, 2016 saw the official 70th anniversary of JBL, and to mark that occasion, let's take a look back at some of the company's landmark products and technologies that helped revolutionized the tour sound industry, as well as some of renowned concerts and festivals they've been a part of.

Long before there were touring sound systems, James Lansing developed components that allowed early audio pros to create unique loudspeaker systems for a variety of live sound applications. First up, released in 1947, was the D130, a 15-inch cone driver. In order to achieve the sensitivity, range and dynamic response that Lansing was looking for, he incorporated then state-of-the-art Alnico V magnets, a cast frame, and a 4-inch flat-wire edge-wound voice coil attached to a curvilinear cone and aluminum center dome. The distinctive sound and exceptional power handling for the time made it a prime component for early sound reinforcement as well as for musical instrument amplifiers from companies like Fender.

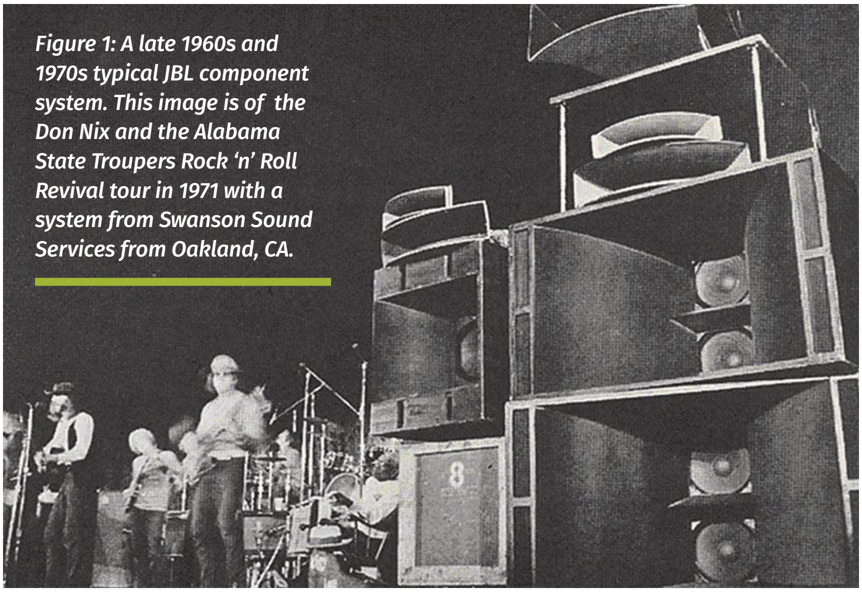


Figure 1: A late 1960s and 1970s typical JBL component system. This image is of the Don Nix and the Alabama State Troupers Rock 'n' Roll Revival tour in 1971 with a system from Swanson Sound Services from Oakland, CA.

By the early 1950s, JBL had unveiled the 2440, a 4-inch-diaphragm compression driver that became an industry standard. It too incorporated an Alnico V magnetic structure (weighing more than 23 pounds) as well as a 4-inch voice coil and aluminum diaphragm that helped it achieve very high levels with accuracy.

MORE PIECES OF THE PUZZLE

Another JBL innovation from the 50s was a series of acoustic lenses. Originally designed for stereophonic audio systems for motion picture theater applications, these components allowed early tour sound professionals to achieve specific high-frequency distribution patterns in different applications.

Looking for a way to better reproduce the dynamic range, the market also turned to other JBL components of the same era like the 075 ultra-high-frequency ring radiator. Introduced in 1955, the 075 "bullet tweeter" was implemented in the emerging high-end high-fidelity

Figure 2: The legendary Woodstock in 1969 utilizing Bill Hanley's custom 4 x 15-inch horn-reflex woofer bins with multi-cell high-frequency horns. Photo courtesy John K. Chester, Fillmore East chief sound engineer and one of the technical staff members that came upstate to work the festival.

Figure 3: Cal Jam II in 1978 at Ontario Speedway in California, with the TFA Electrosound (later Electrotec) five-way all-JBL-component horn-loaded PA. The first Cal Jam was in 1974 at the same location.

market, and it was also adapted for use in larger pro audio systems.

Later, the company's 4550 dual-driver and 4560 single-driver bass horns (both 15-inch loaded) arrived on the scene, capable of handling high output and delivering powerful low-frequency response. As the market for larger concerts emerged (as did the demand for sound reinforcement to support them), audio professionals would build proprietary systems to suit unique applications. A typical component tour sound system from the late 1960s and early 1970s would often incorporate 2220 woofers, 2440 and 2410 compression drivers, 2345 and 2350 radial horns and lenses, and 4550 and 4560 bass bins (**Figure 1**).

Early systems with JBL components were regularly used with many prestigious tours and stages, including the most iconic music event of them all, the original Woodstock in 1969. Sound reinforcement for the festival came in the form of Bill Hanley's custom 4 x 15-inch horn-reflex woofer bins with multi-cell high-frequency horns. The lower systems were loaded with JBL D140F 15-inch bass drivers while the upper "long-throw" systems had 15-inch D130s (**Figure 2**).

Following in the footsteps of Woodstock, the Cal Jam II Festival — a 12-hour rock concert produced by the ABC network and Pacific Presentations — used a five-way PA loaded with all JBL components that was known as the Electrosound or Electrotec system. This system was powerful, with the company stating that it was measured at 105 dB SPL at a distance of one mile. It's also one of the early systems to stack loudspeaker elements in vertical lines (**Figure 3**).

A significant leap forward came in 1975 when Clair Brothers produced the S4 single-box system, a loudspeaker housing 10 JBL transducer components: two 18-inch and four 10-inch woofers in vertical line configurations along with two horns and drivers and two compression tweeters. This "all-in-one" four-way

Figure 4: The first US Festival, Labor Day weekend 1982 in Southern California, prior to the show. With the loudspeaker scrims down, the Clair S4 systems deployed for the event can be viewed.

Figure 5: A closer look at the modular Clair S4, each loudspeaker housing 10 JBL transducer components.

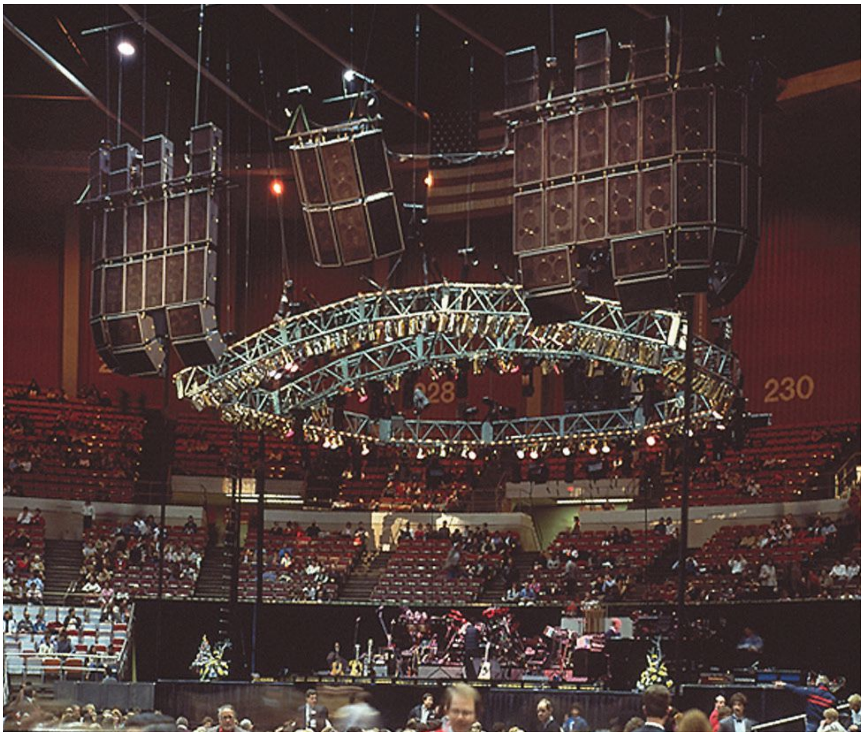


Figure 6: The 1985 Neil Diamond tour with JBL Concert Series. Long-time Diamond mixer Stan Miller and his Stanal Sound company began using JBL in the 1980s, and constructed the Concert Series enclosures for JBL.

touring box was the first of its kind, and allowed production teams to unload and fly PA elements in a way that was superior to the earlier individual system-element component approach. The modular nature of the system also allowed building arrays in different shapes and configurations to better suit individual venues.

The S4 system remained an industry standard for years, deployed at notable events such as the Southern California US Festival in 1982. A total of 180 S4s were stacked in 10 columns of nine on each side of the stage, with an additional row of long-throw horns on top (**Figures 4 and 5**).

DYNAMIC CHANGES

The mid-to-late 1970s brought about a shift of focus. Instead of predominately producing separate components, enclosures and crossover networks, the company established itself as a manufacturer of complete loudspeaker systems. This shift was marked by the 4300 Series of three-way and four-way studio monitors.

Driven by the high-octane rock of the 70s, the 4350 monitor became popular with artists and engineers looking to monitor playback at very high levels. The 4350 was the highest output monitor JBL produced, stated as being capable of a sustained output of more than 125 dB. The Who installed four units in their control room for quadrophonic playback in order to simulate the intensity of their live shows.

Though designed as a studio monitor, the 4350 also gained the attention of touring acts such as the Allman Brothers, who would stack multiple monitors on each side of the stage for smaller venues. It was loaded with dual 15-inch woofers, a 12-inch cone for midrange, a 4-inch voice coil compression driver and horn/lens, and an ultra-high frequency tweeter. After a number of deployments, the 4350 and other large monitors became JBL's first fully accepted professional sound system designs, rather than the previous component part systems.

The Concert Series came along in 1985, the company's first all-in-one tour system (**Figure 6**). It consisted of full-range trapezoidal enclosures and associated subwoofers, with notable component technology including flat-front Bi-Radial horns, extended range compression drivers, and SFG (Symmetrical Field Geometry) cone transducers.

MOVING FORWARD

As the touring market saw the advantages of line arrays, JBL stepped up with the VerTec Series (**Figure 7**). Implementing patented Differential Drive transducers, beryllium-diaphragm compression drivers, the patented RBI-Radiation Boundary Integrator for mids and highs, and a new structural design, it was noted for fidelity and a more lightweight way to suspend long arrays. VerTec was also the first

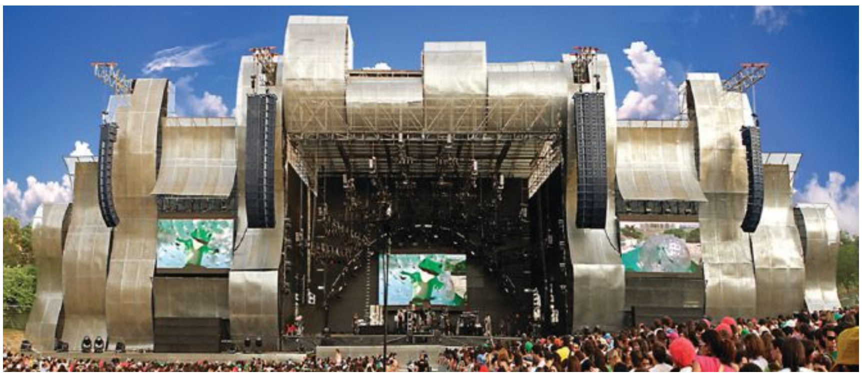


Figure 7: Rock in Rio Portugal 2006 with large hangs VerTec line arrays provided by Gabisom of Brazil.

Figure 8: Thirty years later in 2015, Stan Miller is still mixing Neil Diamond, now with VTX line arrays provided by Sound Image.

system to integrate with sister company Crown Audio's I-Tech HD amplifiers.

In addition, VerTec provided a way to observe estimated directional response and build systems according to needs with a new design program. The series was deployed at a variety of events, including the 2000 Democratic convention, 2002 Super Bowl, 2002 Grammys, and a host of international tours.

The latest JBL tour sound milestone is the VTX Series of line arrays (**Figure 8**). Supported by multiple patents in driver, waveguide and suspension technology, the series is headed by VTX V25 dual 15-inch and V20 dual 10-inch three-way array elements. As with VerTec, VTX integrates with other Harman brand products such as Crown amplifiers as well as dbx and BSS signal processing, optimized to enhance sonic performance as well as system management.

Moving forward, it will be quite interesting to see what's next from the then-fledgling operation established by James Lansing that has gone on to have significant and lasting impact in the world of pro audio.

Editor's Note: For more, check out ProSoundWeb for "James B. Lansing & The Creation Of JBL" by John Eargle as well as various RE/P articles providing greater detail on the systems noted here. LSI

ALLEN & HEATH dLIVE

The backstory on the development of a new digital mixing platform.

by Rob Clark & Nicola Beretta

The three control surfaces available in the dLive platform. Clockwise from the top are the S7000, S5000, and S3000.

dLive is a step change for Allen & Heath in digital mixing, built on a new technology platform: the XCVI core, which incorporates the DEEP processing suite of embedded plugins. dLive also provides an intuitive workflow due to the integration of touch screens and hands-on controls.

XCVI

At the heart of dLive's design is a powerful processing engine, the XCVI core. It was developed by the Allen & Heath R&D team using next-generation FPGA technology and is capable of handling 128 full processing input channels into 64 mixes (all at 96 kHz sampling rate), along with 16 FX engines and processing extensions to accommodate a new suite of algorithms. This represents a considerable increase in signal processing compared to our previous flagship system.

At the beginning of the project, it was clear that we couldn't achieve this goal with traditional DSP technology. Some of the limitations of DSPs are the actual I/O on the chip, the fixed number of cycles, and consequential problems with routing and latency management. There wasn't really a processor on the market that could do everything we wanted, so the move to FPGA was the obvious choice.

The key challenge was how to implement all of the processing algorithms that we wanted to maintain from the iLive and GLD systems on FPGA technology, while also accommodating the new DEEP processing initiative. The team responded by designing a custom processor architecture on FPGA with 36 parallel virtual processing cores. These cores appear as mini DSPs to the DSP engineer, and thus are capable of running existing algorithms as well as providing facilities to develop much more powerful ones. Within the same XCVI processor, six parallel mixing engines calculate more than 10,000 cross points per sample, while the router is capable of 3,000 x 3,000 audio paths.

Aside from this power, the XCVI core also has what we consider to be tangible sound quality benefits. When seeking sonic performance upgrades, it's important to study the bit widths in all parts of the algorithm. As previously noted, the core operates at 96 kHz sampling rate and exploits variable bit depth with much higher numbers of bits where it matters (up to 96-bit in the accumulator) for added precision and noise performance.

Another key benefit is mix coherency. In the analog world, mix engineers could take any route, any path inside a mixer, and the outputs would all be phase aligned. Looking at the digital desks on the market, we found that only two or three are phase coherent on the outputs, and normally it's at the expense of a high system latency. Due to the XCVI core, internal mix paths (such as input to mix, input to group to mix, etc.) are aligned on the output down to the sample. Coupled with very low latency

There's also a choice of (left to right) DM64, DM48 and DM32 MixRacks.

data converters, this has resulted in just 0.7 milliseconds of system latency, analog in to analog out.

DEEP PROCESSING

Allen & Heath DSP engineers are passionate about modeling actual gear and emulating classic modules, FX units, and processors. This obsession is now taken a step further with DEEP processing. The architecture embeds leading processing emulations directly within dLive's input and mix channels. An array of bespoke algorithms, including graphic EQ, compressors, and preamp models faithfully emulate and sometimes "reinvent" popular industry classics.

Use of dynamic or EQ modeling across multiple channels typically requires external, dedicated plugin systems and interfaces, with the associated cost, complexity and hassle. The integration of the plugins in the channel processing provides instant access and comes with no extra latency. Users can pick their favorite model on the fly, without burning FX slots.

The compressor models capture the nuances and non-linear ballistics of industry preferences. For example, the 16T is an accurate model of the ubiquitous engineers' favorite, while the 16VU models the vintage original. Opto emulates a range of opto compressors but with the extra flexibility of faster attack and recovery. All models come with a parallel compression dry path and a dual side-chain filter.

We stopped recreating the experience of using an analog mixer and instead designed dLive as a true digital native.

Four graphic equalizer models are available on all 64 mix outputs. One-third-octave (1/3-octave) "Constant-Q" and "Proportional-Q" are modeled on the American and British industry standards, while asymmetrical "Hybrid" offers a combination and "Digi-Q" is designed for minimal band interaction.

The first of a series of preamp models, Dual-Stage Valve, recreates the distortion characteristics of valve circuits. Stage-1 adds subtle tonal harmonic distortion and compression, and Stage-2 provides either "break up" distortion or continuous overdrive, both utilizing Triode and Pentode valve models.

HARMONY UI

Throughout the design process, the aim has been to create workflows that allow the engineer to focus on the mix, not the mixer. The dLive layout is customizable, allowing users to create an interface that matches their own mental "maps" of the show. Every input or mix can be assigned to any and every bank

Director, the multi-platform editor and control software for dLive.

and/or layer, virtual scribble strips allow inputs and mixes to be clearly named and color coded for "at-a-glance" navigation, and there are 26 programmable SoftKeys plus three pages of six assignable rotaries per screen.

We designed dLive as a true digital native, drawing on familiarity with the smartphones and tablets that we all use daily without thinking. The single or twin 12-inch capacitive touch screens are designed to feel familiar and to be responsive to every pinch, swipe, drag, and drop. Bespoke "widget" areas can also be set up on the screens to keep track of scenes, meters, FX, and other custom controls.

The M-DL-ADAPT interface fits in any dLive I/O port and adds compatibility with most iLive/GLD networking cards, including Dante, Waves, ACE, MADI and more.

Screens are framed by a set of rotary controls intended to enhance creativity and to provide immediacy of tactile control over key processing functions, working in tandem with the visual feedback displayed on the screens. The rotary knobs have been carefully prototyped with the goal of optimal grip and precision control, working with RGB illumination (with colors mapped to functions) for instant visual orientation.

EXPANSION, INTERFACING, NETWORKING

Inter-system and network connectivity is available through five 128-channel I/O ports, giving access to more than 800 inputs and outputs for a range of networking cards, including Dante, Waves SG, ACE, and MADI. With these I/O slots (three in the MixRack and two in the surface), there is flexibility for audio network bridging and interfacing different digital standards to the mixing system. Further remote IO expansion is provided at both the MixRack and control surface through dual-redundant Cat-5 DX ports, which allow connection of up to three of the

DX32 expander units.

Another key area of focus by the R&D team is obtaining an appropriate balance between strength/durability and weight. All surfaces, MixRacks and expanders have dual power supply slots for redundancy as well as the same rugged, hot-swappable PSU design. Dual redundancy is also built into every audio connection throughout the system. Further, control surface illumination faced rigorous trials to deliver optimum performance in both dark environments and bright sunlight.

MIX 'N" MATCH

We followed the same ethos as the iLive range in choosing a distributed system design for dLive, with separate MixRack and control surface. The "processing brain" is housed in the MixRack, all with configurable mix bus, groups, mains and matrixes.

Market research confirmed three sizes, so racks are available as the DM32, DM48 and DM64, with the additional DX32 expansion rack, as well as the S3000, S5000 and S7000 control surfaces. All MixRacks and surfaces are mix 'n' match compatible, with common configuration, set up and show files. MixRacks can also be used stand-alone with the Director control software on Mac and Windows. Finally, there's also a remote app for iOS platforms that provides comprehensive wireless setup control.

The IP8 controller for dLive, programmed using a dLive surface or Director software. It interfaces via standard TCP/IP network connections and can be networked with other controllers, computers and third party devices using a standard Ethernet infrastructure.

With dLive, we believe we've created a powerful engine and user interface capable of forming a flexible mix and control "ecosystem" with the capacity to handle future software enhancements in the months and years to come. LSI

***Rob Clark** is technical director and Nicola Beretta is product manager for Allen & Heath.*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THE RIGHT FIT

A variety of technology deployments in diverse circumstances.

by Live Sound Staff

PHOTOGRAPHY BY STEFAN M. PRAGER

A Meyer Sound LEOPARD array flying in service of Passenger on tour.

ATTENTION TO DETAIL WITH PASSENGER

English folk balladeer Passenger, on tour in support of his new album, is utilizing a main system made up of members of the Meyer Sound LEO Family supplied by UK-based sound company Major Tom. The majority of shows are in large auditoriums and smaller arenas, where the system employs main hangs of 34 LEOPARD line array loudspeakers with bass support from eight 1100-LFC low-frequency control elements.

For the largest venue of the tour, Amsterdam's 17,000-capacity Ziggo Dome, the main hangs comprised 12 LEO and six LYON loudspeakers, with the LEOPARD loudspeakers re-configured as side hangs. Completing the roster under the direction of system tech Ian Hamilton are four M'elodie array loudspeakers, four JM-1Ps, four UPA-1Ps, and two Galileo processors.

Front of house engineer and production manager Simon Kemp, who has worked with Passenger for the past two years, states that he chose LEOPARD in part because of its attention to detail. "My show is all about hearing every phrase Passenger sings. His fans are there to hear his storytelling and anything short of perfect clarity won't be accepted," he explains.

In addition, LEOPARD provided him with the necessary sonic impact while still being a manageable size: "With LEOPARD I can achieve amazing audio coverage and maintain a 100 dB average in 5,000 capacity rooms and my PA system fits in 17 feet of truck space. I'm not aware of anything else that comes close to that accomplishment."

UP AND RUNNING (& MIXING) IN BEVERLY HILLS

The Encore Event Technologies Beverly Hills (CA) location recently deployed a Yamaha RIVAGE PM10 digital console at the Beverly Hilton Hotel's International Ballroom, which hosts more 300 high-profile events per year ranging from televised awards shows like the Golden Globes to concerts by top artists to theatrical productions and more. The console was acquired from Chicago-area sound company TC Furlong.

"Both the Beverly Hilton and Encore treat this room as a theatrical venue and not as a typical hotel ballroom with back-to-back shows all year long," explains Rachel Wolfe, director of event technologies for Encore. "In any given week, we could have a Broadway review on Monday, Diana Ross on Tuesday, a ballet on Wednesday, Ozzy Osbourne on Thursday, and a

Mix engineers Jack Hayback (left) and Clint Rowland with the new Yamaha RIVAGE PM10 at the Beverly Hills Hilton ballroom.

televised awards show on Friday, so we need a console that can handle this demand. The PM10 has shown itself to be able to do so gracefully.

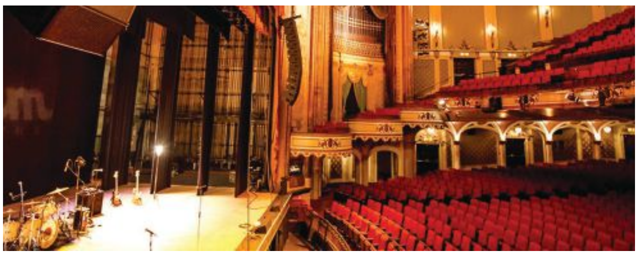
"Our clientele are extremely production savvy; we deal with touring road managers and engineers, film industry line producers, television producers and directors, etc.," she continues. "They want to know that we have the top of the line gear available and that we can pull off their event flawlessly."

Jack Hayback, freelance front of house engineer for Encore, adds, "I've had a lengthy history mixing on Yamaha consoles. Together with help from the folks at TC Furlong and freelance engineers Clint Rowland and Serguei Soloviev, we installed the console and built several templates for the space.

"Although, I was slightly intimidated at the prospect of getting to know another digital console, the PM10 is remarkably easy to use," he continues. "I committed a lot of energy to educating myself on it, but as soon as I started using the console, it became second nature. Literally, within a few days, I knew everything I needed to know. With a lot of great features, the PM10 is very definitely a Yamaha console."

BRINGING AN UPGRADE TO A CLASSIC VENUE

The Orpheum Theatre on Broadway in downtown LA was recently outfitted with a new house sound reinforcement system installed by Audio West (Placentia, CA) that's headed by d&b audiotechnik components. The venue enjoys a long, rich heritage and remains busy these days in hosting TV and special



A perspective of the Orpheum Theatre in LA with the new d&b audiotechnik arrays in place.

events as well as touring acts such as Maroon 5, The Who, Naz, and Adam Lambert.

"I do a lot a touring work and have used many of the current systems; I knew d&b would be perfect for this room for various reasons," notes Jon Bullock, the venue's head of audio. "I still called and asked visiting engineers for opinions; they all came back and said d&b would be great, and certainly more than acceptable to any touring band."

Audio West owner Glenn Hatch adds, "We had already done work with the Orpheum, providing rental systems for certain shows. Although we were both very familiar with how the room worked acoustically, Jon and I did an ArrayCalc model and we settled on a V-Series specification: V8s and V12s supported by V-SUBs and J-INFRAs for some extended low-end weight."

Bullock continues, "With the help of SAS Productions audio engineer Mario Rodriguez, there was some extensive rewiring needed in order to use our existing patch panels and runs from the stage to the amp room. We went from a 4-way active crossover system to a passively crossed system. Because of this, a good bit of the work was in the amp room. We went from 20 amplifiers running 20 speakers in pairs to nine amps running 32 speakers individually. With the (d&b) D20 and D80 amps driving the system, things are a lot smaller and neater down there now and I have more than enough horsepower. I am really, really happy with the system."

EFFICIENT AWARDS SHOW WIRELESS IN VEGAS

Orlando-based Professional Wireless Systems (PWS) provided RF coordination for the recent 17th Annual Latin Grammy Awards at T-Mobile Arena in Las Vegas, managing more than 170 RF frequencies of wireless microphones, in-ear monitors and intercoms. To minimize the impact of the already crowded RF bandwidth, PWS deployed five Radio Active Designs UV-1G wireless intercom systems to meet communication needs, primarily by the stage managers and the audio department. The body packs of the UV1-G operate in the VHF range, which helps

The RF team at this year's Latin Grammys. Left to right: RF tech Jason Lambert, PL tech Dave Nichols, PL engineer Tim Kepner, RF PL engineer Gary Trenda, and PWS general manager Jim Van Winkle.

in minimizing UHF band congestion.

"We had 34 RAD packs on five base stations," explains Gary Trenda, RF PL engineer. "Each base station is equipped with six body packs, but we were able to double up on a few of the channels with folks that only needed to use them sporadically."

The five base stations were rack mounted and located backstage. Two zones — the bowl/main stage and back of house/dressing/green rooms — were covered by separate antennas that expanded coverage.

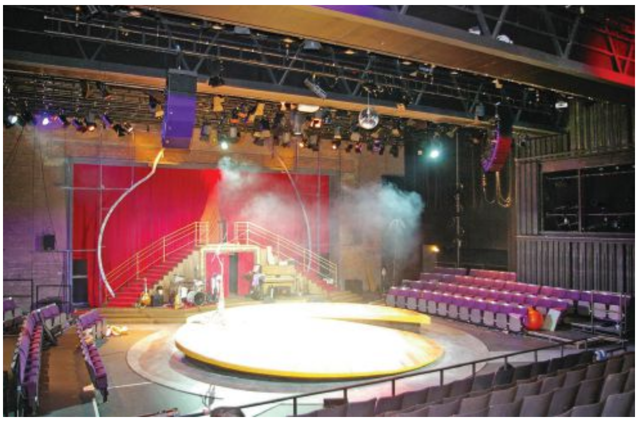
DEVELOPING A SONIC SIGNATURE

Situated in the heart of Newcastle University's main campus in the UK, Northern Stage Theatre offers three performance spaces, with the venue's 450-seat main house recently undergoing a technical infrastructure upgrade under the direction of production manager Chris Durant. He worked closely with locally-based production company Nitelites on a new sound reinforcement system, which includes RCF HDL10-A line arrays serving as mains.

Nitelites director and system designer Andy Magee outlines the specifics of the project: "While the theatre's existing loudspeaker (model) had been a premium system in its day, it was over 10 years old. Its coverage was too wide, it lacked projection and was unable to cover the rear of the room. I knew that with the HDL10-A there would be sufficient energy delivered to the back of the room while maintaining a consistent spectral balance throughout the listening area."

Specifically, two main hangs of five HDL10-A per side are suspended from a bridge truss, while a further pair of HDL10-As, located behind the main arrays, fire down to cover the audience wrapped around the front of the large stage. Low frequencies are extended by two floor-mounted SUB8004 subwoofers, while the system operates under BSS Soundweb control.

Durant notes that he's satisfied with the solution, adding that it has given the venue a "sonic signature." He states, "For a space as wide as this you need projection, and compared with the point source (approach) we had previously, this delivers greater intelligibility with ample bandwidth and directivity, and a more even spread. There is now so much headroom it's ridiculous."



New RCF HDL10-A flying left and right at the main house of the Northern Stage Theatre in the UK.

Lectrosonics wireless was virtually invisible in the production of A Chorus Line at the Stratford Festival. Credit: Stratford Festival

COMPACT THEATRICAL PERFORMANCE WITH AN IMPACT

A recent production of multiple Tony Award-winning musical A Chorus Line as part of Ontario, Canada's Stratford Festival marked the first time that the show has ever been modified; specifically, it was rearranged to take place on a thrust stage with the audience seated on three sides. Sound designer and theatre consultant Peter McBoyle, faced with the prospect of surrounding angles of the production continually in view, chose to deploy Lectrosonics SSM wireless transmitters to help deliver audio while remaining discreetly hidden on the performers.

"This venue is always challenging because of the large thrust stage — at any given time, all the actors are exposed to the audience," McBoyle explains. "A Chorus Line also required more volume with a pop sensibility to it than many shows that I've done at Stratford. And we also needed the smallest transmitter packs possible, as the cast members were wearing audition clothing like unitards and T-shirts.

"We chose the SSM because it really addresses the needs of the theatre market in terms of performance, sound quality and discreet size," he continues. "I beta tested the transmitters when they were introduced last year, and having them perform so well on this production is really proof of concept."

With 19 people in the cast who never leave the stage and no intermission during the show, two transmitters were assigned to each performer for redundancy. In total, more than 50 channels of wireless were used. "In a show like this, the ability to perform properly is at risk of being compromised by a lav mic element getting saturated with sweat or having hair covering it. So we made the decision here to double pack the main cast of performers in case a backup was needed, since they never leave the stage," McBoyle concludes. "As we were confronted with having to hide two transmitters on each performer instead of one, size was a huge priority." LSI

AUDIO-TECHNICA ATM350a

Checking out a new mic system for instruments.

by Craig Leerman

Audio-Technica ATM350a systems include a compact instrument microphone with a range of proprietary UniMount components that facilitate mounting to almost anything. The mic measures just 1.5 inches long and .5 inch wide, with the attached cable that's a little over 13 feet long terminated with the locking 4-pin connector that A-T regularly uses with many of its wireless systems.

A remote power supply unit is included that has a locking 4-pin connector on one end and a standard XLR on the other, allowing the ATM350a to work as a hard-wired mic. The power supply offers a recessed roll-off switch at 80 Hz to help control undesired ambient noise.

The mic, which has a cardioid pattern with reduced side and rear pickup, utilizes a fixed-charge back plate, permanently polarized condenser element. Stated frequency response is 40 Hz to 20 kHz and sensitivity is -49 dB. With a whopping maximum SPL rating of 159 dB, it should be able handle very loud instruments with ease.

The UniMount components are handy, slick and very easy to use. First, the mic is placed in either the supplied 5-inch or 9-inch gooseneck with foam windscreen, with the other end of the gooseneck simply inserted into one of the many mounting base options available to users. These include a magnetic piano mount base, drum clamp, universal clip-on, and woodwind/acoustic bass strap mount. There's also a wireless universal clip-on system. The goosenecks include rubber cable tenders to help route the cable and keep it from getting in the way.

Audio-Technica ATM350a microphone.

A look at how the mic fits into the gooseneck.

POSITIONING OPTIONS

Out of the box I was impressed with how well thought-out the mounting options are. The goosenecks are stiff and will hold their positions but can easily be adjusted. The bottom of the goosenecks are 8-sided and fit into matching hole in the base, so there are a variety of mounting positions. A small hand tightened set screw locks the gooseneck in place so there's no movement once the mic is positioned.

Another benefit was consistency due to the mic remaining at the same distance from the instrument.

The magnetic piano mount base has non-marring felt on the bottom so it won't scratch surfaces, while the universal spring mount has thick rubber jaws so it won't scratch as well. The spring mount includes a hand-operated locking screw to tighten and lock the clamp into position. The drum clamp slips over a tuning lug on any drum, and includes an lug extender to retain tuning ability.

One of the most unique mounts is the strap system. It consists of a small rubber base with a Velcro strap that can wrap around a woodwind instrument such as a flute or around the tail end of upright bass strings, below the bridge. A standard metal gooseneck base mount is attached to the rubber piece for optimum fitting.

I took the mic to my bench and plugged it in, giving it it phantom power. The mic sounded great with a voice. The first test with an instrument came via a snare drum, with the short gooseneck employed with the drum clamp. It sounded great while mounting easily, clamping over the rim but still allowing tuning via the lug extension. I also liked the option of the power supply's 80 Hz roll-off switch in this application.

Next up were some rack and floor toms from my inventory. Again, very nice sound, and with the choice of the 5- or 9-inch goosenecks, no problem with positioning. The clamp worked with every brand of drum that I tried it on, and I don't see why it would not work with every drum on the planet that uses a rim and lugs for tuning.

SOUND & MOVEMENT

It was time to take the ATM350a package out to some gigs. First up was working with a Klezmer band at a large wedding.

(For the unfamiliar, Klezmer is traditional music tracing its roots to Ashkenazi Jews of Eastern Europe.)

Clarinet is a big part of this style of music, so I showed the non-marring strap to the clarinetist in the band and asked if he'd use the mic. He gladly agreed, so we placed the strap well past the keys down the instrument, and then used the 5-inch gooseneck to position the mic at the bell. Normally I prefer to mike clarinets from the side, as sound comes from both the open keys and the bell, but the ATM350a sounded excellent on just the bell end.

In addition, the clarinet player was quite happy to not be tied to a stand mic, instead able to move his instrument around and even dance. He, as well as the band, were impressed with the sonic quality, and again, he really loved the freedom of movement it granted. In fact, I was going to swap the mic over to the drummer but the clarinet player would have none of it!

Another benefit was consistency due to the mic remaining at the same distance from the instrument. I had no problems in the mix, unlike working with clarinet players who move around a stand-mounted mic, which in turn means that I have to ride that fader all night.

The variety of mounts available with the ATM350a.

ANOTHER TAKE

Next up was a corporate show with a pianist playing for the cocktail hour in the lobby. His instrument, provided by the hosting hotel, was a beautiful baby grand. With only one ATM350a on hand, I decided to use it for the mid-highs, joined by another condenser for the lows.

Using the magnetic mount and 9-inch gooseneck, I placed the mic about one-third of the way in from the high end, positioned about 6 inches away from the dampers. It resulted in a very natural sound through the PA.

In addition, this package proved much easier to deploy than setting up a boom stand and a full-size condenser model. A pair of ATM350 mics on a piano will capture the instrument extremely well while also being simple to set up. Also, the compact size and lack of external stands mean that the piano lid can be closed if need be.

Audio-Technica ATM350a systems are highly recommended for virtually any instrument miking application. The sonic quality is top-notch while the choice of mounting bases and gooseneck lengths means there's plenty of flexibility, something that often comes in handy in the ever-changing world of live sound. Also note that the mic, power supply and various mount packages ship with a large padded zippered case.

U.S. MSRP: Ranges from $199 to $349 depending on mount options. In addition, mounts are available separately at prices ranging from $30 to $89. LSI

Dual ATM350a mics deployed on a grand piano.

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

CHANGING THE PARADIGM

Smaller-format digital consoles and mixers.

by Craig Leerman

In the analog realm, smaller-format consoles are pretty well defined by limitations in channel counts and feature sets. Not so with digital – many compact consoles of the more compact variety are loaded with enough features to handle larger shows and enough inputs to handle medium-sized gigs.

Further, an increasing number of newer digital models are easily expandable via add-on cards and stage boxes, and in some cases a mixer can be cascaded with another to double the input count. Most modern smaller-format models also offer a variety of powerful processing, including compressors and gates on input channels, graphic EQs on outputs, matrix buses, and great-sounding effects, including reverbs and delays, that eliminate outboard hardware.

The majority of compact consoles introduced over the past few years include most (if not all) of the features of their larger counterparts, again with the only differences being channel counts and overall physical footprint. Using the same software makes sense because techs and engineers only need to learn one system, while sound/production companies can stock a few models of the same series that can be deployed relative to the size/scale of particular gigs. And, compact (and lighter in weight) models make it easier for lesser-scaled productions that want to travel with their own mixers for consistency.

Let's take a look around at more recently introduced smaller-format consoles on the market from a range of manufacturers. LSI

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

QSC TouchMix-30 Pro | *qsc.com*

Faders: Uses a touch screen with virtual faders, 1 control knob
Mix Inputs: 32
Aux/Group: 14 mono can be paired to 7 stereo
FX: 6
DCA: 8
Mains: LR
App: TouchMix-30 for IOS and Android
Screen: 10-inch multi touch screen
Local I/O: 24 XLR mic/line inputs, 2 main XLR outputs, 14 aux XLR outputs, 2 monitor XLR outputs, 6 TRS stereo inputs, 2 TRS stereo outputs
Also: 32-channel direct to hard drive record/playback; 32 x 32 DAW interface with Mac; MP3 playback direct from USB; anti-feedback/room tuning/gain and effects wizards
Physical: 7.5 x 16.9 x 18.1 inches, 17.5 pounds
Additional Models: TouchMix-8, TouchMix-16

Allen & Heath Qu-32
allen-heath.com, americanmusicandsound.com

Faders: 33
Mix Inputs: 32 + 4 FX + 3 stereo
Aux/Group: 10/4 + 4FX
Matrix: 4
FX: 4
DCA: 4
Mains: LR
App: Qu-Pad for iPad, Qu-You iPhone
Screen: 7-inch, 800 x 480 color touch screen
Local I/O: 32 XLR + 4 TRS, 26 XLR
Stage Boxes: AR84, AR2412, AB168
Also: ME-1 personal mixers; 18 x 18 record/playback to DAW or direct to USB hard drive
Physical: 34 x 19 x 8 inches, 44 pounds
Additional Models: Qu-16, Qu-24, Chrome Edition

TECHNOLOGY FOCUS: *Precision synchronization technology achieves low latency and low jitter as well as high sample accuracy in a network that is fast and simple to set up. Individual IDs assigned to each device on a Dante network allow automatic device detection as well as easy patching. Speedy setup leaves more time for the all-important job of setting up the mix.*

OF NOTE: *Up to eight VCM processors, including the Portico processors developed in cooperation with Rupert Neve Designs, can be mounted in the virtual rack. In addition, renowned Dan Dugan automatic microphone mixing with advanced algorithms is built in.*

Yamaha Pro Audio QL5
yamahaca.com

Gene Kim mixing monitors on a QL5 for the recent tour by Phil Wickham.

Operators coming from analog or digital consoles will quickly become comfortable with the QL5. Years of accumulated know-how and feedback from users worldwide have been applied to creating an interface that is simple while providing refined operation that responds fully to the demands of a broad spectrum of real-world applications.

The ability to respond swiftly to multiple, rapidly changing demands is essential for effective live sound support. QL consoles feature large touch-panel displays as well as "Touch and Turn" knobs that make up a remarkably smooth, efficient control interface. Attention has also been paid to details such as fader feel and channel name display visibility in order to deliver a sophisticated overall operating experience.

Remote control and offline setup capability via an iPad or computer adds even more refinement to an already state-of-the-art operating environment.

KEY SPECIFICATIONS:

Faders: 34 (32 + 2)
Mix Inputs: 72
Aux/Group: 16
Matrix: 8
FX: 8 FX + 8 premium
DCA: 16
Mains: L/R, L/R/C, 5.1 surround
App: QL StageMix for iPad
Screen: 10-inch color touch screen
Local I/O: 32 XLR mic/line inputs, 16 XLR outputs, stereo AES
Stage Boxes: Rio3224-D, Rio1608-D, Ri8-D, Ro8-D
Also: Dugan Automix; 62 x 62 Dante interface
Expansion Slots: 2
Physical: 33 x 23 x 11 inches, 48 pounds
Additional Model: QL1

Avid S3L-X
avid.com

Faders: 16
Mix Inputs: 64
Aux/Group: 24
Matrix: 8
FX: 40 stereo plug-in slots
DCA: 8
Mains: LCR/mono
Screen: XGA monitor
Local I/O: 2 XLR + 2 TRS, 2 XLR + 2 TRS, plus 4 x 4 x 4 XLR I/O on 2RU E3 engine
Stage Boxes: VENUE 4RU Stage 16 (16 x 8 + 4D)
Also: 64 x 64 Gigabit AVB to Pro Tools; 2-track USB playback/record

Physical: S3 control surface is 28 x 15 x 3 inches, 14 pounds; E3 engine is 2RU x 15 inches, 21 pounds; Stage 16 I/O is 4RU x 8 inches, 16 pounds

RWG Spotlight Listing

An S31 playing a key role at the 2016 V Festival in the UK.

DiGiCo S31 | *digico.biz*

Constructed from aerospace graded aluminum extrusion, with durable polycarbonate overlays and RGB LED Hidden To Lit technology, the S31 has all bases covered. It's Ideally suited to applications where more instant control and feedback are critical to the operator.

There are 46 buses, 40 flexi channels and 16 flexi buses in its impressive specification, as well as three of the latest P-CAP multi-touch screens, with 10 channel strips per screen in a new-look design, and integrated USB to DAW connectivity for up to 48 tracks of recording.

The S31 also comes with dual DMI option card slots, so it can interface with industry formats including analog expansion, MADI, Dante, Waves SoundGrid, or Calrec's Hydra2 network. Truly, it opens even more doors, in delivering increased functionality and accessibility, all at an affordable price point.

OF NOTE: *As the larger version of the popular S21, the S31's expanded work surface offers 10 additional faders for more control and two other multi touch screens for faster access and more visual feedback.*

TECHNOLOGY FOCUS: *Using the high-power QuadCore SoC, associated with high bandwidth memory, the S31 connects to a low power 484-ball array FPGA which in turn connects to fourth generation control SHARC DSP, capable of not only controlling the FPGA, but offering the potential for additional processing in the future. The compact footprint of the S31 has no relationship to the scale of processing power going on under the polycarbonate work surface.*

KEY SPECIFICATIONS:

Faders: 31
Mix Inputs: 40 flexi (stereo or mono)
Aux/Group: 16 flexi aux or sub group
Matrix: 10 x 8
FX: 8
Mains: LR
Screen: 3 multi-touch screens
Local I/O: 24 XLR inputs, 12 XLR outputs, AES I/O
Stage Boxes: D Racks, SD Racks, MiNiDiGi Rack
Also: 4 assignable DiGiTuBes; integrated USB2 audio I/O interface for recording and playback of up to 48 channels; optional Waves integration
Expansion Slots: 2
Physical: 11.6 x 40.3 x 23.1 inches, 55.1 pounds
Additional Models: S21, SD11, SD11i

Mackie Axis (DL32R rack-mount mixer & DC16 surface)
mackie.com

Faders: 17
Mix Inputs: 32
Aux/Group: 14
Matrix: 6
FX: 3
DCA: 6
Mains: LR
App: Master Fader app for IOS
Screen: DC16 includes color backlit TFT screens; system uses iPads (not included) for additional visual information
Local I/O: 32 XLR mic/line inputs, 14 XLR outputs, stereo AES output
Also: 32 x 32 direct to disk recording and playback; 32 x 32 to Mac and PC; supports up to 20 IOS devices controlling mains or monitors
Expansion Slot: 1
Physical: DL32R – 5.4 x 19 x 17.5 inches, 18 pounds; DC16 – 3.3 x 36.8 x 17.6 inches, 38 pounds
Additional Models: DL1608, DL806

Michael Demus (left) and Tom Lane with a TL5 at the renowned 16th and Grand Coffeehouse in Nashville.

Yamaha Pro Audio TF5 | *yamaha.com*

The TF5 offers intuitive operation, advanced features and renowned Yamaha reliability, providing engineers free rein of their mix. TouchFlow Operation allows the user to respond to the music and artists on stage with unprecedented speed and freedom via an interface optimized for touch panel control.

The physical Touch & Turn knob is available right beside the touch panel. There are also four User Defined knobs below the panel that can be assigned to control almost any parameter you need fast, direct access to while mixing. In a live situation with musical instruments, QuickPro presets can be searched by instrument type and recalled quickly and easily.

Recallable D-PRE Class A preamplifiers support high sound quality while advanced live recording features and seamless operation with high-performance I/O racks give the TF5 capabilities that make it an outstanding choice for a wide range of applications.

OF NOTE: *Scene memory features banks A and B, each capable of holding up to 100 scenes, for a total of 200 scenes, that can be set up and instantly recalled. The TF5 ships with a full version of Steinberg's Nuendo Live software for multi-track recording.*

TECHNOLOGY FOCUS: *Three dedicated apps – TF Editor for Mac or PC on and offline control, TF StageMix offering iPad control, and TF MonitorMix – further enhance usability. Since up to three devices running TF Editor or StageMix and up to 10 devices running MonitorMix can be connected at the same time, even large bands can have the personal control they need, reducing demands on the engineer.*

KEY SPECIFICATIONS:

Faders: 33
Mix Inputs: 48 input channels
Aux/Group: 20
FX: 8
DCA: 8
Matrix: 4
Mains: L/R
APP: TF Editor for Mac and PC, TF StageMix for iPad, TF MonitorMix for IOS
Screen: Color multi-touch
Local I/O: 32 XLR mic/line inputs, 2 analog RCA stereo line inputs, 16 XLR outputs
Stage Boxes: TIO 1608-D
Also: 34 x 34 digital record/playback channels via USB 2.0 + 2 x 2 via a USB storage device; Quick Pro presets
Expansion Slot: 1
Physical: 8.9 x 34.1 x 23.6 inches, 44.1 pounds
Additional Models: TF-Rack, TF-1, TF-3

Midas M32
midasconsoles.com

Faders: 25
Layers: 4 + 4
Mix Inputs: 32 + 8 FX + 8 aux
Aux/Group: 16
Matrix: 6
FX: 8
DCA: 8
Mains: LCR/LRM
App: M32-Mix (iPad), M32-CUE (iOS/Android)
Screen: 7-inch color screen

Local I/O: 32 XLR + 6 TRS, 16 XLR + 6 TRS + 1 AES; 2 AES-50
Stage Boxes: DL16, DL150 Series
Also: P16 personal monitoring; 32 x 32 recording/playback via USB2/FW800
Physical: 35 x 24 x 10 inches, 54 pounds

PreSonus StudioLive 32
presonus.com

Faders: 33
Mix Inputs: 32 (cascading up to 64 with 2 consoles)
Buses: 26 flex (aux/group/matrix)
FX: 4
DCA: 24
Mains: LR, mono/center
App: UC surface for Mac, PC and IOS
Screen: 7-inch color touch screen
Local I/O: 32 XLR inputs, 12 XLR outputs, additional TRS I/O
Also: Bluetooth wireless tape input; onboard 38 x 38 SD recorder; 38 x 38 USB recording interface; 55 x 55 AVB recording interface
Physical: 6.5 x 32.8 x 23 inches, 37.2 pounds
Additional Models: StudioLive 32.4.2AI, StudioLive RML32AI

Roland Pro AV M5000C
proav.roland.com

Faders: 20
Mix Inputs, Aux Group Matrix: Up to 128 audio paths configurable as mono or 64 stereo
FX: 8
DCA: 24
Mains: L/R, L/R/C and 5.1 surround
App: M-5000 Remote iPad app, M-5000 Remote Control software for Mac and PC
Screen: 12-inch color touch screen
Local I/O: 16 XLR mic/line inputs, 8 XLR outputs, 2 stereo AES I/O
Stage Boxes: S-4000S, S-2416, S-1608, S-0818, S-0808
Also: The 128 audio paths are configurable as inputs, outputs, or buses; integrates with Waves SoundGrid and the company's M-48 personal monitoring system; 16 x 16 USB audio interface for DAW
Expansion Slots: 2
Physical: 13.6 x 29.1 x 28.5 inches, 70 pounds
Additional Models: M300, M200i

Soundcraft Si Impact
soundcraft.com

Faders: 26
Mix Inputs: 40
Aux/Group: 14 (combination mono or stereo)
Matrix: 4 (mono/stereo)
FX: 4
DCA: 8
Mains: L/R, L/R plus mono, L/R/C, LR
App: Soundcraft Remote for iPad
Screen: 5-inch color touch screen
Local I/O: 32 XLR mic/line inputs, 16 XLR outputs, AES output
Stage Boxes: Mini Stagebox 32, Mini Stagebox 16, Compact Stagebox
Also: 32 x 32 USB audio interface; FaderGlow
Expansion Slots: 2
Physical: 6.3 x 29.5 x 19.7 inches, 44.1 pounds
Additional Models: SI Performer, SI Expression

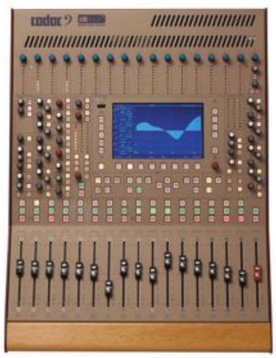
Behringer X32 Compact
behringer.com

Faders: 17
Mix Inputs: 40
Aux/Group: 16
Matrix: 6
FX: Virtual FX rack featuring 8 true-stereo FX slots
DCA: 8
Mains: LCR
App: X32 Edit for PC or Mac, X32 Mix for iPad , X32 Q personal monitor mx for iPhone and Android
Screen: 7-inch day-viewable color TFT
Local I/O: 16 XLR inputs, 8 XLR outputs, 6 line in/outputs, AES out
Stage Boxes: S16, S32, SD16 and SD8
Also: UltraNet connectivity for company's P-16 personal monitoring system; 32 x 32 channel USB 2.0 audio interface; DAW remote control
Expansion Slot: 1, pre-installed X-USB card, optional USB/Firewire, MADI, Dante and ADAT
Physical: 8.5 x 20 x 24.6 inches , 34 pounds
Additional Models: X32, X32 Producer, X32 Rack

Lawo mc²36
lawo.com

Faders: 16
Mix Inputs: 184
Aux/Group: 32
Matrix: 512 x 512
FX: Up to 192 DSP channels
DCA: 128
Mains: LCR, mono, surround
App: mxGUI for Mac and PC
Screens: 2 x 21.5-inch touch screens
Local I/O: 32 XLR mic/line inputs, 32 XLR line outputs, 8 AES I/O
Stage Boxes: mc2 Compact
Also: 3 RAVENNA/AES67 Audio-over-IP ports; Waves SoundGrid integration
Physical: 16.3 x 31.3 x 34.7 inches 93.8 pounds
Additional Models: 24-fader and 40-fader versions

Cadac CDC four:m
cadac-sound.com



Faders: 17
Mix Inputs: 56
Aux/Group: 8
Matrix: 6
FX: 6
DCA: 8
Mains: LR, Mono
App: TabMix Console Remote Manager for iPad
Screen: 7-inch TFT color display
Local I/O: 16 XLR mix/line inputs, 8 stereo inputs, 16 inserts, 16 direct outputs, 4 group outs, 8 aux outs, XLR main and mono outputs
Stage Boxes: CDC I/O 3216, CDC I/O 6448, CDC MC AES3
Also: MegaCOMMS protocol provides up to 128 bi-directional low-latency channels carried on co-axial cable with runs of up to 150 meters (492 feet)
Expansion Slot: 1
Physical: 7 x 10 x 22.8 inches, 32.6 pounds
Additional Models: CDC four, CDC six

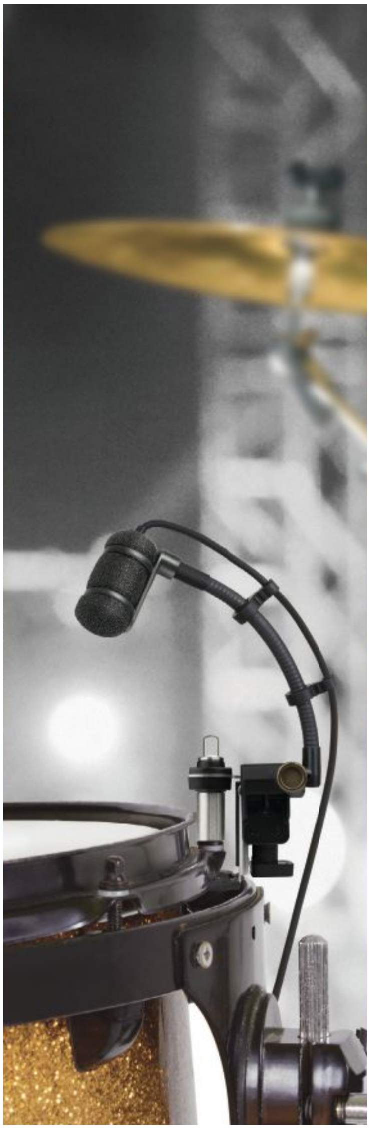
Jon Lundgren has joined the engineering team at **Radial Engineering**, based at the company's headquarters in Vancouver, BC. His primary objective is to expand the digital design presence for the Radial group of companies, and specifically, he is working on the recently re-launched **Dynaco** product line.

QSC has promoted three long-term staff members to senior leadership roles. **Anna Csontos** has been named vice president, chief of staff; **Perry Celia** is senior director, domestic sales; and **Arnie Marx** moves to senior director, software development. (They are pictured here left to right.) In addition, the company announced that it has been named by the *Orange County Register* as one of the Top Workplaces in Orange County (CA) for the sixth time.

Allen & Heath has recently expanded the team based at its U.S. distributor, **American Music & Sound**, with the appointment of **Mike Bangs** (pictured center) as live sound and touring expert, **James Duvall** as commercial solutions specialist, and **Maryam Larki-Bavi** as marketing manager.

Josh Maichele and **Jesse Stevens** (pictured L-R) have joined **L-Acoustics** as North American regional applications engineers, responsible for bolstering active technical support and expertise for the company's Certified Providers, end users, and designers in the Midwest and Northeast regions of the U.S. and Canada, respectively.






ROAD STORIES #4

Fielding questions from the keen kid. *by Ike Zimbel*

In the early 1990s I was serving as a maintenance tech in a large recording studio complex when I got a call from the sound company I'd worked for in the previous decade. Was I available to mix and tech front of house for the upcoming Canada Day (July 1st in Canada, like July 4th in the U.S.) gig at the provincial legislature? I hadn't done live sound in a while, but it was a daytime gig, on a paid holiday, with no shop prep, so what the heck — I said yes.

On the day of the gig, I show up bright and early, greet some old colleagues, including the owner of the company whose client's gig this was, as well as some new faces. One of the new faces I'm introduced to is someone we'll call "The Kid," and he would be assisting me at front of house.

So The Kid and I proceed to the FOH riser and start to set up. The whole time he's bombarding me with questions: "Whaddya' think of this piece of gear? Whaddya' think of that console? Do you like to EQ the system this way or that way? Didja' ever use one of these?" And so on.

Now, I've always been a big believer in teaching the up-and-coming. As long as the questions are reasonably intelligent and respectful, I try to answer to the best of my ability and hopefully impart some knowledge. (I once had a new hire, on his first day, corner me in the coffee room at the shop and demand to know "How many 'K' in a hundred watts?" When I tried to explain that this wasn't a valid measurement, he became increasingly insistent: "No, I really want to know!" — the implication being that I was just blowing him off because he was a newbie and as a result was refusing to answer. The fact that he was from the UK and this all came out in a thick London East End accent made the whole ordeal feel like a Monty Python routine. To this day, I've never really been sure if he was winding me up or not, although he seemed totally serious at the time... But I digress).

All through setup and sound check I continue to field questions from The Kid. Eventually, we're ready to start the event and he moves up to the stage to be part of the deck crew.

The show itself runs smoothly. The only other thing I remember about it was when the host, MuchMusic VJ Monika Deol, unimpressed with the crowd's anemic singing of the national anthem, belted out the most stirring rendition of "Oh Canada" I've ever heard. (And she was standing right next to me on the FOH riser at the time.)

When the strike rolls around, The Kid comes back to FOH to help me tear down. He's still full of questions, but I can tell he's working up to something specific. Finally he asks me: "How well do you know 'X'?" ("X" being the owner of the sound company.)

"Pretty well," I respond, adding, "I've known him for about 10 years. Why?"

And out it comes: "What do you think I have to do to get him to send me out on the really big shows with the big concert rigs?"

I pause, look The Kid up and down, and ask: "How old are you?"

"Sixteen!" is his response, to which I reply, "Well, here's what you have to do... GET OLDER!"

Postscript: This gig eventually led to an offer to return to the company and manage its audio department, which I did for eight years, starting later that year. Meanwhile The Kid, who was also an accomplished musician, went on to some success with a few bands.

A few years later he was visiting the shop on some business and came upon me relating the above story to one of the staff. As I got to the punch line, he says, "Yeah, but here's the best part. When I was on that gig, I lied about my age — I was only 14!" LSI

Ike Zimbel *is a wireless frequency coordinator and tech, based in Toronto. You can find him on LinkedIn.*

"PROBABLY THE BEST PA I'VE USED IN MY CAREER SO FAR"
JON LEMON
FOH- SIA, RAY LAMONTAGNE

"I WAS BLOWN AWAY" BEN "SNAKE" SCHMITT FOH- KIP MOORE

"I WOULD DEFINITELY USE IT AGAIN AND AGAIN"
TIM COLVARD FOH- MACKLEMORE & RYAN LEWIS, MADONNA, LIONEL RICHIE

"FOR THE NEXT TOUR WITH MR. BROWN I'M REQUESTING ADAMSON"
GORDON MACK FOH- CHRIS BROWN, JUSTIN BIEBER

"IT'S THE BEST DELIVERY OF HIGH-MID I HEAR TODAY"
PAUL (PAB) BOOTHROYD FOH- PAUL MCCARTNEY, AC/DC

"I PUT IT NUMBER ONE"
KYLE HAMILTON FOH- PHARRELL WILLIAMS, RIHANNA

"ADAMSON IS MY FIRST CHOICE WORLDWIDE"
SCOTT EISENBERG
FOH- IMAGINE DRAGONS

"THE HIGH END OF THIS PA IS ONE OF THE CLEAREST AND SMOOTHEST I HAVE EVER HEARD"
DEMETRIUS MOORE FOH- DRAKE

"IT SOUNDS THE WAY MUSIC SHOULD SOUND"
JOE HARLING
FOH- ELLIE GOULDING

"IMPRESSIVELY SMOOTH FROM NEAR TO FAR FIELD."
HASSAN (HASS) SIAHI
FOH- DAVID GUETTA

"..ABOUT AS BIG AS I'VE EVER HEARD."
CHRIS RABOLD FOH- LADY GAGA

"PRODUCTS ONE STEP ABOVE EVERYONE ELSE - E15 IS NO EXCEPTION"
PATRICK DEMOUSTIER FOH- NIGHT OF THE PROMS

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