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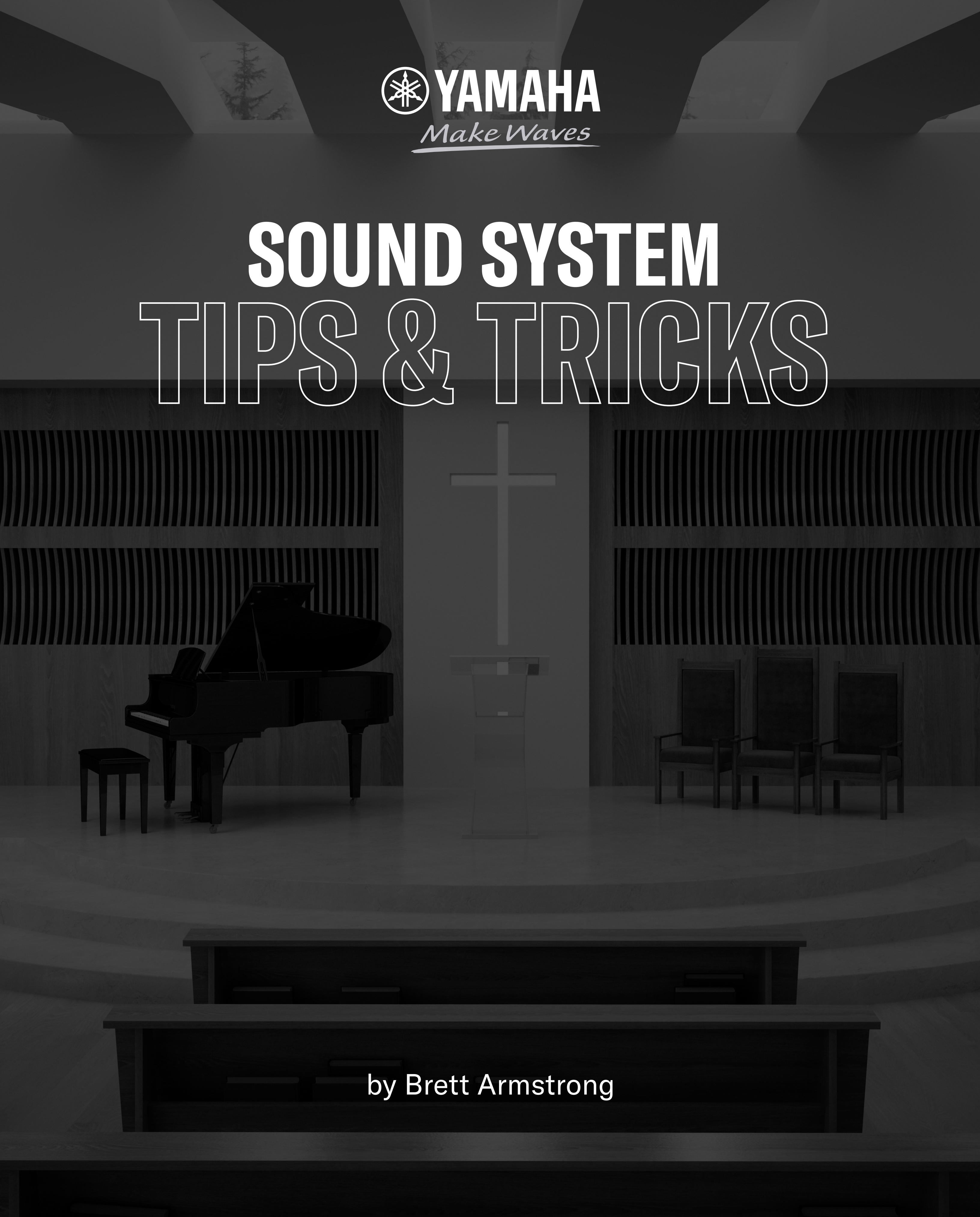
SOUND SYSTEM TIPS & TRICKS



by Brett Armstrong

MIXING LIVE AUDIO CAN BE STRESSFUL. There are times when things will be out of your control. Circumstances may require you to make the best of a lousy situation. However, if you know your stuff and plan ahead, you can "turn that frown upside down" and significantly reduce the risk of audio disasters. Yamaha compiled this collection of tips and tricks to help give you the confidence and knowledge you need to produce a live show that everyone will love.

CONSIDER BUILDING YOUR MIX OFF OF A TEMPLATE

Take a look at all of the instruments and singers in your band, then consider a template of presets with the following in mind:

- Engage the high-pass filter (HPF) for channels that will be helped from HPF — primarily vocals, this may also be referred to as Low Cut.
- The drums channel will likely be helped with adding compression. Set the threshold but don't engage it yet.
- Start all faders at unity. You will move them as you start building the mix live.
- Consider where the vocalists sit in the mix — are they more high, mid- or low-range singers? Now you know where to carve out space for them in the other channels.

This template concept is a great way to build a mix from scratch. Consider your unique situation and add to the above points to create your own template.

USE COMPRESSION FOR PRODUCING A WELL-ROUNDED SOUND

It’s challenging to produce a balanced sound when you have multiple channels that include a wide range of volume dynamics. Use compression to even out volume spikes. Remember, compression is like a third hand — it makes the louder parts softer and the softer parts louder, evening out the overall mix.

HEAR WHAT YOUR LIVE MICROPHONES HEAR

Put headphones on and listen via PFL/SOLO/Cue to a vocal mic. Pay attention to all the sounds it picks up — are you hearing anything other than vocals? This tip shows the importance of proper mic proximity to a sound source.

KNOW WHAT YOU MAY BE BOOSTING

As we mentioned directly above, mics can pick up a variety of background sounds. Therefore, when you boost the high-end frequencies of a vocal mic you might pick up drum cymbals, for instance, and unintentionally accentuate them.

PULL YOUR MALE SINGERS OUT OF THE MUD

Cut your male vocals in the 325-350 Hz range, which is the low-mid, to clear them. Often, this range is where the muddiness occurs.

USE REVERB FOR VOCAL SEPARATION

You can push a singer into the background using too much reverb. On the other hand, a little reverb can make a singer stand out in the mix. Use your ears to figure out what’s best for your situation.

BASS AND KICK DRUM CAN WORK TOGETHER ON THE LOW-END

The bass can add low-end tone while the kick drum wins on the attack. Give it a try with your music.

KEEP YOUR HEADPHONE VOLUME DOWN BY USING DELAY ON YOUR SOLO BUSES

Slow down the solo bus with delay to sync your headphones with the sound system. That way, you don’t have to run headphones overly loud. If you’re 75 feet from a speaker cluster, try a 75-millisecond delay. Run the headphones out to an external delay unit and then into a headphone amp and then back to the headphones.

DON’T LET A POWER OUTAGE TAKE OUT YOUR SYSTEM

Power can go out for any reason. Plug your vital audio equipment into an APC/UPS unit to keep the system powered and your service running.

PLAN ON EQUIPMENT FAILURE

Equipment will fail — it's only a matter of time — so be prepared. Ask yourself questions like, "What might fail?" "How could we get around it?" and "What's the least amount of equipment we need to keep the system going?" Wherever possible, have replacement parts and gear readily available. Sometimes backup gear is not an option, however, so develop a plan, write it down, keep it handy during events, and practice it with your team.

USE A "911" MICROPHONE

Wireless systems go out. Wireless mic batteries die. Even DI boxes can go bad at the worst time. Set up a wired vocal microphone on a stand with a long cable. Place it just off-stage. Grab an extra DI box and 1/4-inch cable and place them at the base of the microphone stand.

The next time a mic or DI goes out during a service, you (or a musician) can use the "911" mic — or emergency setup, whatever you want to call it. If it seems all the equipment is crashing, a church audio system only needs one channel and a mic.

DON'T FORGET ABOUT HPF AND LPF

The high-pass filter (HPF) allows high frequencies to pass through, while the low-pass filter (LPF) allows low frequencies to pass through. If you don't need low frequencies from a channel, engage the HPF. If you don't need highs from a channel, use the LPF. In the case of HPFs and LPFs with controllable frequency points, sweep the point until it's noticeable in the mix, then back off a little.

TAKE CONTROL OF YOUR HOUSE EQ BY CONTROLLING THE Q-VALUE OF YOUR CUTS AND BOOSTS

The Q-value is the same on all standard 32-channel rack EQs, except that it might automatically tighten up if a cut is below 3 db. Therefore, if you run a digital mixer, use the on-board master EQ to alter the house EQ. This also allows you to control the Q-value of your cuts and boosts.

USE A DUCKER FOR BACKGROUND MUSIC AND ANNOUNCEMENTS

A ducker on a digital console will automatically cut the volume of a channel when it detects sound on another channel. This comes in handy if you're a one-person operation. Does this scenario sound familiar? — You're running all over the place and have background music set to play, but the pastor starts talking when you aren't in the sound booth.

You can set the delay for when the music channel comes back up after they stop talking. This way, the background music could stay low in volume if they take a breath before speaking again.

DON'T DISCOUNT FREQUENCY BANDS OF AN INSTRUMENT

When you look at a 32 channel EQ, each fader is a frequency band that you can increase or decrease in volume (up increases, down decreases). When trying to EQ certain instruments in the mix, there will be frequencies that will have higher frequency bands, and others will have lower. Drums are a particularly hard instrument for sound techs to balance and get a good sound from. By boosting high frequencies on the toms, you can often get a cleaner more distinct sound. The same concept applies to bass drum and bass guitar; by differentiating mid and high frequencies between the two instruments, you get a cleaner, less muddy sound.

USE MEANINGFUL DISTORTION

Distortion can work on more than a bass or a guitar — it can even work on a snare drum. Distortion sounds different depending on how you use it and what parameters you set. Only use distortion when it helps the overall sound of the song.

DON'T FORGET ABOUT GATING

Focus your gating around a frequency range, if possible. By doing this, the sound is only broadcast when the input reaches a certain volume level. You'll know you're only getting the frequencies you want. Imagine what you could do with a kick drum or a tom.

THE "SMAART" APP

Tuning a PA system by ear isn't an exact science but tuning it with specialized software is. Programs like System Measurement Acoustic Analysis Realtime Tool (SMAART) use a very "flat" — frequency-neutral — mic paired with sophisticated analysis software to measure the frequency response of a room and make corrections as needed.

Each program has a specific testing algorithm with step-by-step instructions. You'll typically plug in your reference mic and blast pink noise through the speakers to find what sort of build-up the room is causing, then correct accordingly to get as flat of an EQ curve as possible.

As always, check and ensure that these "corrections" didn't do more harm than good — there's always a certain amount of error involved.

WRAPPING IT UP

Remember, live mixing is often an exercise in compromise. You often have to settle for the best you can do in a given room with the provided equipment. What's important is that you practice these skills. You'd be surprised at what a good engineer can do with a bad room.

You've been listening to music your whole life and you know what it should sound like —remember to use your ears and trust your gut!

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.

Cue: This option on a mixer essentially allows a chosen track to be played in your monitor headphones.

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.

PFL (Pre Fader Level): P re fader simply sends a copy of your track before the channel fader, whereas a post fader send will do this after your channel fader.

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.

Solo: Choosing one channel to Monitor via headphones.

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 setting up 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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