YAMAHA

Make Waves

YAMAHA GUIDE

TO

SOUND SYSTEM

TIPS AND TRICKS

by Brett Armstrong

CONSIDER BUILDING YOUR MIX USING 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 usually benefit from HPF, primarily vocals. On some consoles, it is referred to as Low Cut.
- The drums channel would likely benefit from compression. Set their 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

Multiple channels with a wide range of volume dynamics make it challenging to produce a well-rounded sound. Use compression to even out many of those volume spikes. Remember, compression is like a third hand. It makes the loud parts softer and the softer parts louder, evening out the overall mix.

Hear What Your Live Microphones Hear

Put your headphones on, listen via PFL/SOLO/Cue to a vocal microphone, and pay attention to all the other sounds the microphone picks up. This gives you an idea of other "stuff" coming through the microphone and proves why proper microphone proximity to a sound source is so important.

Know What You May be Boosting

Microphones on the stage can pick up a variety of background sounds. In particular, boosting the high-end frequencies of a vocal mic can pick up drum cymbals 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 range, to clear them. Often, the 325-350 Hz range is where the muddiness exists.

Use Reverb for Vocal Separation

You can push a singer into the background using a lot of reverb. On the other hand, a little reverb can make a singer stand out in the mix. Use your ears to find out what's best for your situation.

The Kick Drum and Bass Can Work Together on Your Low-end Sound

Try letting the bass give you the low-end tone while letting the kick drum win on the attack.

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, so 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

Use APC/UPS units to keep the power going to your vital audio equipment. Power can go out for several reasons and using an APC/UPS unit can keep your system powered and your service going.

Plan on Equipment Failure

Ask yourself questions like, "What could fail?" "How could we get around it?" and "What's the least amount of equipment we need to keep the system going?" Plan for equipment failure so that when it does happen, you'll be prepared.

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 microphone stand with a long microphone 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 microphone or DI goes out during a service, you (or a musician) can pull out the 911-microphone or your emergency setup – whatever you want to call it. If it seems all the equipment is crashing down, a church audio system only needs one channel and microphone.

Don't Forget About the 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 out of 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 a standard 32-channel rack EQ, except that it might automatically tighten up if a cut is below 3dB. Therefore, if you run a digital mixer, use the onboard 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

For example, try adding a lot of high-end on your toms. Even the bass guitar has usable sounds that aren't just on the low end.

Use Meaningful Distortion

Distortion can work on more than a bass or a guitar. It can even work on a snare drum. Distortion can sound different depending on how you use it and set the appropriate parameters. 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 broadcasted only when the input reaches a certain volume level, and you know it's when you are getting the frequencies you desire. 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 a PA with a specialized software program is.

Programs like System Measurement Acoustic Analysis Real-time 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. Still, you'll typically plug in your reference mic and blast pink noise through the speakers to identify what sort of buildup 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.

Wrap Up

Remember, live mixing is often an exercise in compromise. You frequently 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 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!

If You Are Ever Prone to Hitting a Piece of Equipment to Make It Work...

…once is maintenance, twice is abuse.

GLOSSARY OF TERMS

ACTIVE SPEAKERS:
Powered speakers, also known as self-powered or active speakers, are loudspeakers with built-in amplifiers. They can be connected directly to a mixing console or other low-level audio signal sources without needing 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 will enable you to add those 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.

GEQ:
A graphic equalizer 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 that are usually placed on the stage in front of the band.

INSERT:

In audio processing and sound reinforcement, an insert is an access point built into the mixing console, allowing the audio engineer to add external line-level devices into the signal flow between the microphone preamplifier and the mix bus.

PASSIVE SEPAKERS:

A passive speaker has no built-in amplifier; it must be connected to an amplifier through an ordinary speaker wire.

PRE/POST:

Pre and Post Sends are AUX sends. They control the sound sent to objects like nursery 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, pre-fader.

PREAMP:

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

SEMI-PARAMETRIC EQ:

Sometimes called pseudo or quasi-parametric EQ, a semi-parametric EQ 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:

The signal-to-noise ratio (abbreviated SNR or S/N) is a measure used in science and engineering 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:

Speakon is a trademarked name for an electrical cable/connector. Initially manufactured by Neutrik, 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 a way to "pre-mix" several channels on a sound console before sending them to the master output mix.

TRS:

Tip-Ring-Sleeve refers to the parts of the jack plug that the different conductors are connected 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:

Unity gain is a term 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.

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.

PINK NOISE:

Pink noise definition is - a mixture of sound waves with an intensity that diminishes proportionally with frequency to yield approximately equal energy per octave.

Q-VALUE:

Stands for "Quality Factor," defining the bandwidth of frequencies that will be affected by an equalizer.

XLR:

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

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