CHURCH OF THE ROCK

CRAVE

AUDIO | VISUAL | LIGHTING MINISTRIES

AUDIO TRAINING MANUAL

Third Edition - Updated 2020.10.08

Contents:

Session 1 – CRAVE Values and Audio Fundamentals

Before we get into how to run audio at Church of the Rock (COTR), we must first answer why. And the why is that it is an act of worship to God. Anything that is done out of a love for Christ is an act of worship. So serving the worship band, or getting a really good audio mix is an act of worship if we're doing it to please God. And when we are serving we are not missing worship because we can't close our eyes and sing our hearts out, our service is our act of worship.

Serving God with audio is part of us fulfilling our purpose in life. Our life finds purpose when we worship Christ and then dedicate our time, talents, and treasures out of response to His love.

This fits in well with COTR's vision which is to help people Know God, Live Free, and Find Purpose.

Values/Expectations

In addition to that vision, the CRAVE team has things it values and expects as well. Much of that is summarized in the following verse:

> *"By the grace God has given me, I laid a foundation as a wise builder, and someone else is building on it. But each one should build with care. For no one can lay any foundation other than the one already laid, which is Jesus Christ."*
> *- 1Corinthians 3:10-11 NIV*

By serving the worship team and pastors, the CRAVE team builds a foundation for them to share the Gospel.

CRAVE Values

At COTR, the technical team places a strong value on the following:

EXCELLENCE

- Using the technical gifts God has given us to achieve a high standard
- Showing commitment to the team by being prepared and ready when scheduled
- Having a continued desire to learn

MOTIVATION

- Seeing all technical expressions as an act of worship
- Realizing your role is part of a greater purpose
- Willing to share your skills, and help train others

ATTITUDE

- Modelling servanthood by showing faithfulness to your position
- Willing to be flexible when unexpected changes happen
- Realizing that receiving input is an opportunity to develop

And one of the ways these values are exercised by a front-of-house (FOH) audio tech is by how he communicates with and serves the worship leader and team.

Audio Fundamentals

Power up / Power down Sequence

Every sound system at any of our venues (and basically all around the world) needs to be powered on and powered off in a particular order.

Power On sequence:

1. Mixing Board
2. Main Speakers
3. Personal Monitoring System (such as Aviom or A&H ME system)
4. Stage Monitors

The Power Off sequence is exactly reversed:

1. Stage Monitors
2. Personal Monitoring System
3. Main Speakers
4. Mixing Board

CAUTION

Powering up, or down in the wrong order can lead to loud “pops” through the speakers which reduces the life of the speakers and can be damaging to listener’s hearing.

Mixing Board Basics

Though there are a wide variety of sizes, brands, and types of mixing boards, at their core they all function in the same way, and therefore once you have a good understanding of mixing boards as a whole, you should be able to understand most any mixing board after only a short while. Fundamentally they all combine (or mix) large numbers of audio inputs into controllable and hopefully good sounding audio outputs.

Main Features

I/O OR INPUT/OUTPUT

All mixing boards feature some combination of jacks to allow you to plug in audio sources such as microphones, instruments, music players etc. as well as jacks to allow you output that audio to some type of speakers.

CHANNELS

Channels are the most important aspect to a mixer. Essentially (and there are some exceptions to this) every cable that was plugged into a jack on a mixer can be controlled with the controls on that channel which is usually laid out in a strip and are often referred to as channel strips. So for example a Kick drum is plugged into input 1 on a mixer it can be controlled with the Channel 1 controls.

BUSES/GROUPS/DCA'S AND MAINS

In order to simplify mixing when there is a large number of channels being used, large mixers usually include either Buses, Groups, or DCA's. Though they work on different principles, they all achieve the same goal of allowing you to adjust the output level of many channels equally while only using one fader. For example for a drum kit with 8 inputs, a DCA can be used to change the level of the entire drum kit.

Channel Strip Controls

Since channel strips are the most important aspect to a mixer, we need to understand what each part of the channel strip does. While there is some differences in controls available on different mixers, and often different names, the fundamentals are exactly the same on all mixers.

HA (HEAD AMP)/GAIN/TRIM/SENSITIVITY CONTROL

Nearly always the top-most knob on any channel strip, this knob sets the input gain into your mixing board. It is crucial to get this set right to achieve proper gain structure.

PAN

This is one of the few controls that is called the same thing on every mixing board. For any stereo or surround sound system of any sort, including mixing boards, pan is used to place the audio more to the left or to the right speakers. This is very important for any kind of recording, or for live sound in which rooms are setup as stereo. But it’s important to note that many live sound setups are run mono, meaning that all the speakers output the same audio. And in this case the pan knob doesn’t do anything.

EQ CONTROLS

EQ stands for equalization, and allows for the boosting or cutting of specific audio frequencies. And while every mixing board will have EQ controls of some sort, how many, and how they function can be drastically different from one board to another so it’s important to familiarize yourself with the controls whenever you work on a new mixing board. But the most common controls are as follows:

Gain
While input or HA gain is to set the level of an entire channel, gain as an EQ control is used to boost or cut a certain range of frequencies.

Frequency
The frequency control is used to determine which frequency will be boosted or cut from low to high frequencies. Usually within the range of human hearing 20 Hz to 20,000 Hz

Q/Width/Bandwidth
This control will not work with all EQ types, but will determine how wide the range of frequencies affected will be by the boost or cut of the gain control. Wider settings will affect more frequencies and usually sound more natural, while narrower settings are more useful in pinpointing problem frequencies.

MIX/SEND/AUX

The simplest way to describe these controls is to say that they send a copy of the audio of that specific channel to another place. Or that you are creating a separate mix that goes to another place. While there are many uses for this control, the most common are to send audio to stage monitors for musicians, to send a different mix to a recorder, as an effect send, or to send audio to another room. One important function of most sends is the ability to send it pre or post-fader. This will determine if the send will be affected by fader changes or not.

DYNAMICS

This control will usually only be found on larger mixing boards. But is often one control of dynamic processors which help to even out audio levels or get rid of unwanted hums etc. Much more on this will be explained in another session.

SELECT

Select buttons are only found on digital mixing boards and usually used to select that channel to apply changes to.

ON/MUTE

On most mixing boards, the channels are on by default, and they have a mute button which essentially turns that channel off to all your outputs including your mix/send/aux outputs.

On some mixing boards, including most we use at COTR, the channels are by default off, and you need to press the *ON* button to turn them on. It's the exact same principle but in reverse.

CUE/SOLO/PFL

This button has 2 main purposes, first, to give you more precise metering of your input levels, secondly, to isolate the channels with this button pressed in your headphones.

FADER

The fader is used to control the output level to your main mix. A couple things to note, all faders roughly cover the same ranges of level, but longer faders will allow smoother changes to the level as it's changes are finer, and changes to the fader usually don't change the level to your mix/send/aux outputs (though sometimes it can be setup this way).

EXERCISE 1

In this exercise, the leader will take the participant through the basics of setting the input level on a microphone channel.

1. Solo/Cue correct channel
2. Turn HA knob until level reaches approximately -12 db
3. Push fader to useable level, should be somewhere roughly close to 0/unity

EQ Basics

EQ can be used for shaping the tone of a source, separating similar sounds, reducing feedback, and other uses. Every mixer will have a different number of, and sometimes differently functioning controls. There are 2 main types of EQ's, parametric and graphic.

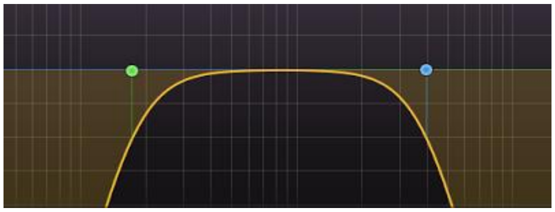
PARAMETRIC

Parametric EQ's are EQ's that usually have a relatively small number of bands, but allow you to sweep which frequency is being affected, and how much you are boosting or cutting it. They often also have some type of control for how many frequencies are being affected usually called bandwidth or Q.

Parametric EQ's have 3 main EQ shapes they can use:

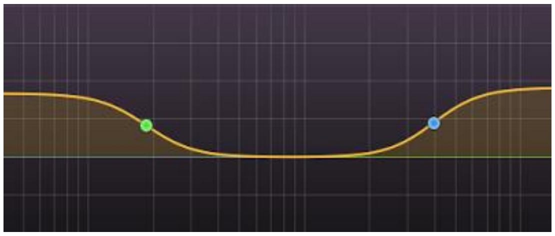
Filter

A Filter is an EQ shape that entirely removes frequencies above or below a certain point. The filter to remove low frequencies are called either low-cut filters, or more commonly high-pass filters (HPF) as they allow high frequencies to pass through unaffected. And the filter to remove high frequencies similarly is called either a high-cut filter, or low-pass filter (LPF).



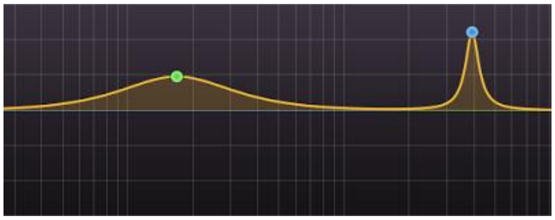
Shelf

This EQ shape is called this because when visually represented it resembles a shelf. Similarly to a filter, there is usually a low and high shelf which allows you to boost or cut the entire low or high frequency ranges.



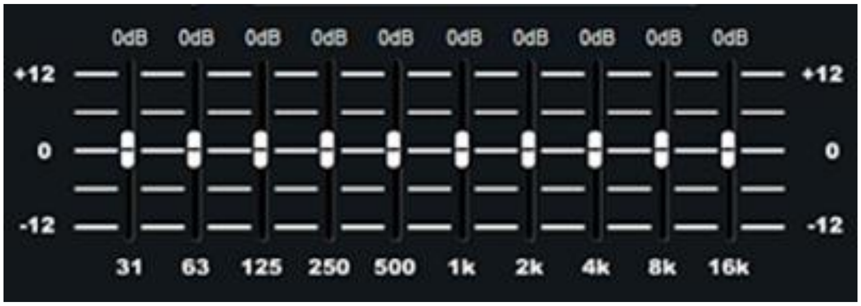
Bell

Called bell because of the shape, this curve is the most commonly used and it will boost or cut a specific frequency and some of the frequencies around it. How many surrounding frequencies that will be affected is set with the Q or Bandwidth control.



GRAPHIC

Graphic EQ's are usually only used for more system setup applications like "EQ'ing a room". They usually consist of a large number of sliders that boost or cut a set frequency.



EXERCISE 2

In this exercise the leader will have pre-recorded music playing, and will demonstrate all the filter types of the parametric EQ being sure to show all parts that were just explained, and then each participant will have a chance to experiment with all the parts of the EQ.

EQ Uses

Tone Shaping

The most obvious use for EQ is for tone shaping, or in other words to slightly change how a source sounds. Two things need to be emphasized on this point:

1. EQ can only slightly change a sound. If an instrument is not sounding very good, EQ will only be able to make a slight improvement. Which leads to the second point:

2. If you want high quality sound, you will need to attempt to get the best sound possible before starting to eq. The best sound will be achieved when you have a good musician playing a well-tuned and setup instrument, that is miked or DI'd properly and with correct gain structure. If all those things are in place, only subtle EQ should be needed.

Separation of Similar Sounds

Another important use of EQ is separating similar sounds. When 2 sounds that have similar sounds play together, they tend to mask each other, or in other words make both sounds more difficult to hear. EQ can be used to help reduce this and allow for more clarity between instruments. The most common use is between the kick drum and the bass guitar, often the very low frequencies will be cut out off the bass guitar, and the exact same frequencies added to the kick. A lot more will be covered on this topic in the training session dedicated to EQ.

Feedback Suppression

System setup is the most important factor in reducing feedback. We're not going to go into a lot of detail here as most of you will be using systems that have already been setup and tuned in advance. But some of the system setup that aids in this is speaker placement, mic choice and placement, monitor placement, gain structure, and room acoustics.

In certain circumstances, even after all these factors are setup correctly, feedback may still occur. It is in this scenario where EQ can be used to reduce or eliminate feedback. This will be demonstrated in the next session.

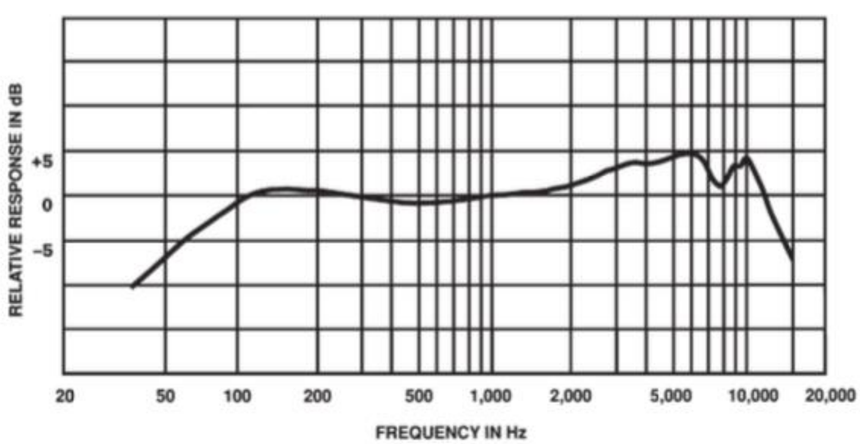
Microphone Basics

As mentioned previously, mic selection and placement is crucial for reducing feedback but it also very important for achieving a good mix.

The 2 main considerations in microphone selection are frequency response and pickup pattern.

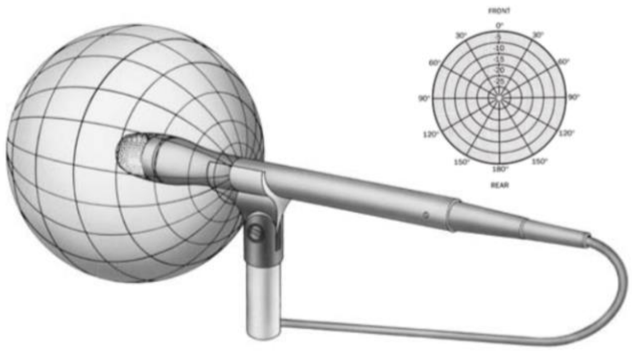
Frequency Response

Every type of microphone has a way in which it picks up sound. Some mics are made so that they pick up every frequency as evenly as possible. But there are many mics that are tuned to be used for specific applications. The most common examples are live vocal and instrument mics. They are tuned to sound best on those sources, and help to reduce feedback and boomy-ness, and sibilance.

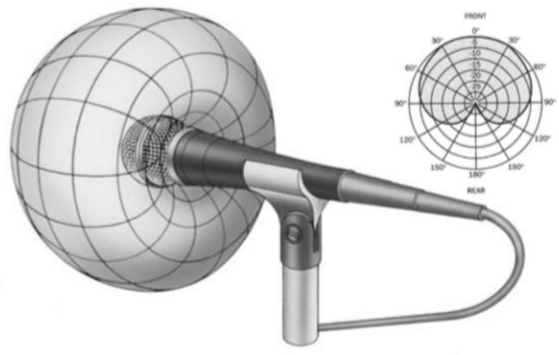


Pickup Pattern

Most people assume that all mics pick up sound from one end and not from any other, but that is not always the case. There are actually quite a number of different ways mics can pick up sound, but the 2 main ones that will ever be used in live sound are cardioid and omnidirectional.



Omnidirectional Microphone



Cardioid (Unidirectional) Microphone

CARDIOID

This is the most commonly used microphone and works how most people expect. It is most sensitive to sound directly in front of it, but picks up nearly no sound directly behind it. This is very useful to reduce feedback and to help separate instruments.

OMNIDIRECTIONAL

This mic picks up sound evenly from every direction around it. This is beneficial for mics such as Lav mics as it reduces the drop in volume when the source is not directly in front of it, and also allows it to be heard if even the mic is not placed correctly. But the tradeoff is that these mics are much more prone to feedback.

DEMONSTRATION 1

The leader will demonstrate the pickup patterns of the mic by turning them as he speaks.

EXERCISE 3

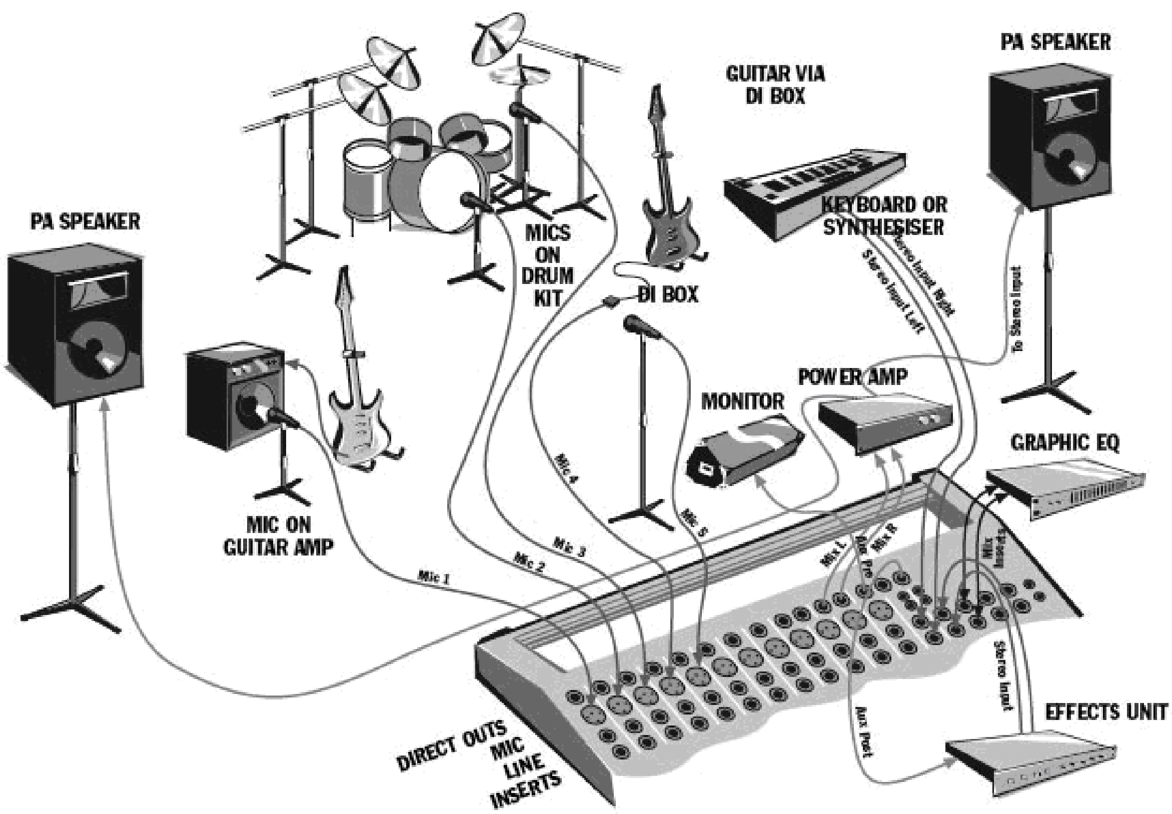
The leader will get each participant to setup a channel from scratch using a wireless handheld mike. The channel will need to be zeroed before this exercise. The participant will need to:

1. Solo/Cue correct channel
2. Turn HA knob until level reaches approximately -12 db
3. Push fader to useable level, should be somewhere roughly close to 0/unity
4. Use EQ to increase clarity and intelligibility.

Session 2 - Sound System & Troubleshooting

Sound System Setup

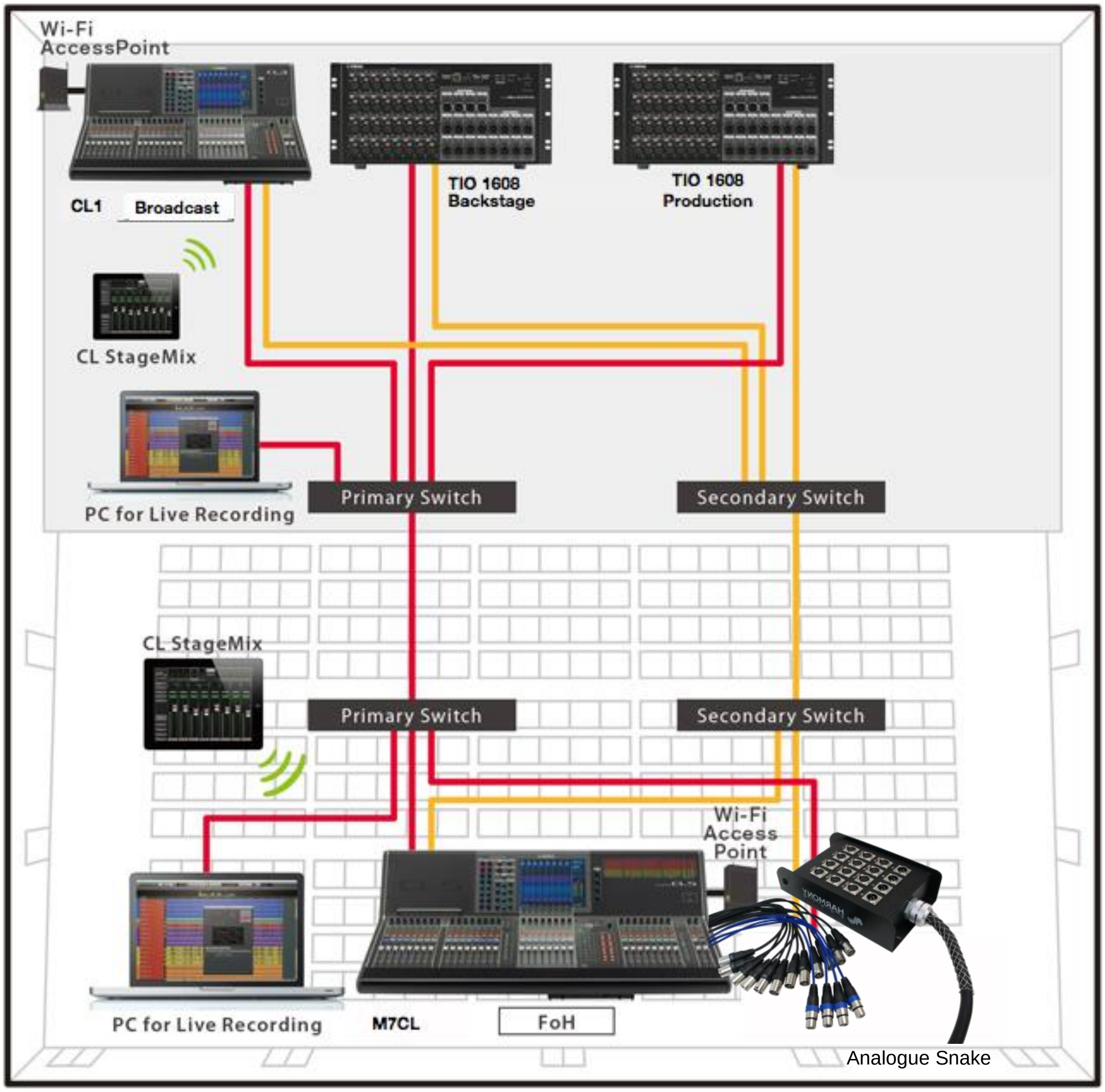
In order to be able to setup and troubleshoot a sound system, a basic knowledge of sound system setup is required. And throughout this session, the leader will demonstrate what he is explaining and/or allow participants to try things out.



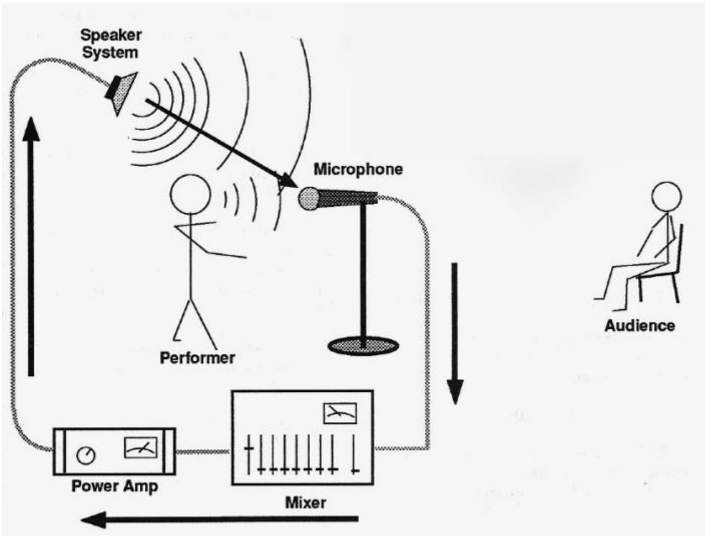
This is a very simplified and generic image that illustrates a basic sound system setup. At its most basic a sound system will have a mixing board as the main hub. Sound sources will be individually connected into it, the main mix will be sent out to main speakers, and other signals can be sent to stage monitors, effects units and the like.

In more advanced systems there is often a digital audio network also being employed, some form of personal monitoring (Aviom, ME 1, Monitor Mix App, etc.), and the ability to use tablets and PC's for control and recording.

The following diagram is like the Dante Digital audio network that is being used at COTR Winnipeg South Campus (WSC):



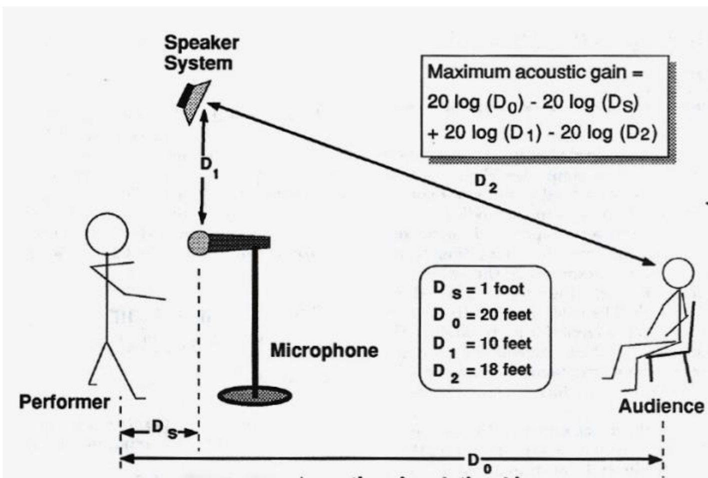
A well set up sound system will sound better, be less prone to feedback, allow for future growth, and will work well for the musicians or speakers on stage.



Feedback

Without getting into much detail, as it's actually a fairly complex topic, reducing feedback is best achieved using some simple principles.

This illustration shows that feedback occurs when sound that is picked up by a microphone, is amplified through a sound system and then the amplified sound is again picked up by the same microphone. It creates a feedback loop.



Acoustic gain relationships

And while this illustration shows a complex formula for calculating how prone your system will be to feedback, what should be taken from it is this: To reduce feedback,

1. Have performers closer to their microphones. (Ds)
2. Do not have main speakers behind microphones.
3. Have main speakers further away from microphones. (D1)
4. Have main speakers closer to the audience. (D2)

DEMONSTRATION 1

The leader will demonstrate how to use EQ to increase gain before feedback.

Sound System Components

This section is a good resource, but the leader will go through this section quickly to allow time for more hands-on training. There are a number of components common to many modern sound systems. Understanding the individual parts will allow you to use any sound system better, allow for quicker stage changes, and for more logical troubleshooting.

Mixing Board/Console

As was mentioned earlier, the mixing board is the central hub of any sound system and careful consideration needs to be made whenever purchasing one.

Mixing boards were introduced in the last session and will be explained in detail in the next one, so we will not discuss them more at this point.

Speakers

MAIN SPEAKERS

Main speakers are the speakers that provide sound to the congregation or audience in the room you are mixing in. They are usually full-range speakers, meaning that they produce sound from the low-mids up to the high frequencies.

SUBWOOFERS

Subwoofers (subs) are speakers that are also for the congregation. They produce only low frequencies, typically below 120-130 Hz.

MONITORS

Monitor speakers are on the stage and they provide sound to the musicians on stage. They will receive a different audio mix from the congregation that is tailored to the person using it.

All these speaker types can be either passive or active. Passive meaning they need a separate power amplifier to produce sound. Active meaning they contain a build in amplifier and can be connected directly into a mixing board. COTR’s different sound systems contain both passive and active speakers in various settings.

Personal Monitoring Systems

Personal Monitoring Systems are devices that allow performers on stage to control their own monitor mix. These often are used in conjunction with wired headphones or wireless earbuds (ears) or can be fed into a monitor speaker.

There are many manufacturers making these types of systems. Aviom used to be the industry standard but many other comparable or better products are now available. Many employ the use of handheld devices and apps.

Cable Connections

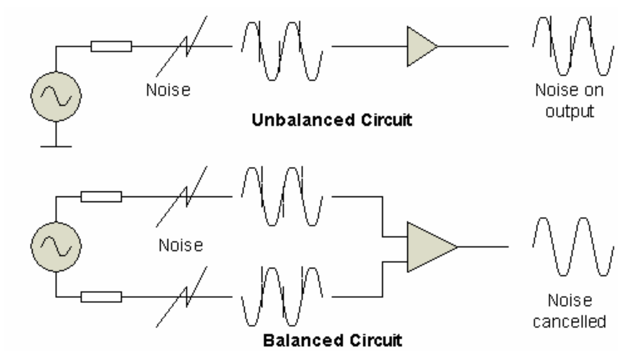
Though there are an ever-increasing number of wireless devices being made, there are still a lot of connections that need to be made with cables. While there are hundreds of different types of cable connections, I will briefly describe the ones most common to modern live sound.

MICROPHONE/XLR

XLR cables are the most commonly used cables in live audio. They are often called mic cables as that is their most prevalent use, but they are also used for instruments, powered monitors, effects units, recorders, amplifiers, and more. They are a balanced cable.

Balanced vs Unbalanced

To greatly simplify, an unbalanced cable is significantly more prone to interference and noise and therefore can only be run short distances while balanced cables are much less prone to these things and can be run long distances without introducing noise. Balanced cables are preferred in professional settings.



INSTRUMENT/TS/¼”

Instrument cables are used primarily to connect an instrument to an amplifier or DI box. The TS stands for Tip and Sleeve referring to the connector on the cable. They are an unbalanced cable.

TRS/BALANCED ¼"

TRS cables are often used in connecting audio equipment, such as recorders or powered studio monitors. The TRS stands for Tip Ring Sleeve referring to the connector on the cable. They are balanced when used mono, unbalanced when used stereo.

SPEAKER CABLE

Speaker cable is used solely to connect speakers to amplifiers. They can have a wide variety of connectors, including Speakon, ¼" TS, or none at all, so it's important to identify what type of cable you are using as using the wrong cable can cause minor to severe problems. Speaker cable is unbalanced and typically unshielded. Speaker cables use a much heavier gauge of wire and should not be confused with instrument cables.

CAT 5E/CAT 6/ETHERNET

Cat 5e or Cat 6 cable has been commonly used for years in computer networking. Now that most live sound systems include a digital audio network and personal monitoring systems these cables are now used for live sound as well. Often the cables are made more rugged for the demands of live use.

Snakes

Snake is the term used to describe a cable that sends large number of signals from one point to another.

Analog snakes are multicore cables meaning than in one cable are many separate wires where each can be used to connect a separate sound source. They are usually balanced and made up of XLR and/or TRS connections.

Digital snakes achieve the same purpose of sending a large number of signals from one place to another without losing quality by converting the signals from analogue to digital audio before sending the signal. All the signals travel digitally to the mixing board often with only one Ethernet cable.

DI Boxes

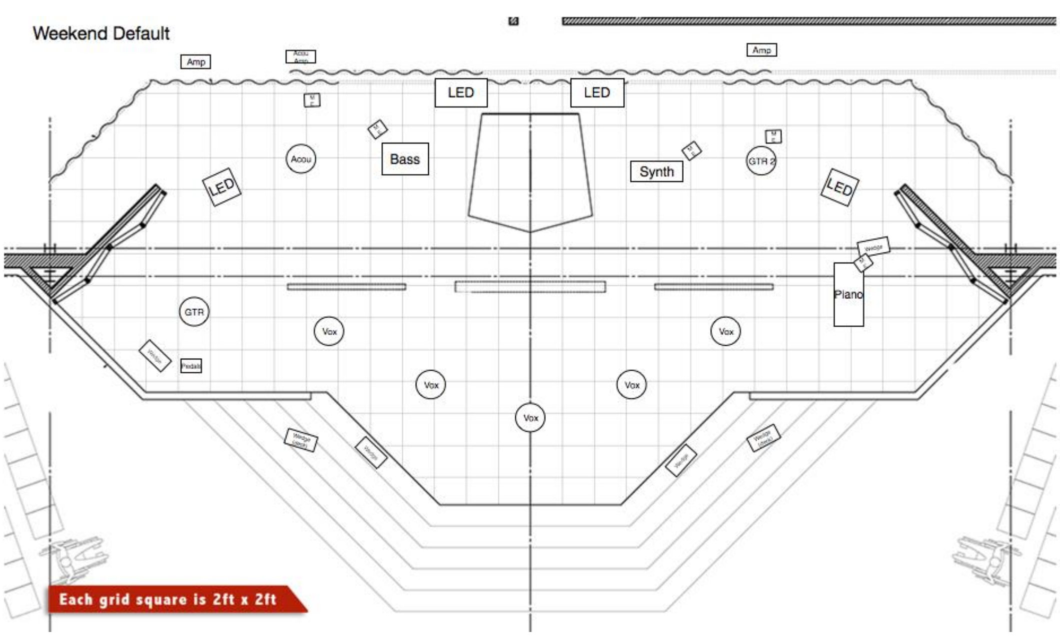
There are typically many unbalanced sound sources on stage that need to travel a long distance to reach the sound board. Unbalanced cables should not be longer than about 20 feet to avoid picking up unwanted signals. The solution is to convert the signal into a balanced signal. That's what a Direct Injection (DI) box does. A ¼" TS cable is plugged into it and an XLR connection goes out. They usually include a ground lift button and a pad. They are used to connect acoustic guitars, synths, laptops, etc. There are also DI boxes that also do some form of tone shaping. These are often used for bass guitars and sometimes acoustic guitars.

Microphones

There are many types of microphones. As we already learned, the two most important features of microphones are their frequency response and directionality. Using the correct type of microphone for the correct purpose is an important part to a well -functioning sound system. For example, this mic is a Sennheiser 835 and it is designed for handheld vocals as it has a cardioid pickup pattern (less prone to feedback) and its frequency response is tuned to the human voice.

Stage Setup

When setting up as stage for a worship band and for a presenter, there are a number of factors to consider, and therefore careful planning will be of great help.



*A slightly outdated stage plot for COTR WSC

Band's Comfort and Sightlines

The worship leader is responsible for the overall presentation of musical worship so before doing any kind of stage setup or change it is critical to discuss it with the worship leader first. Sometimes an important stage setup factor will conflict with a band member's comfort. In these cases, it's important to have agreement with your worship leader and it's important you are able to explain the reason for the conflict.

Some of the things the band or singers will need is the ability to hear their monitor, adequate space, sightlines to the worship leader and other important band mates, proximity to the gear they will use, and enough light to see charts.

Stage Volume

It is important that the worship team be able to hear what they need in order to perform well, so how monitors, amps, or any other loud source on stage is positioned becomes very important. Positioned poorly, a musician either will not be able to hear his monitor very well or he will hear too much of someone else's monitor and it will interfere with his own.

Excessive stage volume will both increase likelihood of feedback and reduce clarity of the mix to the congregation, so thoughtful positioning to decrease stage volume is critical. This is also what drives the use of electronic drums, headphones, and in-ear monitors.

Microphone Placement

Because of the efforts we take to reduce stage volume (using electronic drums etc.) we use fewer microphones on stage than if we had a fully acoustic band, however the positioning of any microphone remains critical.

Here are some considerations for positioning microphones for live sound:

- Less is usually more. More microphones usually add more potential for feedback and frequency cancellation, so use only enough microphones to allow you to boost what is needed for the room and capture clean audio for the broadcast mix. One exception is using 2 microphones on a guitar amp (see Appendix A).

- Feedback is your enemy. For the most part get the microphones as close to the sound sources as possible. You will get a stronger signal requiring less gain, and less of the unwanted sounds around it (including the PA system) will be picked up, all of which will improve your quality of sound and reduce feedback.

- Be thoughtful and use your ears. It's worth spending a few extra minutes finding the best spot to place your microphone, and the best way to determine if you've found it is by how it sounds. See Appendix A for some common mic placement examples.

- 3:1 Rule. If you use multiple microphones on a single source (guitar amp, choir, etc. see Appendix A), you will get the best sound if the microphones are 3 times as far from each other as they are from the source as possible. This reduces the effects of phase cancelation. One way to test this is to have both faders up in your mix, and in the head amp section of your mixer, change the polarity on one channel. You should hear a major change in the quality of sound. It should get quieter and thinner. If you do not hear this change you will need to spread the mics apart further, and/or get them closer to the source. Be sure to change the polarity back to normal after testing.

Cable Management/Stage Appearance

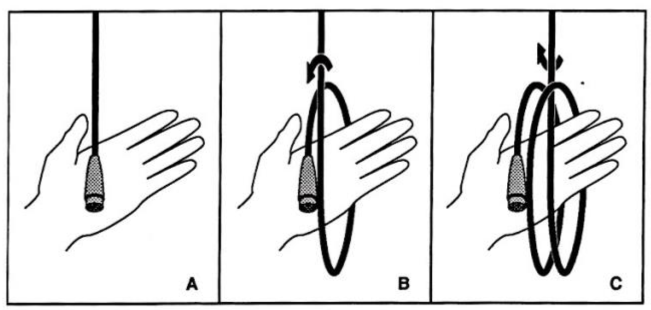
Thought needs to be given as to how to run cables on a stage. It may seem obvious but every piece of gear being used on a stage needs a way to be powered up and connected to the sound system, but you cannot just grab a cable and run it across the stage. If a cable isn't long enough to reach something in an orderly way, take the time to replace it with a longer cable.

The most important reason for that is safety. Poorly run cables can become a tripping hazard and can either injure people or damage gear. Wherever possible, avoid running cables across main walkways. If it is unavoidable, some type of cable mat or gaff tape needs to be used to secure the cables.

Additionally, the appearance of cables is also an important factor. Messy cable runs give an impression of a lack of care on the part of the audio tech and can even distract people from worship. Well run cables are less noticeable on stage and help foster an environment of excellence. This is especially important if video cameras are being used.

CABLE WRAPPING

It cannot be emphasized too much that cable wrapping is an incredibly important skill for an audio tech to have. Most people aren't even aware there is a right and wrong way to wrap cable but there most certainly is. Over/under cable wrapping is the right way to wrap cable. It achieves two important things: It greatly reduces the knotting up of cables and extends the life of cables be relieving internal pressure within the cable.



Over/under cable wrapping technique.

EXERCISE 1

The leader will demonstrate and then coach participants on how to properly wrap cable including how to wrap cable on the ground.

Lighting and Set

While compromises usually have to be made to allow for all the previous factors, being considerate to the people working on the overall look of the stage is important. It is best to avoid blocking light or important set pieces wherever possible or putting an ugly piece of gear somewhere very visible if another option is available.

Troubleshooting

The ability to quickly identify and correct a problem with a sound system is an incredibly important skill. While some of it will come naturally, having a systematic approach will increase the speed and accuracy in fixing problems as they come up. In all troubleshooting, if you make a change and it doesn't fix the problem, be sure to return system to how it was so as to not create new problems.

Here is a helpful approach:

KNOW YOUR SYSTEM

The reason we spent so much time going over the sound system setup is that the better you know the system you are working on the less problems you will have in the first place and you will be able to identify and fix the problems that do come up more quickly.

DETERMINE THE PROBLEM

Before you can fix any problem you first need a clear understanding of what the problem is. Often simple fixes can be overlooked because the problem wasn't identified clearly to begin with.

EXAMINE THE CLUES

The following mental checklist will help you find problems:

- What are the main symptoms?
- Is the problem in the whole system, or in one small part of it?
- If a part, which part of the system?
- Is the problem constant or intermittent?
- Has anything in the system changed since it was working properly?

Often simply asking these questions will already result in you finding your problem.

INVESTIGATION

If you still cannot resolve the problem begin to narrow it down. What parts of the system can be eliminated as not being part of the problem? For example, if a singer can't be heard in the main speakers but can be heard on headphones or in monitors, the microphone is not the culprit. For a systematic investigation start at the source and work your way forward. Follow the signal flow until your problem can be isolated.

ABSENCE OF SIGNAL

For this specific problem the following list should be followed systematically:

- Basics. Check that everything is actually plugged in and turned on and, if applicable, batteries aren't dead. These are by far the most common problems.

- Check equipment settings. Before assuming a more complex problem, check input gain, faders, DCA or groups, master faders, monitor knob, mute or on/off button, solo button.

- Check signal path. Make sure all cables are connected, and into the correct channels. For instance, if there was no signal from a Keyboard, check to ensure the instrument cable is plugged into the output of the keyboard and the input of the DI box. Check the XLR cable is going out of the output on the DI and into the correct channel of the snake. Check to make sure the correct snake channel is plugged into the correct channel of the mixing board. It is important to check all of these steps in order.

- Use a different source (mike, mp3 player, or keyboard) as a way of tracing the signal. For a particularly difficult problem involving an absence of signal or distortion, using a different sound source like a phone may help you isolate the problem.
    - With music playing, you can begin plugging a powered speaker in at each point in the signal path, starting at the mixing board and working your way out.
    - Start by removing the cable currently plugged into the output of the mixing board and plugging a known good cable out of that output and into the powered monitor.
    - If it works, plug original cable back into mixing board and continue up the signal path.

UNWANTED SIGNALS

If the problem is interference or distortion, the approach to troubleshooting differs slightly and will be more dependent on the type of unwanted signals. This is often more difficult to fix. A helpful strategy will be to eliminate what you can and work your way back from the source.

If the problem can be isolated to one input, start at the console. Move the source with the unwanted signal to a different channel input. If the problem is gone, it means the problem was in the channel of the mixing board. If problems persist, switch it back and continue working your way up the signal path.

Here are some common unwanted signals and probable causes:

- Hum. Usually caused by problems with grounding or electrical components nearby. If a DI box is part of the signal chain, the ground lift button may fix this problem.

- Hiss. Almost always caused by poor gain structure. Somewhere in the signal path something is too quiet so something else is gained up too much and causing hiss. Always check the level of the sound source before setting the gain on a mixing board.

- Static and crackling. Almost always caused by bad cables, or poor connections.

- Distortion. Like hiss it is almost always caused by poor gain structure, but in this case something is too loud to and is overloading some part of the system. For example, if the sensitivity of a wireless mic is too high, a loud singer will overload the microphone transmitter and send a distorted signal to the mixing board. Since it originates in the mic itself, there is no way to fix it other than to adjust the sensitivity of the mike. This is why it is important to monitor the input in microphones during soundcheck and rehearsal.

- Wireless frequency conflicts. If two or more wireless transmitters are on the same frequency, a noise will be created that almost sounds like digital birds tweeting randomly. In this case one of the transmitters will need to have their frequency changed.

MISCELLANEOUS PROBLEMS

Sometimes it is not that the signal is absent or completely unwanted, but just that the sound is just not quite right. This is where it's important to develop your ear as an audio tech and become very familiar with systems you are working on. In time you will be able to identify when something is off more quickly.

If the source is present but does not sound correct some things to investigate would be:

- EQ settings
- Compression settings
- Excessive Reverb
- Excessive monitors
- Phase cancellation
- Dying batteries

EXERCISE 2

The leader will guide the participants into setting up a very small sound system including a mixer, powered speaker, microphone, and guitar or synth with a DI box.

Session 3 - Soundcheck

Importance of Soundcheck

At every event where live sound is needed it is critical that a soundcheck is done, even if at times it can only be brief or even done "on the fly". Even if the setup has not changed there are a great number of small differences that can affect the quality of sound or new problems may have developed. It's important to check everything every time. After running sound for a while you will realize that proper system and sound checks will make services run more smoothly.

Additionally, for a FOH audio tech, the soundcheck is a great time to develop relationship with the worship leader and band. The best way to do this is to lead the soundcheck in an efficient yet friendly way. The worship leader, singers and musicians do not know what you need for a soundcheck and will often keep themselves busy with other things or noodle on their instrument until they are given direction from the audio tech. Performing an efficient soundcheck allows the band (and yourself) more time to rehearse. This will always be appreciated and will make the worship team feel taken care of by their sound guy.

You will also build that relationship by being prepared for the band to arrive. At COTR there is a structure in which the band will always rehearse, sometimes just briefly, before each service and soundcheck will always happen just before the rehearsal. On Planning Centre Online (PCO, our scheduling software) the rehearsal time is always listed.

System Check

Before the Band Arrives

The audio tech should arrive at least 45 minutes before the band arrives to do the system check and ensure that:

- FOH area (or case) is setup and powered up.
- Wireless antennas are in place and/or wireless receivers are turned on.
- Wireless mics and packs have charged batteries.
- Soundcheck Pastor/MC mics.
- Audio is being received for any video or stream to be played.
- Pre-service music playlist is available and working.
- Speaker system is turned on and fully functional.
- Personal monitor system (including wedges) are working and appropriate signals being sent to them.
- Every instrument and mic are connected, working, and cabling is in place.
- If there is another audio mix position (such as broadcast mix), communicate with that person to ensure they have signal on every channel.

EXERCISE 1

The leader will guide the participants in turning on the entire sound system.

Soundcheck

Audio techs don't fully understand what musicians need and vice versa, so clear communication is essential. It is really important to listen to the band as they rehearse to better understand what is trying to be achieved with the song.

It is important to explain yourself when you make changes that affect the band. For instance, when you ask a singer to sing for a long time during soundcheck, explain that you are not just making sure the mic is on, but you are setting it up so it works its best.

As the Band Arrives

With all of that in mind, as the band arrives the audio tech needs to do a soundcheck for each performer as soon as they become ready. The soundcheck needs to include:

- Ensuring sound source is setup it's best.
- Setting gain structure of every instrument and mike.
- Setting rough EQ and Compression settings.
- Building a rough monitor mix.

START AT THE SOURCE

As it is with troubleshooting, you will achieve the best sound possible by starting at the source of your sound and working your way back from there.

For example, to achieve the best bass guitar tone: Have a well-playing bassist, with a well setup bass guitar with the best settings, plug into the right kind of bass DI box with the correct settings, which is plugged into the correct channel of the mixing board, with correct gain, EQ, and compression settings. Some of this we cannot control. We cannot control how the bassist plays, or how well his bass guitar is setup. But the rest of the chain we can control. We can even ask them to make adjustments to the knobs on their bass. But it is crucial to work in the order listed above. If you try to improve a poor bass sound using EQ or compression, but the bass DI is set incorrectly, you will never be able to achieve the best sound possible. The settings on the Sansamp bass DI pictured is a good starting point.

This applies to all sound sources. Other examples include making sure electric guitar amps are miked with the best mic at the right location on the speaker, or proper gain structure on a singer's mike, or proper settings on an acoustic guitar's built-in pickup etc.

SETTING GAIN STRUCTURE

When many pieces of electronic audio equipment are used together, correct gain structure becomes very important to achieve good quality sound. Gain structure basically refers to which pieces of equipment are amplifying or reducing the signal and by how much. A properly set gain structure takes maximum advantage of the dynamic range and signal to noise ratio of each piece in the chain. No one piece is amplifying the signal excessively unless it is a piece designed for that function (such as a mic preamp). An example of poor gain structure would be a setup where a mixer's master fader is near the bottom, while all of the individual channel faders are near the top. The resulting level out of the mixer is much the same as it would be if all faders were near unity, but the chances of distortion are much higher because of limited available headroom in the channels before the master fader, while the signal to noise (S/N) ratio of the final output isn't as great as it could be where the master fader at a more appropriate level, which means your much more likely to have noise. Gain structure must be considered to optimize any system where levels can be adjusted in more than one place.

DEFINITIONS

- Gain. How much an electronic circuit amplifies a signal is called its "gain." In most specs or references you will see gain expressed as a decibel value (i.e. 6 dB of gain).

- Dynamic Range. The dynamic range of a sound is the ratio of the strongest, or loudest part to the weakest, or softest, part; it is measured in dB. For electronic equipment, the dynamic range is the maximum range available, which would imply it is the range from the highest level attainable in a unit down to the noise floor.

- Noise Floor. The point in which the signal gets lost in the noise generated by various electronic components.

- Signal to noise ratio (S/N ratio). The ratio of signal to noise. It is the heart of gain structure theory. Put simply, it is maximizing your sound system to enable you to have the most amount of signal with the least amount of noise.

- Headroom. The difference between the normal operating level of a device, and the maximum level that device can pass without distortion. Music generally has wide variations in dynamic range; without enough headroom, you'll find your gear clipping (distorting) far too frequently! This is why on digital mixing boards we set in input level to roughly -12 dB to give ourselves that headroom.

In Practice

Knowing these gain structure concepts will help you achieve better sound. You will want to ensure every piece of gear in your chain is set properly, and again it's helpful to start from the source. In practice it will include things like:

- Setting Mic Sensitivity. Most wireless microphones have a sensitivity setting. It is usually expressed in negative decibels (-dB), therefore a lower negative number means the mic is more sensitive and gain will increase, and vice versa. Most will include an input level meter. When a person is singing loudly you want to see a healthy level of signal generated at the microphone without clipping. If it is too low you will have poor S/N ratio, if it is too high you will be likely to distort the signal on loud passages.

- Correct DI settings. All DI boxes have some form of gain control. For most it is simply a pad button. By pressing this button, you will reduce the gain entering the DI box by a set amount, usually 10 or 15 dB. This is intended to tame very strong input signals. However, if it is depressed for a normal signal it will negatively affect your S/N ratio. On DI boxes like the Sansamp bass DI there is a level control and for the average bass guitar you will want that control at about the 12 o'clock position. If a bass has a particularly weak signal, increasing the level may serve you well.

- HA (Head Amp) Gain. This is the initial gain stage at the mixing board. With the levels set correctly on the wireless mics and DI boxes, you will set the gain on the mixing board channel so that during louder sections you will be getting a level of approximately -12 dB. One thing to note is that during soundcheck singers will never sing at full volume, so it's best to set their gain at around -18 dB during soundcheck.

    - *Mic/Line Level.* Microphones produce a relatively weak signal, usually in the range of -60 to -40 dBu. Line level has 2 main standards: +4 dBu for professional equipment like mixing boards, signal processors, and some electronic keyboards, and -10 dBV for things like phones, streaming devices, and computers. To deal with this massive difference in level, mixing boards will either have a mic/line selector button, or their HA gain will have sufficent range to accommodate these large differences (sometimes including a pad switch). It is important to be aware of the level you are receiving from a device, and how the input of your mixing board works in order to not have either very weak mic level signals, or overloaded line level signals.

- Signal processing gain. All compressors, gates, limiters, de-essers, and EQ's have either an attenuation (negative gain) or gain control. These should be used subtly, as any major increase or decrease at these stages will affect the rest.

- Fader position. Fader position should always be set to achieve a pleasing mix. If your gain structure is correct, with your DCA near unity, and master is at the correct position, most of your faders should end up fairly close to 0 dB or unity.

- Master fader position. Our sound systems are setup so that the power of the amps and speakers are such that if the mixing board had correct input gain and master fader at unity, the output level would be the maximum that would be desired in that room. For example, at the

WSC we nearly always run our master fader at FOH at -10 dB. With the input gain correct, and most of the faders near unity we end up with an output level right around 90 dB SPL. Pushing the master fader up to 0 dB would result in a level at right around 100 dB SPL which is the maximum we'd ever want in that room.

ROUGH EQ AND COMPRESSION SETTINGS

EQ and compression have been covered briefly in previous sessions, and in depth in subsequent sessions, but for the purposes of soundcheck there are a couple concepts.

Assuming you've already ensured you are getting the best sound possible from the source, I would very quickly try to get a rough EQ setting. Your speed in this will improve with time, but quickly try to get rid of any obviously bad frequencies or add some small boosts where needed. This is also a good time to ensure the gain is correct within the EQ.

Again, assuming the gain has been set correctly, quickly ensure that the threshold of the compressor is set so that it's not over-compressing or set so that it's not working at all. This will directly affect the output gain level

SETTING MONITOR LEVELS

Setting monitor levels for musicians is actually not as easy as it sounds. If, for example, a musician says they need more drums in their monitor do they mean all the drums or just certain ones? And how much more do they need? Becoming proficient at setting monitors will come with experience but there is a method to setting monitors well.

In the example above where the musician asked for more drums in their monitor, you would go to the SENDS ON FADER for their specific monitor and make small changes to the drums that they need for keeping time; the kick, snare, and hi-hats. After making that small adjustment, the musician will either say "yes that is good" or will ask you for more drums. This sometimes can take 2 or 3 attempts to get it right, but after a while you will begin to get a sense of the amount of change particular musicians want when they ask for more of something. Sometimes, as a way of trying to expedite this process, the musician may say, "I need 10% more of my own vocal." While it would be difficult to give them precisely 10% more, what they are communicating is that they want a relatively small change.

Since all of our digital mixing boards now have an iPad app that allows you to control the mixing board from the iPad, this process of dialing in monitor mixes can now be done in a more intuitive way. Usually during the first song of rehearsal, with the iPad in hand, the A2 audio tech can now walk up on stage and stand in the path of the various monitors (close to the performers) and listen to what they are hearing. The performers can then tell the audio tech exactly what they would like to hear and the changes can be heard instantly. This will help dial in monitors efficiently for the musician, it will train you as to how they want their monitor to sound, and it will build relationship and trust with the band and worship leader.

Personal Monitoring Systems

Each of campuses now has some form of personal monitoring system. It will be very important to take the time to get familiar with how it works so you can properly assist the musicians who use them.

DURING REHEARSAL

The band rehearsal is a great time to prepare so that by the time the service has started you have a well-crafted mix and a band that has what it needs.

The following can be done during rehearsal:

- Fine-tune EQ and Compression settings.
- Setting reverb levels.
- Testing of recorders or other equipment.
- Rehearsing your mix.
- Creating scenes on the mixing board.
- Fine-tuning monitor levels

CREATING SCENES

In the next session we will explain in more detail how to create scenes and use them on Yamaha digital consoles. Before we do that however, we will explain two scenarios where you may want to use them.

- Different Lead Vocalists. At WSC, it has become the expected procedure that we will create a new scene for each different lead vocalist. This is done for 2 main reasons: To move the correct singers into the lead and background vocal (BGV) DCA’s (to group vocals together correctly), and more importantly, to send the correct lead vocal and BGV to the correct channel on the personal monitors (ME1’s) so that the band will always be hearing the correct lead vocalist. At the other campuses it may not be as critical to create scenes for this scenario as the personal monitors are controlled differently by the band.

- Dramatic changes. If there are dramatic changes between songs where a great number of things will need to change immediately at the beginning of a song it may be a good idea to create scenes for the 2 different songs. Additionally, if you are mixing for a drama which includes many different microphones and people coming on and off stage with different lines, or with many sound effects, you may want to consider create a series of scenes to aid in this work.

When you are using scenes, you need to be very thoughtful as it possible to lose a mix you were working on by recalling a scene before saving the previous one or overwriting a scene by storing it without changing the number. Additionally, you need to be very thoughtful with how your scene changes are affecting the monitors, especially with many people using in-ears. Asking the band members if their monitors are set correctly before changing scenes will help to avoid problems in this area.

EXERCISE 2

With a willing and patient band member and singer, the leader will guide each participant on doing a rough soundcheck for an instrument and vocal including setting their monitor. Once all participants have done this, they will all go on stage and a few will adjust monitor using the iPad.

Session 4 - Yamaha Digital Mixing Boards

Yamaha digital mixing boards are known for their ease of use and ruggedness, and they are the primary mixing boards we use at all of our COTR campuses. For the purposes of training we are breaking down all the mixing boards we use into two categories: CL Series (including M7CL, LS9, and CL1, and CL5) and TF Series (TF5).

CL SERIES

M7CL

LS9

CL5 & RIO3224-D2 Digital Snake

TF SERIES

TF5

TIO1608 Digital Snake

DEMONSTRATION 1

During this explanation, the leader will demonstrate every area he is discussing.

I/O

I/O simply stands for input/output. The number and type of I/O connections on a mixing is one of the main factors that is considered when determining the correct mixing board for an application. For most modern mixing boards this will also include expansion slots for adding additional analogue or digital I/O, and digital snakes.

CL SERIES

M7CL & LS9

- Lots of analogue I/O.
- Not as well suited to use digital snakes.
- Expansion slots that allow a wide range of additional digital or analogue expansion options.

CL Series

- Limited analogue I/O.
- Designed for use with digital snakes.
- Same expansion slot options.

TF SERIES

TF5

- 32 mono + 2 stereo inputs. 16 outputs.
- Designed for use with one specific digital snake, the TIO1608.
- Expansion slot only for TF-specific Dante card for use with TIO1608.
- USB allows multichannel recording

TIO1608 Digital Snake

- 16 input channels, 8 outputs.
- Designed for use with TF series mixing boards.
- Works well with CL1. Software needed with M7CL or LS9.

Dante Network

Along with the advent of digital mixing boards came digital snakes and digital audio networking. The Digital Audio Network we use at Church of the Rock is Audinate's Dante Network (Dante). With Dante we are able to send up to 128 channels of audio to and from each of our digital audio devices. At our WSC we use it for our digital snakes, to send individual channels to and from our broadcast mix room, personal monitoring system, foyer sound system, and digital snakes. Certain campuses use it for their digital snakes.

Fader Banks

On an analogue mixing board there is usually 1 fader per input channel. Digital mixing boards, on the other hand, are not limited this way. A digital mixing board may have significantly fewer faders than input channels. When that is the case the faders are organized into fader banks.

CL SERIES

M7CL

- 62 faders. 96 potential inputs.
- Uses Centralogic system where the 8 faders at centre of console can control any set of 8 input channels, DCA groups, or output channels.

LS9

- 33 faders. 64 potential inputs.
- Has 2 banks for channels 1-32, and 33-64.

CL Series

- CL1 - 18 Faders. 96 potential inputs.
- CL5 – 34 Faders. 128 potential inputs
- Uses Centralogic system with input banks, as well as DCA and custom banks, all with 8 faders. Total mix capacity: CL1 - 48 mono, 8 Stereo. CL5 – 72 mono, 8 Stereo

TF SERIES

TF5

- 33 faders. 48 potential inputs.
- 2 input banks, an output bank, and a custom bank.

Knobs (Encoders)

One of the distinct differences between digital mixing boards and analogue ones is the amount of knobs. Digital mixing boards have significantly fewer knobs, yet they have greater flexibility.

CL SERIES

- When a channel is selected, each of these knobs will change the corresponding setting for that channel.
- User defined knobs can be setup to use for many other functions. One of which is Touch & Turn. This allows you to touch any knob on screen and adjust it using the Touch & Turn knob. Not available on M7CL or LS9.

TF SERIES

- Very few knobs. Instead, most board functionality relies on the multi-touch screen.
- Touch & Turn knob allows you to touch any knob on screen and adjust it. On screen, the knob will be covered by a purple circle.

Screen Displays

Understanding the different screen displays will allow you to navigate the different mixing boards with ease. All of the screen displays for all the different mixing boards in the CL series are basically identical, while the TF series screens have some major differences.

CL SERIES

Overview Screen

- Displays a less detailed overview of all major channel functions
- Return to this screen by pressing any bank button.

Selected Channel View Screen

- Display by pushing in any knob.
- “Medium-level” View of all channel functions.
- Some settings can be changed like DCA assignments and mute groups.

TF SERIES

Overview Screen

- Displays a less detailed overview of all major channel functions
- Return to this screen by pressing home button.

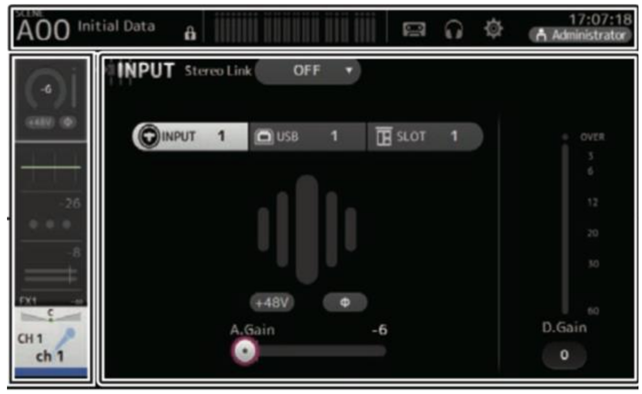
Navigation Area

- Displayed as a bar on the left whenever an area is pressed on the overview screen.
- Scrolls up and down.
- Pressing on any other section will bring up the corresponding configuration screen.

Configuration Screens

- Display by pressing corresponding field on Selected Channel View Screen.
- Most amount of detail and ability to change settings.

Configuration Screens



- Display by pressing corresponding area of overview screen or navigation area.
- Most amount of detail and ability to change settings.

EXERCISE 1

TF Series
The leader will get each participant to press every section of the Navigation Area, with a brief explanation of each configuration screen.

CL Series
The leader will get each participant to press every area of the Selected Channel View Screen, with a brief explanation of each configuration screen.

User Defined Keys

CL SERIES

- Allows quick access to functions we frequently use.
- Set by tech leaders.
- Usually used for quick access to SENDS ON FADER, mute groups, tap tempo, and talk back.

TF SERIES

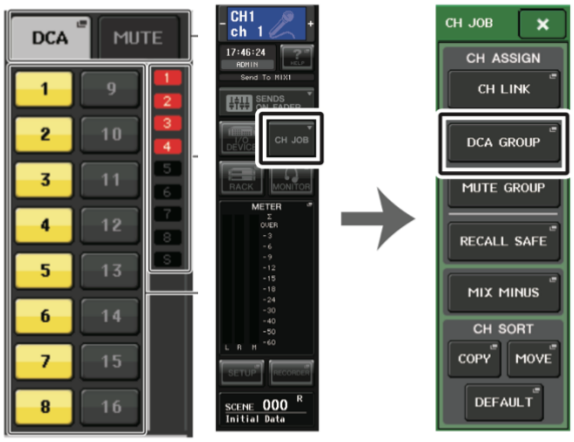
- Allows quick access to functions we frequently use.
- Set by tech leaders.
- Usually used for scene changes and starting/stopping recorder.

DCA Groups

DCA stands for Digitally Controlled Amplifier. What they do is allow you to assign a number of channels to a single fader,and then level changes to that fader affect the level of all the channels assigned to it equivalently. These act in much the same way as a subgroup found on other mixers, but work on a different principle.

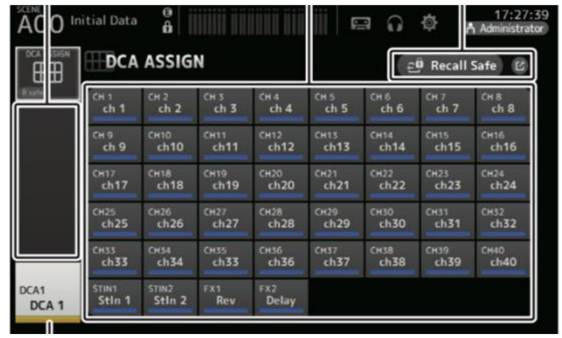
At COTR, DCA assignments are setup on all mixing boards that have DCA's and usually do not require changing. However, there is one instance in which they could be changed. Usually the lead singer is assigned to DCA 7 and the background vocals are assigned to DCA 8. Most weekends have multiple people leading songs. In this case it is very beneficial to change the DCA assignments so that the correct lead singer will always show up in your lead vocal DCA.

CL SERIES



- Can be easily changed from the Selected Channel View.
- Can also be set from Channel Job – DCA Group.

TF SERIES



- Can be easily changed by pressing DCA area of Overview screen.

Scene Management

The ability to store and recall saved scenes is one of the features that make digital mixing boards a joy to work with. For something like a drama presentation with a large number of microphones and instruments it is invaluable as it allows the audio tech to make quick changes to a large number of channels very quickly as the action takes place on stage. But it is also very useful on regular weekend services as well.

The most common time you would use scenes is when there is more than one lead vocalist on a weekend. In this case you would create a new scene for each lead vocalist. This allows for quick changes to things like DCA assignments, monitor mixes, and sends to personal monitors.

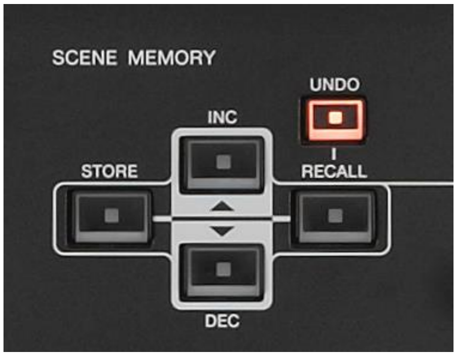
CAUTION

Much care needs to be given in your storing and recalling scenes. Not storing changes before recalling a different scene will result losing those changes, while storing over a scene with changes meant for a different scene number will overwrite that scene.

Scenes are normally created during rehearsal.

Changing Scenes

CL SERIES



- Buttons allow easy access to scenes.
- Further controls with pressing Scene area of overview screen.

TF SERIES

- User defined buttons used for easy access to screens.
- Further controls with pressing Scene area of overview screen.

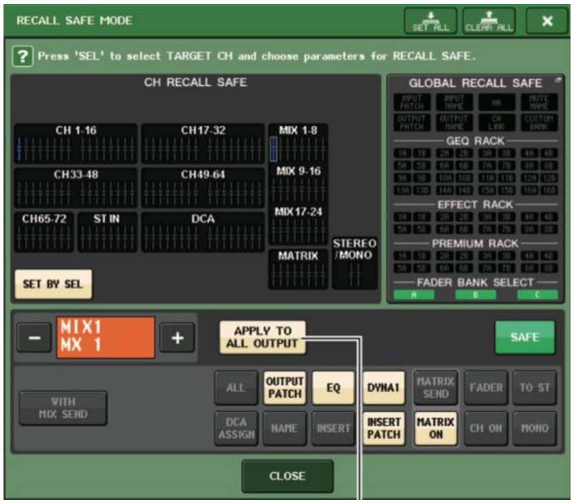
Recall Safe

Allows you to configure which items are recalled and which items are not (i.e., recall safe) when recalling Scenes and Presets.

Using scenes the way we do at Church of the Rock, we have found it helpful to use the recall safe in this way: For all instruments and vocals we recall safe the HA gain, EQ, and dynamics processing. For Pastor’s mic s and most audio sources from computers we recall safe everything. The reason is that for instruments and vocals, if during rehearsal multiple scenes are already created, and a change is made to gain, EQ or compression, having those things recall safe will mean that those changes will not be undone as soon as a different scene is recalled (also causing a noticeable change in level or tone). We recall safe everything for Pastor’s mics and computer audio because if a pastor is speaking, or a video is playing, and we need to change a scene (to the offering song for example), we do not want a sudden loss or change in those channels.

Recall Safe is already setup at each campus, but it is good to be aware of this functionality.

CL SERIES



- Note that what you want to safe and the safe button must both be engaged to safe a channel.

TF SERIES

- Note that what you want to safe and the safe button must both be engaged to safe a channel.

Mute Groups

Mute groups are a simple way to mute a large number of channels at the same time. Be careful using them as they are easy to forget. These are already set at the campuses. Pictures below show where you'd add or remove a channel from a mute group if needed.

CL SERIES

SELECTED CHANNEL VIEW screen

DCA/MUTE field

- Assigned to user defined keys.

TF SERIES

- Every channel is part of mute group unless safe from it.

Sends on Fader

The SENDS ON FADER feature of Yamaha digital mixing boards is very helpful. It allows us to make quick changes to any audio we would like to route to another destination by changing every fader on the console into a send to a particular mix. It is used primarily to create a monitor mix for musicians on stage, as an effect send, and for sending audio to other locations.

For monitor mixing we will select the appropriate SENDS ON FADER and adjust levels of the sources to create a monitor/headphone mix for musicians.

For effect sends, it is used to send multiple instruments to the same effect processor.

For sending audio to other locations we will select the appropriate SENDS ON FADER and adjust levels of sources we'd like to send to other rooms.

CAUTION

Care will need to be taken when using SENDS ON FADER as if you do not get out of the send you may begin making changes intended for your main mix to a monitor.

CL SERIES

- Tap Sends on Fader button on overview screen.
- Most commonly used sends assigned to user defined keys.
- Note how channel displays change when in a send.

TF SERIES

- Button for every send.
- Note how channel displays change when in a send.

Effects (FX)

Judicious use of spatial effects such as reverb and delay can really enhance a mix and Yamaha digital mixing boards have these effects built in. This is one area where experimentation and creativity will pay off.

The signal flow of a reverb is as follows: Using the SENDS ON FADER that corresponds to effect, push up the fader of the channel which you wish to effect. This sends that amount of signal to the reverb for processing. The reverb processes the audio simulating an acoustic environment. That signal is routed to an effect return channel. Pushing up the fader for the reverb return adds the effected signal to your mix. The level of the effect return in relation to the original signal is how you will adjust the amount of reverb.

Premium effects are available on the CL series of consoles. They are mostly emulations of classic recording equipment known for their pleasing sonic qualities. They are mostly EQ and compressors and would need to be inserted into a channel instead of using the signal flow mentioned above. Care must be taken to not inadvertently double compress or EQ a channel by using the channel strip EQ or compressor, and one in the premium rack.

CL SERIES

- Tap the Rack button on the overview screen to access virtual rack.

TF SERIES

- Tap the Edit button in the FX section to access FX screen.

EXERCISE 2

The leader will take each participant through the process of sending a channel to a reverb and blending back in the reverb return.

Metering

Understanding the meters of a mixing board will help you set your gain staging more effectively.

CL SERIES

CHANNEL STRIP METER

- Not very detailed.
- Do not use to set input gain.
- Use as a reference to quickly see if channel has level and it's not peaking.

TF SERIES

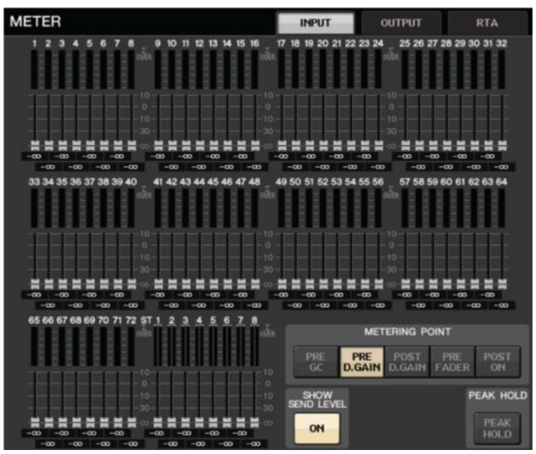
CHANNEL STRIP METER

- Not very detailed.
- Do not use to set input gain.
- Use as a reference to quickly see if channel has level and it's not peaking.

CUE METER

- Displayed when a channel is cued.
- More detailed.
- Use to set input gain.
- Input gain should be approximately -12 dB at peak. This is to give headroom to avoid digital distortion.

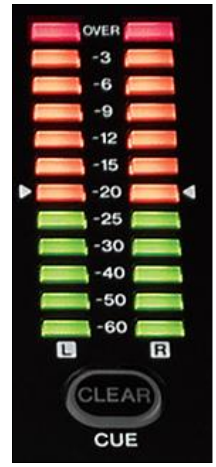
INPUT METERS



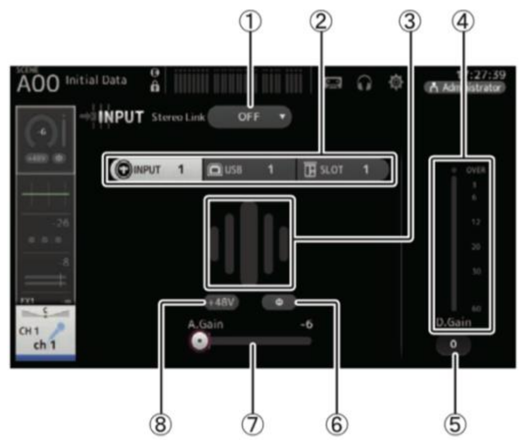
- Accessed by pressing meter area of overview screen.
- Shows non-detailed view of all input gain.

CUE METER

- Displayed when a channel is cued.
- More detailed.
- Use to set input gain.
- Input gain should be approximately -12 dB at peak. This is to give headroom to avoid digital distortion.



INPUT SCREEN



- Accessed by pressing input area of overview screen.
- The “Gain Finder” ③ is less accurate than using cue meter or level meter ④

User Settings

User settings allow for the restricting of parameters that can be operated by each user, or to change the settings of USER DEFINED keys, knobs, custom banks, and preference settings for each user.

These are setup on all mixing boards at Church of the Rock and most of the functions need to remain the same. However, the one user setting that you may want to change is which channels reside in your custom fader banks.

CL SERIES

- Access by pressing Setup from the overview screen, User Setup, Custom Fader tab.

TF SERIES

- Access by pressing Setup from the overview screen, Custom Fader Bank.

Save/Load

Yamaha mixing boards allow you save and load setup data on the mixing board to a USB drive. This includes every function and feature of the mixing board.

It will be a very good habit to save setup files regularly. The naming scheme that we use is: YEAR.MONTH.Type. For example, 2018.05.Weekend for saving the regular weekend data.

CL SERIES

- Access by pressing Setup, Save/Load.

TF SERIES

- Access by pressing Setup, Save/Load.

iOS Apps

Each Yamaha mixing has it's own dedicated Stage Mix app, and the TF series has a monitor mix app as well. This functionality has proven very effective for setup, troubleshooting, but most importantly setting monitors.

CL SERIES

Stage Mix

- Channels displayed by fader bank, which can be selected at the top.

TF SERIES

Stage Mix

- Software nearly identical to mixing board.
- Channels displayed by fader bank, can be selected from top or by swiping.

CL SERIES

SENDS ON FADER

- Press the SENDS ON FADER button to adjust monitors.

TF SERIES

SENDS ON FADER

- Press desired aux in details area on right of screen.

- Choose which monitor by pressing Select Target Mix button.
- Press SENDS ON FADER button again to return to main mix.
- Press Home button to return to main mix.

Monitor Mix (CL and TF Series)

- Available for iOS and Android.
- Can only control a single Aux channel on a TF series board.

- Phone must be signed into correct Wifi.
- Open App then select mixer.

Select appropriate Aux

Adjust mix as required.

Session 5 - Preparing to Mix

Mixing Style

Although there are individual preferences, there are generalized styles of mixing. For example, Adele always has vocals high in the mix, whereas other pop bands have vocals just above the instruments.

COTR has a general standard related to wanting fullness and "big"ness, (think Hillsong) with vocals just above instruments as opposed to minimal instruments with lead vocal jumping out.

We serve the worship leader so build rapport with them take their input — **the worship leader is ultimately responsible for overall sound of worship team.**

Soundboard Layout

When mixing a live service, the audio tech has a lot of things to manage and to watch. Having a logical layout of instruments and other sources on a mixing board is really important to make changes quickly in the moment. Additionally, having a generally consistent layout on mixing boards on consoles in various rooms or campuses makes it easier for a new audio tech to jump in and start mixing.

So here is a normal board layout, from left to right, on a mixing board in any COTR room or campus:

Drums - anywhere from 2 to 8 channels which can include: Kick, Kit, Snare, Hihat, Ride, Tom L, Tom R, OH L, OH R; Bass, Synth, Keys, Acoustic Guitar, Electrics Guitars, Rarely Used Instruments, Lead Vocal, Background Vocals, Pastor Mics, Computers/Phones etc.

Rehearsal and Note-Taking

It is just as important for the audio tech to rehearse as it is for the band. It is during rehearsal where you can get a much better sense of levels and sounds being produced by the individual instruments and vocalists. It is also where you can try out different effect settings, but more importantly, it's where you can learn the cues in the song that require changes in level. For example, verses of songs usually need more vocals and less instruments as a whole.

As you are rehearsing cues in the songs, it's very important to make notes. You will use them to remind yourself of various cues during the service and they can also be used to help other team members in their roles (for example visual techs).

Specifically these things should be noted on PCO:

- Who leader singer/s is/are
- Which instrument starts the song - this is so that your mix can start off right with the right feel instead of starting wrong and everyone hearing you adjust your mix.
- Solo's/Special parts to songs

Instrument Sound and Purpose

While the term "good sound" is subjective, there are certain things that our ears expect to hear. Each instrument has a specific sound — certain things that make it sound like what it is. Before any mixing can be done, an audio tech needs to understand what makes up good sound in each instrument.

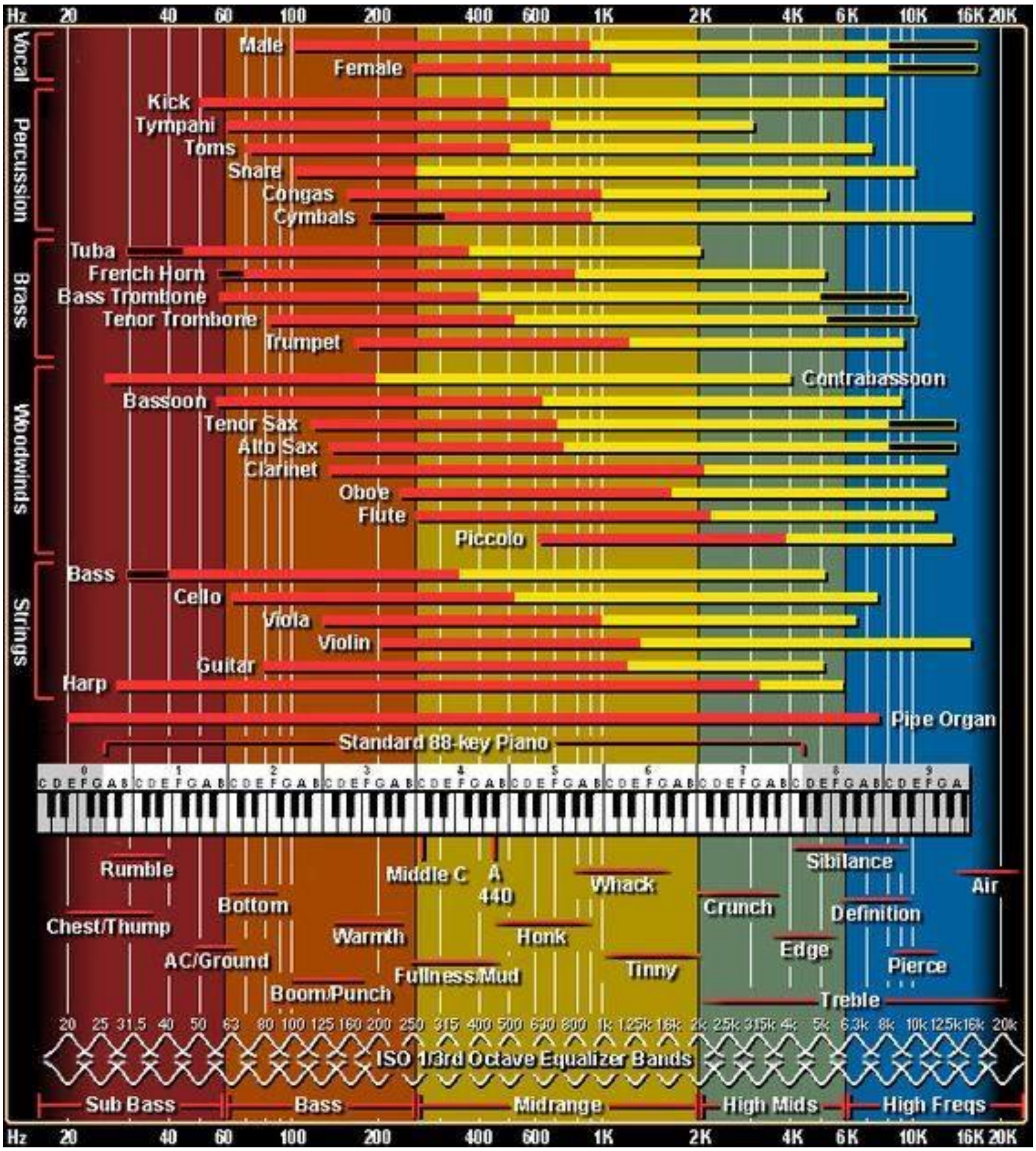
And in addition to sounding good, it's actually very important to know what role each instrument plays within the band and the mix.

DRUMS	
Kick	2 main parts to the sound - Thump and Beater sound Adds body Adds tempo
Snare	2 main parts to the sound - Body + crack (higher in freq than kick) Adds rhythm People clap with the snare (most of the time)
High hat	Open + closed Most constant part of the beat. Most similar to metronome. Often used for the band to stay on time
Ride	"Ting" noise Used in a very similar way to hi-hat, often used in place of hi-hat, rarely are both played at the same time
Toms	Usually 2 or 3 toms used Noticeably pitched, so seemingly play 3 different notes Floor tom contain a lot of low end energy Usually used sparingly and mostly for accents, fills, or transitions.
Overheads (crash cymbals)	Used for accents Only contain higher frequencies.
**Kit*	In some room and campuses, the electronic drums only have 2 outputs: Kick and Kit. All of the drums besides the kick are combined on a channel called kit. While a limitation, it is still helpful to be aware of all the different drums to know how they are contributing to the sound.

BASS	
	Adds body Relationship to kick drum very important. Kick and bass together provide the entire very low end of music. Often used for vocals to hear pitch as bass nearly always playing root notes of chords Works with drums to get "groove"
SYNTH	
	Providing fill to mix. Sometimes used for lead lines, in that case it's important to note that. Often dialed back due to it cutting through - esp. with church organ sound. Can be "annoying" if too loud.
PIANO	
	Louder song = fills and prominent on emptier spots Quiet songs = often leading Piano probably has the largest frequency range of any instrument
ACOUSTIC GUITAR	
	Piano and acoustic have a very important relationship "Melody section" of sorts Often will get muddy with too much piano, especially with Piano playing lower notes
ELECTRIC GUITAR	
	Very wide range of different sounds can be produced with an Electric guitar. Generally though, frequency range mostly in lower and higher mids. Most often will have “clean” and “dirty” (distorted) tones as well as FX. Can overpower a mix easily. Important for audio tech to know the song and how prominent it should be.
VOCALS	
Ranges (same terms as different types of saxophones):	Tenor: mid-range Alto: Mid high end Soprano: high end Fills - singer who sings off melody - need to make sure it is heard
Vocal levels	Vocals should be well balanced: - Lead always on top. - Remainder form “one voice”. Often grouped together on the mixing board. An audio technician will always "ride" vocal levels.
Dynamic ranges	Most vocalists go from singing quietly to loudly, some more than others, so this needs to be accounted for. Compression helps to even this out. EQ'ing and compression always used for all vocals to help balance that out.
Choir	It’s important to note that with a choir, the blending of their voices is a key part of their sound. The microphone used to capture their sound needs to be positioned higher and further away than would normally be used for an individual singer so that it picks up the overall group evenly instead of picking up some individuals more than others. In the mix choir level needs to be around where the background vocals are. Often when choir is part of worship the background vocals sound less distinct.

DEMONSTRATION 1

During this explanation, the leader will play examples of these different instruments and vocals to demonstrate the points being made.



This chart gives a general picture of the overall frequency ranges of various instruments along with descriptive words that are often used to describe sound.

Critical Listening

One major way to prepare for mixing and to develop your "ear" as an audio technician over time is the practice of critical listening. This is where you listen to well produced recordings, or well mixed live audio for very specific things. Try to learn what is making it sound balanced. This is especially helpful to do with songs that you are going to be mixing yourself.

Some things to listen for are:

- Tone of each individual instrument, thinking back to the unique characteristics mentioned earlier.
- Lead and backing vocal level in relation to each other and instruments.
- Specific parts coming through too much or too little.
- What's the main instrument in the song?
- What instrument is accompanying the main?
- Transitions between verses and choruses.
    - Are level of instruments dropping into verse?
    - Increasing for chorus?
    - Moments in song where instrumentation may change, i.e. electric guitar driven song quiets down and then more acoustic guitar and piano comes through.
- Use of reverb and delay or other FX.

EXERCISE 1

In this exercise, the group will listen to a well recorded live worship recording and analyze it together per above.

Session 6 - Mixing

Mixing is where all the skills we've talked about (and more) come together to create a cohesive, balanced, and pleasing mix of all the elements that make up a band. Developing your skills in audio mixing is a pursuit that you can keep refining and developing for a lifetime.

Purpose of a Good Mix

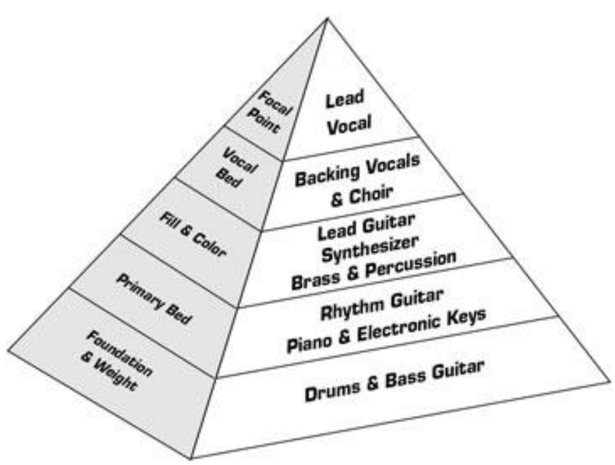
It's helpful to ask ourselves why we want a good mix. There are a number of reasons. First, we want the excellence of our mix to itself be an act of worship to God. Secondly, if we are mixing worship we want to aid the congregation in participating in worship. Third, a church that is dedicated to excellence is a good example to the world. Fourth, a good mix honours the work the worship team has put in. Fifth, the personal satisfaction of achieving a good mix. There are likely more reasons, but knowing the purpose of a good mix will inspire excellence.

How to Approach a Mix

Before you move a single fader, you need to know what a good mix is and what you are trying to achieve in your mix. As was mentioned in the last session, there are actually many different styles of mixing; generally louder or quieter, more full or more spacious sounding, lead vocal very prominent or less so, drum and bass focused or piano focused, etc. The style of mixing may need to change from song to song within the same service.

Do you see how you need a goal to aim for in your mixing? Never change a setting or move a fader without knowing why you are doing it. Just changing stuff to hope it sounds better will never result in a great mix. The aim of your mix should be that it sounds as good as any live album you've heard in the same style.

As was stated previously, COTR does have a particular mixing style. Our style has been developed by our leadership team, and we aim to achieve that vision as a way of serving them, and more importantly, God.



That style focusses on a blend of many instruments to achieve a big or full sound (think Hillsong) with the lead vocals only sitting slightly above instruments, and a large number of background vocalists blended well together roughly at the same level as the band. Some songs will have an electric guitar driving them, while other will have a piano, but in all songs every instrument needs to be heard in the mix.

DEMONSTRATION 1

The leader have a VSC session setup and ready will demonstrate building a rough mix with just levels from the foundation to the focal point as outlined in the drawing. If time permits, he will then have each participant create a rough mix.

Mixing Level

The level of your mix is an important factor in achieving a good mix. For worship, the level needs to be such that it is loud enough so that congregation can sing without feeling self-conscious about their voice, but not so loud so that they cannot hear anyone else in the congregation singing. At that point it begins to feel like a concert instead of corporate worship and people disconnect. And there is actually a dB range where this is achieved: 85-95 dB. At COTR WSC we run our worship at 90 db peak.

The level for spoken word is all about being able to be heard clearly but without being abrasive, and again at WSC we have spoken word peak between 76-78 dB.

People's perception of loudness (including your own) can be unreliable and affected by the frequency content of the music, so every mixing board will be equipped with an SPL meter to display the overall volume in the room so that it can be monitored accurately and adjusted.

With all types of mixing, there is a tendency to get louder as it feels like it makes the mix better, and that is what is called "fader creep". This is where, as you are adjusting you tend to keep wanting one thing louder, but then that makes something else get masked so you push that louder and so on. A much better practice is, when you want a certain instrument louder, is to find another instrument that is in the same range and full its fader back slightly.

Skills for Good Mixing

It's important to have really good skills. Good mixing is part art and part technical. Developing the following skills will help you grow in both of those aspects and is the reason why experience will be the best way to learn.

EAR

Having a good "ear" refers to someone who has the ability to hear small changes and discern things that need to be changed simply by listening. While this comes more naturally to some more than others, it is a skill that can be developed over time and with much practice.

MUSICALITY

While you don't need to be a trained musician to be a good audio tech, having an interest in music will drive you to excellence. It will make you aware that you cannot just "set it and forget it" with mixing. You will basically always be adjusting some fader at every point in all the songs, even minutely, to keep the mix active and engaging. Some people even refer to their audio techs as "mix musicians" for this reason.

ALERTNESS

Live sound is much different than recording in that you only get one chance to get it right, there are no retakes. There are also many things in a worship service that could happen suddenly so being alert and paying attention is key. You never want to miss a cue. Be alert.

SMOOTHNESS

Not like a smooth talker, though that might help too, but someone who is able to make smooth fader movements and will know when to mute things to avoid pops through the speakers. In order to do this, you need to be thinking ahead as to what is coming up so that you are proactive and begin your fader movements or mutes sooner, and aren't just reacting suddenly. This is why it's important to take notes during rehearsal and to know the songs that are being performed.

ATTENTION TO DETAIL

There is a lot of little details to keep track of when mixing.

PROFESSIONALISM

While it might not seem like a mixing skill it is. If you are more professional in arriving on time, how you interact with the worship team, how you setup a stage, and organize yourself while mixing, you will achieve better mixes and make the band and yourself more comfortable, which will help them play better and you mix better.

TECHNICAL KNOWLEDGE

It might surprise you that it is mentioned last, but all the other skills mentioned are actually more important than technical knowledge, though it is essential as well. When all these other skills are combined with great technical knowledge, it makes for an excellent audio tech.

Important Concepts

Mixing is definitely an artform, and there are certain concepts that will help you achieve better mixes.

Mix Along With the Song

When you are mixing a song you can never just “set it and forget it”. Throughout every song you will need to be adjusting the relative levels of instruments and vocals to create a pleasing mix. Most songs have more mellow verses with more upbeat choruses, and/or have different main instruments for the different parts, and often vocalists sing louder during choruses, so your job will be to ensure the levels of the verse and chorus are emphasizing the right things, the overall level doesn’t exceed the limits mentioned earlier, and the transition between parts is smooth.

Mix With Your Ears, Not Your Eyes

This might seem obvious, but since we are mixing sound we must adjust our mix based on what it sounds like. There can be a tendency to think an EQ must look a certain way, or faders should be at certain positions. These decisions and all others need to be based on how they are affecting the sound. If something looks wrong, but sounds right, it’s right.

Cuing (Soloing) is Your Friend

In a live mix setting, we do not have the luxury of hearing individual channels or groups on their own in the sound system. What we can do is cue individual tracks or DCA groups. This can be especially helpful in isolating channels that are having a technical problem, a tuning issue, or an EQ issue. One very helpful technique is to cue your background vocal (BGV) group. This will allow you to blend the BGV’s so their relative level to each other is balanced, then use the DCA to adjust the overall level of the BGV group.

Live Broadcast Mix

At COTR, in addition to our weekend services we also have a live online service, video venue multi-sites, TV broadcast, an app, and on demand video and audio (including podcasts). And all these sources require a good sounding audio mix as well. But the audio mix created by the FOH audio tech is not suitable for all these sources as the balance in the sanctuary is not the balance needed for broadcast. Therefore, we have a separate live broadcast mix position as part of our production room. Most of everything above applies to a broadcast mix, but there are some distinct differences such as:

- Panning. Most live audio is mono, broadcast will always be stereo, and while having panning that roughly matches the band layout on stage usually is best for video, there is room for creativity with panning to improve the sound of the mix.
- Level. While the FOH audio tech sets the volume of sound in the room he's mixing, a broadcast audio tech will set level strictly with output meters to ensure the right level is going to all the sources. The level in the broadcast mix room is only important for the broadcast mix tech to hear the mix accurately.
- Audience mics. Mics are placed around our main sanctuary to capture the congregation during our services. They are not needed for FOH audio but for Broadcast Mix they are used to capture the congregations singing, applause, and reactions. And getting their balance right is important to a good broadcast mix.
- Audio and Video recorders. The broadcast tech is required to start audio and sometimes video backup recordings.
- Less Band interaction. Since the broadcast mix is away from the stage, the tech will have little or no interaction with the band during services or rehearsal. So it's important for them to use downtime to try to get to know the worship team.

EXERCISE 1

With a leader at multiple mix positions, each participant will get a chance to build a mix from scratch including how the mix changes throughout the song and/or between different songs.

Session 7 - EQ Explained

EQ or equalization, is a powerful tool that is a part of every audio system in one form or another. When used well it can help enhance or polish a mix but used poorly it can destroy a mix. A simple description of EQ is this: a system that can be used to boost or cut specific frequency ranges in a sound source.

Before we continue with any further discussion about EQ, it's important to emphasize that before you begin making any changes to an EQ you need to have a goal in mind of what you are trying to accomplish. Many people will start turning EQ knobs hoping to magically make it sound better. Whenever you EQ you need a goal in mind such as, "I need to increase the clarity of this vocal."

EQ Types, Controls, and Applications

As was covered in session 1, there are 2 main EQ types: parametric and graphic, with parametric being much more commonly used. A parametric EQ has 3 major EQ shapes: filter, shelf, and bell.

There are many different controls for EQ but they usually include gain, frequency, and bandwidth (often called width or Q).

Additionally, EQ has many uses; mainly tone control, separation of instruments, and feedback control.

Boost and Sweep

Boost and sweep refers to a method of identifying a frequency to be adjusted. It is accomplished using a bell curve with a fairly narrow bandwidth (Q), boosting the gain substantially, and then using the frequency knob to hone in on a frequency.

This is often used when trying to identify a problem frequency. You'll know you've found it using this technique when it sounds its absolute worst. Then you would normally apply a small cut to this frequency, widen the Q, and improve its sound.

It can also be used to identify a frequency that you want to enhance, an example would be clarity on an acoustic guitar. Using this technique you'll know you've found the frequency when it is excessively clear to the point of being sharp and then you would reduce the gain and widen the Q.

EXERCISE 1

In this exercise the leader will allow each participant to try a boost and sweep with a sound source with his guidance.

Common EQ Uses on Specific Sources

With EQ, it is important to remember that there are no magic EQ settings that will make everything sound amazing, or always sound good on a particular instrument. Every sound source needs to be considered every time. There are, however, some EQ settings that will commonly be more helpful. This is based on what we covered in the "Preparing to Mix" session where we explored how the different instruments have parts to their sound that our ears recognize as good sound.

With all that in mind, here is a guide of some common EQ techniques for commonly used instruments:

All instruments except Kick and Bass: In general, a HPF should be used on every channel of a mix with the exception of the kick drum and bass guitar (they are the only 2 instruments that produce any kind of usable sound in the very low frequency range). For every other instrument only rumbling or hum is usually produced at those low frequencies. Applying the filter gets rid of these noises and also clears up room for the Kick and Bass to be heard more clearly. One exception to this would be if there were a performance where the piano is the only instrument. In this case, you may want all of the low-end to fill in the sound. This works because it is not overlapping any frequency ranges of other instruments.

KICK

Boost very low frequencies (usually between 50-70 hz) to enhance the “thump”, boost high-mids (usually between 1 kHz and 2.5 kHz) to enhance beater sound, and often cut low-mids (usually between 250 Hz and 500 Hz) to remove “stuffiness”.

SNARE

Boost lows (somewhere around 200 Hz) to enhance “body”, boost high-mids to exaggerate “crack”, and cut low-mids to reduce “stuffiness”.

HIGH HAT

Very little EQ is usually needed. Roll off lows more extremely (more important for acoustic kit).
If too “sizzley" cut using high-shelf.

RIDE

Roughly the same as Hi-Hat.

TOMS

Roll off very lows frequencies, boost lows (in the 150-250 hz range). If ringing, sometimes adding a high cut slightly reduces it.

CYMBALS

EQ much like Hi-hats.

NOTE: Drums - At most of our campuses, and in our Youth Room the electronic drums only output 2 channels instead of 8: Kick and Kit. The Kick channel will include only the kick drum, while the Kit channel will include a blend of all of the other drums and cymbals in the drum kit. EQ changes to the Kick channel will only affect that drum sound, while EQ changes to the Kit channel will affect all of the rest of the drum kit. Care will need to be taken when adjusting the Kit channel so as to not adversely affect the sound of one drum by trying to adjust another. For example, trying to add more “snap” to a snare drum may make the hi-hat cymbals sound too sharp. The only way to adjust the individual drums within the Kit channel is to adjust them on the drum “brain”.

BASS

Each bass is unique, but commonly needs some low-mid boost for body and high mid boost for clarity. Interaction with Kick drum is very important, so often would shelf out, or at least cut very low frequencies (somewhere around 40-80 Hz range).

Piano, Synth, Guitars, and Vocal Relationship: The way that piano, synth, guitars, and vocals are blended in a mix is very important as they all contain a lot of overlapping mid-range frequencies which can make them harder to be heard clearly. Often one or more of these instruments will need a cut to the low-mid frequencies to reduce the overlap and increase clarity.

PIANO

Since it has such a large frequency range, EQ affects it more. Often cut lows and low-mids to clear up that area of mix when playing with a band. If piano is on its own, the lows and low-mids will not need to be adjusted nearly as much. Like the synth, it sometimes requires a high mid boost or cut for same reasons.
Synth - Since it is electronically generated it usually doesn't need much EQ. Roll off lows. Depending on sounds used, it may need high mid boost to cut through, or lowered to soften a sharp sound.

ACOUSTIC GUITAR

The sound of each instrument is very unique. Most acoustic guitars benefit from a high shelf boost to add openness and clarity, most sound better with a low mid cut as they can sound boxy in this area. Like a synth, acoustic guitars sometimes need a high mid boost or cut.

ELECTRIC GUITAR

Same as acoustic guitars in their uniqueness and relationship to other instruments. Distortion can fill up frequency ranges so loud guitar heavy songs might need more EQ finessing than quieter ones.

VOCALS

Each voice is unique and changes noticeably from soundcheck to performance. Common EQ's include high shelf boost (for openness and clarity), 3.5 - 4.5 kHz cut for piercing female vocals, low-mid cut to clean up "boxyness", and/or slight 2- 3 kHz boost to cut through mix (although too much sounds harsh and artificial). Always use a HPF.

EXERCISE 2

Through the explanation of this section, the leader will demonstrate and occasionally allow participants to adjust the EQ settings based on the descriptions.

Session 8 - Other Signal Processing

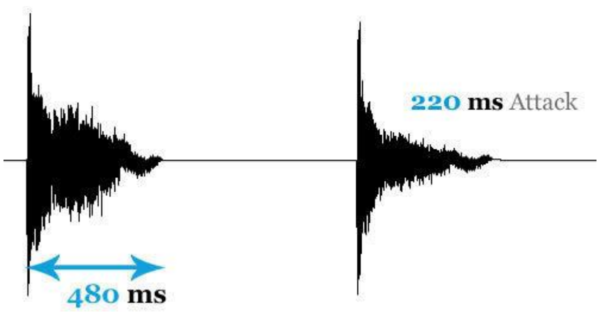
To assist audio techs in getting a good mix, many different signal processors have been developed over the years. In previous sessions we discussed EQ at length, but only briefly touched on the other most common signal processors which we will explore further here. They are compression, gate, reverb, and delay.

Compression

Compressors are in the category of what would be called dynamic range control. When used for audio processing, the term dynamic range refers to the difference in level between the loudest and quietest parts of an audio source. Compressors are used to compress the dynamic range of an audio source. In simple terms, they “even out” the volume of a sound source. They do this by automatically reducing the level when the audio source gets louder than a specified point.

Compressors are very useful for sound sources that have extreme differences between their loudest and quietest level (such as vocals). Without a compressor, a source with a large dynamic range will do 2 things: it will become too loud in the mix if they suddenly get louder, or become too quiet and get lost in the mix if they suddenly get quieter. A compressor, used well, will make this problem more manageable.

Another less common use for compression is for envelope control. Envelope refers to the way a sound changes over time (for more info search for ADSR). It is how quickly a sound starts, continues, and stops. For example, the sound of a person clapping has a very different envelope than the sound of a violin. A compressor can alter this character of a sound by how quickly it begins to reduce gain on the sound (using the attack control) and how quickly it releases the gain reduction (using the release control). Although used less frequently, we sometimes use a compressor in this way to make a kick drum sound “tighter”. The attack time is set slower to allow the initial transient of the kick drum to come through clearly. The gain reduction then reduces the level of the resonance of the kick drum making it sound tighter. The release is set fairly quick to not cut off the initial transient of the next incoming kick drum sound.

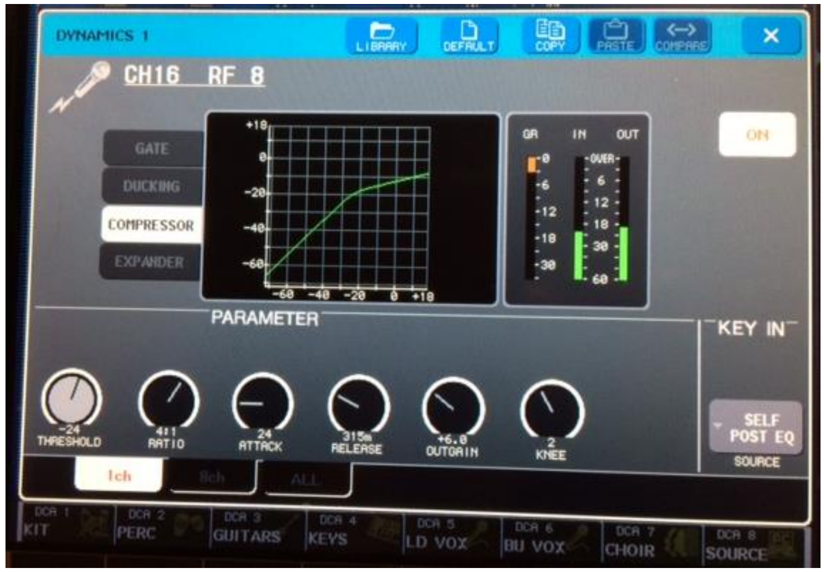


Compression Controls and Meters

While there are a variety of different compressor types, the vast majority of them share the same type of controls and metering. We will explore the most commonly used terms.

THRESHOLD

Threshold is the setting where, when exceeded, the compressor starts to reduce the level of the audio signal.



RATIO

The Ratio control sets the amount of gain reduction. For example: in a 3:1 ratio, every 3 dB of level above the threshold will be reduced to 1 dB.

Lower ratios tend to sound more natural and less noticeable and is why at COTR we tend to set ratios usually between 2.5:1 – 4:1.

ATTACK

The attack control sets how quickly the compressor starts to reduce level after it reaches the threshold.

RELEASE

The release control sets how much time it takes to stop reducing level once the signal has returned below the threshold.

GAIN OR OUTGAIN

This is another gain stage; it is important to be aware of it. Since the compressor is reducing gain, this can be used to add the gain back in. By doing this the average level will be increased.

KNEE

The knee control ranges from hard knee to soft knee and controls the transition from uncompressed to compressed. A soft knee will make the compression less noticeable by starting to compress the signal a bit earlier by a smaller amount.

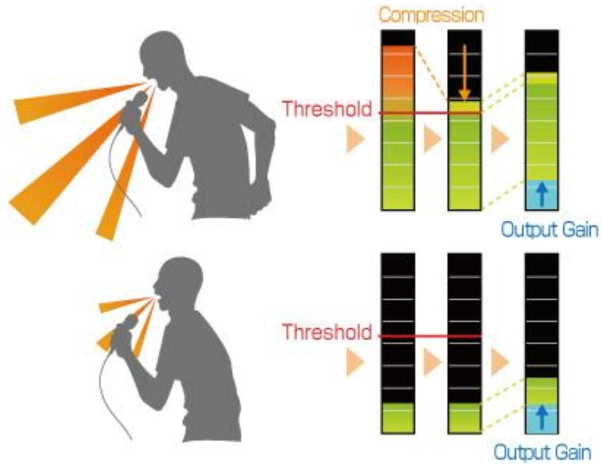
INPUT METER

Often labelled “in”, this meter displays the amount of gain that is coming into the compressor. Usually this should be the same as what is coming into the channel which was set using the HA control.

GAIN REDUCTION METER

This meter is often simply labelled “GR”. It displays the amount that the compressor is reducing the input gain.
The amount of gain reduction will be determined by how the compression controls are set. It is important to monitor the amount of gain reduction happening on all the channels on which you set a compressor. Too much gain reduction will “flatten” or “suck the life out of” the signal, but too little will mean that the compressor is not doing the job of “evening out” your audio signal.

At COTR, for the most part we set our compressors to reduce gain by about 6 dB for most sources. Sometimes up to 8 dB for more dynamicsources. We have found this to still sound natural and retain the life of the signal, but still controls the dynamic range enough to be helpful.



OUTPUT METER

Often labelled “out”, this meter displays the amount of gain that is leaving the compressor. It will be affected by how much gain reduction is being applied, and by how much outgain has been set to counteract some of the gain that has been reduced.

Limiting

Limiting is a form of compression usually with a very high ratio and hard knee. Limiting is usually only used to protect gear, protect musicians hearing (i.e. with in-ear systems), or in mastering. It is rarely used on individual sources in live sound because it sounds too noticeable.

DEMONSTRATION 1

The leader will demonstrate how compression works for level control on a few sources: acoustic guitar and a vocal. The leader will then demonstrate using a compressor for envelope control with a kick drum. In both cases, he will over-compress the signal intentionally to better demonstrate the effect then dial it back to useable level.

Expander/Gate

An expander (or gate) works somewhat in the opposite way as a compressor. Instead of making the louder parts of the sound quieter it adjusts the quietest parts of the sound to be quieter or completely silences the signal. An expander (or gate) will have similar controls as a compressor.

Some uses for this effect include:

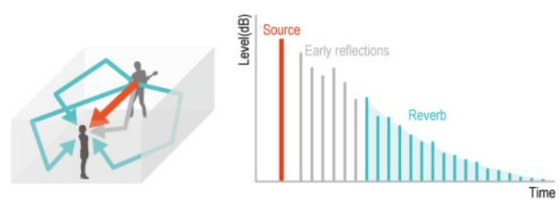
- Eliminating a persistent hum on an instrument when they are not playing.
- Silencing mics used in a drama when the actor is not speaking to reduce feedback.
- When used on acoustic drum kits to try to better isolate microphones from one another; the mic is only open when that particular drum is being hit.

DEMONSTRATION 2

The leader will demonstrate how gates work by using either a source that has a hum, or if that is not available a vocal, so gate will come in after they finish singing lines. It will be explained that this is being as an example and that vocals often are not gated except for potentially in dramas.

Reverb

Reverb stands for reverberation. On modern consoles a signal processor is used to simulate the way sound reverberates in acoustic environments. In most of the rooms we mix live sound in, the room is designed to absorb most of the sound. This can make music sound dull. Acoustics are the natural sounds that would be present in a room, such as the natural echoes, and adding reverb can actually make it sound more natural.



When used well, reverb adds character, body, and warmth to certain sounds. When used poorly it can wash out mix or sound gimmic ky. Reverb should not be used on all sound sources.

Below are some general guidelines on using reverb on specific sources:

- Instruments with a lot of low-end typically should never have reverb on them. The kick drum and bass guitar both tend to sound very washed out and muddy if reverb is applied to them.

- The rest of drum kit sounds good with a room reverb set with a short reverb time, especially the snare drum. Only use long reverb time on drums for special effect and usually only for part of a song as it can decrease clarity (or just sound really "80's").

- A synth or electric guitar usually have reverb in their signal when it arrives at the sound console. In rare cases you may need to add it as an effect.

- Piano and acoustic guitar can benefit from a hall reverb. Room or stage reverbs can sound good as well.

- Vocals usually always benefit from reverb when they are singing. When they are talking or praying it should be removed as it sounds out of place. Plate reverbs as well as halls are most commonly used, but actually many others sound good on vocals as well. Some experimentation is good here, but be tasteful. As a general rule, faster songs should have shorter reverbs and slower songs have longer reverb times. This is because with faster songs the next note or word comes sooner. If the reverb hasn't trailed off, it now interferes with the next note causing it to wash out. On slower songs there is more time for the reverb to fill in between beats. The longer reverb on slower songs adds to the dramatic feel of slower songs. Reverb also slightly covers over vocal imperfections, and vocals generally sound stark without reverb. Almost every vocal you have ever heard live or recorded has some reverb on it. There are a few exceptions with really good singers and when they want it to sound stark.

Reverb Controls

With the wide selection of digital audio consoles, audio software, and outboard units available there are many ways that different reverbs can appear and function. Between them there are a wide variety of controls available, but for our purposes we will touch on the two most critical controls which are common to mostly all of them.

REVERB TYPE

Reverbs are always broken down into what type of acoustic environment they are trying to simulate or method used to create the reverb sound. They include: Room, Hall, Stage, Plate, or Spring.

- Room - Room reverbs simulate the sound of a smaller room with good acoustics such as a recording studio. They complement drums particularly well.

- Hall - Hall reverbs simulate the sound of a concert hall. They usually are full-sounding and complement acoustic guitars, pianos, wind instruments, or vocals on slower songs.

- Stage - Stage reverbs simulate the sound of being on a large stage. These are used to give sound a live feel.

- Plate - Plate reverbs simulate a method of creating reverb in which the sound was passed through a steel plate and the resonance of that steel plate created the reverb effect. Plate reverbs usually sound quite good on vocals.

- Spring - Spring reverbs simulate a method of creating reverb in which the sound was passed through a metal spring and the resonance of the spring created the reverb effect. This reverb is still commonly found in electric guitar amps as that sound has become a standard electric guitar effect.

REVERB TIME

This is the length of time the reverb continues after the initial sound. It is sometimes referred to as decay.

In general, shorter reverbs sound better on drums or on vocals (in faster songs). The reason is that if a reverb hasn't decayed enough from one beat of a song before the next beat arrives it will mask that second beat. This will continue on causing the entire signal to become cloudy and less distinct.

Longer reverbs sound better on acoustic instruments, especially on slower songs. This will give a sense of the instrument being in a nice sounding room, it "fills up" the sound, and adds an ethereal effect.

Reverb time is often preset on the different reverb types, but can be adjusted to achieve these effects. Some examples were given earlier of when you may want to use longer or shorter reverb times.

Delay

Delay is discrete echoes of the original sound. Delay can sound really good if used cautiously, but can become very gimmic ky very quickly.

Many electric or even acoustic guitarists use their own delay. They like to control it as part of developing their own "sound" as a guitarist and we would not need to add another delay.

The only places we tend to use delay at COTR are on: vocals during very slow songs where you are looking to make it fuller and more dramatic (usually only in spots where the vocal is on its own), and on saxophone. Sax sounds much like a vocal, and benefits from tastefully used delay.

Delay often really benefits from being synchronized with the song's rhythm, and so we use a tap tempo to achieve this.

DEMONSTRATION 3

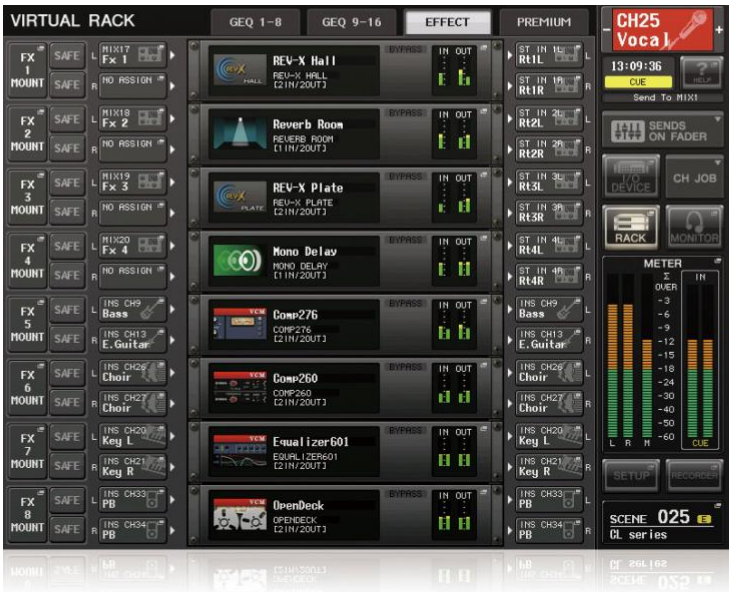
The leader will demonstrate reverb and delay on vocal tracks using different reverbs with different reverb times and by demonstrating tap tempo for the delays.

Signal Routing for Reverb and Delay

While it was briefly covered in previous sessions, it is important to review how to route an audio signal to apply reverb or delay to it.

For any console, audio software, or outboard unit there is a method for routing a signal that will sound better and allow for greater control:

An auxiliary send (bus, mix, etc.) is used on one or more channels on a mixing board to send a copy of the source(s) to the processor. The processor will create the reverb or delay and output the sound to another channel on the mixing board. The fader for that channel is then used to blend in the correct amount of reverb or delay.



EXERCISE 1

The leader will get each participant to experiment with changing the amount being sent to delay and reverbs as well as changing reverb settings.

What's Next?

Shadowing

While these formal training sessions are very informative, some of the best training will come "on the job." So if you are continuing on to serve on an audio team at COTR, we will facilitate this is by having you shadow an experienced audio tech at one of our campuses.

The way we use the term shadowing is to mean that you, as the trainee, will follow alongside of the trainer or another experienced volunteer as they serve in their role on the weekend. This means you will arrive at the same time to go through the rehearsals and services as the person you are shadowing.

This process will start with you simply watching everything, but progressively more and more responsibilities will be added until you reach the point where you could run the position yourself. This progression will start with you helping with equipment setup and teardown, changing batteries, making notes in PCO etc. and then advance to the point where you will mix under supervision. Once you show a proficiency in setup, mixing, and serving the band, you will be allowed to run a service unsupervised. Shadowing can start even before all training sessions are finished and this entire process may take a number of months.

While we try to have consistent setups, each COTR campus does differ in their setup. For example, the WSC is the "sending" campus so it has the most complex system including an entire broadcast mix position. Mobile campuses have equipment designed for quick setup and teardown and other campuses have their own unique setups.

Some of the key things that will be learned during shadowing are:

- Proper communication/relationship with worship team and pastors
    - This includes understanding key elements the worship leader wants in a mix like who is singing lead, and any instrument solos etc.
- Mixing monitors for musicians and singers. Concepts like:
    - Kick, snare, and hi-hat are needed in most monitors for timing
    - Bass often used as a pitch reference for singers
    - Keeping overall monitor level down as much as possible
    - How singers need to hear each other
    - Using iPad to set monitors
- Putting into practice all of the previous training concepts
- Using VSC (Virtual Soundcheck) if applicable.

Continued Learning

It should be evident to you by the time you've completed this class that live audio mixing requires both technical and creative skills. Learning the technical skills are actually the easiest part, but developing an "ear" for music, a rapport with bands, a musical sensibility, and a drive for innovation requires continual learning and experience.

Since most people don't have access to a live sound system to practice on, we are committed to helping those who want to serve in audio at COTR develop those skills. We will make the following available to you:

- Shadowing as previously discussed.
- Mixing for other COTR ministries such as our women's ministry, youth, young adults, at our Power & Praise services, workshop rooms at conferences etc.
- Virtual Soundcheck mixing practice. We will make a room available where we will setup a mixer (something most similar to the one with which you would normally serve with) and use pre-recorded audio tracks to practice mixing with.

We would also strongly encourage continued self-learning. There are many great resources.

- Re-reading this manual
- Shure Houses of Worship guide: http://cdn.shure.com/publication/upload/399/audio-systems-guide-for-houses-of-worship.pdf
- Church on the Move's SEEDS blog: https://seeds.churchonthemove.com/home
- Conversations with other Church Techs
- Conferences etc.
- Home Recording using free software like Garage Band, Audacity, Reaper, etc.
- Critical listening of well recorded music

Passing It Along

As you progress in your skills and experience we will then teach you how to pass along what you have learned to other people who are beginning to learn audio.

I hope that this training has been insightful and helpful. It is very rewarding being able to serve God, advance His Kingdom, and even be a part of helping people coming to know Christ and growing as they get to know Him better by using a skill we've developed and enjoy. Be richly blessed as you keep growing and serving! God bless.

Appendix A – Mic Placement

Placing microphones properly will make a huge improvement in the quality of your audio, and will aid in reducing feedback. It's also good to note that how you place a microphone for live sound differs from how you place it for studio recording.

Below are some guidelines and examples of good mic placement on various sources for live sound, but like all parts of doing sound, always use your ears to determine the best positions. They are not in alphabetical order, but in the order of most common use at Church of the Rock.

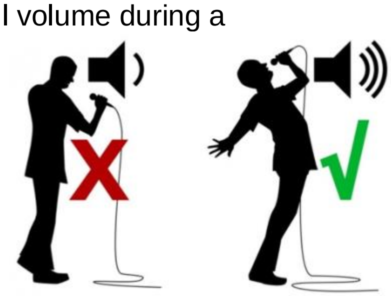
Hand-Held Vocals

The funny thing about placing a vocal mic is that it's not up to you. It is up to the singer to hold the mic in such a way that will produce the best results. So it is up to you and the worship leader to teach vocalists proper mic technique, and for you to know how to make adjustments for when the technique is off.

Hold the mic close, but not too close. If a mic is held too far away, the level would need to be pushed up a lot higher to be heard, greatly increasing the chance of feedback. If this happens the only thing that can be done to avoid feedback is pulling back your overall level so this vocalist can be heard, or selective EQ to increase gain before feedback.

If the mic is too close (eating the mike), this causes proximity effect which is a large buildup of low-end frequencies. The solution for this is to roll off low frequencies.

Singers save their voice for the services. Vocalists will never sing at full volume during a soundcheck. They have to pace themselves to ensure they can sing for the service (or often multiple services). So during soundcheck you will need to intentionally set their gain lower than your normal input level (so -18 dBFS instead of -12 for example) so that as they warm up during rehearsal and sing full out during the service you will not exceed your desired input level (or at least by not as much). Additionally, changing the gain after a rehearsal should be avoided if at all possible as they will impact the level going to every other part of the system (monitors, in-ears, broadcast mix, etc.).



Cupping the mike, or grabbing the antenna. Cupping a hand around the ball of mic actually changes it's pickup pattern, essentially making it omnidirectional. This in turn makes it much more prone to feedback. Additionally, on certain Sennheiser wireless microphones that have an external antenna, if a person's hand is wrapped around the antenna located at the bottom of the mic, the wireless strength is reduced and can cause static noise in the signal.

Walking in front of the main speakers. One of the most important ways that feedback is reduced is by having the main speakers located in front of the stage and the mic's behind them. If a mic is brought in front of the main speakers there is a significantly greater chance of feedback.

Lavalier and Wrap Microphones

Lavalier (lav) and wrap (headworn) microphones allow a person (usually a pastor) to keep their hands free while delivering a message. It is important to know how they need to be positioned to achieve best possible sound quality, comfort, and appearance.

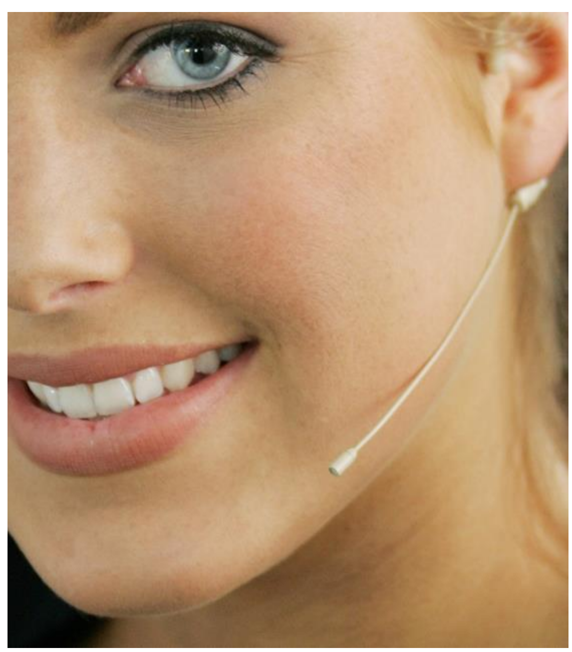
Wrap (headworn) Microphones

Wrap mics are called that because they wrap around one or both ears, and have a short boom to place the microphone capsule quite close to the mouth of the person speaking. As was mentioned previously, getting microphones closer to the source is one major way to reduce feedback.

The specific wrap mics we use at Church of the Rock are either the Que Audio DA12 or Mogan Standard. Both microphones wrap around a single ear with a rubber earpiece. Both have an omnidirectional pickup pattern, and use the same type of connector to our wireless transmitters.

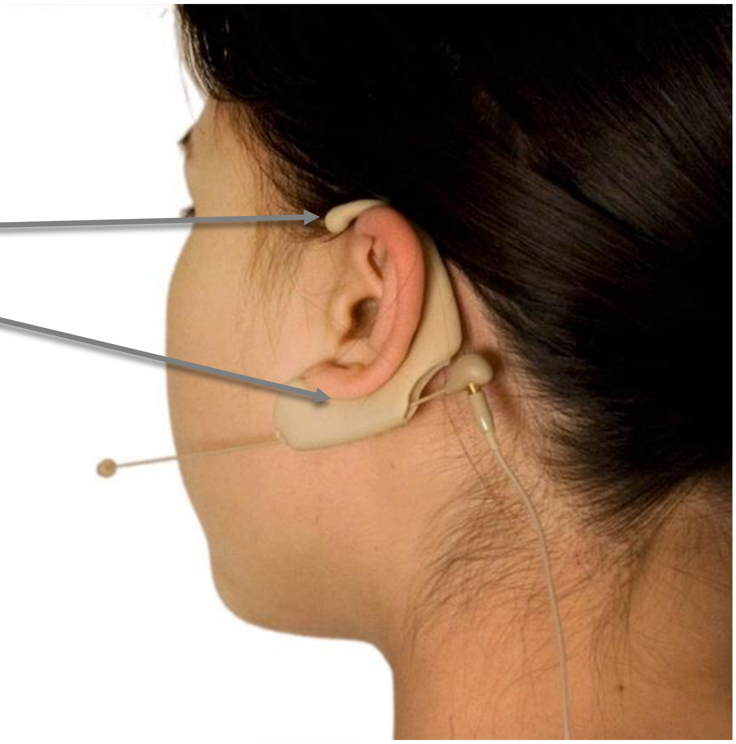
These mics are very difficult for a person to place on themselves accurately so it is usually the audio tech that will do this for the pastor. So it is important to be aware of how to place it correctly.

The boom of the microphone should be positioned to come from under the ear, close to the face without the capsule actually touching it (the boom can touch the face as long as the capsule doesn't), and below the mouth instead of in line with it. If a man has a beard, you will have to ensure capsule will not rub up against the bristles. This positioning helps to avoid the mic rubbing against the face, but also having it below the mouth reduces "essy"ness and plosives.



To achieve this positioning, bend the boom arm to angle it correctly. I suggest pre-bending it before putting it on the person, and then fine-tuning it once it is on.

How it is positioned around the person's ear is very important as well. It is often done incorrectly, and causes it to be uncomfortable for the speaker.



The top end of the earpiece should extend out in front of the ear, and the bottom end should sit behind the earlobe. It should not go all the way around the front of the earlobe.

The woman pictured here has smaller ears, so the earpiece goes underneath her ear, but not all way around front, and for this picture they were focusing on the earpiece and didn't bend the boom at all so ignore the boom's positioning.

Lavalier (Lav, Lapel) Microphones

Lav mics clip on to the lapel of a person, the cable usually goes through their clothing, and attaches to belt-pack transmitter which is usually clipped on their belt, or pants. Most of our pastors (and most public speakers in general) know how to put these on themselves, but as an audio tech it's still very important to know how it should be positioned.

It's also important to note that Lav mics are omnidirectional. This is helpful for 3 reasons: it reduces the drop in level that happens when a pastor turns their head when speaking, omnidirectional mics are less prone to wind noise, and since these are put on by the person speaking themselves, if they put them on upside down the microphone would still pick up their voice well.

They should be positioned a bit more than a fist's length below the chin, which is just beneath the collar of a shirt. It should be positioned as close the centre as possible to reduce the level change of turning the head from one side to the other.

I also recommend making a "Broadcast Loop" to improve the appearance of the Lav. To do this, insert your mic into its clip, then loop the cable back up into it as well. When the lav is attached to the pastor's clothing, he will run the excess cable behind the fabric so all you see from the front is the microphone itself, the clip and a small cable loop.

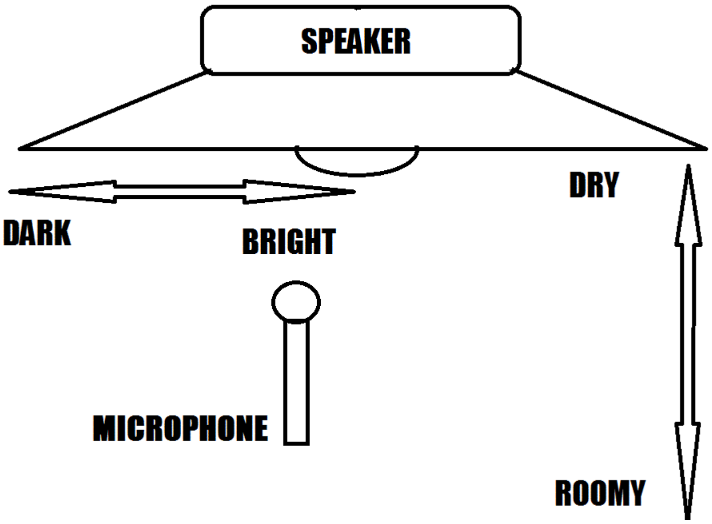


Guitar Amplifiers

In recent years the quality of processing for electric guitars has increased significantly, to the point where guitars being played through amp simulating software or hardware units are very close to sounding as good as through an actual guitar amp. For this reason, many of our guitarists have begun using amp simulating pedals, and this is now what we supply to any new campus for their electric guitarists. The main advantage of using this system is the reduction of stage volume.

However there are some guitarists and campuses that do not use these pedals. Additionally, a well setup amp, miked well still sounds a bit better than these pedals, so it's important to know how to mic them well.

Miking a guitar amp well is all about mic placement. Distance from speaker, and placement on speaker need to be considered.



Distance. Positioning the mic closer to the speaker will give you a "dry" sound meaning it won't pick up much of the room ambience (or other instruments on stage) and it will be less prone to feedback. However it can have a build-up of low frequencies due to proximity effect. The Shure SM57 which we use for guitar amp miking has been tuned to reduce this effect. Having the mic further away will pick up more of how the amp is reverberating in the room. For recording this is useful as they are in isolated rooms, and feedback isn't an issue. But for live sound close-miking is always recommended. At Church of the Rock we recommend having it about a finger's width in front of the mesh in front of the speaker.

Placement. Guitar amp speakers do not send out their full range of frequencies evenly across the speaker. Generally the centre of a speaker contains more of the mid and high-mid frequencies and less of the low and low-mid frequencies, while the edge of the speaker contains less mids and high-mids and more low and low mids.

So we have found that if you want to use a single mic, a very good place is at the cap edge; the place where the smaller circular cap of the speaker meets the cone. This seems to have a very good balance of body and clarity.

Most guitar amps have a mesh in front of the speaker, so in order to position the mic accurately, a flashlight will need to be shone into the amp to see through the mesh.

On or Off Axis. On axis refers to having the mic facing the speaker straight on, and off axis means intentionally placing the mic at an angle, sometimes roughly matching the angle of the cone of the speaker, pointing towards the centre of the speaker. Off axis miking seems to subtly reduce the high end pickup. We would recommend always starting with miking on axis as it seems to work consistently well, but if the signal is too sharp sounding, but already has enough body so it wouldn't need to move the mic closer to the edge, then trying off axis might be appropriate.

Multiple Microphones. Because there are differing tones that can be achieved by placing the mic in different locations with respect to the cone, one technique to exploit this is to use 2 (or more) mics on the same amp.

The technique we use at the WSC of COTR is to place one mic very near the centre of the speaker, and another very near the edge. This results in us having one channel that's very clear and one with lots of body. These seems to fill out the sound very well, as well as allowing the sound tech to change the tone of the guitar simply by changing the relative positions of the 2 faders.

In recording, this technique can be used, but they may additionally place another microphone relatively far from the amp to pick up the room ambience.

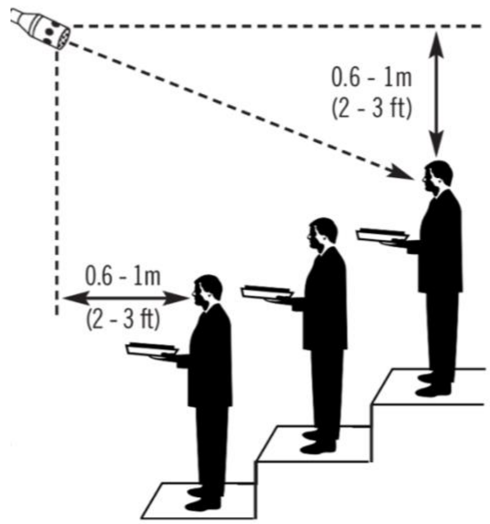
For any multiple miking scenario, be sure to follow the 3 to 1 rule discussed is session 2.

Choir

The main thing that makes a choir sound good is the blending of all the individual voices into a single powerful sound. Proper miking technique is essential to capture this.

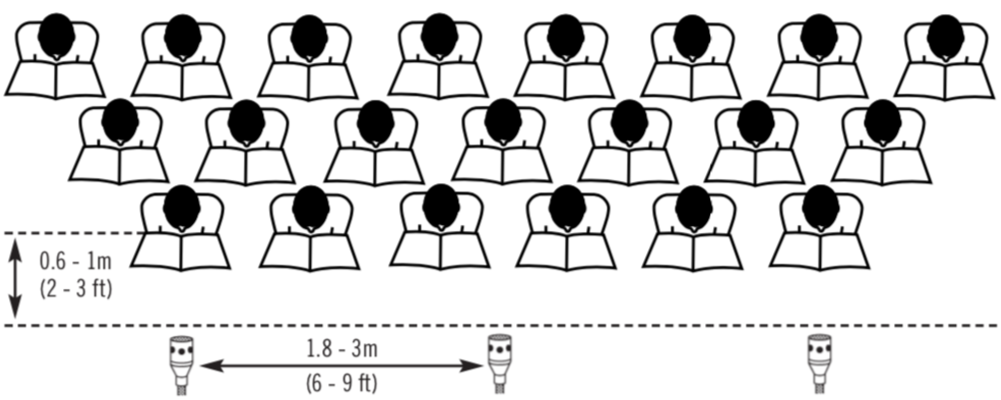
For this reason choirs need to be miked evenly and from a distance. This distance will cause more stage noise to be picked up, and make it more prone to feedback. However miking a choir too closely will result in it not sounding like a choir, but just a few individual voices. Oftentimes a compromise needs to be made.

The ideal placement for a choir mic is a few feet in front of, and a few feet above the heads of the first row, with the mic pointed at the mouth of people in the back row. This will have the most even pickup of the choir due to the rounded shape of a the cardioid pickup pattern; the person furthest away is the most on axis, while the person closest is the most off-axis.



Microphone positions - side view

A compromise would be to move the mic(s) a bit closer and lower. You will have improved gain before feedback and should still capture a well-blended choir.



Choir microphone positions - top view

Both the 3 to 1 rule, and less is more (from Session 2) apply here. Use as few mics as possible to pick up the choir and make sure they are spaced at least 3 times as far from each other as the source.

Mic selection is important. A condenser mic with a flatter, and fuller range response will sound better than something like a Shure SM57 with its tailored response. Something like the AKG C1000 is a good choice, or the DPA choir miking system is very good.

Saxophone

When miking a saxophone, you want to position the mic between the bell and the keys. This is better than simply aiming a mic down the bell because the keys also produce an important part of the tone.

A clip-on mic is always the preference as it moves with the saxophone and therefore remains constant in level.

A mic on a stand will work well as long as the person doesn't move very much.

Acoustic Guitar

The vast majority of the time, an acoustic guitar will have a pickup. For live sound plugging the acoustic into a DI box or wireless transmitter is the best option for simplicity and the ability for the guitar player to move around.

But when an acoustic guitar doesn't have a pickup, a mic will need to be used. Miked well, an acoustic will actually sound better this way, but the guitar player is not stuck to that position.

Mic selection is important. A condenser mic with a flatter and fuller range response will sound better than something like a Shure SM57 with it's tailored response. Something like the AKG C1000 that we often use for choir is a good choice.

There are actually a large number of ways to mic an acoustic. The method with the most consistently good results is to place the mic about 6" away from front of acoustic, positioned near where the neck and body of the acoustic meet, angled slightly toward sound hole. If more body is needed, move the mic slightly closer to sound hole. If you need more clarity, move the mic further away.

Cajon

Miking a cajon is fairly difficult because of the way the person sits on it. For best results use 2 microphones. Something like an SM57 for the front position to capture where the hands hit the top of the box. This essentially is your "snare" sound. Then use either a kick drum mic, or a full-range condenser (like the C1000) in the back, behind the sound hole, not too close and a bit to the side. It becomes excessively boomy otherwise. This is essentially your "kick drum" sound.

Other Instruments

While this appendix covers a broad range of applications, there may be a different instrument that requires miking from time to time. To determine mic positioning do the following: Get the musician to play their instrument and ask the rest of the band to not play. Determine how the instrument should sound, and where most of its sound is coming from. Place the microphone at that position.

Consideration has to be made for both the musician's comfort with the mic and mic stand and the visual appearance. Normally a full-range condenser will sound better on most acoustic instruments. One exception can be percussion instruments with a loud volume. In these cases an SM57 usually works very well.

1397 Buffalo Place
Winnipeg MB R3T 1L6
204.261.0070
CRAVE@churchoftherock.ca