ChurchSound

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JUNE 2016 | PROSOUNDWEB.COM

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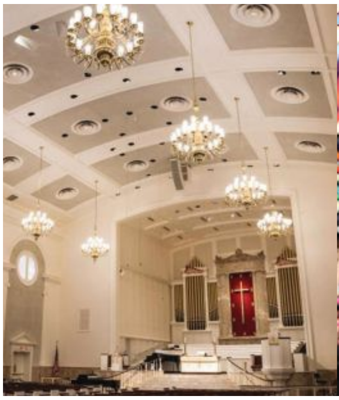

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Ed Note

Welcome to the second *Church Sound* supplement from the editors of *Live Sound International* and ProSoundWeb.

This publication is a true joy to put together. Our authors, with more than 125 years (conservatively) of experience with sound reinforcement in church/worship applications, bring both superior knowledge of the art and science of audio combined with plenty of good cheer.

The results of all of their dedicated work, and the willingness to share what they've learned, is what really makes it all worthwhile – a journal that's full of information of high benefit to anyone working on the front lines of church sound tech.

As you'll see in this issue, the focus is on the technical and non-technical sides of the equation. Simply, it takes success with both to reach an effective and fulfilling role in the quest for excellence. Technical knowledge is powerful and much needed, but it does far less good if it's not shared and combined with an understanding of the bigger picture.

As I noted in this space in our previous issue, also please take advantage of the Church Sound section of ProSoundWeb, where we host more than 700 reference articles, free and available to anyone who visits, 24/7. There you can find all of the articles from our first edition of this supplement as well.

Further, expert advice regarding all aspects of church tech can be had by visiting the Church Sound forum on PSW. And on top of all that, many of our authors also run insightful blogs/websites devoted to sound for worship, and I heartily encourage you to visit them as well.

I hope you enjoy and benefit from this second edition of *Church Sound*, and please don't hesitate to contact me with your thoughts and ideas via my e-mail address.

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ChurchSound

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On The Cover:
Renkus-Heinz VARIA arrays flying and a networked Yamaha CL5 digital console at front of house at The Church at Liberty Square in Cartersville, GA.

When it comes to family members, inheriting similar traits is inevitable. We think that's a great thing, especially when those traits are positive. Though different in size and capabilities, the CL and QL Series share core features like pure natural sound enhanced by unique internal processing effects and built-in Dante networking allowing flexible system configuration. Especially important for Houses of Worship where volunteers are a vital part of the ministry, with this family of consoles, learning how to mix on one series teaches you the fundamentals necessary to mix on the next. Whether transitioning from a remote campus to the main worship center or upgrading your church system, the shift from learning small to going big is quick, user-friendly and seamless.

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IS SOUND SUBJECTIVE?

Going beyond the obvious to a deeper discussion.

by Mike Sessler

The other day I was pursuing one of the many online audio groups and came across a question from someone who I believe to be a volunteer sound tech at a church. He was asking if sound is subjective. He'd been dealing with various other leaders in the church and was struggling to come up with a consensus on whether there was "good" sound or if it's all in the ear of the beholder.

As someone who has been party to many of these discussions, I'm going to provide my thoughts. First, there's the topic of sound system (PA) tuning, because without a well-tuned PA, good sound is much harder to achieve. And as you might expect, there are plenty of opinions on how to tune.

Second, I want to dig into the difference between "subjective sound" and "personal preference" when it comes to mixing. I think these concepts are often confused, and when we assert one as "correct" we get into trouble.

COURTESY OF CAMILLE KING

YOU CAN TUNE A PIANO, BUT...

When it comes to PA tuning, ask 10 people how to do it and you'll get 11 answers. But it basically boils down to two main schools of thought. The first says the system should be as linear as possible; that is, what comes out of the console should also come out of the PA. When you look at a transfer function graph of a system tuned for this goal, it should be pretty close to flat across the audio spectrum.

Before we go any further, note that in an actual live room, a system is never totally flat. There are always slight anomalies. Even if you do manage to get it totally flat, it can change significantly simply by moving the measurement microphone about 3 feet. So when I say a linear system is "flat," I mean in general.

The other school of thought is to build some tonal shaping into the system. This normally includes what some call the "bass haystack" – a 6 to 12 dB bump at the low end that looks somewhat like a haystack, and it usually also includes some roll-off (cutting back) of the high frequencies. How much is rolled off and where it starts will vary, but it's usually in the order of 1 to 2 dB per octave above 1 to 4 kHz.

WHICH ONE IS RIGHT?

Many will argue to the death about which method is correct (or present their own, far superior method), because this is *that* important. (That was sarcasm.) As is often the case, much of it is personal preference. I've heard and mixed on systems tuned both ways, and I (and my ears) prefer the latter approach. I find it to sound more musical and less harsh.

However, good friends of mine will argue that mixing on a system tuned this way is like mixing with blankets over the loudspeakers. I can appreciate that. Their approach to mixing is different from mine and while we achieve similar results, we go about it differently. I think it's possible to get a great sounding mix either way.

Folks in either camp should agree that the overall tune of the loudspeakers should be accurate. Aside from a haystack (or not), and a subtle, linear roll off of the high frequencies (or not), the system should pretty much deliver what comes out of the console. So while we may be able to build consensus on a couple of different ways that are "correct" to tune a PA, there are a lot of ways to really goof it up.

WHAT CAN AND CAN'T BE TUNED

Part of the problem stems from too many badly (or not at all) designed systems that simply cannot be tuned. I've seen seating areas fully covered by two to three

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loudspeakers that are at radically different distances from the seats. The comb filtering that ensues can't be fixed with electronics. I've also seen systems with loudspeakers that are entirely wrong for the space. Sure, they make sound, but it's so uncontrolled there's no way to make it sound good. And I've seen entire seating sections completely off-axis of the loudspeakers, and there's no electronic fix for that either.

Then there are the crazy frequency response traces from systems I've been called in to fix. These are usually the result of someone with just enough knowledge to be dangerous playing with all those "cool" EQ controls inside the DSP. And I've observed, uh, interesting curves on 31-band EQs on the master buses of consoles. Cleaning all that up makes a huge difference.

The bottom line is that while there's some space for preference and individual taste when it comes to tuning a system, putting 10 top-notch sound people in a room with a competent system tech will result in a general consensus of what sounds good and is easy to mix on. And it will be relatively easy to spot ways not to do it. Now let's delve into the tricky world of mixing.

OBJECT OF THE EXERCISE

Generally speaking, the point of live music mixing is to reproduce (reinforce) and make louder what is happening on stage, and to do so as accurately as possible. I say generally speaking because there can be things happening on stage that aren't pleasant, and a good mix engineer either fixes or eliminates them.

In addition to making it louder, engineers can also enhance the audio experience by using things like effects and various mixing techniques. But they're never the goal; no one comes to a concert to hear the engineer's super-groovy plate reverb on the snare – they come to hear the band.

While most engineers will agree with this, in practice there's quite a bit of deviation. For example, there's a movement among some to assault the audience with low end. In my opinion, these mixes aren't pleasant, and if we listened to the band in a small room (where there's no need for a big PA), it wouldn't sound like that.

However, some bands want that sound. As engineers, we're an extension of the band. If we're doing our jobs correctly, we're delivering to the audience the band's vision of its music. This might mean that we mix in a way that's different from what we would prefer.

There's a big "but" in that previous paragraph. While there are preference issues, the truth is sometimes we hear mixes that are just horrid. A bad mix is usually the result of a lack of training, a lack of a musical ear, or someone who just doesn't care. One of those things can be fixed. We also hear mixes that are "OK," but not very good. Sometimes it's an experience thing, other times it's a lack of understanding how music works. Not everyone can do this – that's one thing we should all agree on.

COURTESY OF DEFKREATIONZ

Here's an example. I don't really listen to modern worship music outside of church. I don't like the way most of it sounds or the way it's mixed. However, when I'm mixing in church, I do it the way the band wants, which is usually the way it sounds on the album. I listen to the tracks we're going to do for the weekend for a point of reference and try to enhance that (and maybe move it a little bit toward my preference), but overall, I work hard to deliver what the band (and worship leader) want.

It doesn't really matter that much what we as engineers like. If your personal music preference is the Gaither Vocal Trio, and your church loves to do Bethel, don't try to make Bethel sound like Gaither. Make it sound like Bethel or find a new church.

GETTING WHAT YOU PUT INTO IT

Finally, the mix is only ever going to sound as good as the band on stage. GIGO is a computer term that means Garbage In, Garbage Out. Basically, if you put bad data into a computer, you get bad results. The same goes with putting a bad band on stage – pretty much all the mix engineer can do is make them louder, not better.

I've seen pastors berate the sound operator when the real problem is on stage. I've wanted to tell them, "Pastor, it's not his fault, the band is just terrible." One time a person emailed me to ask what plugins or mics would get that "huge drum sound" heard on some album. I replied, "First, hire that drummer. Second, have him bring his drums. That should get you pretty close."

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THE POINT

Before closing, I want to touch on the concept of politics. Not nation-state politics, but the internal politics of the church. We don't like to talk about this, preferring to believe that everyone is one big happy family. Sometimes that's true, but often it's not.

Where we run into trouble – especially in church – is when there are multiple objectives. Usually this is the case when the leadership isn't on the same page when it comes to the musical portion of the worship service. I've worked in these environments and it's no fun.

When the senior pastor wants All Sons and Daughters and the worship leader wants Hillsong Young and Free, the sound engineer is stuck in the middle. We can't serve two masters, especially if they want very different things. When the pastor keeps saying, "Turn it down!" and the worship leader is yelling, "Crank it up!" it's a losing proposition. To be clear: this is a leadership issue, not a mix issue.

It usually happens when a church finds itself aging, so to appeal to a younger crowd, leadership brings in a younger, cool and loud worship pastor who immediately changes the musical style. This leads to all kinds of conflict, and as noted, the sound people get caught in the middle.

Another big challenge is when a pastor won't tell the sound and worship teams what he wants musically, but will tell them what he doesn't like. You start to hear comments like, "I don't like that, it doesn't fit our culture." When asked, "OK, what would you like to hear?" the reply is, "I don't know, but I'll tell you when I hear it."

This drives me nuts. You're not leading when you're simply saying "no" all the time until the team stumbles on the "yes." Pastors, if you want to continually frustrate your teams and make their lives miserable, don't give them any direction, just shoot down everything they do because you don't like it.

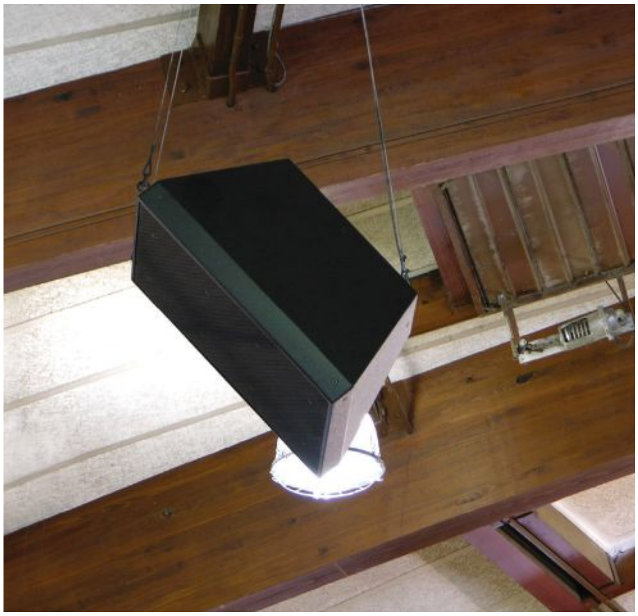
One of the easiest ways to settle the "sound problem" is to choose a musical style that fits the culture of the church. Not every church is cut out for Bethel. Not every church does well with traditional hymns. Find the style that fits the congregation and mix appropriately. If you do that, the other problems get a whole lot easier to solve. CS

Mike Sessler *has been involved with church sound and live production for than 25 years, and is the author of the Church Tech Arts (churchtecharts.org) blog. Based in Nashville, he serves as project lead for CCI Solutions, which provides design-build production solutions for churches and other facilities.*

ARE YOUR LOUDSPEAKERS BROKEN?

The answer is often "yes," and there are some simple solutions.

by Curt Taipale



When was the last time that you listened analytically to the loudspeakers in your sanctuary and perhaps other spaces? I mean really, truly listened to them? The reason that I ask is because easily 80 percent of the church loudspeaker systems that I'm invited to evaluate and re-voice ("tune," "EQ," "optimize," etc.) have something seriously wrong with them – something that the church sound techs and pastoral staff are totally unaware of.

Sometimes sound techs might be suspicious that things don't quite sound right, but they just can't pin their finger on what it is, let alone which loudspeaker has the problem. Often they say that they haven't had time to sort it out. And the reality is that most don't own the test gear to help them dig deeper into the problem.

Think there might be a problem here? (**Figure 1**) The relatively "flat" black trace is the frequency response at a main seating section. The purple trace is the frequency response of the same model of loudspeaker aiming at the adjacent seating area. Those seated in the house left section receive a good quality sound, yet those seated just across the aisle hear no high frequencies at all.

Although this might be just a loose wire, more than likely the high-frequency driver in the loudspeaker aimed at the house right seating section is blown. The worship pastor and tech team just knew that the sound coverage wasn't even, but didn't know why. You can imagine the raised eyebrows when I revealed this fact to them!

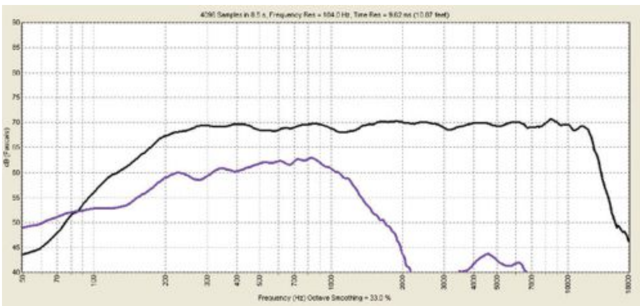


Figure 1: Comparison of two loudspeakers covering adjacent seating areas.

THE RIGHT CONNECTIONS

On the same trip, I visited another church in a town 150 miles away where I discovered that one of the stage monitors was wired out of polarity. I walked on stage with the four monitors lit up and knew within three seconds that one of them was out of polarity. It's an unmistakable sound character that can anyone can identify quickly once they're taught.

Now, I knew that one of the monitors was wired incorrectly, but I didn't know which one, let alone where the incorrect connection was. It could be anywhere. So I took a moment to prove it out with a TEF analyzer.

I've turned off the frequency response trace in this graph (**Figure 2**) to focus on the phase (it wasn't pretty). The green and magenta traces show the phase response of those two floor monitors. Note that one of the two traces shows a polarity reversal as compared with the other.

A little investigation and frankly some head scratching ultimately pointed to the fact that the wiring inside the floor box was inverted on one of the four Neutrik SpeakON connectors. The astute reader quickly grasps the fact that this system had been wired this way for years and yet no one had recognized there was a problem. The vocalists on stage had simply accepted the fact that this was normal.

Miswiring a connector is easy enough to do if one is in a hurry and doesn't have a magnifying glass to read the ridiculously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small numbers stamped into the connector. But that's no excuse for not confirming proper polarity with the wiring. The installer has a 50/50 chance of getting it right the first time!

On top of that, often the company that installed the system didn't pull individual cables to each loudspeaker. As a cost/labor saving measure, many will parallel the loudspeakers if they're flown. Sure, this saves money, but the shortcut makes it impossible to listen individually to each loudspeaker.

PAY ATTENTION TO POLARITY

A couple of years ago a well-known sound contracting company was installing systems I'd designed at two separate projects. Everything they installed in the main sanctuary worked perfectly. But when I went to test the subwoofers in the gymnasium, there was virtually no output.

The two subs were placed next to each other in a solid "bunker" built into the front edge of the platform, right on the center line. So I asked them to pull the subs out of the bunker and check the wiring, and sure enough – one of the two subwoofers was wired out of polarity (i.e., the wires to the "positive" and "negative" connection were reversed). Once the wiring was corrected, the subs delivered the expected output.

As a sidebar: Remember that when a microphone picks up a sound wave, the mic turns the acoustic energy into an electrical signal. Every piece of mic information that I've ever read states something to the effect that "a positive acoustic pressure applied to the microphone diaphragm will produce a positive voltage on pin 2 with respect to pin 3." The sound system should be wired such that when that condition exists, the loudspeaker drivers push "out" towards the listener.

Of course that electrical signal is an alternating current (AC), and the loudspeaker drivers are going to push out and then back in response to that alternating acoustic pressure. But ideally we would like the sound system energy to agree with the acoustic energy it's receiving.

If the entire system were wired in reverse polarity, it could be argued that few listeners would hear the difference. The real problem comes when one loudspeaker is wired out of polarity with respect to the rest of the system, and its sound waves interact with sound from other loudspeakers in that system. Keep reading and I'll share another story that clearly illustrates what can happen from such a simple wiring mistake.

IT HAPPENED AGAIN…

Fast-forward to a second church clear across the country, where a different installation crew from the same sound contractor company was installing another system I'd designed. What are the odds that this other crew would repeat the very same mistake that their counterparts had done one week before? You guessed it – one of the guys had wired one of the two subwoofers out of polarity, and I had to have them pull them out of the bunker and correct the wiring mistake.

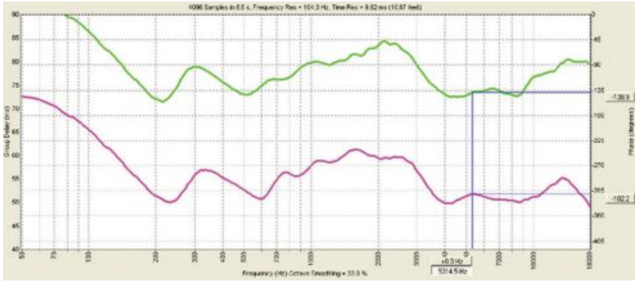


Figure 2: Two loudspeakers wired out of polarity.

The bottom line is that, despite what we might think otherwise, this stuff happens. Far too often. Often enough that I've found that I can't just trust the crew doing the install. I work with some great installers, and they're driven to make it right the first time.

But even the best of us can get in a hurry or get lost in the forest of cabling and accidentally invert the wiring. It happens. But without someone taking the time to check on things before the project is considered "complete," you don't know for certain that everything was in fact done correctly. Worse, if you weren't around when the system was installed and you've "inherited" it, you may have also inherited some surprise mistakes.

CONSTANT ATTENTION

Oh the stories I could tell, but I'll save those for another time. For now, what's a church tech to do? For starters, never stop listening analytically. If at all possible, at least once a year make it a routine preventative maintenance step to go through the loudspeaker system and carefully listen to each one individually.

Further, if possible, turn off all but one of the power amplifiers so you can isolate and listen to (hopefully) just one loudspeaker at a time. At least minimize what you're hearing so that the sound from the other loudspeakers isn't masking the problem.

Loudspeaker drivers tend to fatigue over time. That is especially true for systems driven really hard. And if they're driven too hard, things can break. You may be absolutely convinced that your system has never, ever been driven loud at all. Could it be that maybe, just maybe, someone is in your sanctuary tonight cranking up the loudspeaker system at 3 am when you're sound asleep? Stranger things have happened.

Your congregation deserves to hear the Word preached clearly, as loud as needed, and without concern for feedback or muffled sound. Don't hesitate to hire a qualified professional with the knowledge, the skills, the experience and the test equipment to ensure that loudspeaker systems do their part with excellence. CS

Curt Taipale *of Taipale Media Systems heads up Church Soundcheck.com, a thriving community dedicated to helping technical worship personnel, as well as the Church Sound Boot Camp series of educational classes held regularly throughout the U.S.*

First Southern Baptist Church

Pratt, KS

THE NEW SANCTUARY for this 800-member congregation has bypassed the ongoing shrinking of the UHF spectrum by integrating six channels of Audio-Technica System 10 PRO rack-mount digital wireless systems operating in the 2.4 GHz range. Specifically the church is utilizing four handheld transmitters, as well as two body packs working with earset microphones, for music and spoken word applications.

Advanced Sound & Communication (Kansas City, MO), which provided much of the new church's AV systems, first tried working with the existing wireless system package. "The previous contractor did not perform an on-site scan. The quantity of simultaneous systems was not recommended by the manufacturer, and the frequencies chosen weren't recommended for Pratt, Kansas by the FCC," explains Brent Handy of Advanced Sound & Communication, also noting how the area's agricultural enterprises take up much of the UHF spectrum for field communications and wireless internet. "After doing research and scanning for a week, we could not find enough UHF frequencies to work reliably," he adds.

The situation changed when an Audio-Technica rep visited Handy's office and suggested the System 10 PRO. "It worked," he says. "In fact, it worked so well that I didn't have to break the receivers and antennas out of the chassis and mount them closer to the stage; they're actually in a rack underneath the counter by the front of house mixer. You can take those mics outside the building and they still work. And just in time, really." CS

Faith Bible Church

The Woodlands, TX

THIS NEW FACILITY built for a thriving community just outside Houston offers a fan-shaped auditorium for approximately 1,000 worshippers, with a measured reverb time of about 1.4 seconds, most active in the lower frequencies. "It gives the room a beautiful natural warm sound," explains Bruce Simmons of Texas-based contractors Hairel Enterprises. "As a new build, we advised the church to engage the services of acoustical consultants Jaffe Holden to design a perfect worship environment, allowing for amplified music and occasional choral performances."

The system design offers a left-center-right (LCR) deployment: three identical hangs of the smallest d&b audiotechnik line array model, the T-Series Ti10L, with a pair of J-SUB subwoofers flown immediately behind the center array. "Imaging was the big challenge, but true to d&b's credo of 'democracy for listeners' we found when we modeled in ArrayCalc, feeding the system in mono produces a focus on center," Simmons adds. "The little E6s we arrayed across the stage lip for front fill helped pull the image down for those first rows."

Faith Bible Church house engineer BJ McGeever states, "I don't place a big emphasis on being extremely loud, but rather being extremely clear. One of my highest priorities was making sure that, within reason, every seat in the room received a clear and dynamic sound. Further, I naturally emphasize vocal clarity in my mixes – bringing the vocals front and center was wonderfully easy. Even now, months later, I'm often struck by how well covered the room is, especially in the low end." CS

First Baptist Church Of Redlands, CA

THE CHURCH'S CURRENT HOME, built in 1952, is a beautiful Spanish-style building housing a 480-seat sanctuary, but the room's classic architecture creates problematic acoustics marked by poor intelligibility. A recently implemented new system headed by Renkus-Heinz Iconyx Gen5 loudspeakers, designed and installed by Ireland Sound Systems (Upland, CA), has solved the issues.

"The biggest requirement was superior intelligibility for spoken word," recalls Patrick Ireland, owner of Ireland Sound Systems. "However, the congregation also has a praise band that plays at services, as well as a choir and an organ. The Iconyx Gen5 system was a clear choice because it handles both speech and music extremely well, and it has plenty of power for the praise band."

Iconyx Gen5 is the company's fifth generation of digitally steered array technology, offering a wider selection of configurations and added precision. For the main front-of-house system, Ireland chose a pair of Iconyx IC16-RN digitally steerable line arrays, flown about 11 feet above the floor to the left and right of the stage.

The layout of the sanctuary was a major consideration in choosing Iconyx Gen5. "The room is like so many churches of this era, and covering that kind of space with a conventional system is very challenging – consistent coverage to every seat is difficult to achieve, and getting a system to convey natural sound is problematic," says Ireland. "But with Iconyx beam steering and the added flexibility of the new Gen5 series, we didn't have to compromise." CS

Woodstock City Church

Woodstock, GA

BRANDON THOMPSON (pictured here), production director at this campus of North Point Ministries, has chosen to utilize Waves SoundGrid technology, along with Waves Tracks Live and Waves plugins.

Specifically, he's running the Waves MultiRack on a DiGiCo SD9 digital console, joined by the Waves Live bundle, Studio Classics Collection, Jack Joseph Puig Signature Series, Analog Legends, CLA Classic Compressors, C6 Multiband Compressor, EMI TG12345 Channel Strip, dbx160 Compressor/Limiter, and H-Reverb – all running on a SoundGrid Server One.

"Additionally," Thompson notes, "we're running Waves Tracks Live as our preferred recording and playback solution. The DiGiGrid MGB has been super-useful to use with virtual sound check duties for our DiGiCo SD9."

He concludes, "Utilizing the MGB for 64 channels of MADI audio, we copy all incoming stage rack audio to the MGB. Then we track the rehearsal to Waves Tracks Live running on a Mac Mini, and using the 'Listen to Copied Audio' function on our SD9, and listen back to all tracked audio right after the band has finished. We spend a few hours working on mixes, adding in Waves plugins via SoundGrid, and then bounce out of the SD9 to an MP3 recorder, which is then posted online for the band to listen to." CS

Christ Presbyterian Church

Edina, MN

THE 1,000-SEAT main sanctuary was recently outfitted with a new sound reinforcement system designed and installed by REACH Communications (Champlin, MN) that incorporates NEXO components. A main line array includes four GEO S1210 modules and one S1230 module, and two side arrays each have two GEO S1230s and a GEO S1210.

REACH also added four of the new NEXO GEO M620s for front fill along with ten PS8s for under balcony fill. Three NXAmp 4x4 and two 4x1 amplifiers power the loudspeakers, joined by five NXDT104 controllers. "The NEXO family was chosen for a variety of reasons for this particular project," says Brad Van Voorst, project leader for REACH. "The room is a highly reverberant space that really benefits from the low frequency pattern control that a true line array offers as well as the consistent phase response between the different models within the NEXO family.

"Being able to smoothly transition from the GEO S12 array to the M6 front fills and the PS8 under-balcony fills was an important feature. Dante (networking) integration in the NXAmps kept the entire signal chain digital, which helps to reduce the overall noise floor. Speakers with a white finish were necessary, given the traditional décor of the room."

Last year, REACH had installed a Yamaha CL5 digital console with two Rio3224-D input/output boxes in the sanctuary, and during the recent loudspeaker installation, a BSS London BLU-806 processor was added to the system so that Dante networking can run directly from the console to the BLU806 and then out to the NEXO amps. A Waves Server One and a Waves plugin package were also added. CS

Hillsong

Australia

THIS MULTI-SITE CHURCH with campuses in Sydney, Brisbane, Melbourne, Newcastle and several others, as well as numerous international campuses around the world, has partnered with Adamson Systems for sound reinforcement at multiple venues.

"It started with our looking for a new system that would better meet our needs for the City Centre worship space in Sydney," explains Hillsong production facilities manager Steve Le Roux. "At the same time, we knew that there were existing systems in other venues that needed to be upgraded, plus new churches under construction that would require sound reinforcement. It made sense to be incredibly thorough – we sat through many, many product demonstrations – in order to decide upon one manufacturer to work with for everything. In the end, Adamson came out the clear winner."

The main campus upgrade consists of E12 and S10i line arrays as well as the recently released E119 subwoofer. "The subs are both compact and extremely powerful – ideal for our City Centre space," notes Hillsong head of audio Ricki Cook. "They provide serious low end when needed while blending seamlessly with the PA." The majority of Hillsong sanctuaries planning to install new sound systems in 2016 are mid-sized venues. Most of those systems will rely upon S10i arrays and S119 subs for sound reinforcement. CS

CONSOLE ORGANIZATION

A logical, intuitive approach to setting up the mixing board.

by Chris Huff

It was a mix engineer's nightmare. Vocal microphones for singers were spread out across console channels 2, 7, 10, and 13. To make matters worse, they were labeled across as C, A, D, and B – not even in alphabetical order. The rest of the channel assignments were no better. I wept. (Not really but I felt like it.) What better way to guarantee frustration and a slower mixing process?

The tech at the church in this example had employed a method many use when new to live audio production – assigning channels in an order matching where things are located on stage. Though this might seem to make for a speedy way of finding a channel, there's a far better way.

The best console channel layout approach makes channels easy to find, easy to control, and easy for the next person to use the board. No matter how the stage is set – and musician placement will change – it shouldn't require channel reassignment.

Think of it this way: when a pianist reaches for a key, he doesn't hesitate because the key might have been moved. In the same way, reaching for the snare channel on a console should be as effortless.

LOGIC TO THE PROCESS

The standard order of channels is; drums, bass, rhythm instruments, piano and keyboards, percussion and other instruments, lead vocal, backing vocals, choir, pastor, extra speaking microphones, and finally, playback devices (CD, DVD, or audio feed from the production room).

My personal preference is to have empty channels after rhythm instruments and backing vocals. This enables adding a channel if the band grows or a guest musician or band comes in and extra channels are required.

Drums have a specific order: kick, snare, hi-hat, tom 1, tom 2, tom 3, overhead left, and overhead right. Where a mic isn't used, don't leave a space, so a minimal drum miking setup could be kick, snare, and overhead. An alternative drum order is by frequency (low to high); kick, tom 1, tom 2, tom 3, snare, hi-hat, overhead left, and overhead right. The overheads are capturing everything, but for channel assignment purposes, having them at the end makes the most sense.

Rhythm guitars are arranged by type, so a large band might have acoustic 1, acoustic 2, rhythm electric, and lead electric. Specifying lead and rhythm is great because when the band changes and different musicians take the stage, the channels are routed according to their role. Need to boost the lead guitar? Not a problem.

Guitarists can change roles within a song or within a song set and it's up to the engineer to decide how to label these. Using digital scribble strips and scenes, labels can change per song. My preferred method is to identify the guitarist who plays lead the majority of the time and label that one as "Lead." Then, it's a matter of knowing the song arrangements come mix time.

Piano and keyboards come next. Keyboards can be set for a variety of uses, from melody lines to synth pads. When a musician dedicates keyboards to these uses, label them accordingly for easy mixing.

This is followed by extra instruments, from percussion instruments to violins to whatever else is used. Order these by usage popularity. For example, a band with a regular percussionist and an occasional violinist would have channels assigned from left-to-right, with percussion and then violin.

Now we arrive at vocals (singers), and the lead is first. For churches, I place the worship leader in the first slot position and label it "WL" (Worship Leader). I've had many occasions when the worship leader does not lead a song but will speak when transitioning between songs. Also, this channel will be excluded from mute groups – more on that shortly. All other vocal channels are labeled with the singer's name.

Choir mics are best listed by their order on the stage. And while it's easy to think all choirs mics should be set with the same gain, fader level, and EQ, that's usually not the case. Bass singers might be all on the left side, while sopranos could be off-center. Whatever the situation, using the stage order aides in providing a proper choir mix and easy tweaking. (Trust me on the tweaking.)

Pastor mic labeling can go a few ways. If there's one pastor, the channel can be labeled with the pastor's name or as "Pastor." On analog consoles, I use the generic term, but on digital consoles I use their name. All other speaking mics should be listed next. If there's a dedicated spare microphone, place it at the end of this section. Remaining channels will be assigned to input devices and can be listed in alphabetical order, nothing complicated.

Before starting, review the console for channel layout, mix groups, and mute groups.

MIX GROUPS

Once all channels are assigned, these channels can be assigned to mix groups which allow for easy control over multiple channels like boosting a balanced drum mix, adjusting backing vocals, and making other mid-song adjustments.

Create mix groups for drums, keyboards, guitars, and backing vocals. If there's a percussionist with several mics, add a percussion group. I add the bass guitar to the drum group because I don't want the bass to disappear if I boost the drums. Another option is to add a low-end group just for the kick and bass.

Mix groups for backing vocalists need to be adjusted when the lead singer changes. With the scenes provided by digital consoles this is easy. With analog consoles, I put all singers (lead included) into the vocal group and make sure to tweak the channel level balance between songs.

MUTE GROUPS

An open mic at the wrong time can be disastrous and muting all of the required channels can be a chore in itself. I've seen a sound tech muting channels when the band was leaving the stage but the guitarist unplugged before their channel was muted. Pow!

Mute groups provide the ability to mute multiple channels at the same time. Assign channels to mute groups in much the same way as mix groups with a few exceptions. Keep the pastor and worship leader out of mute groups. A worship leader may lead the congregation in communion or be the last voice of the service while the rest of the band is exiting the stage. Much the same can be said of the pastor.

Use this standard channel layout and mixing will become much like playing a piano – reaching for channels instinctively rather than hunting and pecking.

Mute groups and mix groups are especially helpful with digital consoles that use channel layers (banks), such as when a console has 24 channels but only 12 faders. The less jumping between layers, the better.

UPHOLDING STANDARDS

Use this standard channel layout and mixing will become much like playing a piano – reaching for channels instinctively rather than hunting and pecking. In cases where there's an audio team, make this standard layout mandatory. Not only does it improve mixing speed and provide added controls, but when a sound tech calls in sick (not like that ever happens), another tech can step in and not miss a beat.

I've been called at the last minute to mix an event, and a standard channel layout made my job easy. But I've shown up with the console in complete disarray, making my job so much harder.

Before embarking on this process, review the console for channel layout, mix groups, and mute groups. It might mean a few changes must be made or it's time to start from scratch. Whatever the case, it will lead to easier mixing and bring order to an aspect of mixing and sound operation that demands it. CS

__Chris Huff__ is a long-time practitioner of church sound and writes at Behind The Mixer (www.behindthemixer.com), covering topics ranging from audio fundamentals to dealing with musicians – and everything in between.

Andrew Stone at front of house at COTM in Tulsa.

THE RIGHT FIT

Andrew Stone injects a passion that translates to production success. *by Erik Matlock*

Church on the Move (COTM) in Tulsa is the home to Pastor Willie George, a passionate minister who was first introduced to me through a series of videos that my oldest daughter watched in children's church. That was over 20 years ago, and COTM has since grown into one of the most respected and well-known churches in America.

At the helm of the church's seven-day-a-week schedule, assuming responsibility for all aspects of live production, is another person I've found to be equally passionate about his mission: Andrew Stone. He claims to have never even held a "real job," only audio and production gigs ranging from touring engineer to consultant, and his present position bears the title of production manager for the 300-acre campus with 18 staff member schedules to manage in his department alone.

A LONG ROAD

Originally a drummer who toured internationally with the Oral Roberts University ministry teams, Stone eventually transitioned to the other end of the snake. His skills and passion landed him at the front of house position for a diverse range of bands, both Christian and secular – Third Day, Kitaro, and Psychedelic Furs, to name a few.

Born and raised in Oklahoma, but at that time residing in the Nashville area, he was contacted by an old friend from the ORU touring days, seeking a recommendation. "The church needed help in production," Stone explains, "and basically wanted to know if I knew anyone who might be a fit for them."

After he met with the church staff and spent a week consulting on production changes and improvements, they had a different question: Are you the guy? The timing seemed ordained. He and his wife Natasha were in the process of contemplating the future and whether or not touring was to be their life, as well as thinking about returning to Oklahoma to be closer to family.

Without knowing it at the time, their lives were about to make a drastic change, with Stone repositioned directly in the path of a new career in church production. He took on the new role at COTM.

FOCUS ON WORK & HOME

He also operates Rock Productions to provide freelance production services, and in addition, is involved in helping other churches to achieve a more professional level, working with partners Lee Fields and Jeff Sandstrom on MxU (mxu.rocks),

which travels the country training church technicians on the latest technology and techniques.

"What we began to notice," he says, "was that with many of the church technical conferences, a lot of people who were attending still didn't feel secure about what they were doing. So in creating and providing the type of training sessions that we wanted to attend, we make sure that everyone leaves with something that makes them more confident."

Stone also contributes to COTM's Seeds ministry (seeds.churchonthemove.com) which makes useful tools and techniques available to assist other growing churches. He places a heavy emphasis on training his team as well as others, joined by an intense passion for taking personal matters as seriously as his ministry life.

"One of the biggest challenges for me is the fact that I really enjoy what I do, and I'm also surrounded by a team that feels the same way," he says. "Part of my job, in scheduling these folks, is making sure they're taking time off and focusing on their home life; literally making them go home sometimes.

"Taking a day of rest is still one of the Ten Commandments," he continues. "We have protect ourselves from burnout by separating ourselves from our work and resting. If we fail to follow that simple rule, it affects everything including our families and home life."

GRANTING RESPONSIBILITY

Stone also makes a point of telling tech team members to speak up, whether with problems at church or at home. "We've been purposeful in making them know that this is a very busy place and they have to let us know if there's an issue," he points out. "It's so much better to have a relationship where they can let us know when they're tired or having problems.

"I make sure they know that speaking up is their responsibility," he adds. "It's tough when someone waits five years and all of the sudden they're burned out and just want to quit."

Another piece of the puzzle is offering salient advice to technical directors for leading their teams and recruiting volunteers. "If I interview a potential volunteer, I'm much less enthusiastic about someone telling me how they've done things as compared to someone who is asking about how we do things," he points out. "What I'm most interested in is finding people who are eager to serve instead of someone who is eager to perform. Performing makes it all about 'me' while in production, it's all about serving others."

He has a process that has been developed over years of experience for training new tech volunteers. It starts with finding the lowest common denominator, their baseline of experience, whether it be on camera or with stage crew, and then putting them into a worship service to see how they fit.

"Any kind of training we do, involves immediately immersing them in the real situation," he says. "There's no classroom or blackboard. We will have them shadow someone who's already doing the job and see how they pick it up and respond in real time. They may only be listening in on a headset, but they're part of the action."

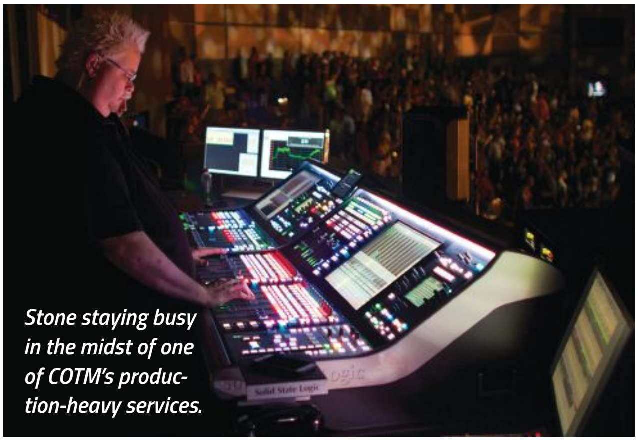
So from the first day, interns are on the job. No courses to graduate from; it's pure on-the-job training. Safety and techniques are covered, skills are assessed, but the real education comes from getting hands on the gear and working. "I don't really know how the other churches do it. This is how we do it, and it works quite well," he says.

STAYING ENGAGED

There's also a transition underway at COTM as Pastor George begins the process of moving his son Whitney into the senior pastor role. This against a backdrop of the church also pulling back on outward projects in an attempt to focus on its own internal health.

Much like Stone's passion for maintaining boundaries and protecting home life, the church itself is taking a season to rest and care for its own. Still, the focus remains on not getting complacent, to work diligently at constant improvement in the pursuit of consistent technical excellence.

"I try to remind everyone that they do have a voice," he notes. "Everyone has something to contribute. Speak up and engage. All these people have great ideas. We're creative beings, and some of the most creative people are on the tech staff. I don't want to see someone just standing there with their hands in their pockets when they might have the exact solution we need."



Stone staying busy in the midst of one of COTM's production-heavy services.

With this expected assertiveness comes responsibility in being proactive to improve the team and production quality. "We never have to ask permission to take responsibility. I don't want to hear anyone tell me that something isn't their job or their problem. It's never someone else's problem. Own it, take responsibility. Start being part of the solution. The people who engage and take responsibly are the ones who find real success. The best leaders are there because they spoke up and contributed."

In the words of a pastor I once served under: "Success without a successor is ultimately failure." Church on the Move seems to be a prime example of success built on developing passionate successors. CS

Senior editor ***M. Erik Matlock*** *has worked in professional audio for more than 20 years in live, install, and recording, including time as a church tech and media director.*

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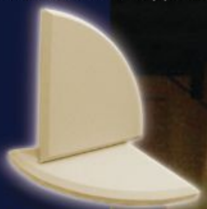

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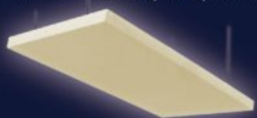

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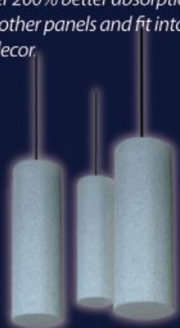

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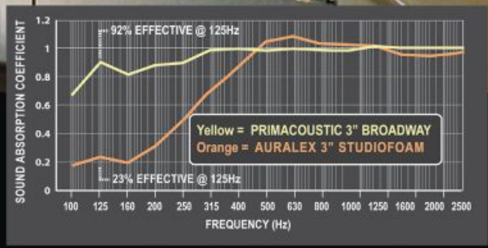

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CHOICES & PLACEMENTS

Techniques for miking a wide range of instruments.

by Craig Leerman

An Audio-Technica ATM650 hypercardioid dynamic mic on a snare drum.

Great sound starts at the source, with the proper microphone in the right position to capture the sound of the instrument and vocals. A "proper" mic doesn't mean the most expensive; many modern mics priced below $200 work very well for live sound applications.

The difference between a good and great mic can be quite audible when heard over quality monitors in a recording studio but may be less audible when heard over a sound reinforcement system in a reflective room full of people. A proper mic is simply one that has the required pickup pattern and frequency range for the intended purpose.

Since the majority of the mics available from reputable manufacturers do what they're designed to, I look for rugged models with low handling noise that can handle high sound pressure levels (SPL) for use on stage. A key is using directional mics and placing them in good positions where they pick up the intended sounds, and reject other sounds onstage like stage monitors or adjacent instruments.

In fact, before even selecting mics for the stage, I take a look at how the instruments are set up and what that will mean in terms of stage volume and the leakage of sound from one area to another. The goal, as much as possible, is to prevent unwanted sounds leaking into mics by moving instrument amplifiers or even pointing them offstage, or placing blankets or plexiglass between amplifiers and acoustic instruments. In addition, monitor wedges should be positioned so they point into the null lobes of mics. (The null lobe is where a mic picks up the least amount of sound.)

PRINCIPLES OF OPERATION

Once the stage is set, it's time to select the mics, a process of selecting the "best" model for each instrument from my kit. But first, here's a quick primer.

Mics are usually categorized by the conversion process that's used to turn acoustical energy into electricity. The most common type used on live stages are dynamic designs that work on the electromagnet principle where a coil of wire is attached to the diaphragm and moves by a magnet, creating electricity as the sound waves push against the diaphragm. Dynamics can be very rugged and resilient to rough handling, but because there's added mass attached to the diaphragm, they may not respond as quickly to changes in sound pressure as other types.

Condensers have gained in popularity in the live world, with many newer models robust enough to withstand abuse. They have two plates with a voltage between them. One plate is made of very light and flexible material, and acts as the diaphragm. The diaphragm vibrates when struck by sound waves, changing the distance between the two plates, which in turn changes the capacitance, producing an electrical signal.

Condenser diaphragms are not connected to a coil mass. They respond to transients and higher frequencies very well, enhancing sensitivity and the ability to capture high-frequency detail with more accuracy. A variation of the condenser is the electret, which uses a fixed charge on one of the plates. Note that condensers require a power source (phantom power or batteries).

The vast majority of mics are unidirectional, with pickup patterns including cardioid, supercardioid and hypercardioid. In general, they reject sounds from directions other than the front (at various degrees), and this is highly desirable given all of the noise on a typical stage as well as from the room.

Cardioid mics have a wider but directional pickup pattern with a large null zone at the rear, so they're a good choice for general miking as well as vocalists. Supercardioids are more focused to favor sounds from the front and have greater attenuation at their sides – about 10 dB – exhibiting minimum sensitivity 140 to 150 degrees off-axis, with a minor secondary response lobe at their rear. Hypercardioids are even slightly more focused and distinguished by minimum response 110 to 120 degrees off-axis at the expense of a slightly larger rear lobe.

Another way mics are categorized is by their intended use on instruments or vocals, but many (applied properly) sound good in either application. One phenomenon that comes into play is proximity effect – the closer a mic is to a sound source, the more the lower frequencies are pronounced. Some mics address this issue with acoustic tuning, low-cut switches, or electronic processing.

MATCHING IT UP

As a general rule, it's preferable to use larger diaphragm dynamics on low-frequency instruments like kick drum, floor tom or bass guitar; medium-sized dynamics on more general instruments (such as guitar amps) and vocals; and small-diaphragm condensers for higher pitched or softer-sounding instruments.

My company's "mic locker" contains mostly common industry standard models that most sound people know. We serve a lot of different types of shows with visiting engineers, and while they may request a particular mic for an instrument or vocalist, if I don't have it, they can still get the job done with my familiar models.

As a result of decades of this work with so many types of performances on a wide range of stages and in an equally wide range of venues, I've been able to formulate mic choices and placements for instruments that work well or that can at least serve to get you started.

Acoustic guitar. Place a small-diaphragm condenser at about the 12th fret position, approximately 8 inches from the guitar, and point it toward the sound hole. This location gets a good balance of the instrument while being out of the way of the musician. Roll off everything below 70 Hz and above 16 kHz, and pay attention to the 100 to 250 Hz region because it may need a bit of a cut as well.

Ukulele. These small instruments have become popular in recent years and range from baritone to soprano. A small-diaphragm condenser pointed between the sound hole and the neck, about 8 to 10 inches away from the body of the guitar, is a good starting point. Roll off everything under 150 Hz and above 16 kHz.

Electric guitar amplifier. Position a dynamic mic about 2 inches away from the cabinet, pointed about halfway between the center of the speaker cone and the edge of the speaker. For amplifiers with more speakers, just mike one of them. Move the mic toward the center of the cone for a brighter sound. As with an acoustic guitar, get rid of everything under 70 Hz and above 16 kHz. With more than one guitar player (and amp) on stage, try to give each instrument its own distinct tone so they both fit in the mix. If a player uses heavy distortion, thin out the sound by decreasing in the

A single-speaker guitar amp and a quad-speaker amp, both miked with a single dynamic.

70 to 150 Hz so there's room in the mix for the kick drum and bass guitar.

Acoustic bass. Normally, upright bass players have a pickup or mic already installed on their instruments if they use an amplifier, so just tap off the mic or use a direct (DI) box for a direct feed. For instruments with no pickups, and if the player doesn't move around a lot, place a large diaphragm dynamic on a short stand, pointed at the middle to top of the sound hole (the F hole) on the low string side so it's positioned above the bridge. Some folks like to position the mic on the treble string side, but I find that the high notes seem to be a bit louder on most of these instruments, so I opt for placement on the low string side.

If the player moves around a lot, clamp a small mic on the instrument (And be sure to ask for permission from the musician before doing this.) I have some older small condenser mics with gooseneck horn mounts that, while designed for brass, work well attached to the bridge of a bass pointing at the sound hole. I've also found a variety of other small gooseneck instrument mics that do the trick.

Multiple mics applied to capture a typical drum kit. And note the sandbags ballasting down the overhead stands.

If you don't have any small clip on instrument mics, a trick using rubber bands can be employed. Loop one rubber band around each side of the bridge so the end loop of each band is hanging toward the middle of the bridge under the strings. Then wrap the band ends around a small dynamic or condenser mic that points up toward the neck. The rubber bands act as a suspension mount and by securing the mic's cable to the bass tailpiece with gaff tape, the mic is secured into a stable position.

Several mics handling a percussion section.

Electric bass. While a DI box is normally placed between an electric bass and the amplifier, my preference is to also mike the bass amp if an extra channel is available, especially if the bass player uses effects in his/her signal chain. A large- and sometimes a medium-diaphragm dynamic usually works well for this. As with a guitar cabinet, position the mic about 2 inches away from the cabinet, pointed about halfway between the center of the cone and the edge of the speaker.

Drums Kits. The only way to attain good drum sound is to start with a well-tuned kit. As a sound person who also plays drums, I've been known to "help" drummers get their kits in tune. The goal with mic placement is to get a quality, natural sound from the drum you're trying to pick up while rejecting the sound of the others as much as possible. This generally dictates utilizing mics with tighter polar patterns. There are a wide range of quality options available in terms of mics that are primarily designed for drum applications. And unless the kit is in a small room/space, my approach is to mike each drum individually and also employ overhead mics.

Kick drum. Employ a large-diaphragm dynamic that can handle high SPL. If the drum has a hole in the front head, position the mic just inside to just outside the hole, depending on the sound, and point it toward the center of the batter head. If there's not a hole in the front head, opt for a batter head placement, positioning the mic between the kick pedal and the edge of the drum head.

Some mics for kick drums have a flat response but many have a contour designed to enhance the sound with minimal EQ changes. No matter the mic, pay attention to the 60 to 100 Hz region, because this is usually where the kick sound will set in the mix. For more punch, try adding a bit of compression, with a 20 to 40 millisecond attack and a 2:1 ratio, and set the threshold for just a bit of gain reduction.

There are entire book chapters devoted to miking pianos, but I stick with a few simple methods.

Toms. Smaller dynamics do the job here, placed 2 to 4 inches from the drum head near the rim and pointing in toward the head, out of the drummer's way. This provides a complete picture of the drum fundamental (center of the head) and overtones (closer to the edge). I prefer to use drum clips and claws for attachment because they present a cleaner look. In addition, I don't like to gate toms (or any drums for that matter), but will gate rack toms if the snare is really loud and bleeds into the tom mics.

Snare. Again, smaller dynamics work well, placed similarly as on the toms, about 2 to 4 inches away, near the edge, and pointing in toward the center. For more snap, a mic for the bottom head can be added, usually another dynamic but sometimes a small condenser. It should point at the drum's snares, about one-third from the edge of the head. Be sure to swap the polarity on the bottom mic (by using an adapter that swaps XLR pins 2 and 3 or the often mislabeled “phase” switch on the console channel) so the sound doesn’t get too thinned out.

Hi-hat cymbal. A small condenser is preferred but if one’s not available, go with a dynamic. Place the mic so it doesn’t pick up a lot of snare or toms, and try to make sure that any instrument amplifier near the mic is in a null zone. Roll off everything under 180 Hz and above 14 kHz.

Overheads. These are useful in helping to pick up cymbals as well as the overall kit sound. Most engineers choose small condensers for this application. Position the mics at a height of 7 to 8 feet, with one pointed at the ride cymbal and the other aimed at the cymbals on the other side of the drum kit. Roll off everything under 200 Hz, and place sandbags on the mic stands so they stay in place.

Percussion. These instruments come in three flavors: struck by hands (i.e., congas and bongos), struck with sticks (i.e., timbales), and handheld (a.k.a., “toys”) that are shaken or struck (i.e., tambourines and maracas). Close mike hand- and stick-played percussion, with the mic positioned 2 to 4 inches above the head near the edge, looking in toward the center of the head. Roll off everything below 100 Hz and pay attention to the 2 to 5 kHz range in the EQ to help bring the instruments out in the mix if need be. For percussionists who sing, their vocal mic can double as the toy mic, and if not, place a cardioid dynamic on a stand for the handheld instruments and roll off everything under 150 Hz.

Grand Piano. There are entire book chapters devoted to miking pianos, but I stick with a few simple methods and they all seem to work well depending on the situation. For a grand piano, if it’s feasible to open the lid to the high-stick position, place two condenser mics about 12 inches above the strings about 8 to 10 inches away from the hammers.

In addition to stand-mounted mics, several manufacturers also offer clip-on miniature mics for instruments that work well.

They can be located together around the middle of the harp, with one pointing at the high strings and the other aimed at the bass strings. If the middle strings seem too loud, separate the mics a bit. If the lid can only be at the low-stick position, try side-address condensers in the positions above but place them a little closer to the strings, then separate them as needed to get a balanced volume across the keyboard.

Another option with the lid at low-stick position (or with a closed lid) is to use surface-mounted “boundary” microphones. This method will work with any type of flat boundary mic. Tape a mic to the lid about halfway between the middle of the keyboard and the highest string, and then tape a second mic about halfway between the middle strings and the lowest string. Try to keep the mics at least 8 inches from the hammers and dampers. Being closer to the hammers results in a brighter sound and attack but if they’re too close, they can pick up mechanical noises from the dampers on soft musical passages.

Once for a theatrical production, the lid had to be closed. I taped a boundary mic inside the lid at about the middle of the piano, and then pointed a medium-diaphragm dynamic at the back of the soundboard to help pick up the bass strings. It worked well, providing a nice, balanced sound.

Keyboards. While electronic keyboards and synthesizers get DI boxes, there can be acoustic keyboard instruments to deal with, such as harpsicord and celeste for an orchestral production. Use a two-mic approach for the harpsicord, placing the mics about 12 inches above the strings and 4 to 6 inches away from the jack and plectrum. The plucked sound of the string is what you want to amplify. A celeste is basically bells that are played by keyboard and not with sticks or mallets. (Tchaikovsky's "Dance of the Sugar Plum Fairy" melody from The Nutcracker is played on a celeste.) Position a pair of small diaphragm condensers about 12 inches away from the resonator chamber, located about one-third in from each side.

Organ. Not a traditional church organ, but rather, a Hammond B3 model with a rotating Leslie speaker. It has a woofer that down fires into a rotating baffle and a compression driver that fires up into a rotating unit with two horns. If possible, utilize three mics, but if not, two can work. One is positioned 8 to 12 inches away from the bottom baffle, and it should be a medium- to large-diaphragm dynamic because it's capturing the lower frequencies, similar to the choice for bass amplifiers. With the top horn, place a pair of small condensers, one at each cabinet corner, making sure they will not be hit by the rotating horn. Panning these mics can result in a rich stereo sound, but a single mic here can also work well if it's moved back from the rotating horn about 12 inches or so.

Clarinet. Some of the sound comes out of the bell and some comes from the open key holes, so a larger diaphragm dynamic is usually my choice but any quality mic can work in a pinch. Position it about halfway up the instrument, a foot or so away. Roll off everything under 100 Hz and above 10 kHz. (This same approach also works well for straight soprano saxophones.)

Saxophone. A medium cardioid dynamic will usually do the trick. While some of the sound does emanate from the keys, a pretty good tone can be captured just from the bell by placing the mic about 6 to 8 inches from it. Roll off everything under 75 Hz.

Brass. My rule of thumb is the bigger the horn, the bigger the diaphragm. Trumpets and coronets get a dynamic that can handle high SPL. I ask performers not to let the mic get inside the bell, and roll off

Don't hesitate to experiment and develop your own optimum "go to" mic placements.

everything under 150 Hz and above 10 kHz. Trombones can use the same mics as a trumpet or step up to a larger diaphragm model. Roll off everything under 75 Hz and above 8 kHz. Tubas and Sousaphones can be captured well with a mic suitable for kick drum. Roll off the low end at about 30 Hz and get rid of the highs above 6 kHz.

Strings. My preference is to "distance mike" violins, violas and cellos so the mics don't get in the way of the performers. Violins and violas are handled with condensers about 2 feet above the performer, with the mic looking at the strings above the bridge. The key is to stay out of the way of the bow. Cellos can be captured with a condenser placed on a short stand about 2 feet in front of the performer, with the mic pointed at the strings above the bridge. Roll off everything under 100 Hz and above 16 kHz for violas and violins, while with cellos, the range is under 75 Hz and above 10 kHz.

Please note that these are all general guidelines, intended to provide a useful outline that serves as a starting point. Don't hesitate to experiment and develop your own optimum "go to" mic placements. CS

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

Alleluia Lutheran Church

Naperville, IL

THE MAIN SANCTUARY on this 80,000-square-foot campus in suburban Chicago is now outfitted with a Yamaha Live Sound TF5 digital mixer, allowing the existing Yamaha LS9 digital mixer to find a new home in the church's youth center. Alleluia Lutheran provides four services each weekend, one traditional and the other three contemporary with various band configurations, running some 22 to 25 input channels to the new mixer.

Audio technical director Ryan Russell (pictured here) notes that the church wasn't originally in the market for a new mixer. "The music staff – the band – wanted the ability to control their monitor mixes from the stage," he explains, and exploring a number of hardware solutions, "I had this crazy idea after reading a little bit about the TF5. We don't need to upgrade our console, but why don't we update to the TF5? It gives us all the iOS app capabilities, so then the band – whoever has in-ear monitors – could be mixing from their iPhones."

A generous donation allowed the church to procure its new TF5 from McCormick's Enterprises (Arlington Heights, IL). Now the worship leaders and members of the music staff are using the Yamaha iOS MonitorMix app to control their individual monitor mixes during performance. And while the TF5 offered a convenient solution to the monitoring issue, it has presented other benefits as well.

"It [the TF5] is way easier to use than the LS9 for my volunteer staff," Russell states, referring to the multi-touch display. "I was able to teach them in 15 minutes the basics of the console, and they are doing far more on this board than they ever did. It's been awesome." CS

Church Of The Incarnation

Dallas

THE NEWLY constructed 500-seat worship facility presents several requirements for the sound reinforcement system – it needs to be heard but not seen, flexible enough to handle a variety of types of music, and because it's a highly reverberant space, sound must stay off the walls to aid vocal intelligibility.

In response, Dallas-based design firm idibri specified a system headed by Eastern Acoustic Works (EAW) Anya loudspeakers, with just two modules left and right proving capable of meeting the specific challenges. "The church is gothic looking, a long narrow room with traditional high ceilings and a slightly cruciform design," explains Ryan Knox, consultant with idibri. "Aesthetics were a priority. They had two insets to the left and right of the alter for sound reinforcement purposes that they hoped would accommodate a contemporary sound system that could handle full range music. I'll admit, it was a pretty tall order."

Anya adapts all of its performance parameters electronically, with the modules hanging straight and without any vertical splay, so it proved the right fit within the limited footprint for loudspeakers. "We specified two Anya modules for each array for two reasons: we thought that would be more than enough system for the room, and also because that was all the space we had," Knox says. "Anya solved a lot of problems from a design perspective and from an audio perspective."

He used EAW Resolution 2 software to dial in asymmetrical output that delivers coherent, full-range response across the coverage area. "I was really pleased when we went to tune the system that it required very little work beyond configuring the Resolution 2 software," Knox adds. "The engineering behind it is really impressive." CS

Canvas Church

Irvine, CA

FORMED IN 2011 "for people who don't like church," the original congregation quickly more than tripled in size and relocated to a larger home, occupying a warehouse space in an industrial park that's outfitted with a sound reinforcement system highlighted by a PreSonus StudioLive AI Active Integration console and StudioLive AI loudspeakers.

"We do a sort of post-contemporary service, and we've typically got anywhere between four and nine musicians up on the stage," explains Norman Gordon (pictured here), the parishioner who specified the system and can typically be found running front-of-house. "It's a full band: bass, drums, guitars, keyboards, and a few vocalists. There's a lot of stuff going on."

Gordon recommended the StudioLive 32.4.2AI console, along with StudioLive 328AI full-range loudspeakers and a StudioLive 18sAI subwoofer. "The AI series really had everything we needed," he says. "We had a 24-channel analog console we'd been using, and even though we really didn't need the extra channels right away, we went for the 32-channel mixer because it was the first console released in the AI line.

"Having been an 'analog guy' most of my life, I really love that warm, English, analog sound," he continues. "It's something I wouldn't have expected with digital. I have to say, though, I was really, really pleasantly surprised at how warm and open the console sounded."

Gordon adds, "While we were waiting for the speakers to arrive, we had a couple of different loaner systems in here. It didn't sound bad, but when we installed the StudioLive speakers and fired them up for the first time, the difference in clarity was just amazing. Articulation is really, really important to me – the ability to understand both the singing and the spoken word." CS

Free Chapel

Spartanburg, SC

THE FOURTH AND NEWEST campus for the Gainesville, GA-based church has a 3,000-seat worship space purchased from another church and completely overhauled, including a new sound reinforcement system. Free Chapel called upon Mankin Media Systems (Franklin, TN), which specified and installed L-Acoustics Kara loudspeakers as the key components.

Developed with input from L-Acoustics Soundvision software, the design utilizes left and right arrays each comprised of nine Kara enclosures backed by subwoofer arrays of three SB18. In addition, four SB28 subs are on the ground, positioned under the stage, while eight coaxial 8XT mounted just under the lip provide front fill. Four LA8 and three LA4X amplified controllers power and process the system.

"First, weight and power requirements were critical because the original facility was not designed with the production capabilities in mind that Free Chapel required to execute their weekend service environment in the space, " explains Tim Corder of MMS. "We needed to maximize performance while minimizing the structural loads on the roof, as well as the current draw. Other manufacturers' options that we considered were both heavier and power-hungry in comparison.

"Second, we knew this was going to be a 'phase one' renovation and, as the church continues to grow, there will be further expansion of the seating layout, as well as additional spaces and systems added to the campus. As such, we needed a flexible system design that would also allow them to grow into the space over the coming years. L-Acoustics provided a complete solution that included sonically-matching main arrays, front fills and subwoofers, as well as future out fills. The system fit the room like a glove." CS

HUM & AWE

Identifying and correcting troublesome noises in sound systems. *by Mike Sokol*

One of the most common questions that comes up on many live sound forums is how to stop noises in a sound system. I've been doing a lot of experimenting on this subject over the past several years, plus I've been battling sound system noise such as hum in the field for more than 45 years, so here's my observations on sound system noise and what to do about it.

Before having any chance at stopping any kind of noise in a sound system, first we need to define what it is. Here's a short list of noises that often appear in live sound systems and what we call them.

Hum – A steady-state bass noise at 60 Hz (cycles per second) without any significant harmonics. It sounds like humming with your lips closed, and is very close to low B-flat on a bass guitar with all the treble turned off.

Buzz – A steady-state noise that's typically at B-flat as well, but it also includes a lot of harmonic overtones with a lot higher pitch. It sounds like a noise you would make with your tongue at the back of your teeth though open lips. Or just touch the tip of your guitar cable while it's plugged into a stage amp that's turned on. The sound you hear will be a buzz.

Hiss – A steady-state white noise that sounds like rain on the rooftop or steam coming from a kettle. You can simulate it by blowing air though your lips without quite making a whistle. This is actually white-noise, not the pink-noise you might use to tune the sound of a room using a real-time analyzer (RTA).

Hash – A pulsating "buzzy" noise that can sound like bacon frying or buzzes being modulated. Sometimes it's steady-state, while other times it will have a modulation period depending on the cause.

SOME FIXES

Believe it or not, you can make a pretty educated guess as to what causing any of the above noises in your sound system by listening to the actual quality of the Hum, Buzz, Hiss or Hash. So here's what *usually* causes each type of noise in a sound system and simple steps to correct them.

Hum is most often caused by some sort of ground loop current between two different pieces of audio gear. This typically occurs when you're forced to plug stage and sound gear into multiple power outlets around the room. For instance, if the main sound system power amplifiers are plugged into outlets next to the stage, while the mixing console is plugged into a convenient outlet at the back of the room, the small difference in ground voltage between the two outlets can cause a rather large current to flow in the shield of your XLR connector. And that current can get into any amplifier or powered loudspeaker and make a lot of hum.

A quick fix is to run a long extension cord from the amplifiers' outlet to the mixing console's outlet. A longer term solution is to add an Ebtech Hum Eliminator inline between the output of the console and the input of the amplifiers, which will break the current loop and stop the ground loop hum.

This same ground loop current can flow between DI boxes on the platform/stage and the mixing console. So that's the time to use the ground-lift switch on the DI box to lift the ground loop hum. (I'll focus much more on detecting and correcting this particular type of noise in my next article.)

Buzz is most often caused by an unshielded audio cable connecting two pieces of audio gear. For instance, use of a non-shielded loudspeaker cable to connect between a bass guitar or keyboard and its stage amp will likely produce a really big buzz. Thus double-check to make sure you haven't accidentally used a loudspeaker cable instead of an instrument cable.

Note: Buzz can also be caused by a floating safety ground in your sound system that can become a shock hazard if you don't pay attention to this warning sign.

Hiss is usually caused by mismatched audio levels between interconnected gear. There's a general design rule that most sound gear's input and output level controls should be run about 75 percent of the way up. (This implies between 7 and 8 on a scale of 10.)

When you turn something on and the noise begins, that's a hint as to its origin.

So if you look at any mixing console's faders you'll probably see a highlighted area right around three-quarters of the way to the top marked as Unity Gain or 0 dB. That's the sweet spot to shoot for. If the input faders are near the top, in order to attain enough sound level, it implies that you didn't give the pre-amp enough gain. On the other hand, if the output faders are down very low to keep from overdriving the amplifiers, that's a hint that you're likely distorting the internal mixing bus within the console.

This level mismatch can occur after the signal leaves the mixer. For instance, if you use a line-level output of an aux send to feed the mic-level input of a video camera, you'll be forced to turn the aux output level control down to near zero to avoid distorting the camera input. That will usually cause all sorts of hiss to show up in the sound going to the video feed.

Is this an instrument or loudspeaker cable? Better check and find out or a "big buzz" in the system might be the result.

Hash is typically caused by switch-mode power supplies contaminating the audio ground. For instance, many laptop computers use internal switch-mode power supplies to generate a bunch of different voltages for the screen, CPU, hard drive, etc. In fact, many times you can hear this hash change frequency based on hard drive access.

The best way to correct this is to either use an external USB audio port such as a Whirlwind pcUSB, or connecting the computer's 1/8-inch headphone output to a special transformer-isolated DI designed for the output of a computer, such as a Radial Stage Bug.

The Radial Stage Bug SB-5, a transformer-isolated DI designed for interfacing computers and sound systems.

With careful listening and a little detective work, you can make any sound system noise-free. Identifying the type of noise you're hearing will put you on the path of discovery.

Of course, there are likely to be multiple types and causes of noise in larger sound systems. In these situations, turn everything off, and then turn on each component one at a time, slowly. When you turn something on and the noise begins, that's a hint as to its origin. Correct that noise, then continue turning on more things, one at a time, until you encounter the next noise. Fix that and continue until the system is quiet with all components turned on. Then it's time to make noise of a different kind!

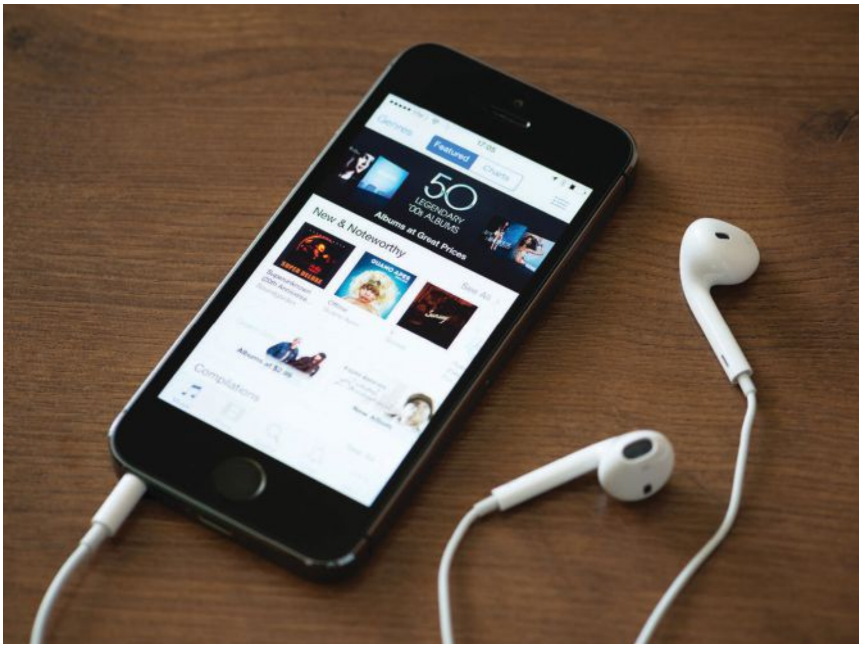
Next time I'll detail more advanced testing procedures using a clamp-amp meter to look for the causes of ground loop hum in your building's electrical system. And we'll also discuss video hum bars that show up on your monitor screens, also with much more. CS

Mike Sokol *is lead trainer for Live Sound Co. in Maryland, and lead writer of the Live Sound Advice blog. For more than a decade, he led the HOW-To workshops, teaching thousands of church sound techs, and he's also an adjunct professor at Shenandoah Conservatory in Winchester, VA.*

THE SPOKEN WORD HEARD

Producing quality voice-based podcasts of sermons and more.

by Mike Sessler



I like to go for a walk just about every day, and ever since buying an iPod years ago, I've really enjoyed listening to various sermon and tech-related podcasts while walking. It's been great – most of the time.

Recently listening to a well-known pastor's podcast (you'd recognize his name), I have to say that the message was great, but the audio was not. It was clear the original recorded audio was rather marginal and the MP3 encoding bit rate was so low that the artifacts were very distracting.

As a long-time audio geek with a great desire for quality audio, I thought it helpful to share the process we use at the church where I serve to put together a really good-sounding podcast.

SQUASH IT

I can't stand music that is overly compressed, with all of the dynamic range taken out (which is why I tend to listen to older music). However, when it comes to podcasts, I really don't want dynamic range. When I'm strolling down the street, or huffing and puffing on the elliptical, or driving down the road, it's annoying to have to keep turning the volume up when the person speaking gets quiet and having my ears blown out when he gets loud again.

Others may disagree, but I want to limit the dynamic range of sermon podcasts as much as possible. It makes the audio far easier to listen to via earphones, on the computer, and in the car. There are many ways to get to a limited dynamic range. A big concern with squashing dynamics is losing the verbal cues that come from varying levels of speech. As the pastor speaks more softly to make a poignant statement, logic dictates that it needs to be quieter. However, I've found that the tonal qualities of the voice can convey those cues regardless of the level.

And let's be honest, people are going to be turning the volume up to hear it anyway if they're in the car or working out. So my philosophy is to do the work for them, and keep the volume consistent.

MINIMIZE SIZE, MAXIMIZE QUALITY

We try to limit sermon podcast file sizes to (and even better, below) 15 MB. They download quickly, even over 3G networks, and don't take up a ton of room on an MP3 player. To get there, we use the LAME MP3 encoder, which is one of the best available. For a long time, the choice was VBR (Variable Bit Rate) encoding of podcasts, with the quality level set to 20 (which equals roughly 48 kbps, average).

However, we've switched to CBR (Constant Bit Rate) encoding. After doing some tests, I found it very difficult to distinguish between VBR at 20 percent and CBR at 48 kbps. Either method sounds more than acceptable on both my Ultimate Ears UE18 earphones and Equator Audio D5 monitors, and further, I hear no difference through the loudspeakers in my MacBook Pro. So for now, CBR it is.

CLOSE TO THE SOURCE

If you're working with a digital console and running a virtual sound check system, you're already in great shape for recording the message. That's how we've done it for years, using an RME MADIFace to run the audio directly after the A/D conversion into a MacBook Pro running Reaper. We record a 2-track board mix of all services, and a discreet track for the pastor on each Sunday morning service. The discreet track is for the podcast; the 2-track is backup only.

Since we're recording speech (for this purpose anyway), there's no concerns about 192 or 96 kHz; 48 or 44.1 kHz at 16 to 24 bits is just fine. Our system runs at 48 kHz, 24 bits, so that's what we record as a series of WAV files. WAV or AIFF files are uncompressed, which is what you want at this stage.

If it's an analog system, don't fret. Use the direct outs – or in a pinch, the output of the insert jacks – to come directly out of the pastor's microphone channel to the recorder. It's best to pick off the output before EQ, compression, or other processing. The reason is that most times, you're making EQ and compression adjustments on the console for the room, which is where most people are listening. However, those same settings may not work for the recording, and there are advantages to making some EQ and compression choices specifically for the podcast.

Material can be recorded to a CD, but it's preferable to record straight to the computer/recorder since you'll be editing and processing the files there anyway. Even an inexpensive USB interface like a Lexicon Alpha provides very good sounding direct recording. Combine that with a laptop, Mac Mini or inexpensive PC – and a copy of Reaper – and you're in business.

The best, most full vocal quality will come from a headset mic (or a handheld if the pastor prefers). Lavalier mics pick up a lot more room noise, and podium/lectern mics can be tricky to get dialed in. Don't skimp on the mic; it's on for more time during the service than anything else.

QUICK EDITING

This is pretty easy. Find the beginning of the message, back up a second or two and put a fade up on the clip. Something that drives me nuts is hearing podcasts with 5 to 8 seconds of silence at the beginning. Limit it to a second or even a bit less. The same goes with the end of the message. If the band is underscoring, I'll look for a musical place to fade out so it doesn't sound abrupt.

NORMALIZE

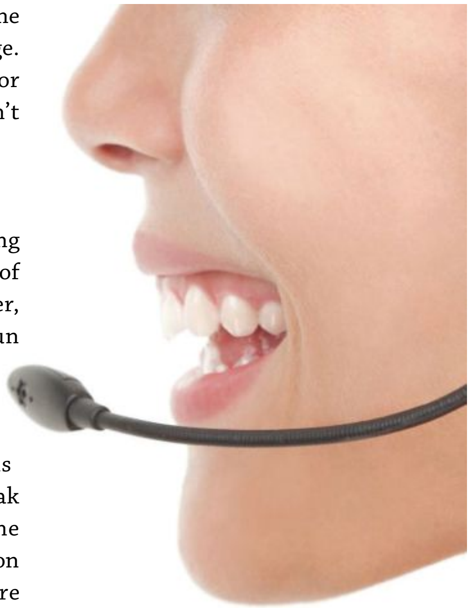
In our recording structure, normalizing is critical. We always leave 12 to 15 dB of headroom when recording. Remember, this is digital recording; once you run out of bits, you're done. Distortion is quick and nasty. So give yourself some safety margin.

Note that recording this way means that there's the need to bring the peak levels up closer to 0 so the rest of the processing goes smoothly. Normalization will bring the peak level of the entire clip up to a level you specify. It's like turning up the volume, only smarter. I normalize to -2 dB so we don't saturate in the next step.

PROCESS FOR THE WEB

What works in the room may not work online. Some experimentation is necessary here, but we tend to high-pass (filter) our pastor fairly high (up around 130 to 140 Hz), and boost the upper mids by 1 to 2 dB. The goal is adding a little bit of clarity to make it easier to listen to in loud environments. But be careful here because it can easily be made to sound annoyingly harsh.

I suggest trying some settings, encoding a section of the sermon, and listening



Headset mics help in capturing a quality vocal signature.

Remember, settings for the room may not be best for recording spoken word.

to it on several platforms to evaluate the results. EQ settings that work with nicer headphones may not work for a cheap set of computer loudspeakers, so check it out. And don't forget that many people will listen to the podcast with Apple's "inexpensive" white earbuds.

We record pre-EQ, with subtle changes to make the voice easier to listen to. Then it's hit with compression. Currently, we're using an R-Channel plugin from Waves to do both EQ and compression; it's one of the more transparent compressors I've heard. In our setup, it hits 12-plus dB of gain reduction and it's really tough to hear any artifacts.

Before acquiring the Waves plugin, we achieved results almost as good with a combination of ReaEQ and ReaComp, both plugins that come with Reaper. Most DAWs (digital audio workstations) have basic EQ and compressors built-in, so play with those first before spending additional funds. However, it's worth signing up for the Waves mailing list because the company often provides super deals on individual plugins.

LIMITING AT THE END

The final step is running the signal through a mastering limiter to really clamp down the dynamic range. We've used JS: LOSER MasterLimiter (included with Reaper) for quite a while, along with Ardaz Maximzer5.

MasterLimiter allows setting a maximum level (I go with -.01 dB); it's a "brickwall" limiter, meaning that nothing gets over that setting. It will also do some compression to keep the signal level up. Maximizer5 supplies some other magic to raise the overall level without driving it over the limit.

Screenshots of Waves L3 UltraMaximizer and R-Channel plugins.

The combination of the two works well and sounds pretty good once things are dialed in. Most recently we've been employing a Waves L3 UltraMaximizer, which is easy to use while delivering pretty stunning results.

Investing a few hours to work on a chunk of the message to get the processing settings right, then tweaking the rendering settings, will produce a solid (and likely better) result.

RENDERING TO MP3

Most DAWs can render out to an MP3 file. If you have the option to use the LAME encoder (which is available in Audacity or Reaper), use it. It's a great encoder that produces better-sounding MP3s at lower bit rates than most other encoders.

As noted, we use the 48 kbps CBR setting and render in mono. I haven't had any problems with mono files anywhere, and as spoken word, stereo unnecessarily doubles the file size. The sampling frequency is set to 22.050 Hz. There's not much useful content in the human voice above 11.025 KHz (the Nyqyist frequency of 22.050), and halving the frequency effectively doubles the bit rate on the frequency range we care about. It's both efficient and better.

EXPERIMENT

We got to these settings by doing a lot of experimenting. I'm intentionally not telling you all of our EQ, compression, and limiting settings because they don't really matter – they're all specific for our pastor. Investing a few hours to work on a chunk of the message to get the processing settings right, then tweaking the rendering settings, will produce a solid (and likely better) result. Then save those settings as presets so you can use them next week.

RECOMMENDATIONS

I've mentioned a ton of stuff in this article, but here's a recap of what I recommend for this process. If you have something else that works, by all means, keep using it. If you're looking for a place to start, consider this list:

- Audacity (free recording/editing software; good, basic and free)
- Reaper (full-featured DAW; $60 for non-profit use, incredibly powerful, still easy to use)
- Lexicon Alpha (simple 2-track USB audio interface; about $65)
- Ardaz Maximizer5 (free maximizing plugin)
- Waves L3 UltraMaximizer (amazing maximizing plugin; $350, but look for it on sale)
- Waves R-Channel (amazing channel strip; $175, but look for it on sale) CS

__Mike Sessler__ has been involved with church sound and live production for than 25 years, and is the author of the Church Tech Arts (churchtecharts.org) blog. Based in Nashville, he serves as project lead for CCI Solutions, which provides design-build production solutions for churches and other facilities.

DIALING IT IN

The beginning of the path to becoming a master of EQ.

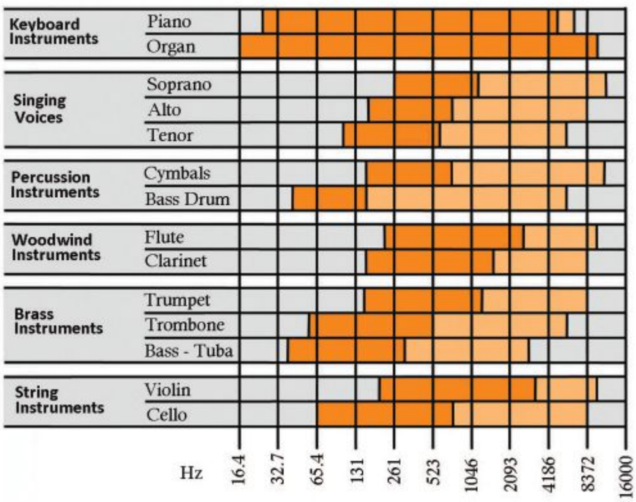
by John Mills

When I first began doing sound, I bought a great set of headphones, thinking that if I was going to be expected to make something sound good, I should probably know what I was shooting for. Then I started listening (like crazy) to CDs. Not just bands or styles I liked, but anything and everything I could get my hands on.

I listened to the lyrics, chords, melodies and harmonies, and also to how it all fit together. I concentrated on the space that each instrument was taking up. I noticed that certain instruments seemed always to be sitting in a certain spot – not to where they were panned, but to the frequencies they occupied.

HOW TO GET THERE

When building a mix, we need to think of the song as a line. Each instrument makes up part of that line. If we have too many instruments or frequencies trying to take up the same space our line gets bumpy and the mix gets muddy.



Where a variety of elements exist in the frequency spectrum.

Listen to each instrument and think of a space for it on the line. Keep other instruments away from it (EQ wise) and you will have an easier time hearing that instrument. You wouldn't want to have a really bassy, heavy electric guitar because it would be taking up a lot of the space the bass guitar really needs. Try to keep each instrument in its place.

Think of each instrument as to what the fundamental piece of it is. For instance, the fundamental of a kick drum will be low frequencies. That's not to say you don't need highs to make it cut, but there really isn't much midrange going on with it. Try to carve out some of the midrange of the kick to make room for the low midrange of the bass guitar.

I always tell new engineers never to be "done" with the mix. Listen for changes, and more importantly, listen to make sure that everything is in the mix and working together. Focus less on the actual sound of the individual instrument and more on how it interacts with other instruments in that same range.

There are no "magic" numbers that work every time because all instruments are a little different. The equation gets more complicated when we use different microphones or the instrumentalist changes patches on their keyboard, but trust me, none of that is really important. What is important is focusing on getting a natural sound that blends nicely with the competitors for the same space. Following are some general guidelines to consider when you're trying to find your space.

20 Hz to 80 Hz: This is the sense of power in an instrument or mix. It's the stuff you feel more then hear. The kick drum and bass guitar are in this range.

80 Hz to 250 Hz: The area where everything comes together. This is where a lot of things can go wrong, and too much in here will make a mix sound sloppy.

250 Hz to 2 kHz: Most of the fundamental harmonics are in this range. These are some of the most critical frequencies to building a solid mix. Learn what instruments are most dominant in these frequencies and clean up around them.

2 kHz to 5 kHz: Here we find the clarity to almost everything. But be careful, too much of a good thing can start to sound harsh. This is an area where subtly is the key.

5 kHz to 8 kHz: Mostly sibilance and "s" sounds. Most of the vocal consonants are defined in this range.

8 kHz to 20 kHz: Brilliance is the word here, the top end of cymbals.

Now let's move along to frequencies with respect to instruments.

Kick Drum and Toms: Cut 500 Hz to get rid of the cardboard box sound. Add 5 kHz to make them cut through the mix. Add a little 60 Hz to 80 Hz to make them really thump.

Snare: Usually I take out a little around 600 Hz and add a little around 4 kHz, and maybe even boost some 200 Hz to make it move a little air, but that really depends on the drum and how it's tuned.

Hi-Hat: I generally cut all the lows and a good chunk of low mids. There isn't anything down there anyway.

Bass Guitar: Players and basses are so very different. Usually if it's muddy I cut 160 Hz to 200 Hz, and possibly add a little 700 Hz to 1 kHz if I can't really hear their notes. But be careful because there are a lot of other instruments fighting for that space.

Piano: This is a beast that can take an entire article to discuss. It depends mostly on how the piano is miked. However, in general, if it's boomy then cut 200 Hz to 315 Hz. If it's kind of barking, then cut more up near 400 Hz to 500 Hz. Judiciously add a little 2 kHz to 4 kHz to make it cut a little more.

Voice: Boomy? High pass at 150 Hz. Too thick? Try cutting at 240 Hz. Need it/them to poke out a little more? Cut other instruments around 2.5 kHz. In my opinion, never add 2.5 kHz to a voice, it sounds harsh and unnatural. Having trouble hearing the syllables? Try adding a little between 4 kHz and 8 kHz.

Background Vocals: Sometimes I like to let the lead vocal shine by dulling the top end of the BGVs. Try a shelf around 4 kHz and cut by 3 to 5 dB. This will let you turn up the BGVs and surround the lead, while the lead still sounds more forward.

TRUST YOUR EARS

The most important question is, "Does it sound natural?" Does it sound like the CDs you've been listening to? More specifically, does it sound like you're sitting in front of the real instrument? Keep this in mind throughout the performance.

I constantly glance down all the channels and think about each input. Kick – does the kick sound right? Bass – does the bass sound right? Guitar – does the guitar sound right? Piano – does the piano sound right? Vocals – do the vocals sound right?

Then I think about it all again and ask if the guitar and vocal are walking over each other. Can I hear the piano? Is it because the guitar has too much midrange near the piano part's midrange? Then try taking a little low-mid out of the guitar instead of turning up the piano. I think you get the idea...

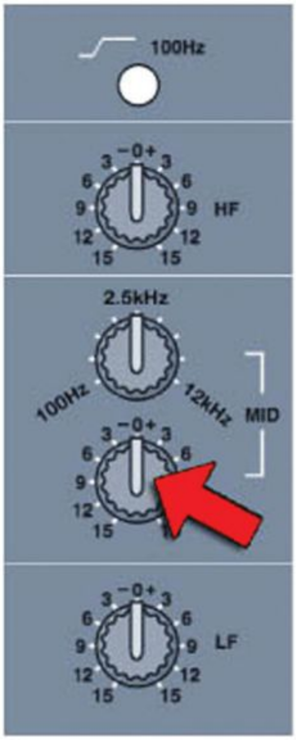


Figure 1

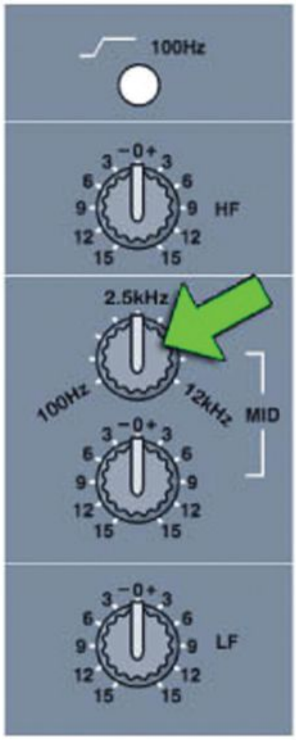


Figure 2

It's almost impossible to make the initial adjustments to instruments or vocals in the mix with the whole band playing. Instead I try to have a mental snapshot, acoustically, of what I think the instrument should sound like, kind of like a target for my EQ choices.

Learning to EQ confidently means you know where you're heading. That's why I recommend listening to CDs with a good set of full-range headphones. And please, no cheap earbuds.

TURN, TURN, TURN

Want to know a bonafide trick of the trade? Turn some knobs! I mean actually get in there and turn the heck out of the EQ knobs and listen to what they do.

Here's simple technique to use in sound check. Grab the gain (**Figure 1**) on the mid EQ of an instrument and crank it up a bunch, then grab the frequency (**Figure 2**) of the mid and sweep it up and down. You'll hear a spot where it makes that instrument or voice sound horrible. Once you find it, take the gain back to zero, listen for a second again, and then cut out about 6 dB of it.

It's amazing how much better that instrument sounds when you "get the junk out" as I call it. This is a great way to learn what frequencies sound like and the technique will eventually train your ear to hear the junk without boosting it first. CS

A 20-year veteran of working live sound everywhere from churches to top touring artists such as Kenny Chesney, ***John Mills*** *is now the education & development manager for Morris Integration in Nashville.*

NOT SO RANDOM MUSINGS...

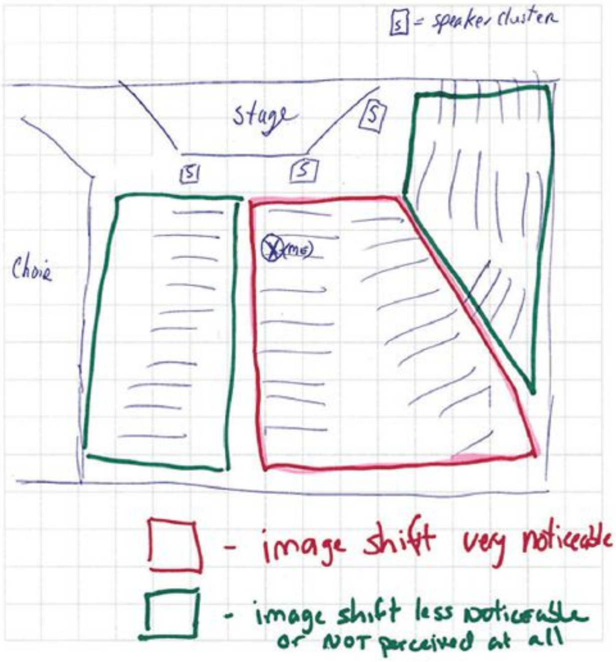
Imaging is everything, IEMs for smaller churches.

by Gary Zandstra

The public image of a church is important, but there's another kind of image – of the sonic variety – that's also significant. During a recent worship service, I got to thinking about image in terms of where sounds are coming from on a stage, and where I actually perceive them as coming.

Specifically in this case, there was a choir singing, and it was located far left. In fact, I had to turn my head almost 90 degrees from the main stage to see the choir. The reinforcement for the choir was coming from loudspeakers flown centrally above the stage, and from where I was sitting, the reinforced sound was louder than the acoustical sound of the choir. Thus my attention was drawn to the stage, which was void of any performing musicians, rather than the choir.

My "sight brain" was telling me the choir was far left, but my "hearing brain" was forcing my attention toward the stage. This is an imaging issue, and it was driving me nuts. Now, if I'd been located closer to the choir, where the acoustic energy of the choir would likely be louder than the reinforced sound, I would have "localized" on the choir and likely would have interpreted the reinforced sound as an "effect" – a quasi-stereo image.



A hand-drawn diagram by the author showing the layout of the church where he found the imaging to be problematic.

Or if I had been located more toward the right side of the sanctuary, the reinforced sound would have seemed more "in line" with the location of the choir. Even though the reinforced sound would still be louder, it would make sense in the sight-sound-brain equation, because the acoustical image would be more in direct line with the choir.

Side note: the room was a small enough that the distance from the choir to the loudspeaker covering the right side of the room was not great enough to cause the listener to perceive much, if any, delay between the acoustical and reinforced sound.

The next time I visit this church, I'm either going to move to my left (closer to the choir) or to my right, far enough over to align the reinforced sound with the acoustical image. And, to improve this situation, my suggestion to the church is to add a secondary loudspeaker(s) that hangs above the choir that will help clear up the imaging problems. The acoustic and amplified sound would be coming from the same direction.

ON THE OTHER HAND...

Fast forward a couple of weeks. I attended a hymn sing/brass concert in a different worship center. During one song, five of the brass players went up to the balcony and played from there, echoing the players on the stage.

It sounded incredible! At times it had the feel of a question and answer session, where the stage musicians would play, answered by the musicians in the balcony. Other times they both played together, and if felt as if I was in the middle of an entire brass section. This was excellent imaging.

The reason the imaging worked is because it was set up as an effect. The antiphonal response of the brass in the balcony was a purposeful image shift that enhanced the musical piece. One on my pet peeves is listening to a band playing over a left-right stereo system, and the toms on the drums are panned to give separation. But when I look at the drums, rather than the sound going from right to left – following the drummer as he runs the toms – it does the opposite. Again, imaging makes a difference.

Another thing that I like to do when I'm mixing in a stereo situation is to try to move the image of the instrument relative to where the musician is standing on the stage. If the guitar player is on the right side of the stage, I will pan the feed of the guitar slightly to the left, and so on.

SHIFTING GEARS...

While recently discussing in-ear monitoring (IEM) with a colleague, he made a statement that struck me: "In reality, it's the smaller church that needs in-ears much more than the larger ones."

A couple of things came to mind. Larger churches/ministries have the funding to get IEM, and they (often, at least) have paid technical staff that can properly set it up. And larger churches also have large stages in large rooms, and stage volume is frequently not as much an issue as it is in smaller churches.

However, here are some (now rather obvious to me) reasons why a small church might invest in IEM:

- Stage volume is a huge issue. In some cases, 70-plus percent of the congregation probably hears more stage volume than sound coming out of the main loudspeakers.
- Because of the stage volume, there are continual complaints about loudness and that the vocals cannot be heard over the instruments.
- Many small churches have singers and instrumentalists that have never played on a stage and are perhaps self-conscious about their abilities. IEM allows them to better hear themselves and other musicians.
- Feedback can be a constant issue because the vocal monitors always need to be turned up too hot so the singers can hear themselves.

So how can smaller churches move into IEM? Fortunately, costs have come down in recent years. Further, an investment can be made in just a few systems, with more receivers added over time as funds become available.


COURTESY OF SENSAPHONICS

Not just for the "big guys"?

I suggest starting with singers; although they're not the loudest thing on stage, in general, their monitors tend to be loudest. So a good starting point might be purchasing one transmitter and enough receivers for all of the vocalists. Then later, purchase a second transmitter and put the band on that mix.

Issues to keep in mind with IEM:

- They do take a bit to get used to, so use them first in multiple rehearsals.
- Mixing for "ears" is different than what is needed with standard floor monitors.
- Without ambient/audience mics feeding IEM, musicians will initially feel isolated and perhaps frustrated.
- Just like conventional monitors, when multiple musicians share a mix, there will be issues!

To help make the use of IEM successful, the person providing the mix (usually the house sound operator) should use a set of headphones (or ear pieces) that match the ones the musicians have when setting up the monitor mix on an aux send of the console.

Also, the operator should try to make sure there is an ambient microphone (or two) to feed that aux channel (and *not* the main mix).

I've found that placing a mic to capture some onstage sound as well as a mic to capture the audience/house sound usually works the best. Dialing the amount of each mic into the mix is a matter of personal taste – work with the musicians on this one.

In-ear monitoring is not a total solution to all stage monitor problems, but it's a valuable tool that can help when deployed carefully and correctly. CS

Gary Zandstra *has worked in church production and as an AV systems integrator for more than 35 years. He's also contributed numerous articles to ProSoundWeb over the past decade.*

CEPAD

Salinas, Ecuador, South America

THE CENTRO EVANGELIST Peninsular Assembly of God, an outdoor church with a seating capacity of 5,000, offers two services every Sunday. Both services blend traditional and contemporary formats, the latter including a band consisting of two keyboardists, electric and acoustic guitarists, drums, percussion, three main vocalists and 12 choral singers.

A recently implemented new sound reinforcement system for the church, designed and installed by Muzeek World (Orange County, CA), is fronted by a Yamaha CL digital console working with companion Rio3224-D and Rio1608-D input-output boxes. “The Yamaha CL console was chosen because of its scalability and network capabilities,” states John Sardari, owner of Muzeek World. “Dante (networking) played a big role in the decision as well, and the fact that Yamaha is the leader in the digital console world.”

The new system also incorporates 14 NEXO GEOS 1210 line array loudspeakers, two GEOS1230s and three NEXO RS18 Ray Subs all driven by NEXO 4X4 amplification. The entire system is covered each week, and in the event of rain, it is taken down and moved to an interior worship center. “The church staff is super happy, and the quality and coverage of the NEXO GEO S12 system is just amazing,” Sardari concludes. CS

Flatirons Community Church

Lafayette, CO

THE CHURCH ATTRACTS an average weekly attendance of about 20,000 people to multiple weekend services at its three Denver-area locations, with the 4,000-seat auditorium on the main campus in Lafayette recently expanding its Lectrosonics Digital Hybrid Wireless system stable.

“We have two new 6-channel Venue 2 frames fitted with 12 channels of VRT2 IQ dynamic tracking filter modules," states Bryce Boynton, audio director for the church. "We also have an original Venue, because we bought our first system over a year ago, so we now have three Venue receivers. We also now have six handhelds – one HHa, the new model, and five HHs – and we have six LT belt packs.

One feature of the LT belt packs Boynton emphasizes is the ability to switch between instrument or microphone mode. In instrument mode, the transmitter presents a 1 megaohm impedance to an instrument input with a piezo type pickup. This improves the sound quality and gives the same effect as plugging into a premium DI.

He adds that he didn’t choose the Venue 2 receivers specifically for the wide tuning range, but the capability has certainly turned out to be useful. “Lafayette, which is east of Boulder, isn’t a major metro area where I’m trying to run 40 channels – although I just did at Easter,” he notes. “The point is that the wide tuning bandwidth of the newer system, bands A, B and C, is really helpful. It was nice at Easter, when we did have 40 channels, that I was able to coordinate the Lectro channels last, because they have the widest tuning range and most flexibility. I don’t need for them to change frequencies in real-time, I just need them to be flexible.” CS

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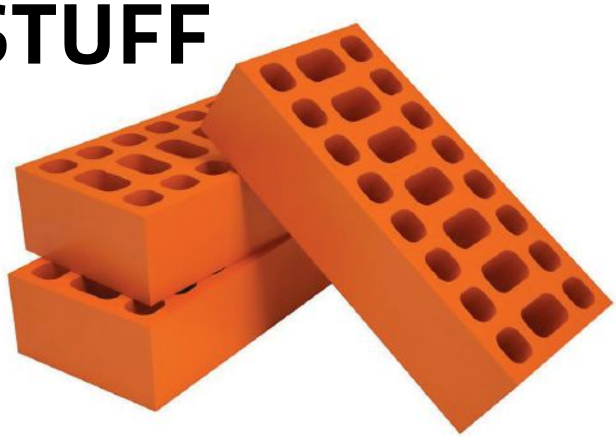
FOUNDATIONAL STUFF

The point of dedication to working in worship tech.

by M. Erik Matlock



Do not read this article. We have many fine authors in this publication. They're the church sound gurus. Not me. I'm just the guy who writes the foundational stuff about survival on ProSoundWeb. Dealing with pastors and leaders and the congregation and your own families. All the stuff we already know but sometimes forget.

But since you've already read this far, perhaps you want that. The first statement told you not to read this, yet here you are.

FOLLOWING DIRECTIONS

It's an area that obviously needs work, since I just told you not to read the article. (That's meant to be humorous.) But the fact is, church sound is a support ministry. Nobody comes to church to watch us work.

I was taught that the only reason anyone knows we're there is because we goofed something up. Forget to turn the pastor's mic on, allow feedback to roll around, leave a bad cable plugged in to make the same annoying noise for two weeks in a row… That's when everyone wants to know who we are.

We exist to support the ministry and help translate the vision of the pastor and worship leader to the congregation. We make the magic happen by doing this professionally and with a good attitude. We do it by learning and growing, by asking questions, and by listening.

TREAT IT LIKE A MINISTRY

To me, at least, ministry really means service. It doesn't mean a spotlight on us, nor does it mean making all of the decisions. It doesn't mean we fail if nobody sings our praises at the end of service.

It's not a foot in the door to "real ministry" or a stepping stone to something better. If that's how you see it, don't get comfortable. You're pretty much guaranteed to move on. One way or another.

LEARNING TO MIX

To new tech people, this seems like the entire reason we're here. Like the person who joins the Air Force to fly planes. There's a whole lot of training that has to happen before just anyone is allowed to take a gazillion dollars worth of jet fighter out for a joy ride.

So unless you really enjoy the thought of crashing and burning, slow down and go through the process. If you aren't willing to roll cables, clean up the stage and sound booth, and show up on time and do the small stuff, then you shouldn't be trained to mix. Show yourself faithful in the details and then, eventually, you'll be shown how to work with the big toys.

Mixing is an art form. It's a balance between multiple skills. Technical skills to understand what's being controlled and how to do it. Musical skills to feel and hear the balance, as well as the blending of tones that's faithful to the musicians. Diplomatic skills to understand the people we're serving, as well as becoming a part of the team. All of that comes together when we mix.

Possessing just one of these skills is not enough, and none of them are learned overnight. And don't just jump into mixing with a full congregation or audience. Assist the front of house tech, watch him or her work. Learn the monitor system and how the sound check works. Mix rehearsals. Work the conferences and classrooms. Earn your way up through the ranks so you're actually prepared when it matters.

TROUBLESHOOTING

Ah, yes. That awe-inspiring ability to find and solve any problem that arises. The spectacular skill-set that allows someone to "just know" what's wrong and spring into corrective action. Reaction time and deep knowledge of a sound system do not happen spontaneously. The ones who solve problems faster than most even acknowledge them have been doing this for a long time.

Troubleshooting comes down to true understanding. Simple. If you've set that system up hundreds of times, you know how it should work and why it doesn't.

THE BOTTOM LINE

No matter how long we've been doing this, we need our foundation to be solid. It's built on everything I've covered here, and doing it right will help in maintaining a position for as long as you desire while being of genuine service. CS

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*Senior editor **M. Erik Matlock** has worked in professional audio for more than 20 years in live, install, and recording, including time as a church tech and media director.*


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