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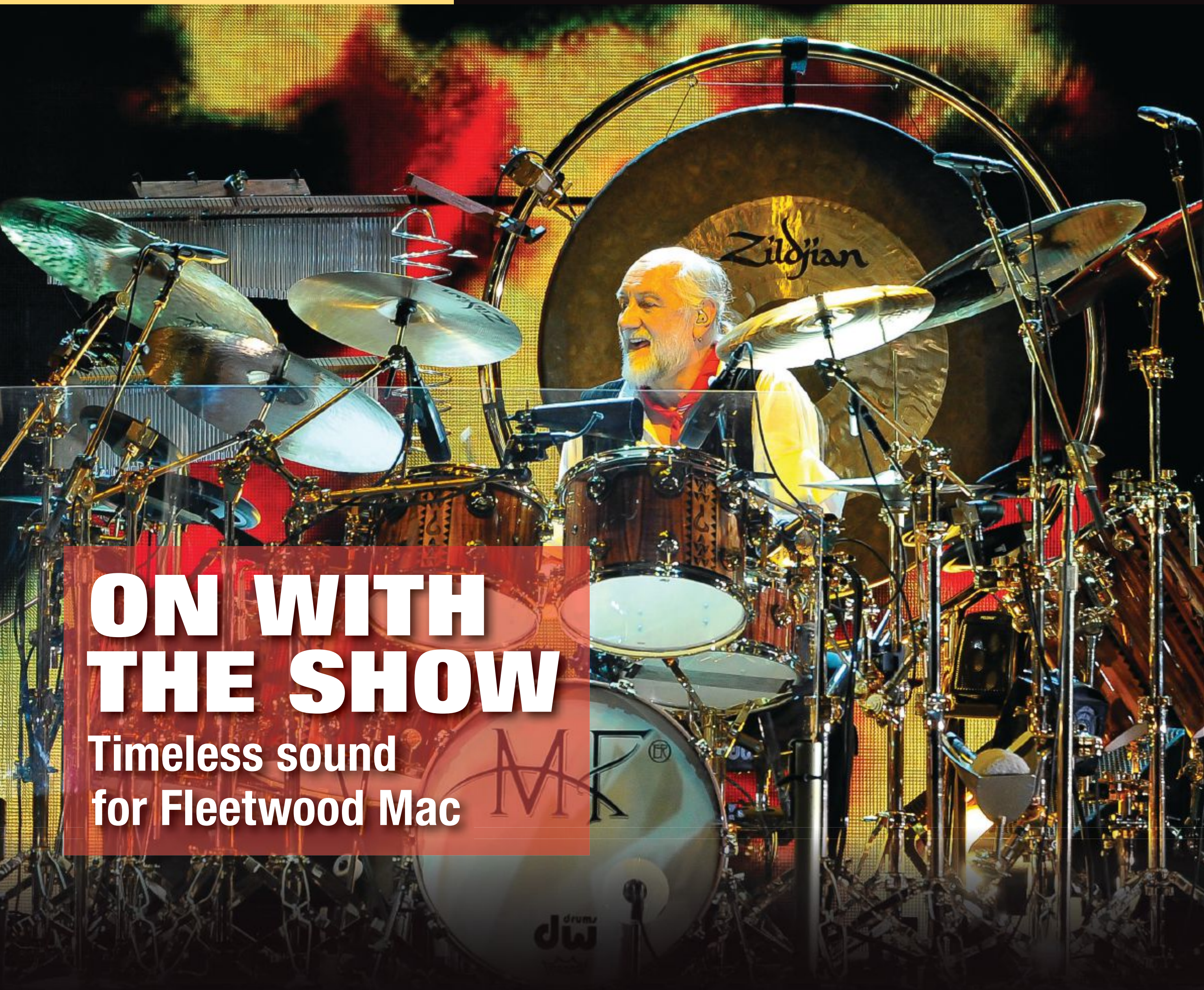

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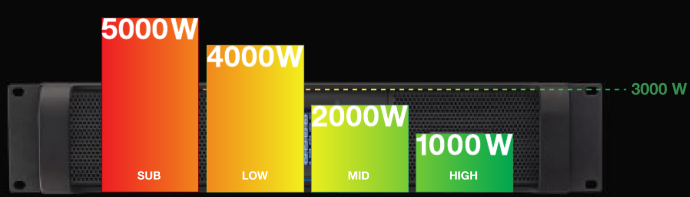

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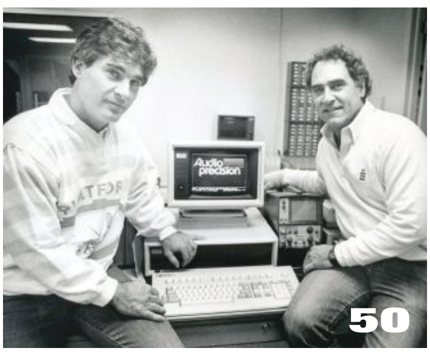

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From the Editor's Desk…

There's something about starting a new year that seems to increase our focus. Maybe it's the optimism of setting fresh goals, or the "clean slate" of putting the previous year in the rear-view mirror. Perhaps it's both.

Whatever the case, in this issue Karl Winkler offers some thoughts on maintaining focus as we deal with the day-to-day reality of responsibilities (and repetition) in the coming year. "The quest to stay sharp and on top of the game," as he terms it, can take on a focus all its own.

In the previous issue (*December 2014 LSI*), I noted in this space the often interesting challenges that installed system projects can present. We continue that thread with a look in this issue at the considerable work that went into implementing a new loudspeaker system at the DeWitt Theatre at Hope College in Michigan. Working in professional audio often involves dealing with much more than, well, audio.

Greg DeTogne delivers details on concert sound for one of the top tours out right now, featuring Fleetwood Mac. The mix engineers provide excellent context in discussing their approaches with this legendary group in the live realm. Both related and in contrast, Andy Coules steps up with part 2 of his look at the history of PA, and Fred Ampel presents key factors in the development of multichannel amplifiers.

Also don't miss Bruce Main's review of venue acoustics, Craig Leerman's overview of selecting vocal microphones in one-off situations, and Mike Sessler's experiences with rechargeable batteries with wireless systems.

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

Keith Clark
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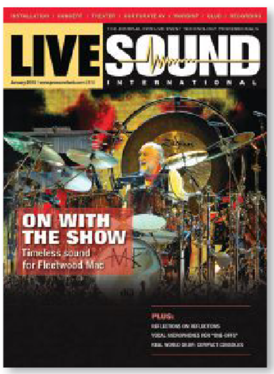

ON THE COVER: Mick Fleetwood at the kit on the latest tour by Fleetwood Mac. ***(Photo by Steve Jennings)***

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XMOS AVB Platform

Semiconductor company XMOS and AVnu Alliance, the industry consortium for open standards-based deterministic networking, have announced the first AVnu-certified Audio Video Bridging (AVB) audio endpoint reference platform. The platform is scalable and production-ready, allowing manufacturers to quickly build a wide range of AVB-enabled audio products, from single loudspeakers and microphones to complex multichannel mixing desks and multi-port conferencing systems. Already in use by manufacturers such as Revolabs and Pivitec, the reference platform enables high-quality transport of A/V streams across mixed-use networks. Because the AVB functionality of the platform is defined in software running on theXMOS family of devices, manufacturers can create the exact feature set required for their products. **www.xmos.com, www.avnu.org**

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AKG APS4

A wide-band UHF active antenna power splitter that can be operated in an extended frequency range of 470 to 952 MHz, available for the company's DMS700, WMS4500, WMS470 and WMS420 wireless receivers. It can feed up to four receivers with the RF signal coming from one pair of antennas, and also supplies power to all connected receivers via BNC cables (each of which is individually protected against short-circuits). Adjustable RF-level attenuation settings help foster maximum operating distance, even when using different cable lengths and types. The ASP4 comes in a half-rack metal housing and works with all active and passive AKG antennas, and ships with 10 BNC antenna cables and a rack-mount unit that includes two antenna front-mount cables. **www.akg.com**

HK Audio USA Lucas Nano 600

A compact, portable PA that can be deployed either as a mono column or as a stereo system with detailed spatial resolution. Both mid-high "satellite" loudspeakers incorporate a 4.5-inch woofer and a 1-inch tweeter, providing dispersion of 90 degrees by +10/-45 degrees. They can be transported via a bay in the subwoofer, which is loaded with a 10-inch cone driver. All transducers are driven by a power amplifier delivering 460 watts. The system also includes a 3-channel mixer with frequency shaping controls, as well as five total inputs, one with phantom power. Proprietary MultiCell Transformer technology maximizes gain, improves directivity and minimizes HF distortion. The subwoofer, which also houses the electronics, measures 15.3 x 19.2 x 18.5 inches and weighs about 30 pounds. A range of accessories is available. **www.hkaudio.us**

Clear-Com LQ Series

A family of connectivity devices for linking intercom and audio systems over LAN, WAN or IP networks. Based on the company's I.V.-Core technology, LQ (formerly known as LINQ) utilizes the low-latency audio CODEC OPUS, which offers a range of different settings to provide high-quality audio on the network bandwidth available. This means that the LQ Series can be used for intercom conversation, networked music performances and audio signals that need to be transported between different facilities. LQ throw-down devices enable connections of 2-wire partyline with call signaling and 4-wire audio over LAN, WAN or internet IP infrastructures. There's a 2-wire (LQ-2W2) or 4-wire (LQ-4W2) option. The LQ-2W2 is both Clear-Com and RTS TW compatible. A maximum of six LQ IP interfaces can be linked together in any 2- or 4-wire combination. **www.clear-com.com**

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STAYING SHARP

Seven tips (and more) in the quest for focus.

by Karl Winkler

AS WE EMBARK on a new year, I thought it might be a good idea to discuss some ways to stay engaged and focused. Particularly for those who have been working in pro audio for several years, there are many who probably come to feel (at least occasionally) bored, uninspired or behind the curve, as one month leads to the next, and to the next, and to the next …

So here are seven things I've found helpful over the years in the quest to stay sharp and on top of the game. Note that this list isn't definitive – you may have additional ideas and methods.

1. Learn. There are a plethora of seminars, workshops, trade shows and other ways to stay on to of current technologies and techniques. Take a look at the SynAudCon roster of in-person and online training courses as a starter. Trade shows like InfoComm and AES provide ample educational seminars, papers and panel discussions, usually with industry leaders at the podium. Then there are the more informal approaches, including books, DVDs and yes, YouTube videos. (Just be sure to read the reviews and comments for some of the less-vetted sources.) Or, download an operating manual for something like a digital mixing desk and start reading!

2. Humble. This is one of my "broken record" topics (along with gain structure). But throughout my years in this business, I've often noticed that the best and brightest show remarkable humility. Even if they're highly opinionated (and get in trouble for it once in a while), the true "gurus" got where they are by keeping their minds open, realizing that even with their accomplishments, there are still many things to learn. There are always new ideas, new technologies, and better ways to do almost anything. Being open to these possibilities keeps things interesting.

3. Avoid the rut. You know the saying about doing the same thing over and over while expecting different results, right? Sometimes we forget that this applies to our careers as well. There's nothing that can make us more bored (and/or jaded) than rote repetition. (See the excellent Bill Murray movie *Groundhog Day* for more on this subject.) Waiting for something different to happen? Then make it so! For example, if you've been mixing rock 'n' roll for years on the same circuit, how about mixing some jazz instead? Or perhaps think about taking on a new role with the same crew just to learn a new skill, like lighting for instance. (Wait – did I just say that?) Or try a new mic technique, or experiment with live recording, or...anyway, you get the idea. Anything to avoid getting stuck in a rut.

4. Volunteer. Donating skills and labor to a church, school, charity event or other worthwhile cause is refreshing. Of course, it can also be frustrating so it's important to know what and who to avoid in these scenarios. But giving of ourselves to a cause that we respect both feels good and provides a unique experience. It doesn't have to be a regularly scheduled activity such as church services every Sunday morning, but perhaps a helping hand at busy times like the Christmas production and Easter pageant. Even a simple role, like loading gear and taping cables, can make a big difference to an organization while providing us with a new perspective.

5. Network. Trade shows and seminars are great for this too, and social media can help as well. Get out there and meet people in our industry to share ideas, techniques, best practices, and of course, war stories. I'm personally always grateful to get new perspectives on our business and ways to do things better. We all (or at least most of us) know a lot of people, and each one of those people knows a similar number of other people. Even though pro audio is a relatively small market, and there are probably only a couple of degrees of separation between any two of us, the goal should be a direct connection to as many as possible. Then as ideas come up, or a job change is inevitable, we have people to call.

6. Plan. One thing I've found to be really effective in getting me off the couch and out into the world is to think about where I want to be in 3 months, 6 months and 12 months. It's easy to belittle this concept since "things always change and our best laid plans then go down in flames." But the truth is A) not all plans are destroyed, and B) even if they are, at least we're working toward something, staying motivated and engaged. And who knows, maybe you'll get lucky and actually achieve your goals! (Crazier things have happened.) Go ahead, make my day and tell me something that feels better than setting a lofty goal and then reaching it. (Just keep your responses clean.)

7. Teach. "If you can't explain it simply, you don't understand it well enough." Some guy named Albert Einstein said that, and what would he know? I've personally found that preparing to teach, or participating in a seminar or panel discussion clarifies my thinking on the matter at hand. Plus, it feels good to give something back and help the next generation, or even our peers, to learn new things. It doesn't have to be calculus, either. Best practices, tips, and road-worn but proven techniques can be a revelation for up-and-coming eager techs looking to get an edge. Plus, teaching can get you noticed by some of the industry associations which may then ask you to, well, teach some more. Embrace it! Who knows, maybe after your back goes out and/or you can't hear so well any more, you'll have set yourself up for a whole new career.

Bonus: Get a life. A life outside of pro audio, that is. Hobbies that engage our interests are a great way to get our heads out of the daily challenges of making things sound good. And of course family and other personal relationships outside the business are a key to mental health. Not only having someone to which we can vent our daily problems, but someone to go on road trips or vacations with and simply enjoy life. After all, that's a big reason why we're here, right? ■

KARL WINKLER *is director of business development at Lectrosonics and has worked in professional audio for more than 20 years. Reach him at karl1@karlwinkler.com.*

On With The Show

Timeless sound for Fleetwood Mac at sold-out arenas.

by Gregory A. DeTogne, photos by Steve Jennings

Fleetwood Mac in concert on the ongoing tour.

Another day, another city, and this time Dave Kob, having just finished tuning the PA, is out at his front of house perch, looking up into the yawning maw of another arena's dome. It's just an ordinary moment prior to a show until he catches sight of a sawzall blade coming through the ceiling above. "What's going on?" Kob asks his crew chief Donovan Friedman, pointing up with a bit or urgency. "I know this is a sold-out show, but I've never seen fans that motivated to get in."

Kob has worked a few shows in his time. Four decades worth, to be more precise, since he first started with Clair back in 1974. With Fleetwood Mac, he was around in the early days as system engineer, working under Richard Dashut, who engineered on Rumours and mixed the first two subsequent tours. By '82 he was in the front of house seat mixing for the Mirage tour. Stints with Madonna, The Eagles, The Who, and others ensued after that, but by '97, Kob was back.

Following another hiatus with the band, he returned again in 2009. The '13 tour followed, and then this year's current On With the Show tour that continues in North America through mid-April, marking the return of singer/keyboardist Christine McVie to the stage for the first time in 16 years.

"Yeah, so basically I've done a number of tours with the band spread over a very long time," Kob says while cautiously looking for traces of sawzall dust that may have floated earthward and onto his Yamaha PM5K console. "Right now, this is the best they've been playing since '97. Having Chris back is just a wonderful thing."

Old & New

The show today is up to two and a half hours of hits. That's what the fans want to hear, so there isn't a lot of experimen-

The legendary quintet, left to right: Christine McVie, John McVie, Stevie Nicks, Mick Fleetwood and Lindsey Buckingham.

tation going on onstage. Crowds are multi-generational, with baby boomers anchoring one end and millennials the other. Drawing upon classic analog components that preserve the band's legendary sound, the tour nonetheless makes use of recent technology – albeit sparingly – where it makes sense.

Clair i-5D cabinets deliver the reinforcement, with power coming from a collection of Crown Macro-Tech amplifiers. "I love the i-5D," Kob admits. "Clair initially only built two systems of it, so I had to step on a few necks to get one out here for me, and I'm not giving it back. The i-5D was basically created by taking the best from the i-5 and i-5b and joining it all together in a single cabinet. I was skeptical of the concept at first, but then I heard it and was amazed."

In larger venues that sell out, the crew usually deploys the whole system, which includes two hangs of 16 i-5Ds left-and-right. A rear hang normally is built around i-5s.

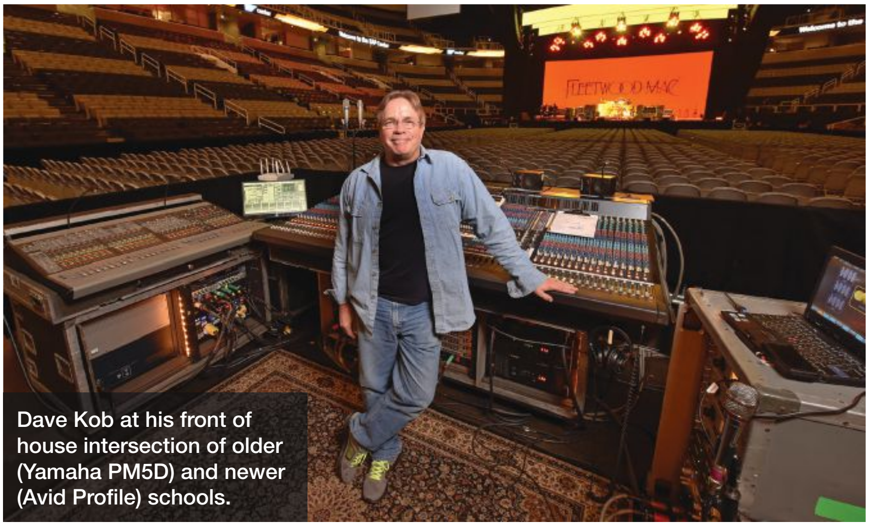
Matter Of Style

An outspoken advocate of analog, Kob's Yamaha PM5000 console is the centerpiece of his ongoing endeavors. "I go to so many other shows these days and the band sounds 'OK' but not great," he notes. "It seems I always find myself wishing I could hear more of this or that. I look around in these situations and there's usually a guy standing there tweaking some esoteric plug-in compressor/dynamic generator or whatever. Rather than mixing the band, he's playing with a video screen.

"Some people are sensible about technology, others get lost in it. A lot of the guys in the latter group have 60 inputs and 110 plug-ins. Well, if it works for them, fine. But it doesn't for me, and that's why I stay with my PM5K. I've mixed tours on digital desks, but I much prefer the ergonomics of an analog console, it just suits my style."

As a concession to the digital world, however, Kob keeps an Avid VENUE Profile at hand serving as a sidecar device. "The sidecar and the PM5K work together," he explains. "I have eight tracks from the USC marching band in there for use on the song 'Tusk,' for example, and some other bits and pieces. I've built Pro Tools snapshots of everything. I fire MIDI from the PM5K that makes all the changes I need on the Profile. There's not really too much to do with the Profile at all. Once it was programmed it was set to go."

His house mix comes together based around tried-and-true formulas, as you'd expect. There is, first and foremost, the matter of overall level. In the Kob canon, level has to be exciting. You can go below that occasionally and it's OK, but remain there and it gets boring. He mixes loud, but a clear distinction must be made between good loud



Dave Kob at his front of house intersection of older (Yamaha PM5D) and newer (Avid Profile) schools.

Clair arrays in the air.

and bad loud. Good loud keeps everyone dancing, and if someone onstage is playing something, it will be heard.

"Then there's balance," he adds. "That's the first critical part of actual mixing. Once you get that together and the EQ, you can animate the sound. Animation, in this sense, takes things and makes them jump out of the mix. I learned the art of animating sound from an engineer working with Elton John back in the '70s, Clive Franks. He would make a tambourine loud for two hits and then drop it back into the mix with reverb – amazing, tasteful things with a truly creative dynamic."

The Collection

House effects on the tour are purposely basic, and period authentic, of course. What worked back in the day still does just as well today. In keeping with those parameters, Kob's outboard selections include four Summit Audio TLA-100 tube leveling amps for vocals, three EL8 Distressors from Empirical Labs for backing vocals, an Avalon Designs stereo tube compressor, and a dbx 160 compressor that he keeps on Mick Fleetwood's vocal. "I use that piece for when he shouts," Kob says, "and he can shout louder than any instrument ever made."

Everything else within the outboard collection resides under the PM5K: a Yamaha SPX2000 for kick drum, an Eventide Harmonizer that runs on Stevie Nicks all the time just as a little "fattener-upper," and a Bricasti M7 vocal reverb, which Kob proudly points out was built with 30-year-old technology and is still a device that many others use as a comparison point. Other than all of this, look around further under the PM5K long enough and you'll find a couple digital recorders and eight channels of Aphex gates on the drums.

On the input side of the equation, there are no wireless microphones on this tour. After a lengthy evaluation of just about every hardwired vocal mic available, Sennheiser 935s were chosen for all vocals except Lindsey Buckingham's, which are still captured by his long-time favorite Audio-Technica AE6100.

Miking elsewhere onstage includes a Shure Beta 52 inside the kick, supplemented on the outside by a Milab kick drum mic no longer made that's one of

Distinct labeling tells the story of this front of house outboard rack.


Monitor engineer Dave Coyle stage-side with his DiGiCo SD10.

Kob's favorites. "The rest of the stuff is what I've used all the time forever as well – (Sennheiser) 421s, Milab overheads, AKG on high-hat," he adds.

Dual Approach

Monitorworld is a split-function affair using in-ear systems and wedges, with dedicated engineers for each: Dave Coyle for the former, and Ed Dracoules for the latter. As unofficial tour historian, Kob recalls that when he first started with the band, "they had an amp line where there was a Leslie cabinet on one side, Lindsey had extension cabinets on his side of the stage, and John (McVie) had his bass cabinets. The only thing they had in their monitors was kick, snare, hi-hat, and vocals. They listened to everything else through the backline."

Today, with Coyle taking charge of the in-ear monitoring systems from behind a DiGiCo SD10 digital console, Dracoules manages wedges with the aid of a VENUE Profile. The artists wear Future Sonics MG5Pro in-ear drivers fed by Sennheiser 2050 wireless systems. Wedges are Clair 12AM models, deployed for Buckingham, John McVie, Fleetwood, and the band's rhythm guitarist.

In keeping with the underlying analog nature of things, Coyle's outboard gear is a Spartan approach that includes such things as Yamaha SPX990s and an

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M5000 from TC Electronic providing effects for Christine McVie and Nicks.

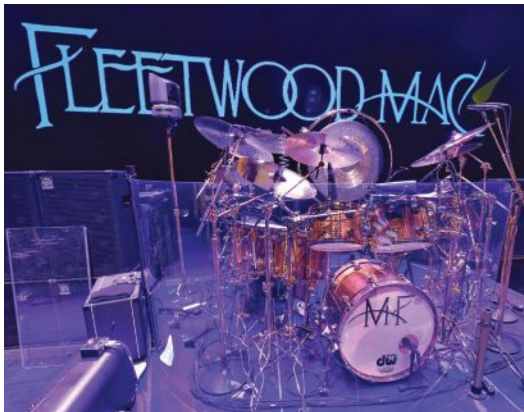
"Everything else comes from the console, which is one of the reasons I like using the SD10," Coyle explains. "That and it offers me the ability to use dynamic EQ and multi-band compression. I don't use a lot of any of that, but given the opportunity to run it on my Pro Tools snapshots is nice. This band likes its older effects, and when you mix older things with newer products on the same input, having this ability is a huge plus."

Staying On Course

The production level of the show is anything but stuck in the past. With motion having become a bigger part of the show over the past two tours, moving video walls now help to visually reinforce the show, along with a large-scale fixed video wall running along the back of the stage. There are about 300 cues for the show including count-ins and clicks.

"Each member of the band essentially likes to hear a mix mirroring the sound of the record," Coyle notes. "And despite their traditional use of effects, they do enjoy newer technologies like the Pro Tools snapshots.

"All that said," he continues, "to mix for this tour effectively, you can't lose sight of the fact that these are actual inputs. They are not heavily compressed, polished, or mastered. When you turn it up, that's what you get. You have to use your VCAs every night because these are live musicians in the most classic sense. There's nothing programmed here, you have to mix if you want it to sound good."



Plenty of mics on Mick Fleetwood's kit, fronted by a plexi shield.

Kob concurs with Coyle's assessment of the importance of staying on the VCAs. "There are a lot of moving pieces on this show," he concludes. "As long as they stay in parameters that are workable for me, it's OK. I just call that mixing, and that's what I love to do. I don't expect this band to come across at the same level every night. Take your hands off the helm here and things will veer off pretty quickly." ■

GREGORY A. DETOGNE *is a writer and editor who has served the pro audio industry for the past 30 years.*

TECH TOPIC

Reflections On Reflections

A quick review of venue acoustics.

by Bruce Main

I BEGAN WRITING this article a few months ago, as I sat in my office here in the Pacific Northwest, looking out the window at the rain making it obvious that, at least in this corner of the world, another summer had come and was long gone. Another year of street fairs, festivals, music in the park, shed lawns, barbeque catering and tarping the arrays to keep them from being waterlogged by the imminent thunderstorm was in the books, and it was time to go back indoors for the winter. And moving a sound system inside of an enclosed space presents a whole different range of acoustical challenges.

Something a younger practitioner of the art and science of live sound reinforcement had said caused me to think that perhaps a review of general acoustical principles was in order. As he walked the venue, he was experiencing lots of reverberation as well as some slap-back echo from a large concrete wall. He asked if we could "turn up the system so that the direct sound would override the slap."

I explained to him that the reverb and slap levels would increase at a 1:1 ratio to the level of the energy generating it. The only ways to mitigate the echo were absorption, diffusion or changing the cluster geometry and using the directivity of the loudspeaker hang so that we didn't excite the wall as much.

Much of the evolution in equipment, especially on the loudspeaker front, is intended to help us cope with difficult acoustic circumstances. New concepts in providing directivity seem to appear every week. Line arrays, beam steering, coverage mapping software and new horn topologies are just some of the tools that can be used to improve the system/room interface. But like any

tools they must be used properly to provide maximum benefit.

An old salt once told me "Put sound where the people are, don't put sound where the people aren't," and in many cases this sums up what we try to achieve with our loudspeaker toolbox. An ideal system would cover each member of the audience at an equal volume level with identical frequency response. Depending on the type of event it might also provide a perfect stereo (or surround sound) image at every seat, excellent intelligibility for spoken word, and seamless localization of the apparent audio source. It would not excite the walls, ceiling, balcony face or any other potential source of destructive flutter echo, but rather would tickle them just enough to create a nice, well balanced but not overbearing reverberant field. That's not too much to ask, is it?

INFORMED CHOICES

Fast forward to the real world. Acoustic spaces play havoc with audio reproduction. Reflections clutter up our nice open mix. Room resonances color the system's perceived frequency response. The room geometry and pick point locations won't cooperate with the inverse square law to allow uniform levels in all the seating areas.

The key to taming the acoustics beast is to understand the fundamentals of the phenomena we are dealing with, how the physics work and then making informed choices on how to use the tools we have to get as close as possible to that idealized response. Reflected sound is one of the primary factors. This can be divided into two categories; reverberation and echo.

Reverberation is a sound field created by multiple reflections from the surfaces in a room. It is statistically diffuse which means that in theory it has the same frequency and time characteristics everywhere within the space. It is not location specific.

W. C. Sabine was a young Harvard physics professor in the late 1800s who did some of the pioneering work in analyzing and describing the characteristics of reverberation. His equations used the internal volume of the space, the internal surface area and the average absorption coefficient of the surfaces to predict RT60 when the room was excited by an omnidirectional source. Later scientists realized that other factors were also important in how humans perceive sound in a reverberant room, such as the ratio of direct to reflected sound and the time delay between the first arrival of a sound and the onset of the reverberant field.

A certain amount of reverb is usually considered pleasing for musical reproduction. The spectral content over time is one of the main factors that our ears use to define how a room "sounds." An overly long low-frequency decay can create the perception of a room as "boomy" while a long high-frequency decay time will be perceived as "bright." The absorptive and reflective characteristics of the surfaces in the room will contribute to this spectral balance.

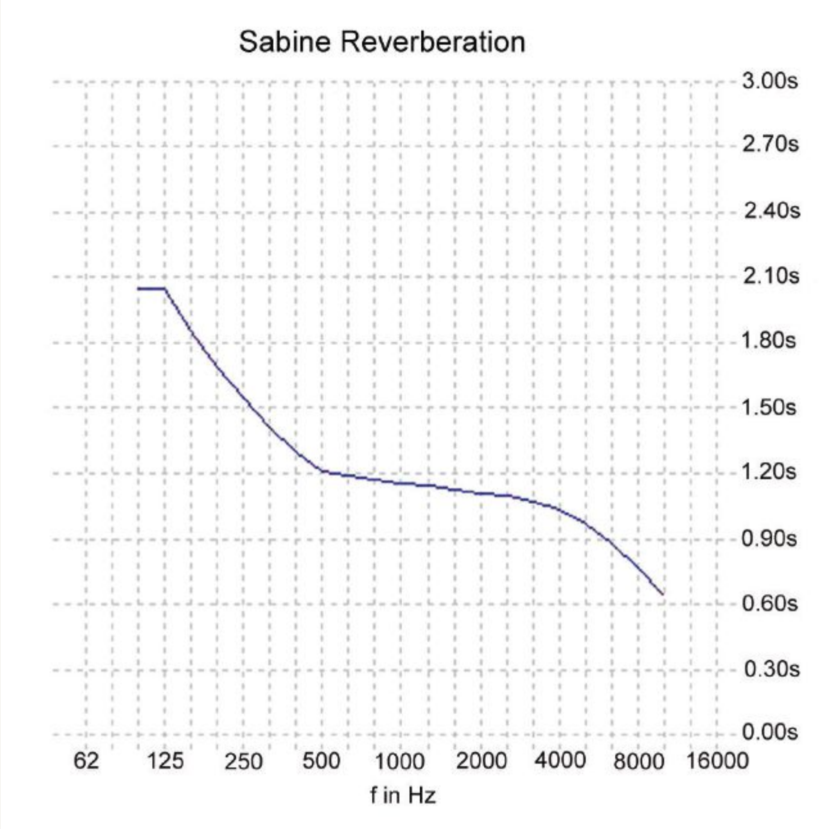


Figure 1 – Reverberation vs. frequency.

The most common way of expressing a measurement of reverb time is RT60. RT60 is the length of time that it takes for the reverberation created by an impulse in the room to decay 60 dB at a given frequency. A higher volume of the initial impulse creates a higher starting point for the decay but does not change the time it takes to drop 60 dB or the frequency content of the reverb tail. Usually (but not always) the reverb decay time at low frequencies is longer than at high frequencies. Figure 1 shows the RT60 response by frequency for a "typical" (if there is such a thing) live theatre.

An echo is a distinct spectral reflection from a non-absorbent or non-diffusive surface. When an echo repeats because it is bouncing between parallel surfaces it is often called flutter echo. Unlike reverberation which is diffuse, echoes are typically localized. The delay time to the first reflection as well as its strength and frequency content can be dramatically different in different parts of a room.

Like almost every other facet of audio, it's important to understand the wavelength component of frequency to understand how reflections behave. A surface's size governs which sound waves will reflect and which will wrap around an obstruction. For example, a 1000 Hz wave has a size of approximately 1 foot. If it encounters a one foot square surface that is reflective most of the energy will reflect back from the plane, and so will any higher frequency components of the sound that have shorter wavelengths. However, any longer wavelengths, say 100 Hz, will ignore the surface and continue on without reflecting.

EARLY & LATE

For the sake of more easily understanding the characteristics of an echo, they're often separated into the categories of early reflections and late reflections. There's a marked difference in the way human auditory response perceives the two.

If an echo is within approximately 50 milliseconds of the direct sound arrival, it's not heard as a separate arrival but it affects the perceived frequency response and the localization of the direct sound. This is widely described as the "Haas zone" or the "precedence effect."

If a listener is in front of and equidistant from two identical sound sources playing at the same volume, he will hear a phantom image halfway between the two sources. If the sound from one of the sources arrives slightly before the other. the sound will be perceived to come from the first arrival. Even if the level of the later arriving loudspeaker is louder, the ear will key on the first arrival as the sound source, although the localization may shift somewhat if the delayed source is much louder than the primary source (**Figure 2**).

Although this trait can be used to our advantage when using delayed loudspeakers to maintain localization on the apparent sound source, early reflections from side walls or surfaces behind the loudspeakers can also cause anomalies in frequency response. When they recombine with the direct sound from the loudspeakers out of phase due to the path length differences, they cause comb filters.

Late arrivals are much easier to identify and their effects are much more obvious. After about 50 milliseconds if they are high enough in amplitude the reflections are heard as discreet arrivals.

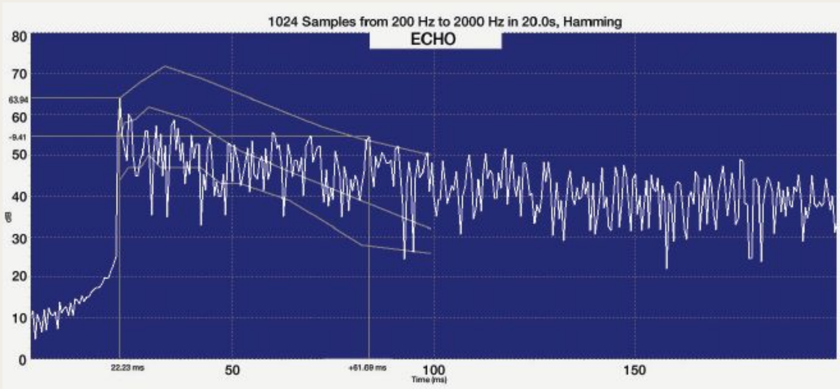


Figure 3 – The yellow cursor on the right is selecting a reflection that is late enough in time and high enough in amplitude to be heard as a distinct echo.

These echoes are very distracting for the audience and performers alike and can ruin an otherwise good mix (**Figure 3**).

Since we are rarely in control of the acoustics in the venue, what can we do to minimize the destructive effects of reflected sound? The first key is to be aware of the nature of the reflected sounds. We all spend countless hours mastering the intricacies of our digital console's signal flow, programming up presets for our favorite processing and learning to use the predictive software that allows us to maximize the benefits of line array technology. Since the room is a large contributor to the final results of any system deployment, we should spend time learning about acoustics as well.

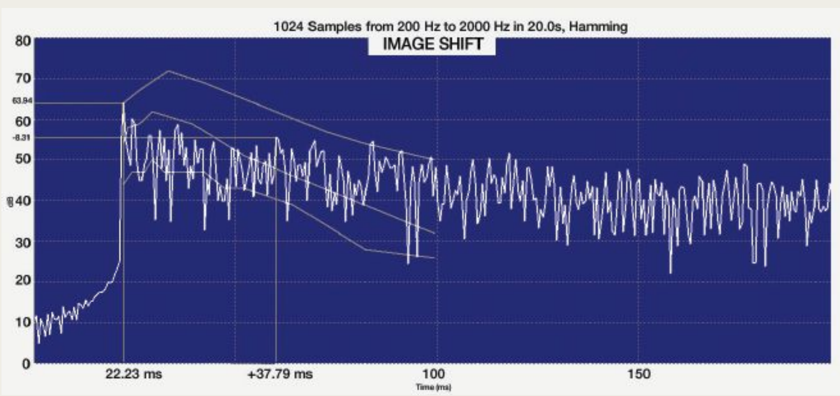


Figure 2 – An energy-time curve. The vertical axis is amplitude, the horizontal axis is time in milliseconds. The yellow cursor on the left is the first arrival. The gray lines represent the Haas zone. The yellow cursor on the right selects a particular reflection and the text at the top tells us what effect that reflection is having on the perceived sound.

EYE ON THE PRIZE

Modern loudspeaker systems have given us great improvements in pattern control, and one of the most effective ways to deal with reflected sound is to create as little of it as possible by using the modeling tools that the manufacturers are providing to minimize how much we are "lighting up" reflective surfaces such as big flat back walls and balcony faces.

When using software to optimize coverage in the audience areas, create room boundaries as well so that you can see how much you're exciting the beast. Many operators get great looking coverage maps on the seating areas but forget what the top half of the dispersion pattern is doing.

Remember, knowledge is power. Having the right gear is important but knowing how to deploy your equipment to create an optimal room/system interface is vital to the end result. ■

BRUCE MAIN *has been a systems engineer and front of house mixer for more than 30 years, and has also built, owned and operated recording studios and designed and installed sound systems.*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VIABLE OPTION

A fresh look at rechargeable batteries.

by Mike Sessler

I'VE BEEN USING rechargeable batteries in live production settings for more than eight years now, and writing about it for almost that long. In that time, I've come across three camps in the rechargeable battery debate.

The first group has used them for years, loves them and wouldn't consider going back to alkaline batteries. The second group has never used them, though most have heard the horror stories and are leery. The third group has tried them and had bad experiences. The phrase, "I've been burned too many times, I'll never trust them," is one I hear often.

This discussion is primarily for the second and third groups. If you've been burned, I'm going to explain why, and if you've heard the stories, I'll show you how to reach a different (and much happier) ending.

By way of introduction, I've tested many brands and types of batteries, and have gone through thousands of church services using rechargeable batteries without any significant problems. Yes, thousands of services. Far too many have had one bad experience, don't investigate the cause, and swear them off. But in my experiences, I've seen just as many Duracell Procells die at the wrong time as rechargeable cells. In almost every case, it's been my fault – I simply forgot to change them.

There are four primary things to know in order to get the most out of rechargeable batteries. Failing to understand and implement these things will guarantee bad experiences, just like trying to stretch an alkaline battery beyond its capacity. Keep in mind that rechargeable batteries behave differently than alkalines. Expecting them to be the same will only lead to frustration. But following these guidelines will lead to a successful result (and save a lot of money in the process).

START WITH QUALITY

This is one of the key factors in getting good results. In the past 10 years, a lot of people have gone to the local Walmart, picked up a $10 pack of four nickel-cadmium (NiCd) AAs and a charger, popped them into their wireless microphones and/or in-ear monitor beltpacks, and went down in flames. That's about 50 percent of the "bad experiences" I've heard.

Rechargeable batteries for wireless mics and IEM packs need to have a more modern chemistry – nickel metal hydride (NiMh), lithium ion (LiOn) or lithium polymer (LiPo). Further, AA batteries for these applications should be rated for at least 2500 mAh, and 2700 or 2850 mAH is even better.

The abbreviation mAh stands for milliamp hours, and indicates how much energy the battery can store. A Procell is rated at about 1800 mAh, while modern NiMh rechargeables from providers such as Sanyo, Powerex and Ansmann have ratings from 2700 to 2850 mAh. They absolutely blow Procells away – I've done real-world testing, and most NiMh batteries will out last a Procell by 25 to 50 percent.

Figure 1 shows my test results for sets of batteries we used every weekend at my church over the course of a year. In a Shure UHF-R wireless transmitter with an SM58 capsule picking up audio roughly comparable to a vocal, the Procells were spent by 8.5 hours, while the "worst" NiMh batteries came in at more than 12 hours. (And these batteries were a year old at the time of testing.)

PROPER CHARGING

The second key component of rechargeable batteries is proper use of a quality charger. A cheap "rapid" charger will not fully charge the cells, and it can also overheat them, shortening their life. Modern "smart" chargers are readily available, easily affordable and fully charge the batteries at the correct rate. They will also "trickle charge" the cells to keep them at peak capacity.

So what is a proper charge rate? Most manufacturers recommend a charge rate that's between 0.5-1.0c. That is to say, the

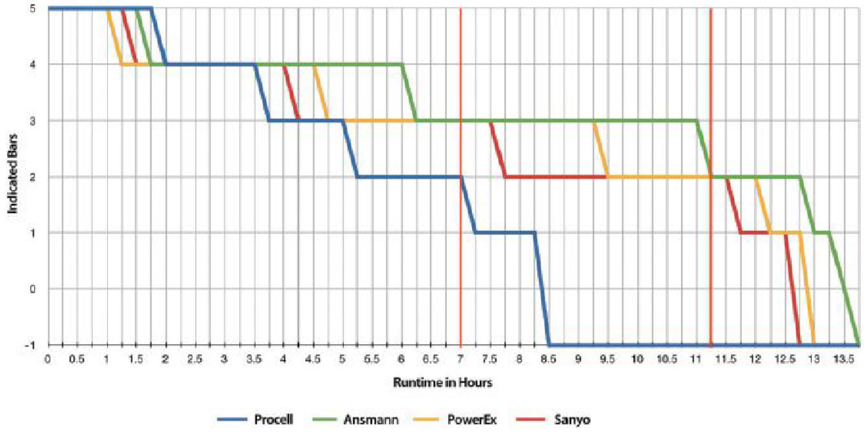


Figure 1: Test results compiled by the author over the course of a year show used rechargeable batteries last longer in use than new alkaline cells.

charge rate should be one half to full capacity of the battery. So if a battery is rated at 2000 mAh, the recommended charge rate is 1000 mAh and the max charge rate is 2000 mAh. A full charge will take from 1 to 2 hours at those rates, respectively.

I've had excellent results with chargers from Maha and Ansmann. Once the batteries are well charged, it's important to know how to utilize them properly. And that brings us to …

PROPER CYCLING

To ensure good results – "good" defined as the mic not dying in mid-use – it's important to use the batteries properly. Fully charged batteries should always go straight from the charger to the mic/beltpack. I like to have enough batteries and charging slots to keep one full set on the chargers while the other set is in use.

At my previous church, we had a Saturday set and a Sunday set. This helped rotate the stock, and we always had a full set ready to go in case something went wrong. It rarely did, however. We always went directly from charger to mic, and at the end of the service, back to the charger.

Note that NiMh batteries will self-discharge over a period of 30 to 60 days. So while you may not lose a lot of capacity from Sunday to Saturday, you'll be down 10 to 20 percent or so. By keeping a set on the charger all week, you'll always have fully charged batteries to work with.

Some users are concerned about the "memory effect," the loss of capacity that happens when NiCd cells are recharged before being fully depleted. Meanwhile,

NiMh cells have no significant memory effect, so charge them when you're done using them –don't run them down too low during a show/service to try to cycle them. Good chargers include a refresh cycle that will fully discharge each cell, then fully charge it again. It's a good practice to do this every 3 to 4 months. The procedure will prolong the battery's life and ensure top performance.

DISCHARGE CURVES

Another factor from the "I've been burned" camp is the different discharge curves between alkaline and rechargeable batteries. An alkaline drops off in a pretty linear fashion. Rechargeables, on the other hand, quickly drop from full voltage to about 75 percent, then hold there for a long time. When it drops off the second time, it drops pretty quickly.

Many battery meters in wireless mics are calibrated to the discharge curve of an alkaline. As the voltage drops off, the meter can predict approximately how long the battery is likely to last. However, as we just noted with rechargeables, the voltage holds and then falls off rapidly. This is why it's not uncommon to see a rechargeable go from 60 to 0 percent within 5 minutes – the meter is not calibrated to read it accurately.

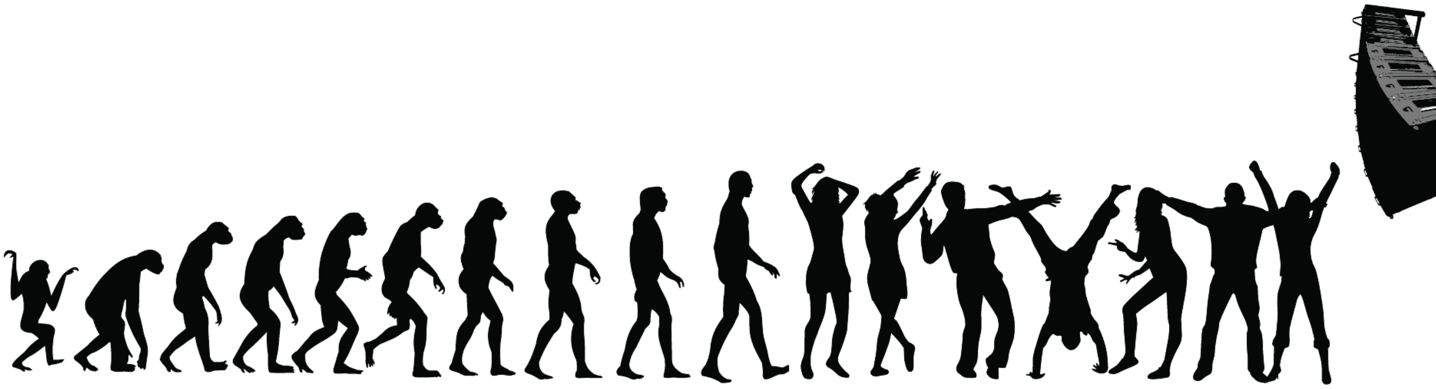
Some newer mics have battery meters that can be switched between alkalines and rechargeables, and it almost goes with out saying that these should be used. However, there's still no substitute for doing testing to find out how long rechargeables will last in your specific mics. This provides valuable information on when to change them, regardless of metering.

For example, I know that Ansmann 2850 mAh batteries will run a solid 12-plus hours in our Shure UHF-R mics. I feel confident putting them in on Saturday afternoons for rehearsal and letting them run through the end of services (about 4.5 hours total). They're then replaced with a fresh set on Sunday morning, running through the end of services (about 5 hours total). If one goes down during a service, it's a fluke and/or the battery is at the end of its lifecycle and needs to be replaced.

One more thing to note is that many of the premium wireless manufacturers are now offering rechargeable batteries and chargers, and these are providing excellent results as well. So if you've been burned, perhaps the information presented here will prompt you to give modern rechargeable batteries another shot. By following guidelines for proper use, I'm confident you won't be sorry – and your budget will thank you as well. ■

MIKE SESSLER *works with Visioneering, where he helps churches improve their AVL systems, in addition to encouraging and training the technical artists that run them. He has been involved in live production for more than 25 years.*

And then there was MLA...



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SERIOUS ADVANCE WORK

Selecting vocal microphones for "one-off" performances.

by Craig Leerman

IN LIVE AUDIO PRODUCTION, choosing a vocal microphone for a singer you've never worked with can be a bit challenging. The "correct" choice is the one that complements the voice and how the particular singer "works" the mic. Another factor is providing a comfort level for someone who may be wary of using an unfamiliar mic, which shouldn't be overlooked – a timid, unsure performance usually isn't a good one.

Let's first look at what we have to work with. Mics come in many types, including dynamic, condenser and ribbon. Dynamics are the most common type for live vocals because many models offer good (and better) performance characteristics in a rugged package. Condensers have become increasingly popular in the live realm because of an often more-detailed high-end response along with a nice full sound across the frequency spectrum. They've also become much more durable, able to handle the usual bumps and jolts that occur on stage.

The same goes with ribbons, which were usually relegated to the studio due to fragility and lack of ability to handle high sound pressure levels. And, most were only available with a figure-8 pattern. That's changed as well, with ribbons able to withstand normal live use while being available in directional patterns that are a necessity on louder stages. Some engineers describe ribbon mics as sounding "warm" and natural, but some models may not exhibit a flat high-frequency response like condensers.

Speaking of patterns, directional patterns are usually the first choice for vocals because they reject the often considerable output of stage wedges, fills, instrument amps, drums, etc. Cardioid is probably the most common pattern for singers, offering a wide pickup pattern but with good rejection off to the sides and rear. Hypercardioid and supercardioid patterns also work well for singers, but may be more affected by poor mic technique – not singing directly into the mic and/or holding it, at least in part, so that the element is obscured. A subcardioid pattern can be a good choice in this regard, with a wider pattern that still offers some rejection at the sides and rear.

Omnidirectional patterns are usually not advised for handheld vocal applications, but they're useful for lavalier or other miniature mics for theatrical performances. Singers in this genre may also be outfitted with headworn miniature mics that mount around one or both ears and use a small boom arm to position the mic closer to the mouth. A similar approach is preferred by concert singers who also dance, except these mics tend to be outfitted with a larger mic element located close to the mouth since the performers are not trying to hide the fact that they're using a mic.

SMART CHOICES

With so many different types, styles, models and brands, it can be a bit daunting as to where to start. The first thing I do is advance the show, find-

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Try to account for less-than-optimum mic technique.

ing out who is performing and if I can speak with them (or their audio folks) in order to get an idea of preferences and expectations.

Sometimes the internet is handy as well, with artist websites and YouTube offering videos/photos of them in concert that can offer clues on their mic choice(s) as well as specific vocal style and mic technique. The bottom line is that I don't want to get to sound check and waste their time and mine in checking out a dozen different mics, and I don't want to carry a ton of extra mics to the show to start with. Advancing the show is the way to go.

Since I've been in business for quite some time, I'm fortunate to have amassed a pretty decent selection of vocal mics, but even if you only have a few choices, there are (generally speaking) certain types that work better for certain singers. For example, for a singer with good mic technique, I start with hyper- and supercardioids in order to attain as much gain before feedback as possible, especially if the stage is going to be loud. For those preferring a wireless handheld, my choices narrow significantly because I only have six different mic elements to choose from in my inventory – and all except two of them are cardioid. Still, this usually has sufficed.

With selections narrowed down, I take the most likely models to the show, and first test each one through the PA and monitors. I then present the one that's best-suited, and if I'm undecided, let the singer try out the two choices and make the final selection.

Every mic has its own sonic signature, and knowing the frequency response and proximity effect (a characteristic of a directional mic when used close that tends to accentuate the bass frequencies) of various models will help in selecting one that flatters a particular voice. Also note that some mics exhibit a fair bit of handling noise. This may not be a problem for a singer in a heavy metal band on a loud stage, but it can be quite noticeable (and distracting) for a folk singer holding the mic on a very quiet stage.

RELATED ISSUES

Most vocalists on my stages get handed mics with an additional foam windscreen. I'm a big fan of these for a several reasons. First, they help in blocking wind noise, which is a particular problem outdoors. (I put windscreens on most/all mics when outside.) Second, they can help lessen the problem of "plosives" (hard consonants like P, B, T that can really pop).

Windscreens also come in various colors, so on a festival or a show with numerous singers, they help my crew to identify what mics are being used. Finally, windscreens help with "hygienic issues" – they can be quickly replaced between acts.

"Allowing singers to hear what they need in order to be in time and on key boosts confidence."

Speaking of hygiene, it definitely should be standard operating procedure to make sure mic grills aren't caked with crud and grime. It's gross for the performer, unprofessional of the provider, and the gunk can also change the sound and frequency response of the mic. I carry a basic cleaning kit to shows, consisting of a toothbrush and Listerine, to perform a quick clean-up between acts (when we're not using windscreens).

To clean, we unscrew the grill, remove the inner foam with needle nose pliers, and then scrub the grill with Listerine on the toothbrush.

Narrow down the choices, and note that sometimes windscreens can be your friend.

It's best practice to also provide optimum stage monitoring.

Blow any excess liquid off the grill with a can of compressed air, towel dry it, and screw it back on. If the inner foam has seen better days, we replace it with one of the spares we bring to the gig. The used foam can be washed in soapy water back at the shop and then dried for re-use – don't put damp foam back on the grille because it can corrode metal parts.

Microphome, a foam disinfectant/deodorizer, is a handy alternative to the toothbrush and Listerine method. The foam evaporates in about two minutes, allowing the mic to be refreshed and ready for use pretty quickly.

THE STAGE SOUND

Choosing the mic is not the only piece of the puzzle. Good monitoring practices are crucial for vocal performances, and it also means singers aren't as likely to blame the mic for any perceived deficiencies. Allowing singers to hear what they need in order to be in time and on key boosts confidence. It also reduces the need to sing loudly, which can strain the voice in trying to "get over" the other things onstage so they can hear themselves. In-ear monitors are great, but the majority of one-offs we work still rely on stage wedges.

For vocal monitors, I usually roll off the low end between 80 Hz and 125 Hz, depending on whether it's a male or female singer. This eliminates low frequencies bouncing around the stage and bleeding into open mics.

Placement of wedges is usually at the discretion of performers, but if they don't have a stage plot or preference, I usually provide two wedges on a single mix and space them a few feet apart to provide a large "sweet spot." For festivals, especially ones with amateur performers or track acts, our go-to is to place four to six wedges across the front of the stage on the same mix. That way we provide good coverage for vocalists who want to wander about as they perform.

Stands are another part of the equation.

Not all wedge angles work for everyone, so we use "Acoustic Aiming Devices" (a.k.a., black-painted pieces of wood) to make sure the angle can be tailored for specific performers. It's best for vocalists to be fully in the wedge pattern so they're hearing the entire mix, not just the woofer or the horn.

Vocalists who walk all over the stage may be better served with side fills shooting across the stage to provide a "wash" of sound. Just make sure the singers are aware of them because if they wander too close, feedback can ensue.

ONE MORE THING

Mic stands are important as well. They come in two primary types, round weighted base and folding tripod style. I carry both styles to gigs because a singer may also want to use the stand as a prop, and I want them to be comfortable.

Regardless of type, make sure that any rubber feet are firmly in place so the stands don't wobble or transmit stage noise and rumble up into the mics. Also check the clutch mechanism to verify it's in working condition. Many vocalists like to play with the adjustments while they sing, and a broken stand can be a real distraction from their performance.

I prefer rubber clips to attach mics to stands because they don't break as easily, and "seem" (not verified) to transmit less noise to the mic element. For very noisy stages, I also carry a few isolation clips that can help eliminate most of the stage vibrations from reaching the mic. ■

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

Looking out into the seating areas from the DeWitt Theatre stage.

MAKING IT INVISIBLE

Sonic quality that's "out of sight" at the DeWitt Theatre. ***by Keith Clark***

MORE OFTEN THAN NOT, there's more than readily meets the eye when it comes to purpose-designed-and-built sound reinforcement systems in performing arts venues. Certainly these systems include traditional audio components and concerns, but they also come with a host of structural and infrastructure issues that present additional challenges – often formidable – to the system design and installation team.

Case in point is the culmination of a system project for the Main Theatre at the DeWitt Center on the campus of Hope College in Holland, MI. Originally opened in 1972 and renovated 25 years later, the 464-seat venue offers both thrust and proscenium configurations, hosting rehearsals and performances by the college's renowned theatre department as well as a variety of regional groups.

It's an intimate space affording excellent sightlines, with gently raked main floor seating on three sides of the thrust stage (which can be removed) and a balcony (also with raked seating) opposite the proscenium. Devoid of acoustic treatment, all wall surfaces are painted a very dark gray, and there's no artistic adornment around the proscenium, signifying the stage as the primary focus. A grid of catwalks and support structure spans the front half of the room, about 30 feet above the main floor, and it's here where most of the venue's production equipment resides, including audio components and lighting fixtures.

Resident sound/lighting designer Perry Landes, who also serves as facilities manager in addition to being an associate professor of theatre at the college, worked in tandem with Ann Arbor Audio to develop and implement a new main loudspeaker system within this space. Led by John Malek, the design-build firm, which is based in Brighton, MI, brought more than 30 years of experience to the project, proficient in sound design, installation, acoustics, modeling, measurement, and as we'll see, much more.

SERIOUS HOMEWORK

The loudspeaker system is the culmination of a 5-phase multi-year sound system replacement project at the theatre, involving infrastructure upgrades, power conditioning, source and signal processing devices, wireless microphone systems, hearing impaired systems and communications.

Landes had done a lot of homework on loudspeaker solutions before Ann Arbor Audio was brought on to the project, formulating a design concept of either main left/right or left/center/right line arrays, accompanied by a voice ring of loudspeakers. "John and I were able to get to realizing that plan quickly, which was one of the truly marvelous things about working with him," he notes. "After spending time there with me and looking at the challenges, he basically agreed with the ideas I'd come up with in my 25-plus years at Hope."

In presenting an informed, comprehensive proposal, Ann Arbor Audio first performed an analysis of coverage and structural issues, using EASE Focus 2 modeling along with, as Malek notes, "a whole lot of CAD drawings." All of this work paid off, resulting in a better grasp of the true scope of the project.

A key factor was loudspeaker location; they were to be heard but not seen, expected to deliver full-bandwidth coverage to all seats while residing at the catwalk level, at or above the top of the proscenium. But first, the specific loudspeaker types, and then models, needed to be selected, with several top candidates evaluated within the space. Sonic quality was a primary factor, along with coverage patterns fitting the unique configuration and issues such as overall footprint, rigging, and adaptability.

Alcons Audio loudspeakers emerged as the consensus choice from this pro-

(Above) John Malek of Ann Arbor Audio at the system rack in the catwalks that contains new Alcons Audio Sentinel 3 amplifiers/processors. (Below) Resident sound/lighting designer Perry Landes at the Yamaha M7CL console at front of house.

cess. They're noted for a proprietary high-frequency approach that utilizes a ribbon driver in tandem with a "Morpher" lens to provide almost total frontal radiation, resulting in a seamless cylindrical (Isophasic) wavefront. "These are very precise loudspeakers, with the ribbon transducers resulting in absolutely no coloration," Malek states. "The additional result is a wallpapering of sound, a 'wall of sound' – as opposed to lobes of sound – that is consistently solid and present."

Specifically, the Alcons LR7 (4-inch ribbon, 6.5-inch cone, vented) line array platform and footprint played into a key design aspect on the project "wish list," which was to locate main left and right arrays within custom-fabricated acoustical chambers attached to a 1-inch thick steel I-beam running directly behind the proscenium arch. The location is optimal for evenly distributing sonic energy to the majority of the venue.

Working with an engineering firm, Landes had done an analysis to verify that large-enough openings could be cut out of the beam without compromising its structural integrity. These cut-outs would be backed by custom-built chambers housing the arrays. All of the data, as well as the dimensions for the arrays (including enough depth to accommodate a "J" array shape), was provided to the engineering firm, which assembled the custom chambers, constructing them of very dense wood that's both cross- and diagonal-braced.

The chambers were secured to the I-beam by the engineering firm, followed by the sound team using the contractor's scissor-lift to hoist and assemble the arrays. The painstaking preparation proved to pay off, with the arrays fitting within the chambers just as intended, their positioning optimized with final mechanical adjustments.

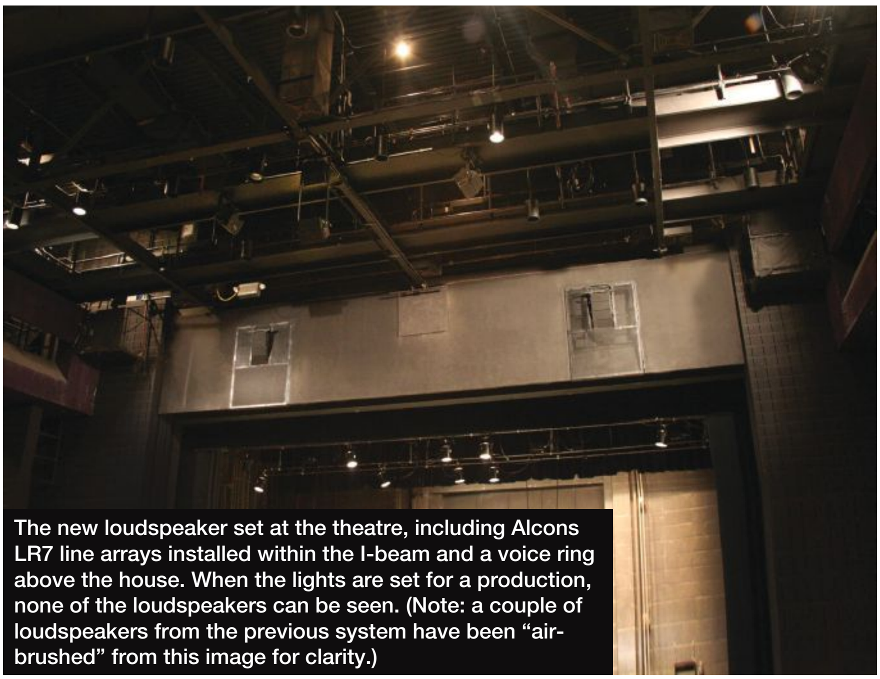
"The success of this project really started with getting very accurate measurements, and then translating them to the CAD realm," Malek says. "We had to be meticulous and precise, including exact data for all of the custom mounting frames, speaker brackets and rated hardware associated with the arrays."

Alcons LR7 modules are very compact, measuring 6.9 x 14.1 x 10.7 inches (h x w x d). For this project, modeling had shown optimal coverage could be attained with arrays of six modules, with 90-degree horizontal dispersion for the (longer throw) top modules covering the balcony and 120-degree horizontal dispersion for the (wider throw) lower modules covering the main floor.

The full-range loudspeaker complement is completed by a "voice ring" of additional loudspeaker modules optimally positioned via custom fabricated steel mounting structures attached to the catwalk grid – dual sets of LR7s for the balcony and main floor, as well as each of the side seating sections.

COMPLETE ISOLATION

Subwoofer selection and location proved to be another challenging facet of the project. These too were required to be "up and out of the way," but the I-beam couldn't be further modified, and the

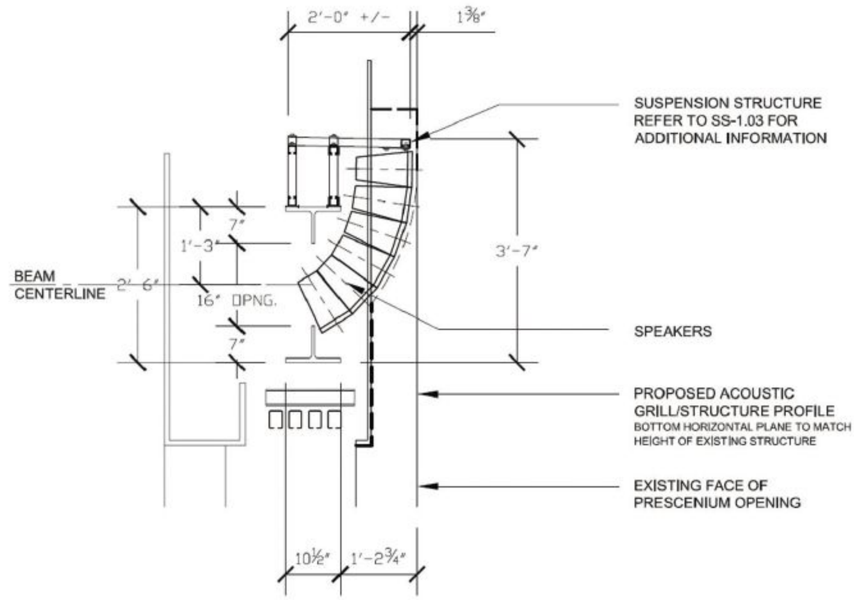


The new loudspeaker set at the theatre, including Alcons LR7 line arrays installed within the I-beam and a voice ring above the house. When the lights are set for a production, none of the loudspeakers can be seen. (Note: a couple of loudspeakers from the previous system have been "air-brushed" from this image for clarity.)

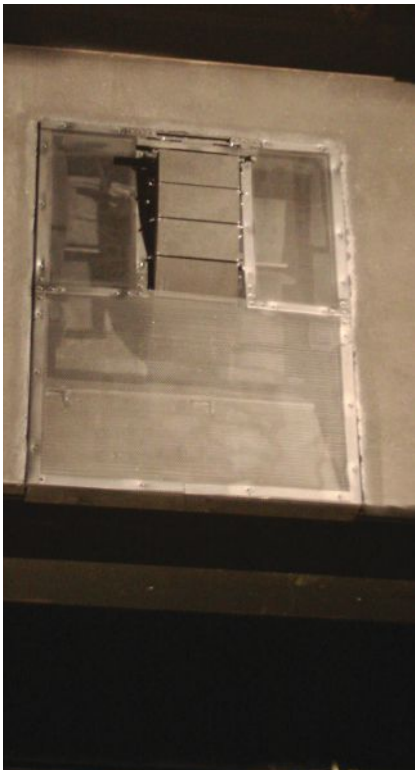
structure immediately above it was too congested. Malek scouted for alternative locations, considering various subwoofer models and configurations, again using modeling to help predict results.

A centrally located position, both in relation to the catwalks as well as the theatre below, was identified and found to be feasible from an acoustic perspective, and further, it didn't impede or otherwise impact any of the lighting fixtures and other elements in the grid. However, it too needed infrastructure, this time in the form of a custom frame of back-to-back channel strut beams across the span.

The steel frame was designed to accommodate four Alcons BF115i (install version) subwoofers, each with a single front-loaded, 15-inch long-excursion cone driver, tightly packed, two next to two, firing downward. Every surface of the frame that comes in contact with a subwoofer is lined with 1/4-inch rubber gasket so that



A drawing showing the structure and arrays, as well as plenty of other details.



A closer look at one of the line arrays in the I-beam.

there's no metal to wood vibration. Further, heavy-duty springs at the attachment points to the beams serve to provide additional vibration isolation from the surrounding structure.

"From day one, wanted to make sure this didn't cause any problems and so went all-in with the structural design," Malek notes. "It's completely isolated from the structural steel of the building, so it's absolutely silent in terms of any additional noise and sympathetic vibrations."

Getting these larger cabinets in place at the ceiling level wasn't exactly a cakewalk. They were brought up to the catwalk level via a tall scaffolding erected on the stage, and from there were wrangled into the steel frame. It also shouldn't go overlooked that a whole lot of planning and subsequent work went into cabling the entire loudspeaker contingent.

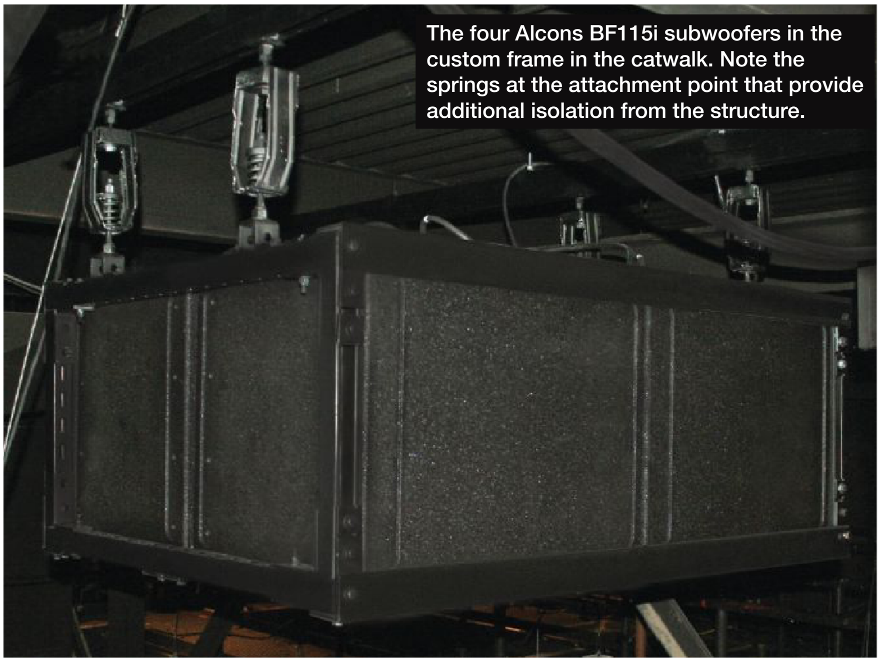
Four recently released Alcons Sentinel 3 amplifier-processors drive all loudspeakers, mounted in a rack positioned by the main door to the catwalk area to keep cable runs at an acceptable length while also being easy to access. All parameters can be dialed in and accessed via the front panel on these units, and they can also be remotely adjusted and controlled via networked PC.

At the heart of each 2RU Sentinel 3 is a SHARC digital processor joined by proprietary class D amplifier stages. It has 192 kHz capable AES/EBU inputs, custom-designed sample-rate conversion, and remote-selectable input sensitivity. Loudspeaker DSP functionality includes a 6-band parametric per channel, delay, and factory presets for all Alcons systems and configurations.

"Sentinel is just a phenomenal package, and it's perfectly matched with the loudspeakers," Malek states. "There aren't many amplifier/control options with this type of capability, and they significantly enhance clarity."

SMOOTH TRANSITIONS

The system's remaining primary component is a Yamaha M7CL digital console that anchors the front of house position that's centrally located on the main floor. Landes notes that this unit continues to



The four Alcons BF115i subwoofers in the custom frame in the catwalk. Note the springs at the attachment point that provide additional isolation from the structure.

perform well, but may be subject to an upgrade to a newer model at a later date.

Via an option card on the console, all audio signal (12 channels) is sent via AES/EBU to the Sentinel amplifier/controllers. "It's a very crisp, sharp, clean way to handle the signal, especially in relation to analog," Malek says. "Digital not only enhances things sonically, especially with respect to speech intelligibility, but it's also cut way down on the infrastructure that would have been required to run analog signal cabling through this EMI & RFI jungle that comprises the catwalk space."

The digital advantage also applied to the system tuning and optimization process, with the sound team able to make adjustments from within the venue rather than needing to run back and forth to the racks or posting someone full-time at the racks and communicating via radios.

"We performed extensive measurements – edge to edge, top to bottom, across the balcony – and there's very little variation in the overall sound pressure level of the entire area," Malek says. "It's amazing how smooth it is. With compression driver/horn systems, you typically see a lot variations and major notches due to coupling in the crossover range, as well as the coverage of the horn versus that of the woofer as they narrow in the upper range of the crossover – it's the typical 'wooshing' sound we've all heard while walking through coverage. This isn't as noticeable when you're seated, but it's still a factor.

"Then in the effort to eliminate this lobing problem, you typically work it, and work it some more, crossover points, slopes, delays and so on, but it's a challenge and rarely feels quite 'done'," he concludes. "What we've got here is invisible transition between the loudspeakers, appropriately as invisible as the actual loudspeakers are to the eye. It's what the ribbon tweeter brings to the occasion, and in our view, it's spectacular." ■

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

INNOVATION & SERENDIPITY

The development and capabilities of multichannel amplifiers. *by Frederick J. Ampel*

THE TWO DECADES FROM the mid '60s through the mid '80s were a time of evolution and experimentation in the professional power amplifier world. The key factors at work during this timeframe were development of higher power solid-state devices and the maintenance of the mono or 2-channel status quo configuration. The available power per channel rose slowly towards the 500-watt mark and then exceeded it as newer high current power devices appeared, but the mono and stereo framework stayed put.

It was a situation that needed a better solution. The amount of space taken up by power amp racks kept rising exponentially as larger demands were placed on sound system power requirements. Monster tour systems were driven by amps housed in dozens of road cases, and mega stadium installs had rack room(s) full of several hundred amps.

Weight also became a serious problem (especially on the live/tour side of the business) as power output increased. If you wanted higher power, you needed a bigger power supply to produce it, meaning bigger and heavier transformer(s) and larger capacitors to provide that raw current and voltage. The immutable laws of physics just wouldn't get out of the way, yet neither would the demand for more power.

Everyone knew there had to be a better way to achieve these goals, but it wasn't until the mid '80s that a set of solutions combined to answer the key questions.

LOGARITHMIC JUMP

As is often the case with major technological leaps, it took two separate developments and the fortuitous timing of one of them to bring about the next step in power amplifiers: the move from 1 or 2 channels to 4, 6, and 8 channels in the same or just barely larger chassis, and the logarithmic jump in output power.

But the beginnings of this major shift were quietly hidden in the Swedish countryside just outside the small city of Kungsbacka. In 1979, working out of a local electronics repair shop, Kenneth Andersson and Dan Bävholm founded what would become Lab.gruppen. The first products were not amplifiers, but consoles. It wasn't until 1986 that the first of the two breakthroughs occurred.

Twenty years after the Crown DC300 forever changed the perception of solid state (transistorized) "high power" amps, Andersson and Bävholm developed the Regulated Switch Mode Power Supply (R.SMPS). The basic concepts behind a switched mode power supply were known at the time, but difficult engineering challenges had prevented successful implementation in high-power amps; the components required to make it work were not yet available. It would take the second and serendipitous development to make the idea feasible.

Engineers at Siemens in Germany had devised the ability to produce high-current-capable Metal Oxide on Silicon Field Effect Transistor (MOSFET) devices, although the target customers were not the small pro audio industry's power amp manufacturers. Nevertheless, these devices allowed Andersson and Bävholm to implement a new amp topology and introduce the 2-channel SS 1300, followed by the first true high power multichannel amp, the 1200 Quatro, a 4-channel model capable of generating 380 to 450 watts per channel from a single 2RU chassis.

This was followed by the even higher power SS 1400, which also led to the development of more advanced finned heat sink, forced air cooling tunnel designs to enhance stability and reliability. How-

The 2-channel Lab.gruppen SS 1300 with Intercooler technology.

ever, this design had an inherent flaw: increasing temperature as air moved through the tunnel resulted in uneven cooling of the output devices.

The problem led to development of the Intercooler (first implemented in the SS 1300), wherein the heat-sink comprises thousands of tiny copper fins that dissipate heat far more efficiently than large aluminum fins. Also, the Intercooler – with output devices embedded – is mounted transverse to the airflow, so all devices benefit from uniform cooling. Like its automotive and aircraft engine namesakes, this technology allowed the rapid development of amps where the heat produced by 2, 3, and 4 kW worth of output devices could be effectively managed.

Above: QSC PLD 4.5. Below: Ashly Audio nX 3.04.

WE HAVE LIFT-OFF

It didn't take long for power amp manufacturers around the globe to develop and introduce their own unique variations on this theme. Within less than five years there were no less than a dozen high-power 4-channel products available, and very rapidly after that, 6- and 8-channel variations.

Suddenly it seemed as if a single amp chassis capable of 10 kW was no big deal. Well, actually there were some significant birthing pains, mostly centered around handling the massive current draw placed on mains power feeds. It took a while to develop the needed hardware and software to effectively manage that set of issues, but the upside is that it also produced "smarter" amps with built-in intelligence and eventually DSP capabilities.

So in the end, the initial pain was a positive force in producing a new level of capability. But as with any new technological development, there are benefits and cautions to be aware of.

Crown Audio I-Tech 4x3500HD.

To expound on those issues we asked several manufacturers to address what they see as important technical challenges of multichannel amps.

Marc Kellom, senior director, engineering and marketing, **Crown Audio**: "As channel count increases, the number of components and interconnections increase, and the number of 'eggs in one basket' also increases. A failure in a common power supply will take out more of the overall audio system than in the days of 2-channel products. An 8-channel amplifier conceivably has 4 times the opportunities for a defect compared to the 2-channel version of the same product. Internal temperatures are also increasing as power density goes higher.

"Our proprietary DriveCore technology is designed to directly address this challenge through integrating hundreds of previously discrete parts onto a single device. By tapping into

Within less than five years there were no less than a dozen high-power 4-channel products available, and very rapidly after that, 6- and 8-channel variations.

the expertise developed by the automotive group of Harman, we have access to extensive information and tools to help enable our reliability to dramatically improve even as multichannel amps become the norm."

Matt Skogmo, director of hardware engineering, **QSC**: "If there's one common trend in the install and contractor market space it's that installations are getting more complex, not less. One strong theme in this trend is the mix of high performance and distributed audio. It's very common to have portions of an install that are geared toward higher output, higher fidelity, low impedance loudspeakers, and other portions that are geared toward 'acres of speakers' or 70/100-volt distributed systems."

To deal with these requirements he notes that QSC has come up with a method of reducing the number of amplifiers needed to address a variety of power points. Called Flexible Amplifier Summing Technology (FAST), it allows channels to be configured in both bridge and parallel mode.

"For example," he says, "you can bridge channels (doubling the available voltage), or you can also place them in parallel (doubling the available current). These options are not limited to a pair of channels. This means that amps equipped with FAST can be deployed as a 4-channel, 3-channel, or 2-channel amp, or even a mono block. Basically it means that no matter the load impedance – from 100 volts distributed all the way down to 1 ohm – a single amp can be configured to deliver maximum power."

Mike Updaw, Eastern sales manager, **Ashly Audio**: "As you increase channel count you need to consider many issues, such as heat, wattage output, consump-

Above: Lab.gruppen PLM20000Q. Below: Powersoft X4.s

tion, rack space, and options such as remote control or digital transport, to name just a few. Our research has shown that utilizing class D designs, Ethernet bi-directional data transfer capabilities, and integrating selectable output impedances into an amp are effective solutions to many of the most common problems in the field."

Klas Dalbjörn, product research manager, install & tour, **Lab.gruppen/Lake**: "As more amp channels are packed into a single product, a major challenge is just being able to fit everything inside the product. On top of this, the internal cooling solution is critical as there are so many hot spots inside a densely packed multichannel product. In our 4- and 8-channel designs, we've used front-to-back parallel cooling to ensure that all channels are effectively cooled equally."

He adds this important point: "When packing amp channels so the summed output power together with the internal losses exceeds what you can continuously pull from the mains outlet, it's essential to both have good internal energy storage to avoid pulling the peak power from the mains as well as a good-sounding 'mains current limiter concept' that avoids the risk that the amp will trip the mains breaker after a few seconds/minutes of extreme program material. Our approach is to design in different schemes to avoid the risk. Over the years these have been called AFS (Automatic Fuse Saver), PAL (Power Average limiter) and BEL (Breaker Emulation Limiter)."

Claudio Lastrucci, R&D director, **Powersoft**: "All of our amps implement fixed frequency (FF) switch-mode technology for both power supplies and output stages. Any stage in an FF switch-mode amp is driven and synchronized by a global fixed frequency signal; this entails a more complex electronic design with respect to a variable frequency system (where the switching frequency of each stage is independent from each other), but it guarantees lower crosstalk and perfect matching between the power supply and the output stages."

Editor's Note: Be sure to check out the full version of this article on ProSoundWeb, easily accessible by simply entering "multichannel" in the search tool. ■

FREDERICK J. AMPEL *has been involved in the pro A/V industry for nearly 40 years. His career has included work in live sound reinforcement, broadcast audio production, systems design and installation, systems integration, hardware design and development, and residential small room acoustics.*

THE SMALLER THEY ARE...

Detailing recent compact loudspeaker applications. ***by Live Sound Staff***

LARGE-FORMAT loudspeakers get a lion's share of the glory, and while it's true that the "big guns" often deliver impressive big PA results, there's a world of models of a more modest scale that are invaluable in meeting the needs of numerous critical applications. Let's take a look at some recent projects.

RMB AUDIO, SELMA, NC

Front of house engineer E. Wayne Sowder reports on some ongoing application work with Martin Audio MLA Mini line array enclosures, in affiliation with NC-based sound company RMB Audio: "When RMB added MLA Mini enclosures to an inventory that already included the MLA Compact system, the sound techs started evaluating the system using the factory presets for basic 'speaker on a stick' configurations," Sowder explains. "It wasn't until RMB tech Matt Johnson began testing the VU-NET network features that we discovered the software had presets for 'single' and 'double' front fills.

"Typically, we'd used Martin Audio W8LM cabinets equally spaced on the downstage edge. Substituting the powered MLA Mini speakers for lip fills would allow the entire system to be controlled on the same network, saving setup time and facilitating control, eliminate the need to carry an amp rack, occupy a smaller footprint on the stage and in the truck pack.

"At the time, the next show requiring lip fills was at the Red Hat Amphitheater in Raleigh, NC for the 'Rock and Roll Marathon.' We decided to use the Mini in the single configuration since it was the closest to our normal front fill package. The first thing we noticed was the voicing similarity between the MLA Compact main hang and the front fills. Walking from the coverage of the main hang into the coverage of the front fills, the clarity remained constant without adding additional EQ. Moving onstage, we found that the Mini also mirrored the excellent rear rejection characteristics of larger MLA systems.

"We chose to deploy the Minis in double-stacked mode for artists with a louder stage volume, such as Parliament Funkadelic and J Roddy Walston & The Business (pictured above). The results were impressive; using the 'double' program in VU-NET compensated for any variations caused by adding additional cabinets and provided a front fill package that caused several FOH engineers to comment that they'd mixed shows where the mains weren't that strong or sounded that good."

DAVID C. COOK BUILDING, COLORADO SPRINGS

Sight and Sound Technologies of Colorado became the first to deploy new PreSonus Dante-enabled loudspeakers, configuring a left-center-right system (pictured below) using a Dante-equipped mixing console with three StudioLive 315AI Active Integration loudspeakers, each with the new SL-Dante-SPK option card, at the Colo-

rado Springs publishing building of nonprofit global ministry and Christian resource provider David C. Cook.

"This space is used for many different applications, including business meetings, overseas live video conferencing, and chapel services," notes Sight and Sound Technologies chief operating officer and co-founder Kris Johnson. "[Christian record label] Integrity Music, a division of David C. Cook, will also use the space as an artist showcase venue. So we had to get creative in the way we designed the system.

"We wanted enough power to crank out some of Christian music's top artists, yet be able to simplify and dial back the sound for a staff meeting," he continues. "The PreSonus StudioLive 315AIs were a perfect fit. With the Dante card in the mixer and the Dante cards in the speakers, it couldn't have been easier to route the audio. The rear of the speakers is so clean, with just a Cat-5 cable and a power cord."

He adds that the loudspeakers' SL Room Control wireless control software "required no instruction. It's very intuitive and easy to connect and use."

ALDEN HALL, WORCESTER, MA

This mulitpurpose venue at Worcester Polytechnic Institute (pictured above) regularly hosts a wide range of performances and programs, from lectures to concerts and theatrical productions. As Richard O'Connell of Boston-based AV Design Build explains, the 10,000-square-foot, rectangular venue employs different seating arrangements, depending on the event. The hall's growing popularity and increasingly busy schedule had made clear the limitations of its aging sound system, with university officials frequently bringing in portable systems to supplement or replace the main system.

With this in mind, AV Design Build recommended a solution based on dual independent Renkus-Heinz ICONYX digitally steerable arrays to provide enhanced coverage and intelligibility, regardless of the scheduled event. Specifically, a pair of ICONYX IC32 arrays were installed left and right of the stage for larger events, seamlessly covering the floor and balcony areas. "With the IC32s, we were able to create a beam and point it right at the balcony, without hitting the back wall," O'Connell states. "We used seven beams on the IC32s, and that was all that was needed to address the room's intelligibility and coverage issues."

For the hall's lecture and spoken word programs, a pair of ICONYX IC8-R-II columns are mounted on the wall stage right, to each side of the podium. "The IC8s are perfect for lectures and presentations," notes O'Connell. "They provided the intelligibility and coverage we needed for a smaller, more intimate setting."

CP HOLIDAY TRAINS

Each year as the holiday season rolls in, two separate 14-car trains (pictured below) travel west across Canada and the U.S. to deliver free live music shows in more than 150 communities, encouraging attendees to bring donations for local food banks. As the crowds have continued to grow annually, the organizers decided to look for upgraded audio reinforcement for the 2014 edition.

To accommodate audiences gathered on either side of the train, the stage car is a double-sided affair, with a drop-down door and stage extension on each side, and a rotating drum riser in the center. On each side of the car this year, two loudspeaker stacks comprised one Meyer Sound UPQ-1P and UPJ-1P VariO loudspeaker atop a 500-HP subwoofer. A Galileo loudspeaker management system with one Galileo 616 processor provided

system drive and optimization, while MJF-210 stage monitors augmented the musicians' in-ear monitoring systems.

"The difference the Meyer Sound systems provide is stunning, despite their smaller footprint," says Randall Prescott, president of Rip Roar Productions in Eastern Ontario, and producer of the trains' entertainment program for the past 13 years. "I can go to the back of an audience of nearly 10,000 and still hear a crisp hi-hat and punchy vocals that have the whole crowd singing along."

Ken Stone, the Ottawa-based independent audio engineer who helped select the audio equipment, adds, "We're gaining at least another 150 feet of throw, and we're not getting near to the maximum level of what the boxes can do. They're little workhorses, that's for sure."

HANNA THEATRE, CLEVELAND

A recent upgrade of the sound system that supports live performances at this 550-seat venue (pictured below) in Cleveland was led by NEXO GEO M620 line arrays, installed by NAC Technologies, also of Cleveland. The theater opened in 1921, and is now managed by PlayhouseSquare. No stranger to NEXO GEO, which has been installed (by NAC) in three other PlayhouseSquare theatres, Rick Galbraith of NAC explains that for this project, they started by placing loudspeakers in the NS1 prediction software, virtually arranging the array size, coverage, and SPL based on available rigging points.

"Available rigging points was our biggest challenge," he notes, "so after using the prediction software, we determined that three clusters of three M6 speakers would be ideal for the theatre. Considering the size of the clusters, the NEXO GEO M6 system sounds amazing, with zero complaints from the theatre or patrons."

Robert Mingus, director of production for PlayhouseSquare, adds that the loudspeakers solved several challenges. "The M6 sounds fantastic. Our 3-box center cluster, which weighs only 76 pounds, virtually disappears, and is not in the way of lighting. With special brackets from NEXO, we were able to mount the 3-box arrays for left and right onto the proscenium for a very clean look. NAC then tuned the system to perfection, avoiding the many problems of getting the sound into the house and off of the thrust stage." ■


M-5000 LIVE MIXING CONSOLE

QSC TouchMix-16

Evaluating a (very) compact digital mixer platform.

by Craig Leerman

THE QSC TOUCHMIX compact mixer series caught my attention from the moment it was introduced. The control surface is without physical faders – inputs and adjustments are accomplished via the large (6.1 x 3.5 inch) color TFT touch screen and rotary knob on the surface, or wirelessly via a tablet running the TouchMix app. And it's been designed to meet the needs of both novice and experienced users.

The company recently sent me the 16-channel model, appropriately named the TouchMix-16, for evaluation. The smaller TouchMix-8 offers 8 channels, and both units are very similar in terms of feature sets, including a library of channel presets as well as onboard effects and gain wizards that help less experienced operators focus on the mix, not the mixer.

QSC TouchMix-16

Multi-track recording is as simple as connecting a USB hard drive to one of the USB ports, and all inputs plus a stereo mix (22 tracks total) can be recorded as wave files. TouchMix can be used as a mix-down desk or the tracks can be imported into a DAW. Users can also save the channel presets or complete mixer scenes to internal memory or a USB device.

The TouchMix-16 provides 20 full-function inputs with 12 XLR inputs, 4 XLR/TRS combo inputs, 2 stereo line TRS inputs, and a talkback microphone input with XLR. Outputs include 6 aux XLRs, 2 stereo TRS (that can drive wired in-ear monitors directly), main L/R XLRs, cue TRS and a headphone jack.

Each channel offers 4-band full parametric EQ, variable high-pass and low-pass filters, gate, compressor and access to 4 DSP effects. Outputs have a 1/3-octave graphic EQ, limiter, delay and notch filters. Eight DCAs and 8 mute groups makes it easy to keep a handle on things. All of this is housed in a very compact package measuring just 3.5 x 14.2 x 11.7 inches (h x w x d) and weighing only 5.9 pounds.

A cool zippered hard-sided bag holds both the mixer and the power supply, which is a "wall-lump" type. A standard IEC cord connects the power supply to an outlet. The mixer's metallic finish is attractive, and controls are all well labeled.

Form & Function

My first impression when I unpacked the TouchMix-16 and put it on the test bench is that there's a lot of interconnect flexibility. Each of the 16 channels has a rotary trim, and the rotary knob is plenty big and has a nice feel. Pressing on a channel highlights that channel, and the large knob can then be used to adjust the volume. Pressing down on the knob prompts a "fine" adjustment mode.

Further, any parameter can be adjusted by simply touching the parameter on the screen and turning the knob. I found this easier in fine-tuning things like EQ rather than trying to move my fat fingers around the screen. It also adjusts headphone and cue volume.

Above the knob are a series of buttons providing a wide range of functions. They're very well-spaced and clearly labeled, so there's no confusion or chance of hitting the wrong button. These include user buttons, as well as phase switch, talkback mic, assign to phones, an aux overview page, mute groups, the effects and gain wizard pages, topped off by an effects mute button that I think should be included on every mixer! Another button is simply marked "Info." Pressing it brings up an option screen where the user can select additional information on any function – no need for a manual when all of the information is available right on the mixer.

To the right of the touch screen are three more buttons. “Home” brings up the fader banks in 8-channel segments, and at the top of each bank are tabs to take you to the other faders banks for channel inputs, stereo inputs, FX masters, aux outputs and DCA groups. “Menu” provides access to pages to set up the general console settings as well as scenes, DCAs, and phantom power. (This menu also includes 2 overview pages for the aux and FX routing to view all routing for each bus.) And “Record/Play” brings up the built-in recorder functions, including channel arming and transport controls.

As noted earlier, these mixers are very well designed for both less- and more-experienced users. In “simple” mode, many functions are hidden and turned into basic controls. For example, in simple mode the channel EQ is a 4-band fixed frequency unit providing boost or cut at 80 Hz, 400 Hz, 2 kHz, and 6 kHz, along with a fixed low-cut at 120 Hz and a fixed high-cut at 8 kHz. In advanced mode, this EQ is now a 4-band fully parametric with optional shelving, with adjustable low- and high-pass.

A wireless USB dongle facilitates use of an iPad for remote control, so I grabbed mine and downloaded the app to check it out. Setup is very easy. In the mixer menu there’s a network section where you can rename the mixer and give it a new 10-digit password, then you simply connect the iPad to that net-

A lot of easy-to-use capability between the buttons, large rotary knob and touch screen.

work and the mixer links up on its own.

Finally, with a mic plugged, I checked out operation and sonic quality. The mixer sounds great, and I was particularly impressed with the FX units. There are 4 slots, and the user can choose between a dense reverb, lush reverb, mono delay, stereo delay, basic chorus or pitch shifter for every send. The reverbs and delays have a ton of control parameters. Confident in it’s capabilities, I added the TouchMix-16 to a system staged for a multi-day festival that would be loading in the next morning.

In The Field

At the festival we provided main stage sound reinforcement, distributed audio for the grounds, and audio for an outdoor movie and a tent. The first day we used the TouchMix-16 for the movie system. The supplied Blue-Ray player only had RCA outputs, so we used a pair of adapters to plug them into the combo inputs on channels 15 and 16. The main outputs were linked to the loudspeakers, and I booted up my iPad to use the mixer to fine-tune the system in the listening area. Perfect.

The next day we moved the mixer to the tent and used it to route theme music from a laptop to powered loudspeakers. It was easy to optimize EQ via the “touch and turn” operation provided by the touch screen and rotary knob. Fast, intuitive and accurate.

From there we took the TouchMix-16 to a corporate event with presenters, video playback and live music bolstered with backing tracks. We were given very little setup time, so I decided to place the mixer next to the stage and mostly mix on the iPad.

The music act had two different laptops with backing tracks, a keyboard with stereo out, keyboard module with stereo output, acoustic electric guitar and two vocal mics. They also required two monitor mixes for floor wedges and wanted reverb in their monitors. It’s an application right in the wheelhouse of this mixer.

When the musicians asked me for reverb in their wedges, I hesitated for a second and then remembered seeing FX sends on the aux overview page. Sure enough, one button press was all it took to get me to the right page where I could dial in any of the 4 effects to any aux bus. The artists were happy, the audience had a great time, my client was pleased – and so was I, impressed with the capabilities of this mixer, in addition to really enjoying mixing on it.

A closer look at some of the capabilities provided on the screen.

Both TouchMix models are a perfect fit for a wide range of needs for rental companies, as well as houses of worship and performing musicians. Based upon my evaluation, these mixers are highly recommended. ■

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

The "36kW Flying System" from Malcolm Hill Associates of the U.K., circa the mid-1970s.

MODERN PIONEERS

The history of PA, part 2.

by Andy Coules

IN THE FIRST HALF of this article (*December 2014 LSI*), we explored the history of the modern electric PA system and it's application in the reinforcement of live music and events over the first 90 years – from the invention of the microphone in 1875 to the first stadium concert in 1965.

That first stadium concert, featuring the Beatles at Shea Stadium in Queens, showed that while there was the demand for bands to play large shows, the sound reinforcement systems of the time were simply not up to the task. Most bands carried their own small self-operated systems, which were little more than glorified vocal amplifiers.

CLAIR & WATKINS

Thankfully there were pioneering individuals on both sides of the Atlantic willing to up the ante and usher in the era of the large PA. Notable pioneers included Charlie Watkins of WEM (Watkins Electric Music) in London and Clair Brothers in Lititz, PA.

Gene and Roy Clair received their first system as a Christmas present from their grocer father in 1954, and for the next few years they hired it out to local dances and events. In 1963 they purchased a loudspeaker re-coning business that gave them access to the latest developments and led to them providing the sound system for regular headline acts at the 4,000-capacity Franklin & Marshall College in Lancaster, PA.

In 1966 the brothers impressed Frankie Valli enough for him to hire them to go out on tour, thus forming one of the first touring PA hire companies. What made Clair stand out from the competition was a knack for piecing together the right bits of equipment to provide the loudest and clearest output. It led to the company working with Iron Butterfly, Jefferson Airplane and Cream – for whom they did a prestigious gig at the Spectrum in Philadelphia for an audience of 18,000 in 1968.

Meanwhile over in the U.K., Watkins unveiled his "Slave" PA system at a festival in Windsor in 1967, so called because one amp is the slave that feeds into another to increase the output of the second. Using 10 amplifiers that generated 1,000 watts of power to 20 loudspeakers, the system was able to produce a volume level that saw it's creator in court for a breach of the peace – thankfully the case was dismissed.

In 1968 WEM introduced the Audiomaster mixer and it instantly became a classic, even though it only offered 5 channels (each channel had controls for volume, bass, mid, treble and a spring reverb). More importantly, the development of the mixer encouraged the move away from passive mixing (i.e., "set and forget" type operation) and paved the way for a generation of roadies to be elevated to the position of live sound engineer.

A REVELATION

However, due to the use of high-impedance microphones, signal loss was a major issue so cable runs were kept to a minimum, which meant the mixer was typically located close to the rest of the system. Fortunately in New York at the Fillmore East, Bill Hanley developed the first multi-core "snake" that enabled the mix position to be moved away from the sound system. This is something we take for granted now but at the time it was a revelation, as Dinky Dawson (Fleetwood Mac's engineer) noted: "Up to that moment I had never seen any group mixed from anywhere other than the side of the stage. This was revolutionary!"

Hanley and technical engineer John Chester came up with the idea that placing transformers at the ends of a cable run would facilitate the ability to send high-impedance microphone signals down a balanced line over greater distances without picking up any interference or losing level. At the time this

The WEM Audiomaster 5-channel mixer.

Roy and Gene Clair.

A portion of the Crown DC300 spec sheet.

approach was too large to take on tour, but the development of low-impedance mics that had a built-in balancing transformer, such as the Shure Unidyne III (1965), SM57 (1965) and SM58 (1966), allowed multi-core snakes to become portable. Shortly thereafter, Dawson became one of the first touring engineers to set up shop in the new "front of house" (FOH) position, located centrally amidst the audience.

The next key step was the development of bigger amplifiers. Despite being 20 years since the invention of the transistor, the most reliable large amplifiers of the late 1960s were still vacuum tube designs where a unit providing 50 watts per channel was considered "hefty" and one capable of delivering 100 watts per channel was "industrial."

Then in 1967, Crown released the solid state (transistor) DC300 amplifier, so named because it utilized a Direct Coupled (DC) design that was capable of delivering 300 watts of power. What was key to the success of the DC300, above and beyond power, clarity and low distortion, was it's size at 7 inches tall and weight of 45 pounds, less than a quarter the size and weight of an equivalently powered tube amplifier.

MAKING A WAY

At this point most concert loudspeaker systems were quite simple,

A Heil Sound traveling system circa 1973.

comprised of either multiple instances of the same model of loudspeaker (often arranged in a column) or combinations of cone loudspeakers and horns (to handle the low and high frequencies, respectively). Being as any given loudspeaker has difficulty delivering a range of more than three to four octaves with clarity and low distortion, these systems could not be considered high fidelity.

What was needed was a way of splitting up the signal into multiple frequency bands that could be more efficiently handled by individual amplifiers, and then recombined in a manner that was pleasing to the ear. And the emerging PA companies were falling over themselves to accomplish just this.

In 1969, Watkins unleashed the Festival Sound System, one of the first 4-way sound systems, splitting signal into bass, lower-mid, upper-mid and high-frequency portions. A short while later (early 1970), Bob Heil was called into service to build a monstrous 4-way system for the Grateful Dead at the Fox Theatre in St. Louis. He estimated that the total audio power of this system was around 20,000 watts (despite being powered by home hi-fi amplifiers!).

Clair Brothers developed its first 4-way system by splitting the sound up into bass, mid, high and super-high bands, and in 1971 McCune Sound Service (San Francisco) employees John Meyer and Bob Cavin created a loudspeaker known as the JM-3 (named after John Meyer). This was a 3-way system that also enclosed the amplifiers and all of the integrated electronics in an external equipment rack. Thus at the dawn of the 1970s, we entered the golden age of the ground-stacked PA system – so called because the bass loudspeakers were placed on the ground or stage, with the mid and high loudspeakers stacked on top.

Many of the those early big PAs were based on aging loudspeaker models such as the Altec "Voice of the Theatre" Series (from the small A8 through to popular A4 and the large A1), which were designed more for use in cinemas and theatres. Gradually PA companies such as Clair and Heil Sound shifted toward use of transducers (cone and compression drivers) developed by JBL Professional, a company with a rich heritage extending back to its origin in 1945. Notable developments include the D130 15-inch cone (introduced in 1947), which was the first commercial use of a 4-inch voice coil (a variant of which remained in production for the next 55 years), and the model 375 (introduced in 1954), the first commercially available compression driver with a 4-inch diaphragm.

MIX & MONITORS

Live mixing consoles as we know them today came with a big assist from the recording side of audio. Bob Heil adapted a Langevin studio console for his fledgling large-scale PA in 1970, and he was also an early user of the Mavis console from IES in the U.K. as well as working closely with the Sunn engineering department on the first 8-channel Coliseum mixer (which was hand built).

A few years later, Ron Borthwick and Bruce Jackson, working with Clair Brothers, developed an unusual mixing console, the CBA 32, that went on to become the company's mainstay into the 1980s. It was the first console to offer plasma bargraph meters, which displayed both average and peak sound levels, and the meters were conveniently located beside the faders. In addition, it was the first live console to incorporate parametric EQ.

Another key step was the development of more complex stage monitoring systems. Prior to this point musicians had either relied on carefully balancing their levels so they could hear everything they needed or they placed loudspeaker stacks at the side of stage which "folded back" the main mix (as those early mixers didn't have auxiliary outputs capable of providing monitor mixes).

But the increase in the size of the PA systems meant monitors were becoming more of a necessity for the performers to not just hear and feel the performance, but also to be able to sing in tune. Pretty soon, curious wedge-shaped loudspeakers started appearing on stages, and once musicians heard them, they all wanted one (or more).

It's interesting to note that such additions were not always welcomed by front of house engineers. For example, Pink Floyd engineer Mick Kluczynski observed that wedge monitors spread "like a virus" and he quickly found himself battling a secondary sound system, struggling for control of the main mix.

BASIC TEMPLATE

Loudspeaker capability developed and grew throughout the 1970s, with the 3- and 4-way ground stacked system dominat-

Now that's a ground stack!

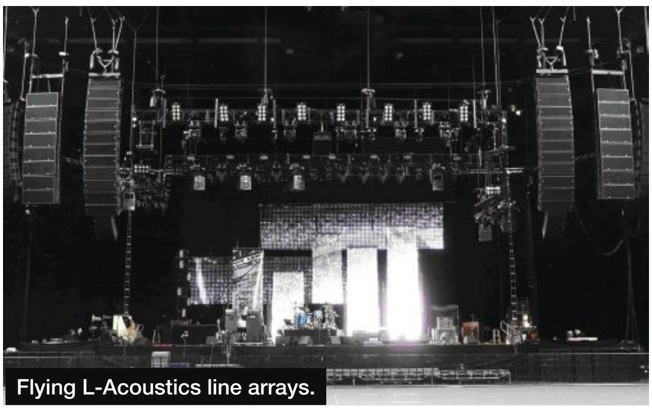
ing and then gradually taking to the air over the next 25 years or so. While hanging or suspended systems were developed, the basic template remained the same.

These systems are often referred to as "point source," meaning they propagate sound equally in all directions, but that can be misleading. In reality, they only behave like a point source at lower frequencies and gradually become more directional as the frequency goes higher. As a result, they tend to "spray" a lot of bass and lower mid range energy in all directions – against the ceiling, floor and side walls – causing delayed reflections that can muddy the sound and make it difficult to manage.

Point sources also conform closely to the inverse square law, meaning that the sound level drops by a quarter for each doubling of the distance from the source due to the fact that the expanding sphere of sound energy is spread over an increasingly larger area. This is another easily observable property as we all know that the sound is typically louder at the front, near the loudspeakers, and quieter at the back.

During this period, numerous manufacturers entered the concert market, presenting increasingly sophisticated loudspeaker systems. JBL Professional, Meyer Sound, Martin Audio, EAW and Turbosound are just some of the names, with the growing number of local and regional sound companies now having the opportunity to purchase "off the shelf" packages instead of designing and building their own loudspeakers.

EAW in particular enjoyed huge success with the KF850, a 3-way "arrayable" loudspeaker designed by Kenton Forsythe – for several years it was tough to find an equipment rider that didn't include the KF850. Meyer Sound lead the way in developing self-powered live systems, with amplification and electronics matched to the specific loudspeaker parameters and physically housed in the same enclosure.


Flying L-Acoustics line arrays.

LINE 'EM UP

Then in 1993, Christian Heil and his team at a company based in France named L-Acoustics launched the "modern" line array era with the introduction of the V-DOSC system. I say modern because the principals behind the line array have been known and used (in a limited sense) for quite some time. Harry Olson first published his findings on the subject in 1957, and the benefits of column loudspeakers (where vertically aligned drivers in a single enclosure produce mid-range output in a wide horizontal and narrow vertical pattern) had been utilized in the Shure Vocal Master PA and Marshall columns that were available in the 1960s and beyond.


EAW KF850s deployed for Cheap Trick in the late 1980s.

L-Acoustics refined the science and exploited the constructive interference caused by closely aligned loudspeakers to push sound energy further, and with a more even frequency response. The company's "cylindrical wave generator" also focused output more in the horizontal plane and wasted less energy in the vertical plane, resulting in a more even distribution of sound throughout the space.

Line arrays are very much a product of the computer age, with precise modeling and precision deployment key to getting the required result. However, it's important to note that line arrays are not the ultimate sound system; there are still many situations where a traditional flown or ground stacked approach is better suited.

The driving force behind every pioneer highlighted in this fascinating journey has always been the desire to provide a pleasing experience for the audience. As the technology advances, we as audio professionals are more able to place the sound precisely where it's needed to provide the visceral experience that only a live performance can provide.

But it's important to remember that the PA is merely the conduit for the performance. If you were able to directly compare the concert experience of someone in 1964 with someone in 2014, chances are they would express the same degree of joy and excitement at what they had witnessed, despite the obvious gulf in fidelity. Because at the end of the day, it's all about the music. ■

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Compact Benefits

Small and mid-sized digital consoles. ***by Mark Frink***

IN THE BEGINNING there was resistance to digital consoles, but the tide has turned and the advantages of digital live sound workflow are so numerous that we now take many for granted. An obvious advantage is size and weight. Compact digital desks weigh 50 pounds or less, and a 100-meter Cat-5 snake fits in the back of a stage box rack. Where an analog console and its associated outboard gear, multi-core snake and splitter would take a couple rows in a truck, a compact digital mixing system fits in the back seat of a crew cab.

With digital snakes, the hums and buzzes of yesterday are mostly gone, while putting preamps on stage improves sound quality. Gain-sharing means consoles don't need a splitter, and mix engineers don't have to listen to their mixes through transformers.

"In-the-box" mixing, where mix processing is performed entirely within a digital console, means engineers can save shows and open them on identical or similar consoles. They can e-mail that file to the next gig, whether across the country or around the world, "cc-ing" themselves as a backup.

Though layered consoles with fixed architecture have been an annoyance to those who grew up with analog, newer designs improve workflow with the ability to "spill" control group members onto fader banks and customization of layers can improve mixing. Remote control of a console from something as inexpensive as an iPad has become a standard feature and a powerful tool, allowing monitor engineers to stand beside performers while front of house engineers check the entire listening area – both with controls in hand.

USB 2.0 has enabled digital console manufacturers to provide simple multi-track I/O for recording and playback to laptop-based DAWs – and even direct to hard drives in some cases – providing affordable "virtual sound checks" to many applications. This allows engineers to easily check a previous show, test sound systems with a previous performance, practice mixing or tweak a show file, teach others to mix, and afterwards, easily mix a show's multi-tracks down for distribution using the same console.

Similarly, 2-track recording and playback using USB "thumb drives," allows engineers to walk away from a console with a board mix they can easily listen to and then e-mail to others if they like. It also provides simple and foolproof playback of walk-in and intro music with no moving parts.

Thirty years ago, 32-input channels was a standard that covered any band and even festivals. Of course, this was before the days of double-miking nearly everything on stage. When I worked at Sun Sound Audio, we took our flagship 32-channel Soundcraft 800 to many New England concerts, colleges, festivals, and even on my first tour with Crystal Gayle.

By 1989 I found myself at the Capitol Theater in Passaic, NJ helping Jonny Podell put Duane Allman's and Dickie Betts' bands back together (again), with two drum kits and The Band thrown in as an opener, all mixed on that same Soundcraft 800-32. Yes, we used a few Y-cords, but today I'm sure that show would "require" 96 channels. Point being, 32 channels can do a lot, and with today's digital workflow, so much more.

Yamaha first introduced the "0" series in the '90s, and the youngest sibling, the 01V, has long been a benchmark for compact digital consoles. Stan Miller and Bernie Becker took 14 of its predecessor, the Pro Mix 01, on Neil Diamond's tour as sub-mixers feeding stereo stems into a 24-channel Yamaha PM3500. In honor of this trailblazer, we lead off the listings here with the 01V96i – to this day thousands are used in smaller venues and recording studios. Enjoy this review of today's small- and mid-sized digital consoles. ■

MARK FRINK *(livesound@markfrink.com) is an independent engineer who has mixed monitors for numerous top artists, and he's also a consultant, author, and editor.*

Yamaha 01V96i » www.yamahaca.com

Faders: 17
Layers: 4
Mix Inputs: 32 + 4 stereo
Aux/Group: 8/8
FX: 4 SPX
Mains: LR (3.1, 5.1 or 6.1)
Screen: 320 x 240 monochrome LCD
Local I/O: 12 XLR + 4 TRS, 2 XLR + 4 TRS, ADAT, S/PDIF
Expansion: 1 MY card: 30+
Also: 16 x 16 USB2 DAW interface
Physical: 17 x 22 x 6 inches, 31 pounds

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

Faders: 33
Layers: 3
Mix Inputs: 32 + 4 FX + 3 stereo
Aux/Group: 10/4 + 4FX
Matrix: 4
FX: 4
DCA: 4
Mains: LR
GEQ: 24 + 2
App: Qu-Pad (iPad), Qu-You (iOS)
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 and Qu-24

DiGiCo SD11i » www.digico.biz

Faders: 12
Layers: 4 assignable
Mix Inputs: 32 (mono or stereo)
Aux/group: 12 (mono or stereo)
FX: 6
Matrix: 8
DCA: 8
Mains: LR/LCR
GEQ: 12
Dynamic EQ: 6
Multi-Band Comps: 6
App: SD App (iPad)
Screen: 15-inch color touch screen
Local I/O: 16 XLR + 1 AES, 8 XLR + 1 AES, and D-Rack Cat-5 ports
Stage Boxes: D-Rack (32 x 16)
Also: Aviom D-16e for D-Rack, 48 x 48 record/playback via UB MADI
Physical: 19 x 23 x 9 inches, 53 pounds

Avid S-3R » www.avid.com

Faders: 16
Layers: 12 assignable
Mix Inputs: 64
Aux/Group: 24
Matrix: 8
FX: 8
DCA: 8
Mains: LCR/5.1/7.1
GEQ: 16
Screen: 15-inch XGA monitor
Local I/O: 2 XLR + 2 TRS, 2 XLR + 2 TRS, plus 4 x 4 x 4 XLR I/O on 2U 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: 28 x 15 x 3 inches., 14 pounds; E3 engine is 2RU x 15 inches., 21 pounds

PreSonus StudioLive 32.4.2 AI » www.presonus.com

Faders: 37 non-motorized
Layers: 1
Mix Inputs: 32 + 2 stereo
Aux/FX/Group: 14 + 4 + 4
FX: 2 reverb, 2 delay
Mains: LR + M
GEQ: 16
App: SL Remote-AI (iPad), QMix-AI (iOS)
Screen: 36 x 6 character monochrome LCD
Local I/O: 32 XLR/TRS + 4 TRS, 3 XLR/2 TRS, 14 + 4 gr/aux TRS
Also: Dante, AVB in Q2, 2015; any two StudioLive consoles can be daisy-chained via FireWire
Physical: 32 x 22 x 7 inches, 50 pounds
Additional Models: StudioLive 24.4.2 AI and 16.4.2 AI

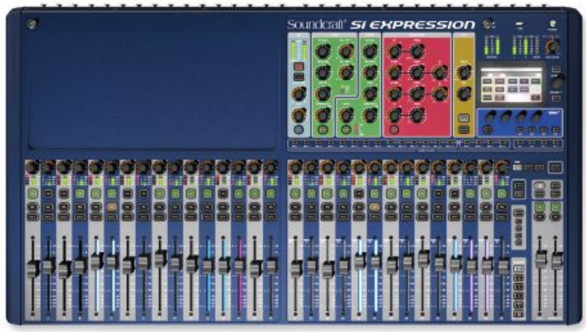
Yamaha Commercial Audio QL5
www.yamahaca.com

Faders: 32 + 2
Layers: 2 input, stereo/VCA, mix/matrix, 4 custom
Mix Inputs: 64 + 8 stereo
Aux/Group: 16
Matrix: 8
FX: 8 FX + 8 premium
DCA: 16
Mains: LR/LRC
GEQ: 8 GEQ + 8 premium or Dugan automix
App: QL StageMix (iPad)
Screen: 10-inch color touch screen
Local I/O: 32 XLR, 16 XLR + 1 AES, 64 x 64 Dante
Stage Boxes: Rio3224-D, Rio1608-D (16 x 8), Ri8-D, Ro8-D
Expansion Slot: (2) 16 x 16 MY: AES/EBU, ADAT, Aviom A-Net, CobraNet, EtherSound, MADI, Riedel RockNet, Optocore, Waves and more
Physical: 33 x 23 x 11 inches, 48 pounds
Additional Model: QL1

Midas M32 » www.midasconsoles.com

Faders: 25
Layers: 4 + 4
Mix Inputs: 32 + 8 FX + 8 aux
Aux/Group: 16
Matrix: 6
FX: 8, including dual GEQ
DCA: 8
Mains: LCR/LRM
GEQ: up to 8 Dual GEQ in FX rack
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
Expansion: ADAT, MADI, Dante
Also: P16 personal monitoring, 32 x 32 recording/playback via USB2/FW800
Physical: 35 x 24 x 10 inches, 54 pounds

Soundcraft Si Expression 3 » www.soundcraft.com



Faders: 32
Layers: 4
Mix Inputs: 66 inputs to mix
Aux/Group: 14 + 4 FX
Matrix: 8
FX: 4 Lexicon processors
Mains: LR/C
GEQ: 31 BSS GEQ
App: ViSi (iPad)
Screen: Small color touch screen
Local I/O: 32 + 1 AES x 16 + 1 AES and 4 TRS
Stage Boxes: Mini Stagebox 32, Mini Stagebox 16
Expansion Slot: 1 FireWire/USB/ADAT, Aviom, CobraNet, BSS "BLU Link," Dante, Cat-5 or optical MADI, Riedel RockNet
Also: dbx PMC16 personal mixers using BLU Link, 40 x 40 USB to DAW
Physical: 37 x 21 x 7 inches, 42 pounds
Additional Models: Si Expression 1 and 2

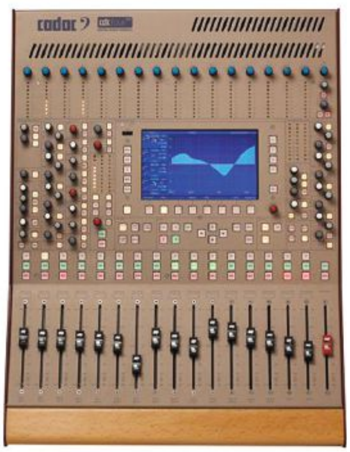
Mackie DL1608 » www.mackie.com

Faders: 9 "virtual faders" on iPad
Mixer Layers: 5
Mix Inputs: 16 + 2 FX + iPad (stereo linkable)
Aux/Group: 6/4 (stereo linkable)
FX: Reverb + delay
DCA: 4
Mains: LR
GEQ: LR + 12 aux
App: Master Fader 2.1 (iPad), My Fader 2.1 (iOS)
Screen: User-supplied 9.7-inch multi-touch iPad
Local I/O: 16 XLR (4 Combo), 2 XLR + 6 TRS
Physical: 12 x 16 x 4 inches, 8 pounds (plus iPad)
Additional Models: DL806

Roland Pro AV M-200i » www.proav.roland.com

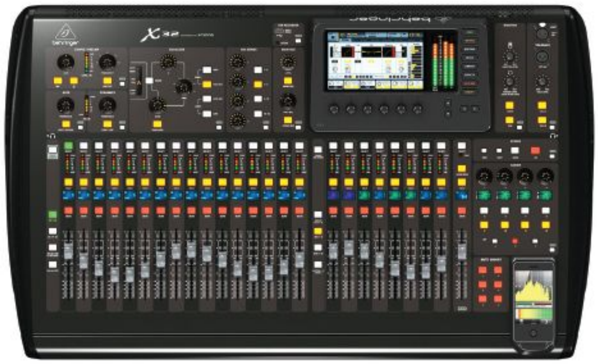
Faders: 17
Layers: 5
Mix Inputs: 32 + 8 + 8
Aux/Group: 8
Matrix: 4
FX: 4
DCA: 4
Mains: LR
GEQ: 4
App: M-200i Remote (iPad)
Screen: Monochrome 132 x 64 LCD screen ; user-supplied iPad
Local I/O: 16 XLR + 6 TRS, 8 XLR + 4 TRS + 1 AES, REAC, iPad LR
Stage Boxes: REAC S-1608, S-0816, S-2416, S-3208, S-MADI Bridge
Also: M48 personal mixers with S-4000D REAC splitter
Physical: 20 x 20 x 8 inches, 22 pounds
Additional Models: M-480, M-300

CADAC CDC Four » www.cadac-sound.com



Faders: 17
Layers: 5
Mix Inputs: 56
Aux/Group: 8/4 + 2 FX
Matrix: 6 x 4
FX: 6
DCA: 8
Mains: LR
GEQ: 14
App: TabMix (iPad)
Screen: 7-inch color screen
Local I/O: 16 XLR, 3 XLR + 31 TRS
Stage Boxes: CDC I/O 3216
Expansion: MegaCOMMS, 32 x 32 FireWire interface
Physical: 19 x 23 x 7 inches, 33 pounds

Behringer X32 » www.behringer.com



Faders: 25
Layers: 4 + 4
Mix Inputs: 40
Aux/Group: 16
Matrix: 6
FX: 8, including dual GEQ
DCA: 8
Mains: LCR/LRM
GEQ: Up to 8 dual GEQ in FX rack
App: XiControl (iPad)
Screen: 7-inch color screen
Local I/O: 32 XLR + 6 TRS, 16 XLR + 6 TRS + 2 RCA
Stage Box: S16
Expansion: XUF FireWire 400/USB card
Also: 32 x 32 recording/playback via USB2/FW800, P16 personal monitor system
Physical: 36 x 21 x 8 inches, 44 pounds

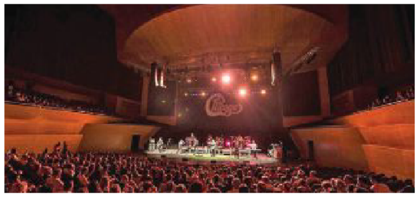
QSC TouchMix-16 » www.qsc.com

Faders: 8 "virtual faders" on touch screen
Layers: 6
Mix Inputs: 16 + 2 stereo + 4 FX = 28
Aux: 6 + 2 powered stereo, 4 FX
FX: 4
DCA: 8
Mains: LR
GEQ: 8
App: TouchMix (iPad), WiFi included
Screen: 6.1 x 3.5-inch color touch screen
Local I/O: 16 XLR + 2 stereo TRS, 8 XLR and 4 stereo TRS
Also: USB3 record/playback 22 tracks plus LR
Physical: 14 x 12 x 3 inches, 6 pounds
Additional Models: TouchMix-8

APPLICATIONS

Originally designed for classical music, L'Auditori de Barcelona in Spain is now welcoming international pop stars to its stage, with the expanded programming requiring upgraded sound reinforcement capability. The venue hosted extensive demonstrations of brands that appear regularly on riders, with **d&b audiotechnik** emerging as the choice for the "long and lively" performance space.

Juanma de Casas of **d&b audiotechnik Spain**, working with **Imaginamusica**, provided a design headed by main left-right V8/V12 arrays joined by T10s in point source mode for front fill. LF is supplied by flown V-SUBs. A dis-



crete pair of out fill V12s rigged upstage behind the mains and subs cover some lateral balconies. Loudspeakers are driven by D80 amplifiers, with optimization via R1 Remote Control software.

Iconic comedy group Monty Python recently took to the stage for a run of shows at London's O2 arena, where **Countryman** H6 headset microphones were the choice of **Rory Madden,** sound designer for the shows and owner of **Sonalyst**. **Graham Paddon** of **Amber Sound** supplied the H6 mics.

All five of the surviving Monty Python cast were equipped with the mics, along with the rest of the cast, chorus and dancers. Madden also wanted back-up wireless mic transmitters on each of the main cast members, with Amber Sound working with Countryman on a solution that enabled B6 lavalier mics to be clipped to the booms of the headset mics, giving the option of a main and backup capsule.

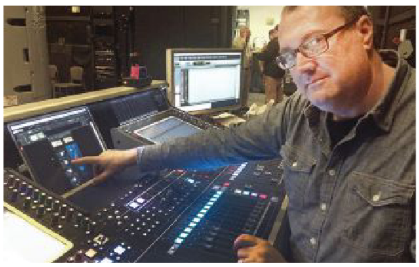
Heavy metal artists Five Finger Death Punch recently toured arenas in the U.S. with Escondido, CA-based **Sound Image** providing stacks and racks that included 48 **Adamson Systems** Energia E15 and E12 line source enclosures joined by E218 subwoofers. **Bruce Reiter**, the band's front of house engineer and production manager, first heard an Adamson rig when visiting a colleague mixing another band.

"I thought it translated heavy music in a smooth and powerful way," Reiter says. "After speaking with a few respected front of house mixers, we decided it was the perfect fit for what Five Finger Death Punch is doing. The production manager

in me also loves how quick and easy the system goes up and comes down." **Cameron Whaley** of Sound Image served as system tech for the tour.

Front of house engineer **Greg Nelson** utilizes **Waves** plug-ins and a Waves MultiRack SoundGrid when touring with Pearl Jam. "I started using Waves in 2002 while mixing some live bootlegs for Incubus," Nelson notes. "I think it was just the Waves L1 Ultramaximizer and the Waves C4 Multiband Compressor on the master to smooth out the mix a bit."

His current setup on the road with the Pearl Jam includes a DiGiCo SD5 console with a Waves MultiRack SoundGrid and Waves Server One. "My favorite



Waves plug-ins when working with Pearl Jam are the MV2 for kick and especially snare, the H-Comp Hybrid Compressor for vocals, the C6 Multiband Compressor for vocals and slight master bus tuning, and the CLA-3A Compressor/Limiter for guitars," he adds.

The Blue Marlin Ibiza beach club in Abu Dhabi at the Golden Tulip Al Jazira Hotel & Resort offers guests a line-up of concerts every weekend during beach season, heard via a new system with **EAW** KF740 line arrays and SB1002 subwoofers. The system was designed by **James Field** of **Imagine Audio Visual**.

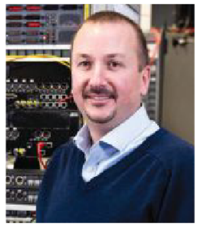
Rectangular truss defines the performance stage, with four line arrays – each made up of four KF740 enclosures – hung from each corner of the structure, firing outward. Two sub arrays of six SB1002s hang from the two corners located closest to the stage, and more subs are ground stacked. Field employs three EAW UX8800 DSPs on the mains and subs, and a UX3600 DSP for EAW QX loudspeakers serving the VIP zone.

PEOPLE

Doug Carnell has joined the executive staff of **Full Compass Systems** as chief operating officer, where he is overseeing daily operations and identifying development opportunities for all of the company's business channels, helping further future growth and streamlining processes. Carnell has 28 years of experience in AV, managing and working with companies ranging from $2 million to $570 million in annual revenue, most serving as executive vice president of operations for AVI-SPL.

"This is very exciting news for us," states Full Compass president **Mark Nash**. "As our business continues to evolve and grow, we benefit greatly by adding Doug's experience and knowledge to our already top-notch staff. We look forward to a spectacular future."

Riedel Communications has appointed **Jake Dodson** as director of product management, where he is responsible for guiding the strategic direction of the company's product range. Dodson has more than 25 years of experience in developing product strategies and bringing new products and technologies to market. His career began at Marconi Research in the U.K. working on high-speed optics for military and telecoms.



"Over Jake's career, he has been a founder and leader of successful leading-edge technology firms," states company CEO **Thomas Riedel**. "His unusual and exceptional track record in guiding companies toward ever-greater success will certainly be an asset to Riedel as we build on our acclaimed product portfolio and move further into the realm of software-based solutions, which offer our customers many new possibilities." ■

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PIZZERIA
Extra Cheese
Hamburger
Mushroom
Peppers
Sausage
Onions
Pepperoni
Anchovies

Cast Of Characters

Only the names were changed to protect the guilty. *by Craig Leerman*

OVER THE YEARS I've had the pleasure of working with many great folks in our business... and then there were these people.

TWISTY KNOB GUY: Wouldn't stop turning knobs, even when everything sounded great. Either he was just never happy with his own mix, or didn't want to sit still for a minute and have the promoter think he wasn't earning his pay.

THE PICKUP ARTIST: Spent more time at catering trying to get a date with the servers than he did setting up the show. I kept waiting for him to grab a wireless and do the line from the Mr. Microphone commercial: "Hey good lookin', I'll be back to pick you up later!"

THE LITERARY SNOB: Why mix or even pay attention to the performers when you can sit at front of house and read a magazine? On the plus side it was an audio magazine, but on the minus side it was devoted to hearing aid technology.

TWEETY BIRD: Spent more time updating the band's fans about the gig on Twitter then he did making the band sound good for the fans that actually showed up to hear them. (#BadMix)

MARATHON MAN: Constantly walked the room, checking the PA. Unfortunately he spent so little time at the console that he missed important things like setting delay tempo or turning off the reverb when the vocalist was just talking to the crowd. It was funny, however, watching him morph into "Sprinting Man" when a guitar lead started.

TEXTING DUDE: A person of the lighting persuasion (not that there's anything wrong with that) who was more interested in his text screen than following show cues. Wait – the show's over here, but the lights are over there … what's going on? Oh, texting again. (Perhaps consulting with an optometrist?)

WE DON'T NEED NO STINKIN' VOCALS GUY (A.K.A., INSTRUMENTALS GUY): Buried the vocals so far down in the mix that the singers looked like they were doing an elaborate pantomime of singers. All of that frenetic motion, none of that hot, velvety vocal action. (At the time it was amusing in a "we'll all laugh about this years from now" way. And we do.)

BAKER'S DOZEN: Always offered more than a dozen reasons why he didn't get things done like the rest of the crew. In return, I had just one reason why he never worked with us again. (I prefer a minimalist approach.)

MONITOR MONARCH: Didn't like to give performers what they wanted in their own monitors. Never having performed onstage or even played an instrument, nonetheless he still knew best and doled out to the peasantry only what he thought they really needed.

CAN YOU HEAR ME NOW NERD: Took a call at front of house during the show, talked for a minute, put his finger in his ear and angrily looked up at the stage, and then wandered out of the venue into the hallway to finish the call. Returned 20 minutes later. (Perhaps it was a conference call with the Monitor Monarch and Baker's Dozen Guy, strategizing new ways of how not to serve a client.)

BDS (BRUCE DICKINSON SYNDROME) SUFFERER: Could attain a pretty good mix with one exception: one instrument not normally at the forefront of the mix was obnoxiously loud. And it was obvious to everyone except him. In the words of *The* Bruce Dickinson himself, "I got a fever, and the only prescription is more cowbell!" Or in this case, more congas. ■

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


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