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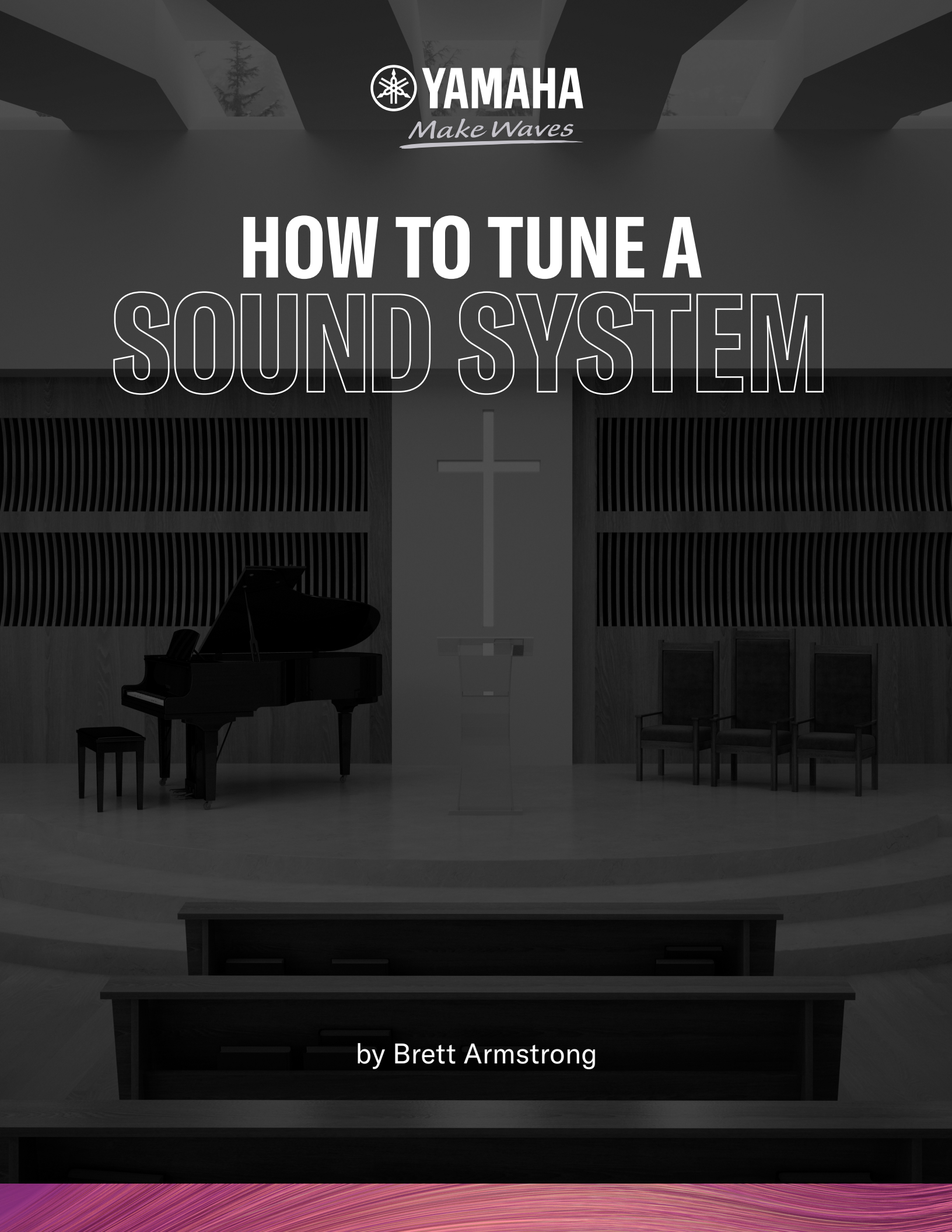
HOW TO TUNE A SOUND SYSTEM



by Brett Armstrong

EVERY CHURCH IS DIFFERENT. Mega-churches have sanctuaries that look and sound like concert halls — complete with ornate architecture and elaborate acoustical considerations. Others are small rooms with a barely-working PA system and no acoustic treatments. While most churches fall between these two extremes, they all have in common the need to make their space sound as good as possible.

If you're mixing in a 1,000+ capacity church, you're most likely using a state-of-the-art digital console with full parametric EQs to fine-tune your mix. The room itself has probably already been tuned by a trained acoustician as well. We should all be so lucky. This guide is primarily for those who are not.

Most of us work in small spaces with minimal to no acoustic treatment, a bargain basement analog console and a rack of graphic EQs. The key to a good-sounding mix is a good-sounding room, but if you don't have one, you can fix it by using a graphic EQ to "tune" the PA system.

REFERENCE MATERIAL

Start by testing the system with a few high-quality tracks you know well. It's beneficial if the tracks are similar to the type of music you'll be mixing for the service — if you'll be working with the latest CCM songs, it doesn't help much to check the system with an AC/DC tune.

Get a feel for what your reference tracks sound like in the room compared to your studio, home stereo, car, etc. Walk around and see how it sounds up front, way in the back and on the sides. Is the low-end overpowering? Do any particular frequency ranges stick out? Is anything missing from the mix?

If you're lucky, the person who tuned the PA last did a good job and you won't need to make any adjustments. However, the person might have had no idea what they were doing and just made a smiley face on the EQ. Either way, it's best to listen first and spin knobs second. As someone else once said, "Never judge an engineer by the curve of their graphic equalizer."

GRAPHIC EQUALIZERS (GEQ)

GEQs are most commonly used to remove problem frequencies caused by the room. Most GEQs have 31 bands. That might be shocking if you're used to the typical four-band semi-parametric EQs on most live sound consoles, but GEQs have a very different purpose. Each of the bands has only one control — gain. Instead of knobs, there are 31 tiny faders that you can use to add or subtract gain for each band.

The bands themselves have fixed frequencies. They're separated into 1/3 octave increments, starting at 20 Hz and going up to 20 kHz. When using an analog GEQ, each band has a Q-value (quality factor, aka resonance) of about four relatively narrow notches.

When cutting frequencies, start small. You don't need to kill a full band over a small bit of feedback. Most GEQs have a dynamic range of ±15 dB, which is a pretty drastic cut. Start with something small, like a 2-3 dB cut, and slowly increase if needed.

It's unusual to add frequencies via GEQ, but if you need to boost a specific range, it's better to do it with small boosts across multiple bands. Don't push 80 Hz up to 12 dB — that's just going to create a new feedback problem. You're better off making small boosts at 63 Hz, 80 Hz and 100 Hz.

It's important to periodically bypass the GEQ and listen to the sound without changes. It's easy to make your mix sound worse by over-correcting, and you'll never know if you don't do A/B testing and compare your new settings against the original.

After making adjustments and confirming that you've actually improved the mix, give your reference tracks another listen. If they sound better, you did things correctly. If they sound worse, it's probably from over-correcting. Try reducing the cuts you made and see if your mix improves.

PLAYING IT BY EAR

The most challenging part of tuning a sound system is identifying frequencies that are feeding back. It's not hard to hear the feedback — it's tricky to find which of the 31 bands on your GEQ will fix the problem. At the end of the day, the only thing that will make the process easier is good old-fashioned ear training. Luckily, several downloadable apps can help. For instance:

- Tenuto
- Teoria
- Good-Ear.com
- EarMaster
- Quiztones

As you work to improve your hearing, you can use a Real-Time Analyzer (RTA) to help identify problem-causing frequencies. RTAs are spectrum analyzers that graph the frequency response of your space. Louder frequencies display more prominent peaks on the graph, giving you a visual cue for problem areas. Many digital consoles include RTAs, but you can also find apps for most smartphones, tablets and laptops. Some good examples:

- Audio Analyzer
- Ultrasonic Analyzer
- Audio RTA
- Audio Spectrum
- Audio Tools

To properly use an RTA to tune the room, you need to generate a "pink noise" signal through the sound system. Most RTA apps have a pink noise generator that you can connect to a channel on the console.

Start by setting your master fader to unity and the channel fader to unity. Set the four-channel semi-parametric EQ on the console to 0 dB. Then, using a dB level meter, bring the channel gain up until you're at the overall dB level you'll be running at during services. It's important to take readings at several room positions: front, back, center, back-left, back-right, center-left and center-right.

The RTA will show you the frequency level everywhere you take a sample. The purpose is to see where each frequency is in relation to the "flat" line on the GEQ (0 dB). You want to create as flat an EQ response as possible, which will go a long way towards preventing feedback issues and give you an overall "tuned" sound system.

Some frequencies will need to be boosted and some will need to be cut. After you make your adjustments, go back and repeat the procedure. (You may have to do this several times to get the EQ settings at the correct levels to achieve a flat response.) Then play your tracks and listen to see if the adjustments have improved the overall sound. You can now use the four-band semi-parametric EQ on the console to add more depth to the track if needed.

FEEDBACK TEST

Now you can move on to your live mics. You need to test them to reduce the possibility of feedback as much as possible:

1. Start by setting up your vocal mics downstage.
2. Bring the master fader on your console up to unity.
3. Bring the channel faders for the vocal mics up to unity.
4. Slowly increase the gain of each mic until it starts feeding back, then turn it down just enough to make it stop.
5. With all of the mics "live," slowly turn up the gain on each vocal mic and identify the frequencies that cause feedback.
6. Find the band for that frequency (or the closest one to it) on the GEQ, and slowly lower the fader until the feedback stops.

Remember, this is a feedback test — you're not "gain staging" the console here. You're identifying frequencies that can cause possible feedback issues. If a mic causes feedback, one of the frequencies is likely causing problems in your room. Make a note of it and cut that frequency on the GEQ.

One of the most common mistakes front-of-house (FOH) techs make when tuning a sound system is over-EQing. Don't go crazy! Cutting a frequency more than you need to isn't going to make it any less likely to cause feedback. It's just going to make your mix sound worse.

RINGING OUT THE STAGE MONITORS

The mains aren't the only speakers in the room. Stage monitors also have the potential to cause feedback. Thankfully, the process for eliminating offending frequencies, or "ringing out," is very similar.

One key difference is the positioning of stage monitors. You have control over the relationship between the mic and the monitor, so make sure they're working together to prevent feedback by pointing the rejection area of the vocal mic toward the center of the stage monitor driver. If possible, try to ensure the monitors aren't so far back that the audio signal bounces off the stage's back wall.

Start by setting up at least one vocal mic in front of each stage monitor. Turn the AUX Masters up to unity. Turn the AUX sent to each monitor up to unity for each vocal mic. Ensure the console's semi-parametric EQ channels are set at 0 dB. Slowly increase the gain of each mic until it starts feeding back, then turn it down just enough to make it stop.

With all of the mics "live," slowly turn up the gain on each vocal mic and identify any frequencies causing the feedback.

Find the band for that frequency (or the closest one to it) on the GEQ and slowly lower the fader until the feedback stops.

WRAPPING IT UP

Hopefully, these techniques will help you get the best sound possible for your church. Consider bringing in an acoustician if you have many hard surfaces, wood or tile floors, and/or windows. Acoustical treatments can sometimes make all the difference in the world.

Set aside a whole day so you'll have the time you need to get the best sound response possible. Remember: once you have the global settings locked in, write them down, save them somewhere prominent and try to lock that part of your system down to prevent anyone from coming along and changing the settings on you.

Good luck and happy mixing!

GLOSSARY OF TERMS

Active Speakers: Powered speakers, also known as self-powered speakers, are loudspeakers with built-in amplifiers. They can be connected directly to a mixing console or other low-level audio signal sources without an external amplifier.

AUX: An AUX send is an output used on most live sound and recording mixers. It allows you to create an "auxiliary" mix in which you have individual-level control over each input channel on your mixer to your AUX send output. This enables you to add effects to an output or channel on your mixer.

DAW: A digital audio workstation is an electronic device or application software for recording, editing and producing audio files.

DI: Direct boxes are often referred to as DI — or "direct injection" — boxes. Their primary purpose is to convert unbalanced and/or high-impedance instrument signals into a format suitable for direct connection to a mixing console's mic input — without using a microphone.

Gain Stage: Gain staging is the process of managing the relative levels in each step of an audio signal flow to prevent the introduction of noise and distortion.

Graphic Equalizer (GEQ): This device is used to alter the frequency response of an audio system using linear filters. Since equalizers adjust the amplitude of audio signals at particular frequencies, they are, in other words, frequency-specific volume knobs.

IEM: An in-ear monitor is used in place of monitor speakers, which are placed on stage in front of band members.

Insert: In audio processing and sound reinforcement, an insert is an access point built into the mixing console. It allows the audio engineer to add external line-level devices — such as compressors or FX processors — into the signal flow between the microphone preamplifier and the mix bus.

Passive Speakers: These loudspeakers do not have a built-in amplifier; they must be connected to an amplifier through an ordinary speaker wire.

Pink Noise: A mixture of sound waves with intensity that diminishes proportionally with frequency, yielding approximately equal energy per octave.

Pre/Post: Pre and post sends are AUX sends; they control the sound sent to objects like nursery room speakers, stage monitors, or anything other than the main house speakers. A pre-AUX send delivers the signal out of the mixer BEFORE it passes through the channel fader (also known as pre-fader).

Preamp: A preamp is a "preamplifier." As the name suggests, it prepares the signal from a pickup or microphone for further amplification.

Q-Value: The "quality factor" defines the bandwidth of frequencies that will be affected by an equalizer.

Semi-Parametric EQ: Sometimes called pseudo- or quasi-parametric EQ, this is a parametric equalizer with one or more missing features. This term is sometimes used to describe a single band of equalization, which generally means a parametric EQ that does not have a Q control — the Q is fixed.

Signal-to-Noise Ratio (SNR): A measure that compares the level of the desired signal to the level of background noise. SNR is defined as the ratio of signal power to noise power, often expressed in decibels.

Speakon: This is a trademarked name for an electrical cable/connector. It is mainly used in professional audio systems for connecting loudspeakers to amplifiers. Other manufacturers make compatible products, often under the name "speaker twist connector."

Subgroups: Subgroups are groups of channels that you can "pre-mix" together before sending them to the master output mix. For instance, if you use multiple mics for a drum set, you can then combine the channels into a drum subgroup so you can then control it as if it were one input.

Tip Ring Sleeve (TRS): The parts of the jack plug that the different conductors connect to. A TRS cable has three conductors vs. the two on a standard guitar cable. (A guitar cable is a TS or Tip Sleeve cable.)

Unity Gain: This term is used when establishing the balance between pieces of audio equipment. The idea is that input should equal output, level-wise. Audio that goes into a device at one level and comes out of that device at the same level is said to be at unity gain.

XLR: XLR refers to a three-pin locking connector that is used in audio applications.

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