First Baptist Hammond

Audio Engineering Handbook

First Edition

Contents

PART 1: SOUND 101

1. Introduction

1.1. Forward

At First Baptist Hammond, we believe that excellence honors God and inspires people. We want to give our very best to God so as to honor Him, because He is worthy of our best. This training manual is designed to serve the members of the production team and enable them to do just that: give God our very best and strive for excellence in our work for Him.

Sound engineering is a vast and complex topic, and it would be unreasonable to expect volunteers to invest the time and effort to become truly experts at it. However, we believe that devoting ourselves in service and committing to becoming better at our ministry is a worthy cause, and we believe God honors that devotion and commitment. We do not strive for perfection, only progress.

That being said, this manual contains an enormous amount of information. Do not be discouraged or overwhelmed by the volume of material presented in this text. When learning and perfecting this craft, it is not expected that you should master it quickly. A more recommended approach would be to view it with a long-term outlook and consume the material one piece at a time. Developing the skills necessary to be a successful audio engineer will take time – *a lot* of time. And it is a process that cannot be rushed. Embrace the learning experience and remember that we do this as a ministry – in service to our Lord.

In case no one has told you this before, engineering audio is an art form. This is, by any definition, an *expressive* ministry – just as creative as writing a compelling sermon and just as artistic as playing music. This is a technical position, to be sure, but without embracing the creative aspects of it, one will never be able to excel. The mix you create behind the sound board is very much an artistic product, and you should know that God values and honors your creative expression. It is our hope that this manual will serve as a guidebook to help cultivate and channel that creativity in a productive way.

This text will be broken into several volumes, each focused on a unique aspect of sound engineering. We begin with an introduction which covers the basics of sound generally and the sound system, then move onto concepts in mixing, operation of the sound board, and techniques to make your mix as good as possible. You are encouraged to digest these concepts at your own pace: quality is much preferred over quantity when it comes to studying these ideas.

"Let us not get tired of doing good, for we will reap at the proper time if we don't give up."
Galatians 6:9 (CSB)

1.2. Goals and Responsibilities

We begin with a discussion about the sound system itself. This is not directed at the individual, but rather the physical system itself. Understanding the goals of the sound system will help the engineer better understand their role in facilitating these goals being achieved. These goals are for sound systems generally, and not specific to systems in a worship environment, so the wording may be a bit generalized.

Goals for Sound Systems:

- Go unnoticed – a perfect sound system should not be distracting to the audience
- Allow the performer(s) or speaker(s) to feel comfortable – a sound system should not intimidate the people on stage
- Sound as natural as possible
- Help the audience hear portions of the speech or music which are naturally quiet and would otherwise go unnoticed
- Produce an even volume throughout the room
- Make sound louder to aid in the artistic expression of performers
- Allow a speaker to communicate with a large group of people without having to raise their voice
- Have every sound source reproduced at the volume which is appropriate for that sound

To help the sound system do its job and achieve these goals, the following responsibilities are given to the audio engineer. Notice that most of these responsibilities are spiritual items, not technical ones.

Responsibilities of the Audio Engineer:

- Get any training that is required to effectively and confidentially operate the equipment used
- Be, first and foremost, a servant. The audio engineer is there to serve, not be served
- Be humble and do the job as a service to the Lord
- Pray over the service and your role in it anytime you are serving
- Pray after the service, giving thanks and glory to God for his provision and for the privilege of serving Him
- Support and serve the worship team and pastor
- Serve the congregation through your work on the sound desk
- Monitor for issues in the sound system and address them as necessary
- Be prepared for the service – mix proactively, not reactively
- Give guidance to the worship team when necessary on the proper use of sound equipment (e.g. proper microphone placement)
- Set mix levels, monitor for tonal quality, and adjust monitor mixes as necessary

- Pay attention during the worship service and actively mix the sound in the room and on the live stream
- Take excellent care of equipment and supplies – treat it all like it belongs to God (because it does)

2. Sound and the Sound System

2.1. Sound System Components

We'll now spend some time getting familiar with the anatomy of a sound system. Don't worry about memorizing everything right now; we'll be spending time going over each of these items in more detail when we discuss the signal chain in ***Chapter 3***. More important, at this time, is that you begin to recognize the different pieces of the "puzzle" that is a sound system.

Generally speaking, a sound system can be broken into three major sections:

1. Capture and Transmission

Before you can send a sound through a speaker system, you must first capture it and transmit it to be processed. We use various devices and tools to help us make this capture and send the sound further along in the signal chain. Some examples of capture and transmission devices are pickups, microphones, playback devices, direct input boxes, adapters, ¼" and 1/8" TRS cables, XLR cables, category 5/6 cables, and cable snakes.

2. Processing

Once a sound is captured, it is transmitted for processing. This is the most critical part of a sound system and, depending on the setup and needs of the system, can range from being very simple to extremely complex. Some of the equipment and tools used during this section include the mixing console ("mixer" or "sound board"), internal and external signal processors, internal and external effects processors, and output processors such as normalizers and limiters.

3. Output

After the sound has been processed, it is transmitted to an output chain for broadcast. This section of the sound system is where we find elements such as audio matrices, amplifiers, speakers, monitors, and line outputs.

2.2. How Sound Works and Why You Should Care

It's important, at this point, to understand *what* it is you are actually managing as an audio engineer. Too often, people make the mistake of thinking that a mix is the result of some combination of arbitrary, intangible noises. This couldn't be further from the truth. While it is indeed the case that, with modern technology, we take advantage of tools that allow us to transform an acoustic sound signal into a digital one, the end result is always converted back into acoustic sound. Whether being played through PA speakers in an auditorium, through headphones from a live stream or video, or through a car stereo from a CD or Bluetooth device, an acoustic audio signal is always the end product.

With that in mind, we arrive at a critical truth: ***to be a good sound engineer, you must understand sound***. Not sound in the abstract or used as a shorthand to describe the mix, but rather the physical properties of sound and how it interacts with the world. You would be doing yourself a great favor by spending a good amount of time reviewing this section; a thorough grasp on how sound works will be a tremendous advantage to you as you learn more about operating a sound system. Things like signal processing and Sonic Real Estate (covered later) make much more intuitive sense when you understand how sound behaves.

Sound Waves

Sound waves are mechanical vibrations that travel through a medium. Typically, we think of sound in the air, but it can also travel well through liquids and solid materials. These waves are created when any object vibrates, causing the surrounding molecules to bounce back and forth. This oscillation generates pressure changes that expand throughout the material.

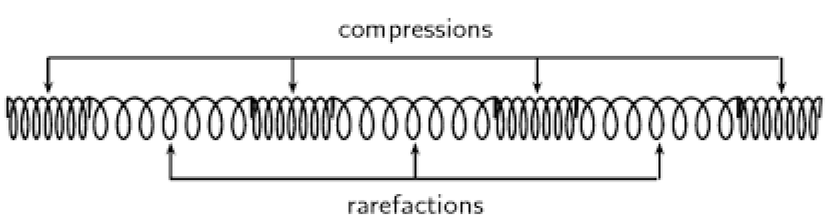
Sound waves are often compared to ripples in a pond, which radiate outward from the disturbance in the center. However, unlike ripples in a flat pond surface, sound waves will travel in all directions. Water is often used to describe the motion of sound and it may be helpful for you to think about sound like water for the purposes of developing intuition. When thinking about how sound is interacting in a space, consider how water would flow through the same space. While not perfect, this is a great analogy and can help you better visualize how the sound waves are moving.

Sound is a mechanical energy transfer, which means it is made from the energy of movement. We use our muscles to push air, vibrate our vocal cords, and make sound when we speak. This sound travels through the air and eventually vibrates the listener’s eardrum.

At the end of the journey, this vibration is translated into electrical nerve impulses to be decoded by the listener’s brain. If we replace the eardrum with a microphone, we can create an alternating-voltage electrical signal that we call *audio* – the primary topic of concern for the audio engineer.

How Sound Travels

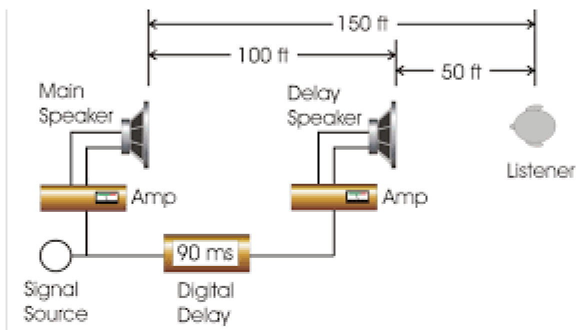
When sound is produced, the vibrating object creates compressions and rarefactions in the surrounding medium (primarily air). Compressions are areas where particles are closely packed together, while rarefactions are areas where particles are spread apart. As the sound wave moves through the medium, these areas of compression and rarefaction travel outward from the source.



Think of how a guitar string will vibrate when picked. The string will first slam into air particles nearby, forcing them to bump into the next molecules in line. All this commotion causes the nearby air to be compressed into a high-pressure zone. After the string reaches as far as it can in one direction, it will swing backward and pull the air back with it, creating an opposing low-pressure zone. If the string is under enough tension, it will swing back and forth for several seconds, sending a continuous series of alternating high and low-pressure waves across the room.

The speed of sound depends on several factors, including the medium and its temperature. For instance, sound travels faster in water than in air and even quicker in solids. At room temperature, sound travels at about 1,125 feet per second (343 meters per second) in air, but in water, this speed can exceed 4,869 feet per second (1,480 meters per second). Since there are 1,000 milliseconds in one second, as a rule of thumb, you can estimate sound will travel 1ft per ms.

This number becomes valuable to audio engineers who must precisely align (or re-align) sound waves for a listener. Often, large music shows will have multiple locations for their speaker systems, each at a unique distance from the audience. To make them work well together, these systems will be time-delayed to reinforce the original wavefront traveling from the stage to the back of the venue.



Wave Properties

Sound waves possess several fundamental properties that define their characteristics. Let's look at a few of the important terms.

Frequency

Frequency refers to the number of cycles a sound wave completes in one second, measured in hertz (Hz). The frequency determines the pitch of the sound; higher frequencies correspond to higher pitches, while lower frequencies yield lower pitches. The audible range for humans typically spans about 20 Hz to 20,000 Hz (20 kHz).

Frequency is a critical aspect of sound waves that influences our pitch perception. Audio engineers often work with different frequencies to achieve desired sounds in music and other audio recordings. Listeners perceive low frequencies at the bottom of the mix, nearer the floor, while high frequencies naturally lift towards the ceiling. Most people perceive 1000 Hz as roughly at eye level – which is why, you will discover, the 500-1500 Hz range is often perceived to be the most congested within the frequency spectrum.

Here is a (very) basic overview of the frequency ranges within the typical audible range of humans:

Sub-Bass (20-60 Hz)

The low rumbling sounds are often felt rather than heard. They are common in genres like electronic and hip-hop music since there is not much sub-bass in typical acoustic instruments.

Bass (60-250 Hz)

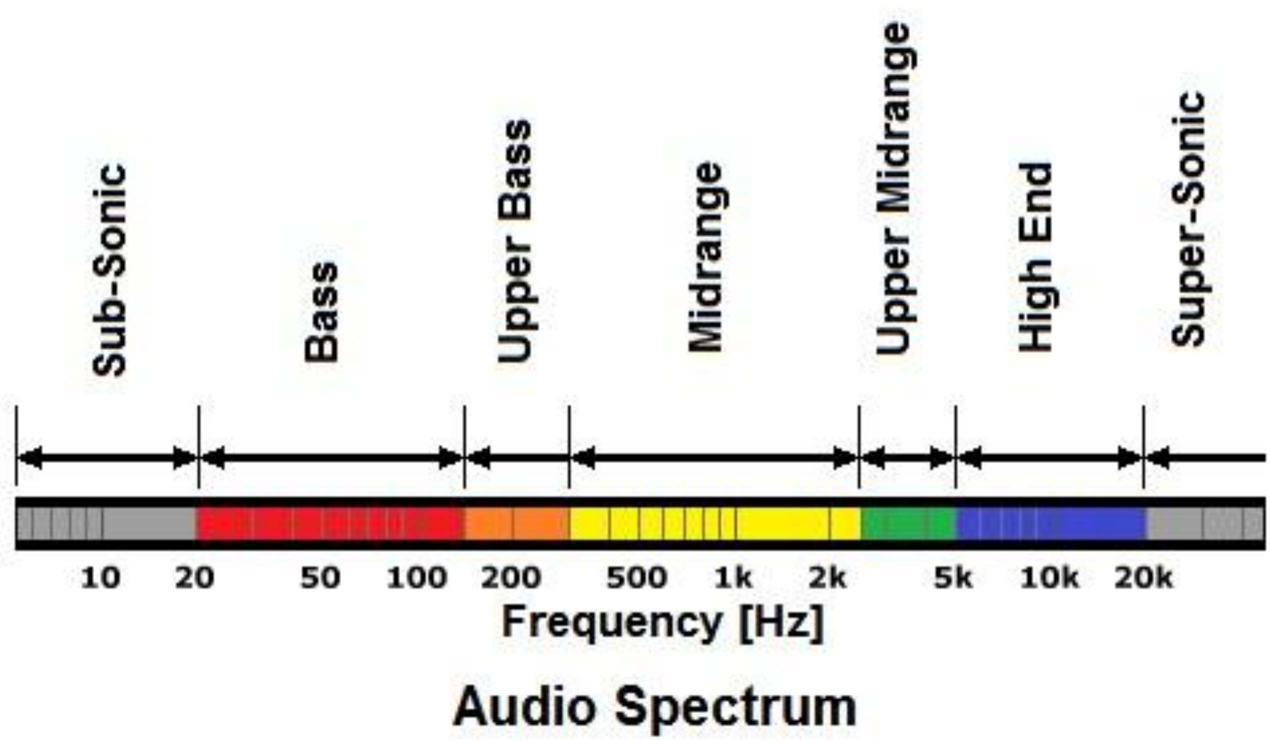
Fundamental for the rhythm and feel of music, providing a foundation for harmony.

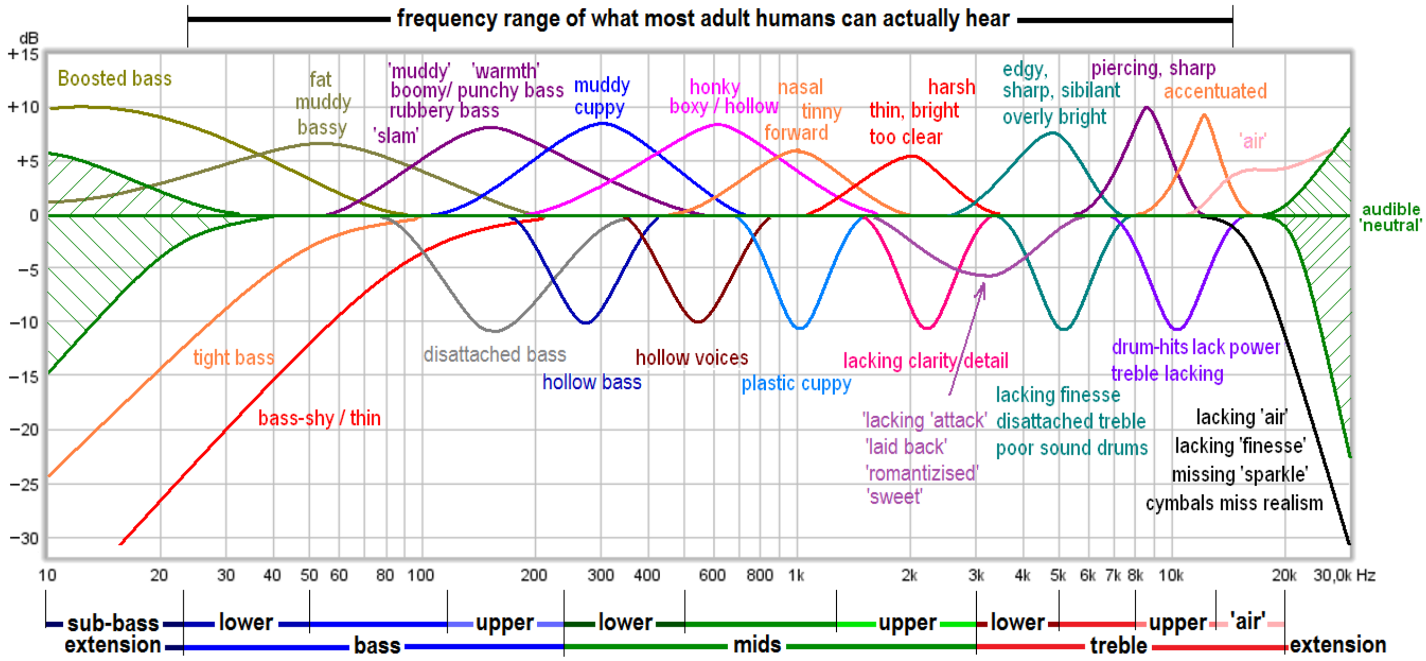
Midrange (250-5,000 Hz)

It contains most vocal and instrumental elements, where clarity is essential.

Treble (5,000-20,000 Hz)

High-frequency sounds add brightness and clarity to music but can also cause ear fatigue if overemphasized.




frequency range of what most adult humans can actually hear
dB
+15
+10
+5
0
−5
−10
−15
−20
−25
−30
Boosted bass
fat
muddy
bassy
'muddy' 'warmth'
boomy/ punchy bass
rubbery bass
'slam'
muddy
cuppy
honky
boxy / hollow
nasal
tinny
forward
harsh
thin, bright
too clear
edgy,
sharp, sibilant
overly bright
piercing, sharp
accentuated
'air'
audible
'neutral'
tight bass
bass-shy / thin
disattached bass
hollow bass
hollow voices
plastic cuppy
lacking clarity/detail
'lacking 'attack'
'laid back'
'romantizised'
'sweet'
lacking finesse
disattached treble
poor sound drums
drum-hits lack power
treble lacking
lacking 'air'
lacking 'finesse'
missing 'sparkle'
cymbals miss realism
10 20 30 40 50 60 80 100 200 300 400 600 800 1k 2k 3k 4k 5k 6k 7k 8k 10k 20k 30,0k Hz
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'air'
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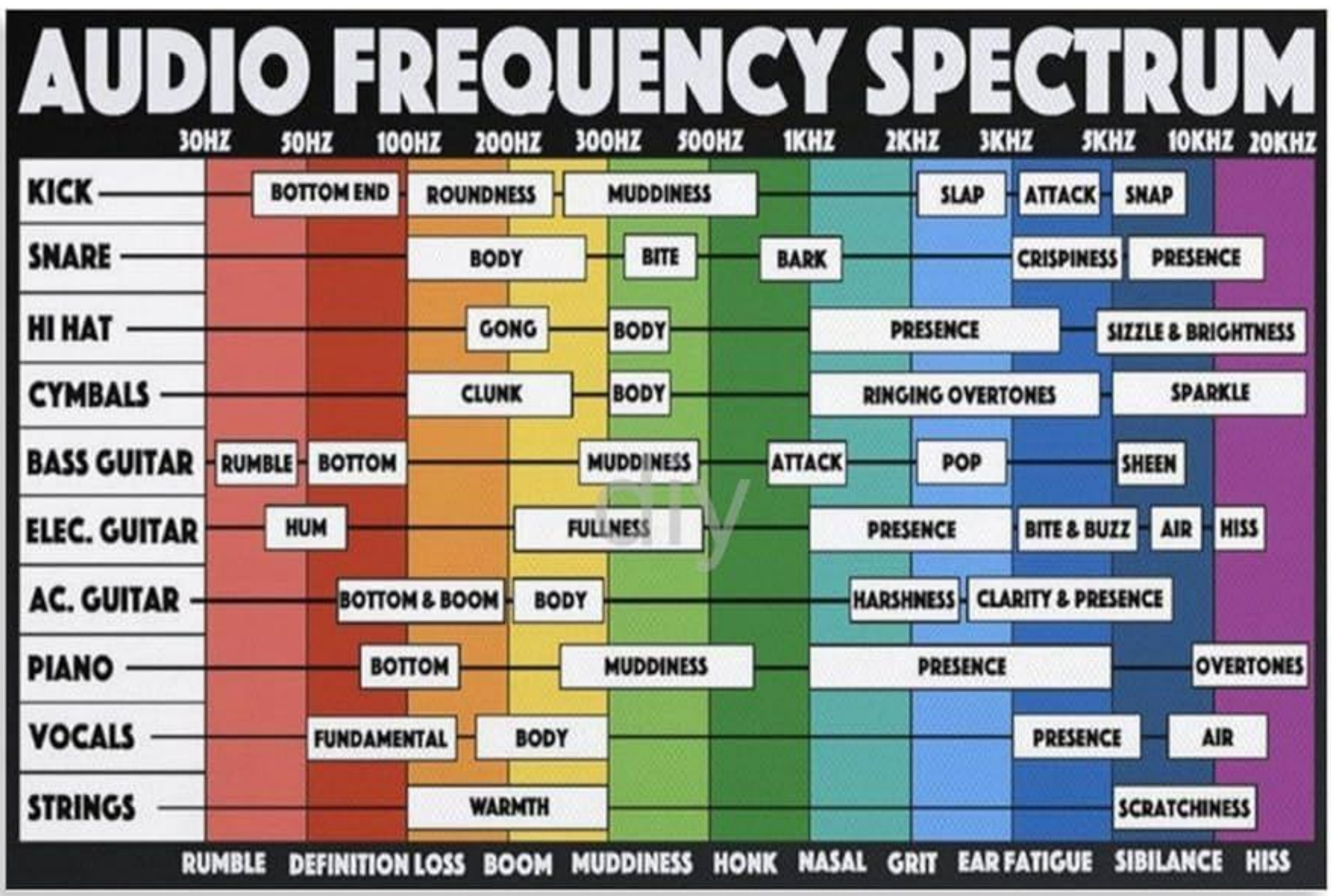
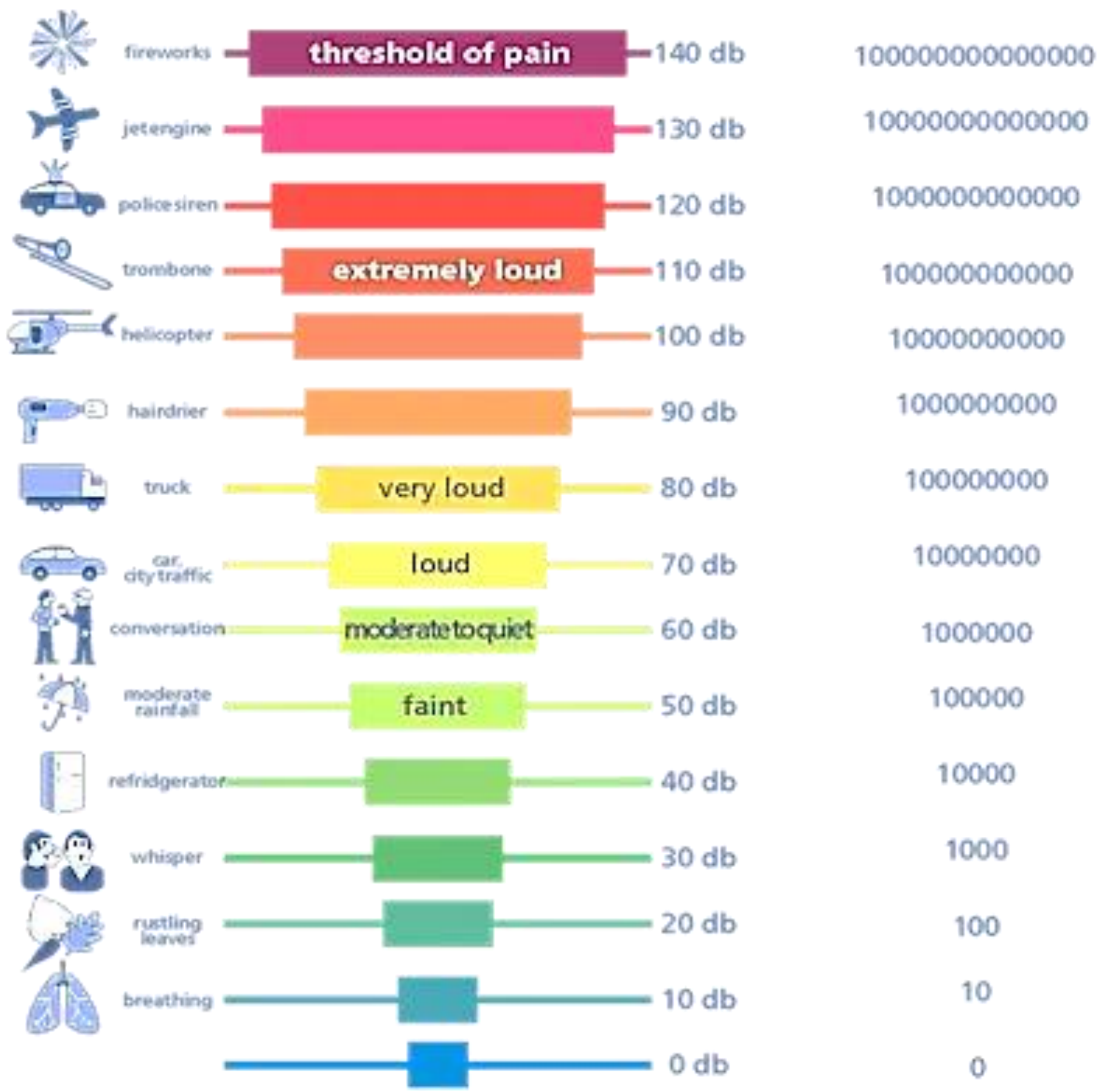

AUDIO FREQUENCY SPECTRUM
30HZ 50HZ 100HZ 200HZ 300HZ 500HZ 1KHZ 2KHZ 3KHZ 5KHZ 10KHZ 20KHZ
KICK
BOTTOM END
ROUNDNESS
MUDDINESS
SLAP
ATTACK
SNAP
SNARE
BODY
BITE
BARK
CRISPINESS
PRESENCE
HI HAT
GONG
BODY
PRESENCE
SIZZLE & BRIGHTNESS
CYMBALS
CLUNK
BODY
RINGING OVERTONES
SPARKLE
BASS GUITAR
RUMBLE
BOTTOM
MUDDINESS
ATTACK
POP
SHEEN
ELEC. GUITAR
HUM
FULLNESS
PRESENCE
BITE & BUZZ
AIR
HISS
AC. GUITAR
BOTTOM & BOOM
BODY
HARSHNESS
CLARITY & PRESENCE
PIANO
BOTTOM
MUDDINESS
PRESENCE
OVERTONES
VOCALS
FUNDAMENTAL
BODY
PRESENCE
AIR
STRINGS
WARMTH
SCRATCHINESS
RUMBLE DEFINITION LOSS BOOM MUDDINESS HONK NASAL GRIT EAR FATIGUE SIBILANCE HISS

Amplitude

Amplitude measures the maximum extent of a vibration or oscillation, indicating how loud a sound is. Higher amplitudes correspond to louder sounds, while lower amplitudes produce softer sounds. Amplitude is usually measured in decibels (dB), a logarithmic scale that quantifies sound intensity.

Understanding amplitude is crucial for audio engineers to balance sounds in a mix. The louder sounds will tend to be perceived as closer to the listener, and quieter sounds will fall away from the listener.

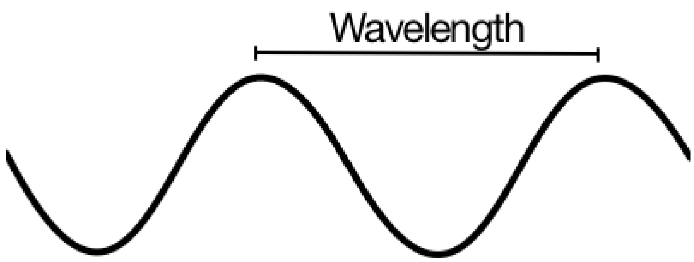
Below you'll find a description of some amplitude ranges to give you an idea of how loudness is measured.



Since it's logarithmic, dB changes might seem unusual at first. The rule of thumb is that a 6dB change equals double (or half) the original volume. So, doubling the volume of a 100 dB concert would bring it to 106 dB, not 200 dB as you might assume. The logarithmic properties of amplitude will be important later when we discuss faders on the mixing console.

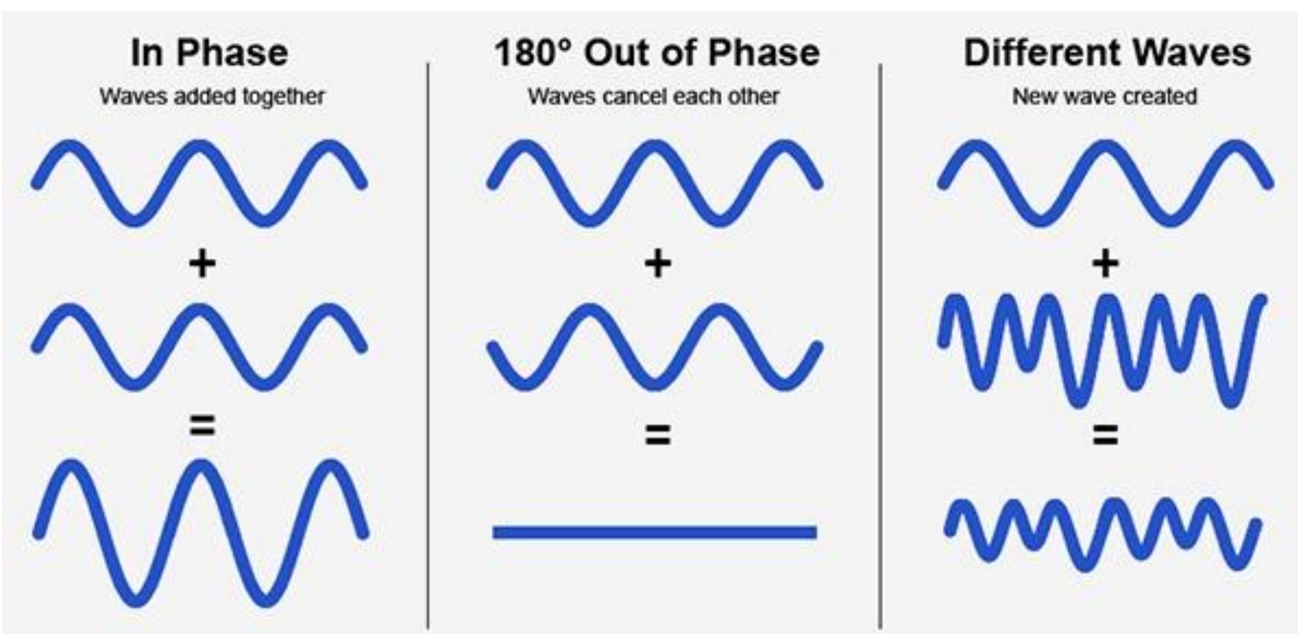
Wavelength

Wavelength is the distance between successive crests (or troughs). Wavelength is inversely related to frequency; higher frequencies have shorter wavelengths, while lower frequencies have longer wavelengths. This becomes very important when designing acoustics in studios or even building instruments to specific sizes to resonate in tune with the music.



Phase

Phase describes the position of a point in time on a waveform cycle, measured in degrees. Phase is a crucial factor in audio engineering, especially when dealing with multiple sound sources or microphones.



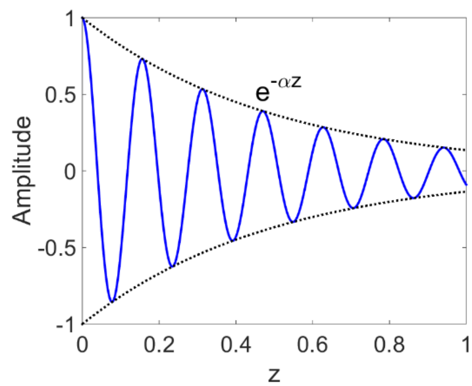
How Sound Waves Change over Time and Distance

As sound waves travel, they undergo various changes that can affect their quality and perception. Several factors contribute to how sound waves alter over time and distance, and audio engineers use these changes as part of the mixing process, enabling them to create the illusion of depth and space in a mix – a subject we'll be exploring more deeply when we discuss *Sonic Real Estate* in later volumes.

Attenuation

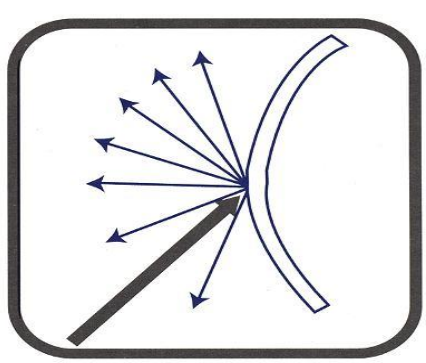
One of the primary factors affecting the propagation is *attenuation*, which refers to the reduction in amplitude (volume) as sound travels. Most attenuation is the result of either wasted energy (spread across large areas) or absorbed energy.

This happens partially because some of the sound is spread out in every direction and isn't focused on any particular listener. Low frequencies are omnidirectional (travel in every direction equally), and higher frequencies tend to be more directional. Another cause of volume drop over distance is due to energy loss as it pushes itself through the air. High frequencies are particularly fast and low power, so they tend to get absorbed by the air faster than low frequencies.



Diffusion

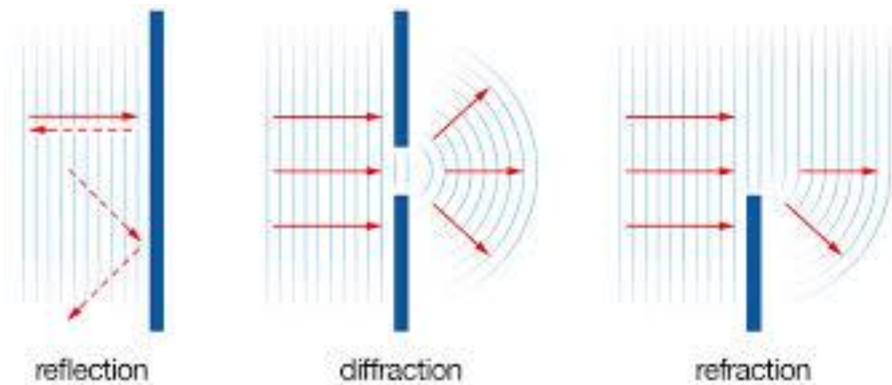
When sound waves encounter obstacles, they can scatter in different directions. This phenomenon, called *diffusion*, can lead to a loss of energy and intensity.



Refraction and Reflection

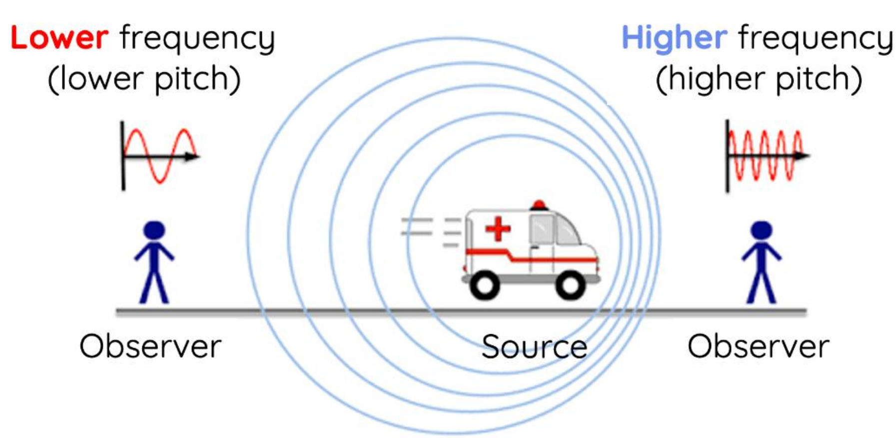
Sound waves can also change direction due to *refraction* and *reflection*. Refraction occurs when sound waves travel through media of different densities, causing them to change speed and direction. For example, sound travels faster in warmer air than in cooler air, which can cause the waves to bend.

Reflection happens when sound waves bounce off surfaces. An echo occurs when the sound waves reflect off a distant surface and return to us. When the reflection is nearby, it creates a *first reflection* echo, which helps your brain echo-locate in space. In fact, if you walk into a new room blindfolded, these early reflections will give you a surprisingly accurate interpretation of your environment. Understanding these phenomena is crucial for audio engineers, especially when designing spaces for optimal sound quality.



Doppler Effect

Another fascinating phenomenon related to sound waves is the *doppler effect*. This occurs when there is a relative motion between a sound source and an observer. If the sound source moves towards the observer, the frequency appears higher (resulting in a higher pitch), and if it moves away, the frequency appears lower (resulting in a lower pitch). This effect is commonly experienced in everyday life, such as when an ambulance passes by with its sirens blaring.



Applications in Audio Engineering (Why You Should Care)

Understanding sound waves is vital for audio engineers, as it directly impacts their work. Here are a few practical applications – we'll be discussing most of these in detail later on.

Microphone Placement

Knowledge of sound wave propagation helps engineers determine optimal microphone placement to capture the desired sound while minimizing unwanted noise.

Room Acoustics

Understanding reflection, absorption, and diffusion aids in designing recording studios or performance spaces that enhance sound quality. In your role, being familiar with these ideas can help you anticipate and troubleshoot issues with your mix.

Equalization

By adjusting frequencies, engineers can shape sounds to achieve a balanced mix, highlighting some aspects while suppressing others.

Conclusion

Understanding sound waves is fundamental to audio engineering and enriches our appreciation of sound in all its forms. By grasping concepts like frequency, amplitude, and how sound changes over time and distance, we can better appreciate the complexities behind the sounds we experience daily.

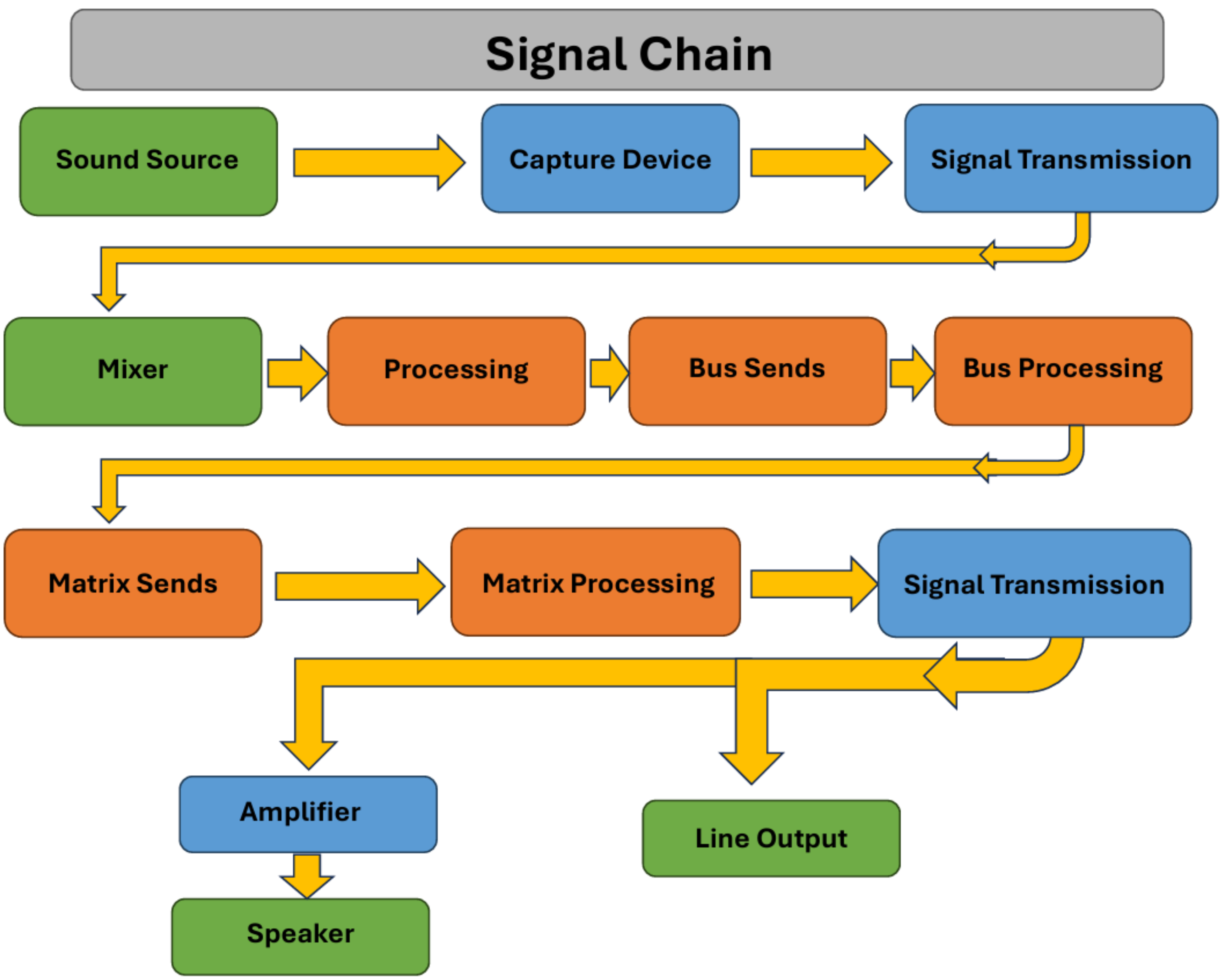
Possessing a good understanding of these elements will be priceless as we dive further into the technicalities of sound production and will give you a good intuition when exploring topics such as signal processing, effects, and balancing your mix.

3. The Signal Chain

3.1. Overview

The *signal chain* is the complete path that sound travels from source to listener and involves a series of interconnected hardware and software components that capture, transmit, shape, and broadcast the sound.

The role of the audio engineer is to carefully order and process these various stages in order to generate the desired quality, tone, and consistency. Understanding signal flow is crucial for troubleshooting and achieving professional results, regardless of the application. Below is a basic flow chart describing the path that a sound signal takes through the signal chain. In the following sections, we will explore each of these stages in a bit more detail.



The signal begins with the source of the sound. Capture techniques are then used to turn the sound into an audio signal. This signal is then transmitted to the mixer where it is processed before being sent to the appropriate output.

A common misconception is that a high-quality sound comes from the processing done at the mixing desk. Of course, accurate mixing is crucial in making the overall mix sound pleasing. And it is certainly true that effective signal processing can dramatically improve certain properties of a sound, such as tone, clarity, and where it sits in a mix. **However, the *quality* of the sound is directly dependent on the quality of the source.** You can add all the compression, equalization ("EQ"), and effects ("FX") you want, but if a signal is of poor quality at the source, no amount of manipulation will make it sound good. Therefore, we start our discussion at the beginning of the signal chain: the source.

3.2. Source and Capture

Source vs. Capture

It is necessary to distinguish between the source of the sound itself and the source of the audio signal. Therefore, we will refer to them as the *source* and *capture*, respectively. It is safe to assume that, generally, we are referring to the captured audio when discussing topics in this chapter and beyond. However, it is important to remember that when we discuss the *source*, we are referring to the actual sound being captured and not the audio signal generated by the capture.

As an example, consider a vocalist singing into a microphone. The physical sound coming from the vocalist's mouth would be the *source* whereas the audio signal transmitted by the microphone would be the *captured audio*, or, simply, the *capture*.

Types of Sources

We can classify sound sources into two main categories: electrical sound sources and acoustic sound sources.

Electrical sound sources are those where the sound originates as an electronic wave generated by some piece of electrical equipment. Examples of electrical sources include recorded sound from playback devices such as computers as well as electronic instruments like keyboards and electronic drums.

Acoustic sound sources are those which naturally make sound that can be heard without being amplified through a speaker system. Examples of acoustic sources include vocals (both singing and speaking) and acoustic instruments such as drums, acoustic guitars, pianos, horns, and strings.

You may have noticed that we failed to include electric guitars as an example. Some instruments, such as electric guitars, can be classified either as electrical or acoustic depending on the capture method.

On their own, when captured directly from the guitar's output or from a pedal board, for example, electric guitars are treated as an electrical source (even though the guitar's pickups are technically capturing an acoustic sound). However, suppose the guitar is run through an amplifier and speaker cabinet before being captured. Notice that it is no longer the *guitar* that is the source – the speaker cabinet from the amplifier becomes the source and, therefore, this sound would be considered an acoustic source. This may seem like a nuanced difference, but it actually provides some insight into how the source affects the quality of the capture. Adding the additional variable of the amplifier changes the source and, therefore, adjustments must be made when capturing the sound.

Electronic Sound Sources

Capturing audio from an electronic sound source is usually very simple – most of the time it's as easy as "plugging it in." That is, electronic sound sources usually output an electronic signal suitable for direct connection to the sound system.

Playback Devices

Playback devices usually come in the form of a computer, iPad, cell phone, or CD/DVD player. Connecting these devices is as simple as using the appropriate cable *(Section 3.3)* to connect the device to the input channel transmitting to the mixer.

Stereo and Split-Track Signals

Often, we will use an RCA cable or splitter to separate the left and right audio signals into two separate captures. The benefit of this method is it allows us to process and broadcast the signal in stereo (more on mixing in stereo in later chapters), but it comes at the cost of taking up two input channels on the mixer. If mixer channels are scarce or if the speaker system is being run mono, both capture signals can be combined into one channel (either by using a single cable or using a Y-adapter to combine the inputs).

Some playback devices (such as the iPad we use for backing tracks) play what's referred to as *split-track* audio. This allows one playback device to send two separate recordings through one output using a stereo cable – one track is played through the left channel and the other through the right channel. In some cases, one channel will have instrumental tracks while the other has backing vocals. In the case of our playback iPad, one channel plays the actual backing track while the other channel plays the click track for the band members. When using a device with split tracks, it's important to pay careful attention that the appropriate tracks are going to the appropriate outputs – otherwise you may end up hearing the click track in the house speakers when it should only be playing in the musicians' ears.

Electronic Instruments

Electronic instruments such as keyboards and electric drums output an electronic signal. Electric guitars (when inputted directly, as discussed earlier) have a built-in microphone or magnetic "pickup" that outputs an electronic signal. All of these signals may be connected directly to the input channel transmitting to the mixer using the appropriate cable and, when necessary, a DI box *(Section 3.3)*.

Acoustic Sound Sources

Capturing audio from an acoustic source can be a bit trickier than electronic sources. Usually, a microphone is used to capture the acoustic sound and convert it into an electronic signal that can be sent to the mixing desk.

Vocals

To get a person's voice (whether sung or spoken) into a sound system, you need a microphone. There are many different types of microphones which can be used depending on the particular situation.

Wired/Wireless Handheld Microphones

Most microphones require a cable (usually XLR) to carry the captured audio from the microphone to the input channel. Frequently, this cable is supplemented with an FM transmitter and receiver that allows the microphone to be "wireless." The only cable required with wireless microphones is to connect the FM receiver to the input channel. Placement of the FM receiver (and its antenna(s)) is very important; it should be close enough to the microphone to receive a clear signal and not near any sources of electronic interference such as CD players, effects processors, keyboards, or computers. All such devices can interfere with the FM signal and degrade the quality of the captured audio. If multiple wireless microphones are being used, each mic/receiver combo must have its own dedicated FM frequency.

Unidirectional/Omnidirectional Microphones

Unidirectional (also called "directional" or "cardoid") microphones "hear" – or pick up – much better from the front than they do from the sides or rear. They're usually best for sound reinforcement to prevent feedback since they are less likely to pick up sound from speakers that aren't directly in front of them.

Conversely, *omnidirectional* microphones pick up sound from all directions easily. While omnidirectional mics can sometimes provide a more "natural" sound quality, they are prone to feeding back easily and, therefore, are not used as frequently in a live sound setting.

Lavaliere Microphones

A *lavaliere* (la-vuh-leer) microphone is a small microphone which usually has a clip to attach it to the performer or speaker's clothing, offering a "hands-free" operation with a high freedom of movement. They usually do not sound as natural as handheld mics and are not considered desirable for capturing vocals from a singer.

Depending on the clothing being worn, the location of a lavaliere mic is sometimes a problem. It should be located as high as possible and centered with the mic pointed upwards (towards the speaker's mouth). The best location is clipped on a tie, just below the knot. If the speaker is not wearing a tie or button-up shirt, it may be difficult to find a place to clip the mic. For aesthetic purposes, the mic cable is usually run underneath the speaker's shirt and the FM transmitter for the mic is clipped out of sight on the speaker's back, either on a belt or back pocket.

Lavaliere mics are notoriously difficult to mix, as they are severely prone to feedback.

Headset Microphone

A *headset* microphone provides the same "hands-free" operation of a lavaliere with superior sound quality and better feedback control. Like with the lavalier, the cable is usually run underneath the speaker's clothing and connected to the FM transmitter clipped behind the speaker's back.

Acoustic Instruments

Acoustic Guitars

If at all possible, audio from acoustic guitars should be captured using a guitar pickup. Many acoustics have built-in pickups with an onboard output that can plug straight into a channel input or DI box. Alternatively, external pickups can be used that cover the guitar's sound hole and transmit the captured audio much the same way. As a last resort (or if the guitar pickup simply doesn't sound good), a directional microphone on a boom mic stand can be used to capture the audio. The mic should be placed as close to the sound hole as possible, leaving only enough space such that the musician doesn't hit the mic while playing the guitar. Occasionally, a microphone will be used intentionally instead of a pickup in order to capture the finger noise made by the musician, however this is more common in a studio environment than in live sound.

Pianos

Sound from acoustic pianos (such as the baby grand in our Sanctuary) should be captured using an omnidirectional microphone (usually a *condenser* microphone – but this is certainly not required) on a boom stand. The lid should be opened (usually on the short stick) and the stand should be positioned so that the mic hovers near the middle of the strings. The tone of the capture can be adjusted by moving the mic towards higher or lower strings.

If equipment and channel availability on the mixing console allows, it is recommended to use multiple microphones (usually 2, sometimes 3) positioned such that each microphone captures a different frequency range coming from the strings. This allows maximum control of tone and volume from the mixing console.

Due to the wood construction of an acoustic piano, sound from nearby speakers tends to get directed back inside the piano and towards the microphones being used, so you should be prepared to deal with higher amounts of feedback than with other acoustic instruments.

Additional Considerations

While on the topic of sound sources and capture techniques, there are some related areas that are worthy of discussion.

Phantom Power

Some microphones (such as the pulpit mic in our Sanctuary) contain circuitry that requires power to make the microphone operate. Condenser microphones are a classic example of such a mic. The mixing console can provide *phantom power* for this purpose – sending power in reverse back through the signal chain to the microphone. It's important when using phantom power to make sure you are using the correct cables – all XLR cables can carry power, but almost no ¼" instrument cables have this capability. Only ¼" cables specifically designed to carry power can be used to carry phantom power. This is a minor concern – just about every conceivable instance where you would be required to use phantom power will be with a microphone using a standard XLR cable.

Phantom power is controlled on the mixing console under the settings for each input channel. You should only turn on phantom power to the specific channel or channels that require it. An important warning: sending phantom power to a channel that is connected to a DI box or a microphone with an on/off switch can cause problems, including damaging the equipment.

Microphone Proximity Effect and Microphone Techniques

Directional microphones have what is known as a *proximity effect*. When a person is in close proximity to the microphone (about 2 inches or less), there is a *dramatic* increase in the low frequency response. The proximity effect can be used to your advantage; if the artist constantly stays close to the mic, it can generate a capture that feels more powerful.

However, when a large number of microphones are being used, the proximity effect is usually a disadvantage. Most of the time, singers tend to stay too far away from the mic – usually somewhere between 8 and 18 inches – for the proximity effect to be a factor, so we set up the sound system in such a way that it sounds best *without* the proximity effect. On solos or when leading a song, singers tend to get in much closer – typically 2 to 4 inches – and the proximity effect can cause the captured signal to have more low end than is desired (too much "bass").

This is a great example of why the source tone is so important. **Ideally, microphones should be 4-6 inches from the vocalist's mouth**. A time-tested rule-of-thumb is that singers should keep their microphone about a fist away from their mouth, with only very small deviations from this for desired dynamic effects. *As the audio engineer, you should pay attention to where your singers' mics are at, and you should **never** be afraid to jump in and make corrections as needed*. This is probably the most important thing you can take away from this section. Mic proximity is the single biggest factor in the tone of a vocalist's source sound and a poor capture cannot be fixed from the mixing console.

Equally important to the distance is the angle of the microphone. When using a handheld microphone, it should be positioned just below the mouth, even with the chin. The microphone should be held almost vertical with only a slight tilt toward the mouth. Source sound should be captured by singing or speaking *across the top* of the microphone. Singing or speaking directly into the top of the microphone will cause excessive breath noises and "P-popping."

Thoughts on Microphone Stands

In practice, most vocalists will not be holding their mics. Whether because they are simultaneously playing an instrument, or because they are not a solo vocalist, many of them will be using microphones on stands. The proper choice of stand can make a big difference. You shouldn't use too much mental bandwidth worrying about mic stands, but you should be aware of the different types and how they can be used.

Straight stands are usually suitable for a speaker or when capturing sound from a small group of singers near one another.

Boom stands help to get the microphone closer to a person than a straight stand otherwise would and are most useful when the vocalist is also playing a piano, keyboard, guitar, or other instrument.

Goose-neck stands are often used to get a microphone close to an acoustic instrument. They are not used very often in practice and, when they are used, are usually holding a mic capturing sound from an acoustic guitar. Some microphones have goose-neck stands built into their design, such as the pulpit mic in our Sanctuary.

Hanging microphones are often the best way to mic a choir. Hanging the mics keeps them above and in front of the group.

3.3. Signal Transmission

The Snake

So, you have decided on which methods to use to capture the audio from the various sources at play. You've used microphones or electronics to turn raw sound into an electronic audio signal. How do you get that signal up to the mixing console – 150 feet away and upstairs?

The answer is a *cable snake*. A traditional cable snake bundles many individual audio (or video/data) cables into one sturdy outer jacket for easier management. Most snakes will contain between 16 and 64 individual channels. Until recently, this heavy, bulky, and expensive snake would then be run – usually hidden in ceilings, in walls, or under floors – from the stage to wherever the mixing console was located.

Thanks to advancements in technology, the bulky snake of antiquity has been replaced by one single category 5 or category 6 (CAT-5/6) cable.

In our stage setup, we have a rack on stage left that contains, among other things, what is known as a *stage box*. The stage box has 32 input and 16 output channels available. By communicating through the CAT-5 cable, these inputs and outputs can be routed for use with any channel on the mixing console. Simply get the captured audio signal into the stage box and it can be "heard" by the mixing console, just like if you were plugged straight into the back of the board.

A Note About Channel Routing

At this point, it is important to clarify a small, but critical, detail about the channels on the stage box. These inputs and outputs can be routed to *any* input or output channel on the mixing console. Said another way: The channel numbers on the stage box *do not necessarily* correspond to the channel numbers on the console.

You don't need to concern yourself with the details of routing at this time, but it's an important consideration to be aware of. If someone on stage tells you they're "plugged into channel 5" on the stage box, this does not necessarily mean their signal will automatically be available in channel 5 on the mixing desk.

In future volumes, you'll learn how to edit the routing as needed, but it shouldn't be something that comes up with any regularity. Our stage plot is set up in such a way that adjustments to routing should rarely have to be made.

Cables

So now you've got your capture device ready, you've determined what input you need to get it to on the stage box, and you're ready to make the connections. Let's spend some time discussing how to get the captured audio from the source to the input channel.

Types of Cables

It is important to understand the different types of cords and cables that are used to make the connections between various parts of the sound system. Ideally, all cables used to connect inputs to a sound system or to connect the various parts of the sound system together should be shielded to prevent the system from picking up hum, noise, or interference. In practice, this is not always practical, however it's a good best-practice to strive for.

All cables used to connect sound sources to input channels fall into one of two categories: unbalanced cables and balanced cables.

Unbalanced Cables

The term *unbalanced* refers to a two-conductor cable where one conductor is a grounded shield and the other conductor carries the sound signal. However, the sound signal must also return via the shield. The connection is referred to as unbalanced because one conductor is grounded and the other is not.

Unbalanced cables may use RCA jacks or quarter or eighth-inch plugs and generally should not be used for distances much greater than 25 feet. At greater distances, an unbalanced cable picks up too much electric “hum” and excess noise, degrading the quality of the audio signal. Although the shield protects the inner conductor from pickup up hum and noise, since the sound signal must return via the shield, it is affected by the hum and noise picked up by the shield.

As a general rule of thumb, unbalanced cables are fine for short distances such as from an electric guitar to a pedal board - or from a computer to an auxiliary input on the sound board - but should be avoided when possible.

Some examples of unbalanced cables include RCA cables and ¼”/1/8” TS (tip-sleeve) cables.

Balanced Cables

In contrast, the term *balanced* refers to a three-conductor cable where the outer conductor is a grounded shield and two inner conductors carry the sound signal. The connection is referred to as "balanced" because the two inner conductors that carry the sound signal are "balanced" at the same voltage level.

Balanced cables can carry sound signals much further than their unbalanced counterparts - typically several hundred feet. Balanced cables are not nearly as susceptible to picking up hum or noise because the sound signal only goes through the inner two conductors and never uses the shield, therefore, hum and noise picked up by the shield is simply grounded.

Classic examples of balanced cables are XLR and ¼” TRS (tip-ring-sleeve) cables.

Converting Cable Types

Often, you will need to change the type of end on your cables in order to get the audio signal from one place to another. To accomplish this, we use adapters and interface converters.

One example you might encounter is when you need to use headphones (with a standard 1/8” headphone cable) to listen in to something, but the output is only available from a ¼” jack. In this case, a simple 1/8-to-1/4” adapter on the end of your headphone cable will get the job done.

We use adapters on stage to convert monitor outputs, which are XLR, to 1/8” headphone jacks so that our worship team can use in-ear monitors on stage.

However, the most common, and most important, need for such devices is when you have sound that is captured on a ¼" device (guitar, keyboard, etc.) and need to send that signal into an input channel on the stage box – all of which are XLR. While it is possible to use an adapter to simply change the end of a ¼" cable into an XLR end, this is not the recommended method. Adapters are typically not adequately grounded and shielded and have a tendency to pick up unwanted noise and interference. Using a ¼" cable (even if it's balanced) with a ¼"-to-XLR adapter on one end to send audio from the keyboard into the input channel will result in a hum that will haunt your sound and your mix for eternity.

Instead, in these situations, we use a *direct injection box*. A DI box (or simply DI) converts an instrument's high-impedance, unbalanced signal – like the signal from a keyboard, guitar, or bass – into a low-impedance, balanced signal. In other words, you can plug an instrument cable into the input of a DI and get a balanced XLR output to send into the input channel.

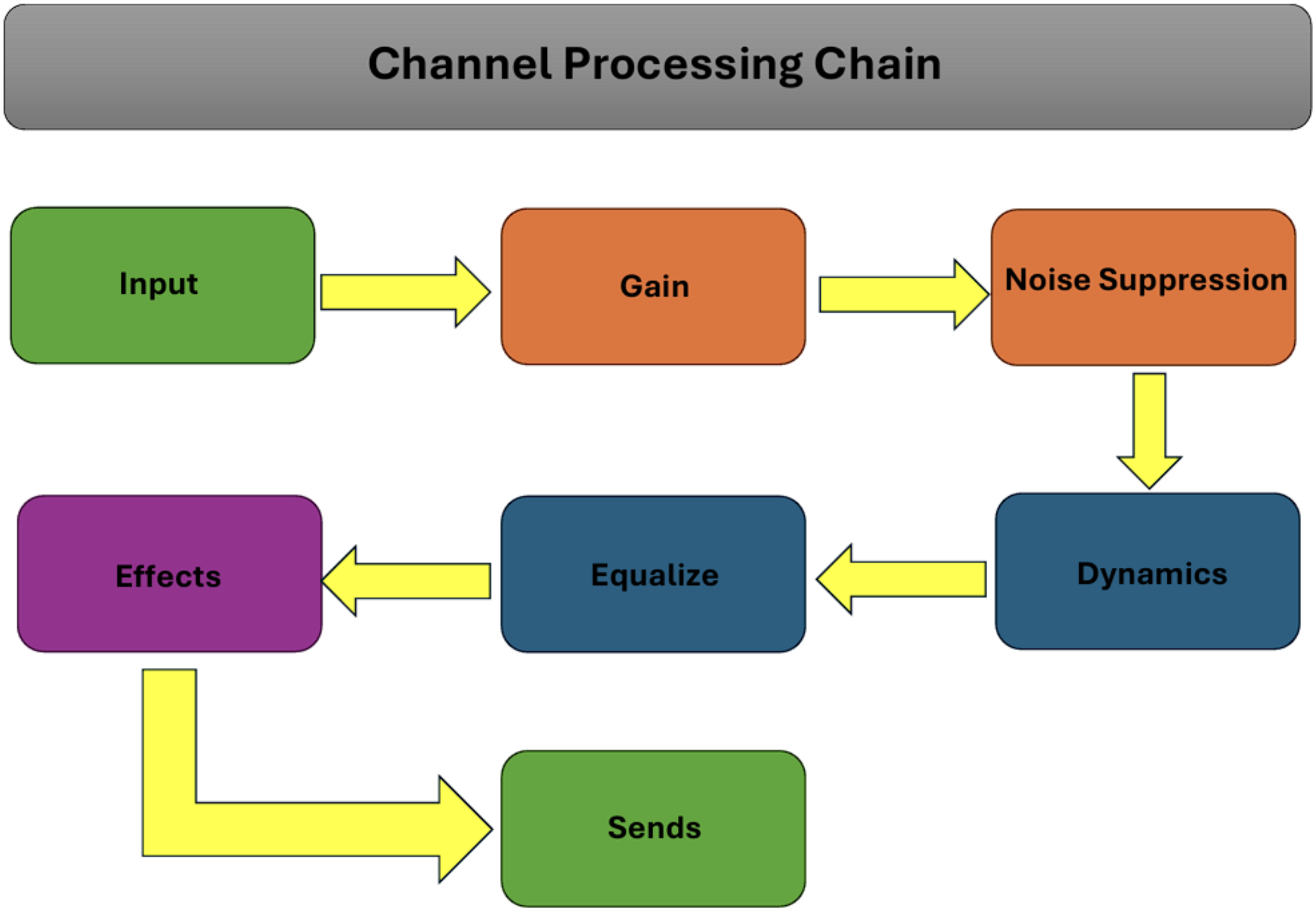
As a best-practice, you should always use a DI anytime you have an input signal and need to either (1)convert from ¼" to XLR or (2)balance the signal from an unbalanced cable.

3.4. Channel Processing

Now that your signal is getting sent through the input channel, you are ready to begin processing it at the mixing console!

From the input channel, the audio signal passes through the *channel processing chain* before being sent to the next step in the signal chain. This is where the "magic" happens. We will not be going into much detail on the different processing functions at this time; later chapters will be dedicated to the nuances and specifics of processing. Besides, the best way to properly learn channel processing is by practicing it at the board. Rather, this section is designed to simply expose you to the various processes and give you a basic understanding of what they are used for, as well as to expose you to the definitions for some of the language used to describe the processing tools.

On the following page you will find a very basic flow chart identifying the various steps an audio signal goes through during channel processing, followed by a very brief description of what each step does and its purpose.



Faders control output volume logarithmically. The closer the fader is to *unity* (0 dB), the greater the control of the volume. Conversely, as the fader gets further away frum unity, it produces less and less control of the volume. In order to mix efficiently and professionally, faders should, ideally, be as close to unity as possible. A common mistake made by inexperienced operators is failing to stage a channel's gain properly, resulting in faders that are too far down on the pot to be effective. Setting up the channel's gain correctly is essential in order to set your mix up for success.

Gain Staging

The first five steps of the channel processing chain (from input through EQ) collectively form a process known as *gain staging* and serve the purpose of ensuring that the audio signal that ends up being sent to the fader is balanced. This allows for the greatest amount of control when mixing. A channel with proper gain staging results in a fader that rarely has to move very far away from unity.

Input

The audio signal is sent, via CAT-5, from the input on the stage box to the mixing console (or, alternatively, from a source other than the stage box and sent directly into the input on the back of the board). This signal is then routed to the specified channel on the sound board to be processed.

Gain (Preamp)

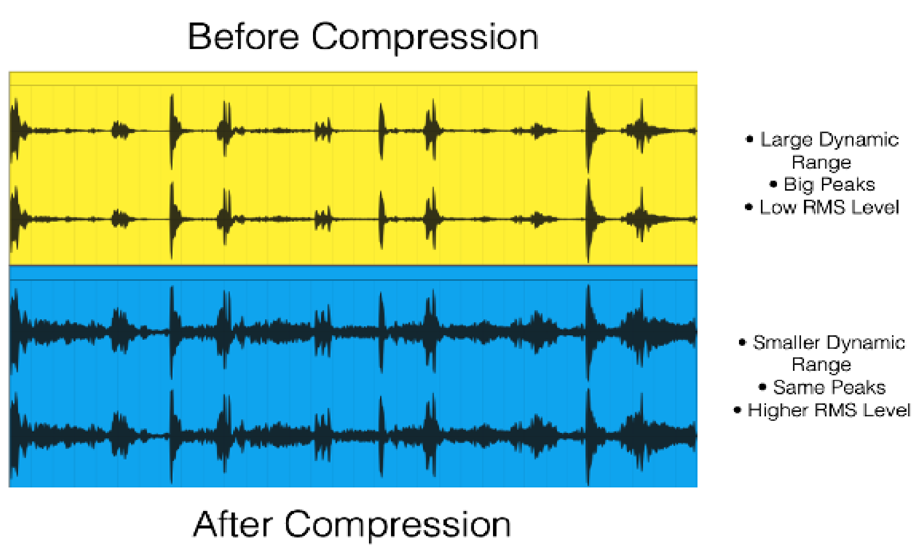
Before any work can be done to the audio signal, its *gain* must be adjusted. The gain (or preamp) function is designed to adjust the volume level at which the signal is sent into the channel. Setting the proper gain is necessary to ensure that the signal being received by the mixer is at an appropriate amplitude. Too much gain, and the signal becomes distorted and unusable and, additionally, results in faders that are way below unity. On the other hand, too little gain will cause the signal to become weak and lack clarity and can also create situations where the fader is too far above unity, removing all headroom while you are mixing.

Noise Suppression

Noise suppression is the process of telling the console at what amplitude to start "listening" to the channel. Tools such as noise gates can help in preventing unwanted "background noise" from getting through the channel by, essentially, turning the input off whenever the signal is below the specified level. This is especially useful for reducing noise bleed when mixing microphones or acoustic instruments.

Dynamics

Dynamic controls such as compression and normalizers are used to restrict the amount of variance in a channel's input volume. Dynamic control tools can be used to make the "quiet" parts of an audio signal and the "loud" parts closer to the same volume. The major benefit of utilizing dynamic controls is to prevent sudden and distasteful changes in the volume coming from a particular channel.



Equalizer (EQ)

The *equalizer* is used to equalize the gain of the signal across the frequency spectrum. EQ tools are used to adjust specific frequencies and frequency ranges to better define the color and flavor of a signal's tone, as well as to quiet down – or remove altogether – undesirable frequencies.

High-pass filters ("low cut") are used to remove sub-bass frequencies to control the "rumble" of the sound's tone. Similarly, *low-pass filters* ("shelf") are used to brighten the higher ends of the frequency spectrum to add air and clarity to an audio signal. *Makeup gain* can be added to restore the signal's gain back to pre-EQ levels.

Effects (FX)

After gain staging, the signal gets sent to desired *effects processors*, which can make many different and distinct changes to the tone of an audio signal. Examples of frequently used FX processors include *reverb* (to add fullness), *delay* (to add depth), *enhancers* (to add stereo width), and *exciters* (to add brightness). FX processing adds color and flavor to an audio signal, and is highly dependent on personal taste and preference. To become good working with effects, there is no alternative to practice and experience.

Sends

Once signal processing is complete, the channel's audio signal is sent to the various *sends* that are selected. Sends can include direct outputs, various FX channels, and several other options. Most often the channel will be sent to one or more of the mix busses *(Section 3.6)*.

3.5. Groups

Within the setup options, each individual channel can be assigned to one or more groups. Groups allow the engineer to combine channels together in such a way that one adjustment effects all the channels equally. The two main types of groups you should be familiar with are *mute groups* and *DCAs*.

Mute Groups

A *mute group* is exactly what the name implies. Channels can be assigned to a mute group then, by pressing a single button, the mute functions of all the channels within the group are toggled.

A classic example is assigning all instrument channels to a mute group - let's say we use Mute Group 5. By pressing the Mute Group 5 button, all instrument channels can be muted or unmuted simultaneously.

DCAs

A *Digitally Controlled Amplifier* (DCA) is a convenient way to adjust the levels for several channels at the same time. A DCA will adjust the level of each channel individually, as opposed to summing all the channels into one signal and then adjusting the level of that single signal.

When you move a DCA fader, the volume of each channel assigned to that DCA will change, but the individual channel faders will not move.

DCAs are most commonly used to group like-channels together. For example, by assigning all vocal channels to DCA 1 and all instrument channels to DCA 2, you can adjust the volume of all vocalists by moving the single DCA 1 fader (likewise, instruments can be adjusted using the DCA 2 fader).

3.6. Mix Busses

A *mix bus* routes multiple tracks into one channel, allowing them to be processed as a group with one single audio signal. You can think of busses as the "middle man" between the channels and the outputs.

Some of the busses we use are monitor busses, effects busses, broadcast busses, and the left-right (LR) bus.

The same processing tools from the channel processing chain can be used to process individual busses. The key difference here is that the processing will affect the audio signal of the bus as a whole rather than single tracks. This is a useful place for tools such as limiters and stereo wideners.

Just like with channels, the output of a bus is controlled using sends. Monitor busses are sent directly to outputs that go to the floor and in-ear monitors on stage. Effects busses are sent back to the channel strip as an additional input channel. The left and right broadcast busses, as well as the LR bus, are sent to what's called a *matrix*.

3.7. Matrices and Outputs

A *sound matrix* refers to an audio routing system that sends specified input signals to various outputs with independent level and processing control. It allows mixing pre-mixed signals sent from the busses for custom destination feeds, allowing greater flexibility beyond simple outputs.

In our setup, the signal from the broadcast busses is sent to matrices controlling the outputs for the livestream feed as well as the assisted hearing devices.

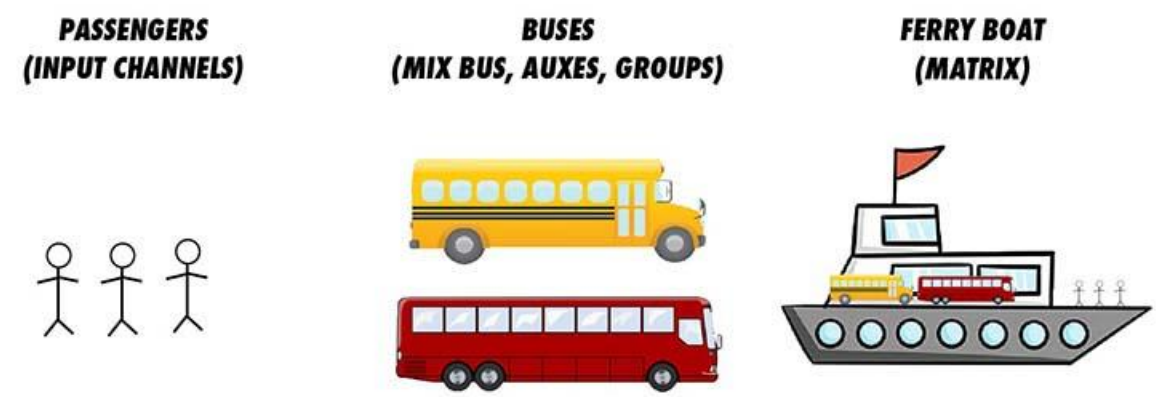
The LR bus is sent to a pair of stereo matrices controlling the output signal being sent to the left and right main speakers. It is also sent to a matrix that controls the sub. This allows us to process the signal being sent to the sub separate from the signal being sent to the main speakers. We can then use tools such as an EQ to send low-end frequencies to the sub while cutting those frequencies from the mains.

As with channels and busses, matrices can be processed using the same tools already discussed. Most commonly, matrices will be processed with an additional EQ and/or some type of limiting device that controls the maximum allowed threshold of the output volume.

After processing, the signal from a matrix is then routed to a specified output. In the case of the broadcast matrices, the signal is sent from the board to the streaming computer where it is used as the audio source for the livestream. The L/R and sub matrices are sent back down to the stage box, where they are connected to their respective speakers.

Final Thought

Below is a helpful diagram showing a great way to think about the relationship between channels, busses, and matrices. The input channels are grouped together and sent to busses. The busses are then grouped together in a matrix to be sent to their final destination.



PART 2: THE BEHRINGER X32

4. Navigating the X32

4.1. Startup and Shutdown

Operating a sound system is not simply about turning equipment on and off. Every action taken at the console has the potential to affect the entire audio system — and by extension, the congregation's experience of worship.

For this reason, startup and shutdown procedures must be performed consistently, carefully, and in the correct order. Developing good habits in this area will prevent unnecessary noise, protect equipment, and ensure a smooth and distraction-free service.

Purpose of Proper Startup and Shutdown Procedures

Before discussing the specific steps, it is important to understand *why* these procedures matter.

Audio systems amplify electrical signals. When equipment is powered on or off improperly, it can create sudden spikes in those signals. These spikes can result in:

- Loud pops or thumps through the speakers
- Potential damage to speakers and amplifiers
- Disruption during quiet or sensitive moments
- Loss of confidence in the production team

A well-executed startup and shutdown process ensures that the system powers on *silently* and *safely*, all components initialize correctly, and that the engineer is able to begin with a known, controlled starting point.

General Power Sequence Principle

There is a foundational rule that governs nearly all audio systems: *Power **ON** from source to output; power **OFF** from output to source.* If you have been on the team for a while, you might notice that this is the *opposite* sequence that has been used in the past. Please note this change in your startup and shutdown routines.

In simpler terms,

- When turning the system on, start with devices that generate or process signal (like the console), and end with devices that amplify sound (like speakers or amplifiers).

- When turning the system off, do the reverse.

This sequence prevents unwanted noise from being amplified through the system.

Startup Procedure

Follow these steps in order each time the system is powered on.

1. Verify System Readiness
   Before applying power, take a moment to check the environment:
   - Ensure all cables are securely connected
   - Confirm no microphones are dropped, unplugged, or damaged
   - Verify stage equipment (instruments, DI boxes, etc.) is in place
   - Make sure all channels on the console are muted
   - Confirm the main LR fader and output matrices are down or muted

   This step helps prevent surprises later.
2. Power On the Console
   Turn on the X32 console using the rear panel power switch. Allow the console to fully boot before proceeding. Do not press buttons or adjust controls during startup and watch for any error messages or unusual behavior. Power on the FOH headphone amp and monitor.
   The console should initialize to its last saved state.
3. Power On Stage and System Components
   Once the console is fully powered:
   - Power on stage box
   - Power on wireless receivers and other input devices

   These devices provide signal to the console and should be active before audio is sent to speakers.
4. Power On Speakers
   Finally, power on the main speakers, sub, and stage monitors. This step should always happen last to avoid sending unwanted noise to the speakers and causing potential damage to the equipment.
5. Load the Appropriate Scene
   Once the system is fully powered, load the correct scene on the console and confirm that all channels, routing, and effects are as expected. This should place you at a familiar and comfortable starting point for the mix.
6. Perform a System Check
   Before the rehearsal begins, verify that everything is functioning properly:
   - Check input signal on key channels (vocals, instruments, playback)
   - Verify that microphones are producing clean signal
   - Confirm monitor mixes are active and correct

- Listen to the main output through headphones or monitors
- Walk the room briefly if time allows

This step ensures that any issues are identified early — not during the service.

Shutdown Procedure

Shutdown is just as important as startup and must be done with equal care.

1. Mute all channels and mute or lower the LR bus and matrix outputs to prevent residual noise during shutdown.
2. Power off main speakers, sub, and stage monitors.
3. Power off external devices such as the stage box and wireless systems.
4. Power down the FOH headphone amp and monitor.
5. Power off the console by selecting “Setup” – “Shutdown” – “Go”. Then, when prompted, turn off the console’s main power switch.

Final Thoughts

1. Never rush startup or shutdown. These are routine but critical processes.
2. Always follow the same order. Consistency prevents mistakes.
3. When in doubt, mute outputs first. This provides an additional layer of protection.
4. Treat the system with care. The goal is not just functionality, but reliability and excellence.

In a live worship environment, distractions can quickly pull attention away from what matters most. Something as simple as a loud pop through the speakers can break focus in an otherwise meaningful moment. By taking the time to power the system on and off correctly, you are contributing to an environment that is orderly, intentional, and free from unnecessary distraction. That is the goal of everything we do in production.

4.2. Console Callouts

Before an audio engineer can confidently operate the X32, they must first become familiar with its physical layout. At first glance, the console can appear complex — filled with buttons, knobs, and faders that seem overwhelming. However, this complexity is only on the surface. The X32 is intentionally designed in clearly defined sections, each with a specific role.

The goal of this section is not to memorize every individual control, but to develop a sense of orientation. As you begin learning the console, the most important question is not “What does every button do?”, but rather: ***“Where should I look to accomplish a specific task?”***

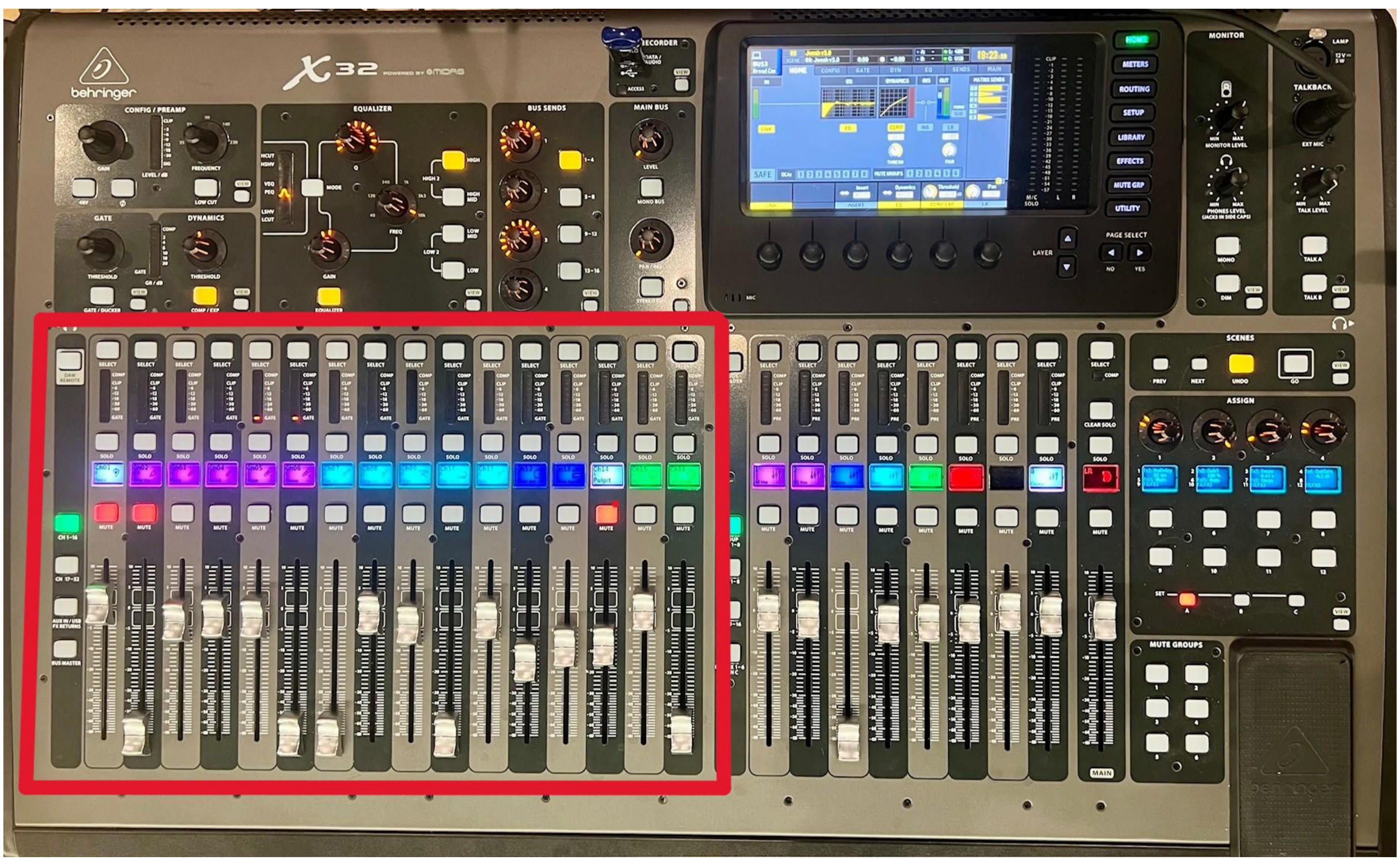
Once you can answer that question, the rest becomes much easier.

Input Channels

The left side of the console is dedicated to input channels, which is where most mixing begins. Each vertical strip represents a single channel and provides direct control over its level and basic functions. Here you will find the fader, which controls volume, along with mute and solo buttons that allow you to quickly remove or isolate a signal. Each channel also includes a select button, which plays a critical role in how the console operates.

Unlike an analog console where each channel has its own full set of controls, the X32 uses a different approach. At any given moment, only one channel is actively selected. Pressing the select button determines which channel you are working on, and that selection directs the behavior of other parts of the console. This is one of the most important concepts to understand early on, as it affects nearly everything you do.

Because the console has more channels than physical faders, it uses layers to provide access to additional inputs. The same set of faders can represent channels 1 through 16, then switch to channels 17 through 32, and then again to auxiliary inputs and effects returns. This allows the console to remain compact while still offering a large number of inputs.



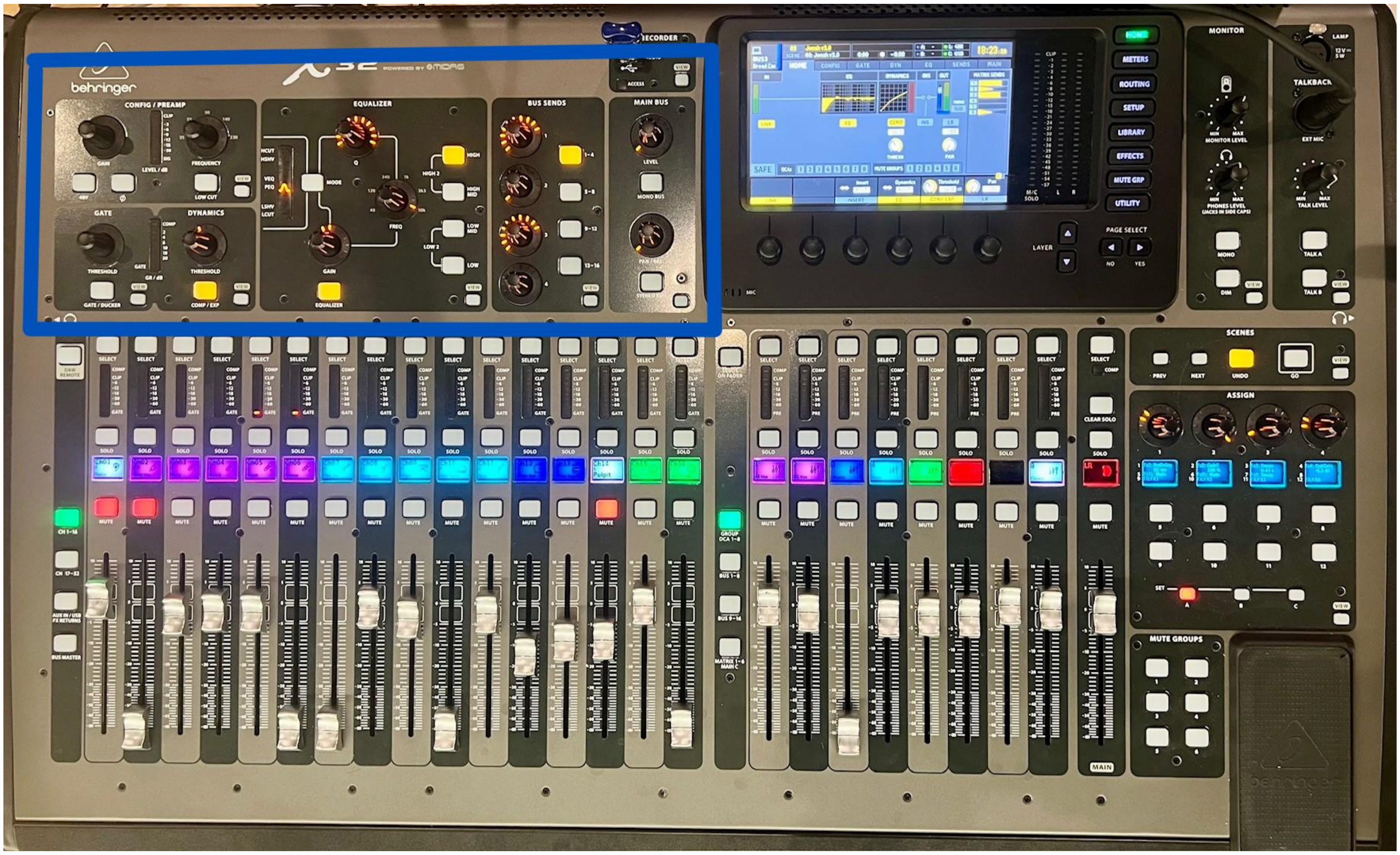
Input Channel Array (red box)

Channel Strip

Near the center of the console is the channel strip, which serves as the primary control area for shaping the sound of the selected channel. Once a channel is selected, this section allows you to adjust its gain, apply equalization, control dynamics such as compression, and determine where the signal is sent.

Each part of the channel strip corresponds to a specific stage in audio processing. As you move from top to bottom, you are effectively moving through the signal path of that channel. While the individual controls will be explored in more detail later, it is important to understand that this section is always focused on the currently selected channel, not the entire mix.

This design allows a single set of controls to manage many channels efficiently, but it also requires intentional awareness. If something changes unexpectedly, one of the first things to check is which channel is currently selected.



Channel Strip (blue box)

Main Display

Above the channel strip is the main display and monitoring section, which acts as the console's central interface. The display provides detailed information about whatever part of the console you are currently working with, and it allows for more precise adjustments than the physical controls alone.

One of the most helpful features of the X32 is the consistent use of "View" buttons throughout the console. Pressing a View button immediately brings up the relevant controls for that section on the display. This eliminates the need to search through menus and makes navigation much more intuitive. Instead of digging through layers of settings, you are taken directly to what you need.

Surrounding the display are controls for adjusting parameters, navigating between pages, and monitoring audio levels. This area also includes controls for headphones and talkback, which are essential for communicating with musicians and verifying what is being heard.

Main Display (green box)

Outputs

On the right side of the console is the bus and output section, which controls where audio is sent after it has been processed. While the left side of the console focuses on inputs, the right side focuses on outputs.

This is where you manage mix buses used for stage monitors and the broadcast mix, as well as matrices that distribute audio to different destinations. You will also find the main output fader here, which controls the overall level being sent to the primary speaker system.

Understanding the difference between inputs and outputs is a key step in navigating the console effectively. When adjusting what is being heard in the room, you are typically working with outputs. When adjusting what is coming into the console, you are working with inputs.

Output Array (purple box)

Scenes and Assign

In the upper right portion of the console is the scenes and assign section, which provides tools for organization and control. Scenes allow you to save and recall complete console configurations, making it possible to move quickly between different setups. This is especially useful in environments where multiple services or events share the same system.

This section also includes assignable controls and mute groups, which can simplify complex tasks by allowing multiple channels to be controlled at once. While these features may not be used constantly, they become increasingly valuable as you grow more comfortable with the console.

Scenes and Assigns (orange box)

Conclusion

When viewed as a whole, the layout of the X32 follows a consistent and logical pattern. You begin by selecting a channel on the left side of the console. You then shape and adjust that channel using the controls in the center. Finally, you determine where that signal is sent using the controls on the right.

This flow — from input, to processing, to output — is the foundation of how the console operates.

At first, navigating the X32 may feel unfamiliar. Over time, however, these sections will begin to feel natural. The goal is not immediate mastery, but steady familiarity. As you continue through this manual, you will build on this understanding and learn how each part of the console contributes to the overall system. Understanding the layout of the mixing desk is the first step toward using it with confidence.

4.3. Main Display

While the physical layout of the X32 provides immediate, hands-on control of audio, the main display serves as the console's central interface for deeper interaction. It is here that you will access detailed parameters, configure routing, and monitor the overall state of the system. If the channel strip is where adjustments are made quickly, the display is where those adjustments are understood and refined.

At first glance, the screen may appear dense with information. However, like the rest of the console, it is designed with a clear and consistent structure. Learning how to read and navigate the display is an essential step toward operating the X32 with confidence.

The main display is located at the top center of the console and provides real-time feedback about the currently selected channel or function. Whenever a channel is selected, the display updates to show its settings, including configuration, processing, and routing. This means the display is always context-sensitive — it reflects whatever you are currently working on.

Across the top of the screen, you will see a status bar that provides important system information. This includes the selected channel name and number, the currently loaded scene, and the status of connected devices such as stage boxes or USB interfaces. While these details may not always require your attention, they can be helpful when troubleshooting or verifying that the system is operating as expected.

Below the status bar is the main workspace of the display. This area changes depending on which section of the console you are interacting with. For example, if you are adjusting equalization, the display will show EQ controls. If you are working with routing, it will show signal paths and assignments. This dynamic behavior allows the display to serve many functions without becoming cluttered.

Navigation within the display is designed to be both fast and intuitive. One of the primary ways to move between different views is through the use of the "View" buttons located throughout the console. Pressing a View button immediately brings up the corresponding screen on the display, allowing you to access detailed controls for that section without searching through menus. This design is intentional. Rather than forcing the user to navigate through layers of options, the console brings the relevant information directly to you. As you become more familiar with the layout, you will find that these View buttons become one of the fastest ways to move around the system.

In addition to the View buttons, the display can also be navigated using the page selection buttons located beside the screen. Some sections contain multiple pages of parameters, and these buttons allow you to move between them. When additional pages are available, the display will indicate this, making it clear that more options exist beyond what is currently visible.

Once you are on the desired screen, adjustments are made using the row of encoders located directly beneath the display. Each encoder corresponds to a parameter shown on the screen. Turning an encoder adjusts the value, while pressing it can toggle or confirm certain settings. This direct relationship between the screen and the controls helps maintain a clear connection between what you see and what you are changing.

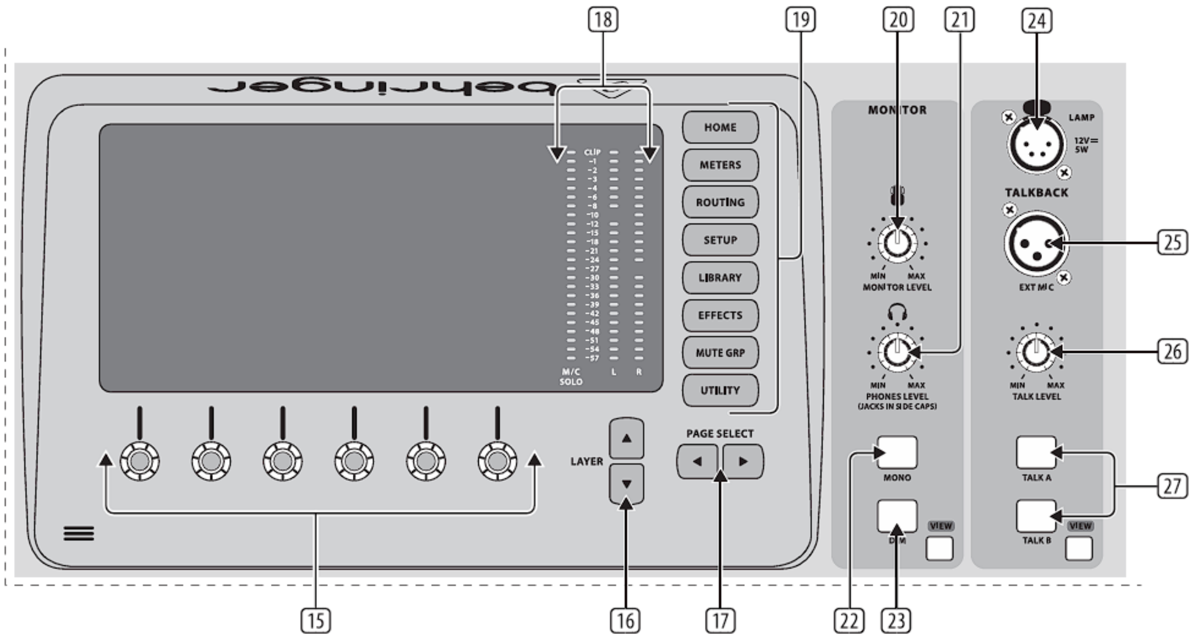
Another important function of the display is monitoring. In addition to the main meters located on the console, the display provides detailed visual feedback about signal levels, processing activity, and system status. This can be especially useful when identifying issues such as clipping, lack of signal, or incorrect routing.

Because digital audio systems rely heavily on internal routing and processing, visual feedback becomes an essential tool. The display allows you to "see" what is happening inside the console, even when it is not immediately obvious from the physical controls alone. As you begin working with the X32, it is helpful to think of the display as an extension of the console rather than a separate component. The physical controls and the display are designed to work together. The controls provide speed and tactile feedback, while the display provides precision and clarity.

Over time, you will develop a rhythm between the two — selecting channels, pressing View buttons, adjusting parameters, and observing the results on the screen. This interaction is at the heart of operating a digital console.

At first, it is not necessary to understand every screen or parameter in detail. The goal is simply to become comfortable navigating the display and recognizing how it responds to your actions. As you continue through this manual, you will revisit many of these screens in greater depth and learn how to use them effectively in a live environment.

The main display is one of the most powerful tools available to you on the X32. Learning to use it well will significantly improve both your efficiency and your confidence as an audio engineer.



15 **Push Encoders** – These 6 controls adjust the parameters presented at the bottom of the Main Display. The editable function will show a circular icon in the display when continuous control is available. The function will show a broad rectangular icon to indicate that a switch or toggle can be accessed by pushing the encoder.

16 **Layer Buttons** – Some screens in the Main Display have more than 6 editable parameters which can be accessed by pressing the Layer Up or Down buttons.

17 **Page Select Buttons** – Use these to scroll through the available screens or to confirm/decline certain actions.

18 **Main/Solo Meters** – The main stereo output level is displayed here along with the solo level of all channels whose Solo button is active.

19 **Category Select Buttons** – Press one of these buttons to jump directly to the subject you wish to edit or configure.

20 **Monitor Level** – Adjust the level of the Monitor outputs with this knob.

21 **Phones Level** – Adjust the volume of the headphone outputs, located inside the left and right side caps.

22 **Monitor Mono** – Press this button to monitor the audio in mono.

23 **Dim** – Press this button to reduce the monitor volume. Press the View button to adjust the amount of attenuation along with all other monitoring-related functions.

24 **Lamp Input** – Connect a standard 12 V, 5 Watt gooseneck lamp here.

25 **Talkback Input** – Connect a talkback mic via standard XLR cable to this input.

26 **Talk Level** – Adjust the level of the talkback mic with this knob.

27 **Talk A/B** – Select the destination for the talkback mic signal with these buttons. Press the View button to edit the talkback routing for A and B.

Main Display Panel

4.4. Signal Flow Inside the Console

Up to this point, we have focused on what the console looks like and how to navigate it. In this section, we turn our attention to something less visible, but far more important: how audio actually moves through the system.

Understanding signal flow is one of the most important steps in becoming a confident audio engineer. Nearly every task on the console — adjusting levels, creating monitor mixes, troubleshooting problems —

depends on knowing where the signal is coming from, where it is going, and what is happening to it along the way. If the physical layout of the console tells you where to look, signal flow tells you what is happening.

When a sound enters the system, it begins as an electrical signal at an input. This might come from a microphone, an instrument, or a playback device. Once that signal enters the console, it does not go directly to the speakers. Instead, it travels through a series of processing stages, each of which can shape, control, or redirect the sound. Although the X32 is a digital console, the path that the signal follows is structured in a logical and consistent order. This path can be thought of as a chain, which you should recall from our discussions in earlier chapters:

Input → Processing → Routing → Output

Everything you do on the console fits somewhere within this flow.

Processing

The signal first enters the console through an input and passes through the preamp stage. This is where the initial gain is applied to the signal, bringing it up to a usable level. Setting this level correctly is critical, as it affects everything that follows. If the signal is too weak, it will be difficult to work with. If it is too strong, it may distort or clip.

After the preamp, the signal moves into the processing stages of the channel. This is where most of the tone shaping takes place. The signal may pass through a gate to reduce unwanted noise, then through an equalizer to adjust its frequency balance, and then through a compressor to control its dynamic range. Each of these steps modifies the signal in a specific way, and each one builds on the previous stage.

It is important to understand that this processing happens in a defined order. Changes made earlier in the chain will affect everything that comes after. For example, adjusting the input gain will influence how the compressor behaves, and EQ adjustments will affect what is sent to monitors and outputs.

Sends and Outputs

Once the signal has been processed, it is then routed, or “sent”, to one or more destinations. The most common destination is the main output, which feeds the speakers in the room. However, the signal can also be sent to mix buses, which are used to create monitor mixes for musicians, alternative mixes (such as for the broadcast mix), or submixes for specific groups of channels.

Each channel can send signal to multiple buses at the same time, and each of those sends can be controlled independently. This allows the same input signal to be used in different ways throughout the system. For example, a vocalist’s microphone can be sent to the main speakers, to a stage monitor, and to a recording feed — all at different levels.

These routing decisions are a major part of what the audio engineer controls during a service.

From the buses, the signal may continue on to additional processing or routing stages before reaching its final destination. One of these stages is the matrix, which allows multiple signals to be combined and sent to different outputs. We use matrices to send the LR bus to the house speakers and the broadcast bus to the broadcast computer and radio.

Finally, the signal reaches the output stage, where it leaves the console and is sent to amplifiers or powered speakers. At this point, the signal is converted back into sound and heard by the audience.

Conclusion

Although this process may seem complex at first, it follows a consistent pattern. Every signal begins at an input, moves through processing, is routed to one or more destinations, and then exits the console. A helpful way to visualize this flow is:

Input → Preamp → Processing → Bus Sends → Matrix Sends → Output

Keeping this sequence in mind will make it much easier to understand what is happening at any given moment.

One of the most common challenges for new engineers is troubleshooting when something does not sound right. In these situations, signal flow becomes especially important. Rather than guessing, you can follow the path of the signal step by step:

- Is the signal present at the input?
- Is it being processed correctly?
- Is it being sent to the correct destination?
- Is it reaching the output?

By asking these questions in order, you can quickly identify where the problem is occurring.

As you continue learning the X32, you will begin to see how the physical layout of the console reflects this signal flow. The input channels on the left represent where signals enter. The channel strip in the center represents processing. The bus and output sections on the right represent where signals are sent. This is not accidental — it is designed to match the way audio moves through the system.

Understanding signal flow does not require memorizing every detail immediately. Instead, it is about developing a mental model of how sound travels through the console. Over time, this model will become second nature, and you will be able to make decisions more quickly and confidently.

This concept will continue to appear throughout the rest of this manual. Whether you are building monitor mixes, adjusting EQ, or solving a problem during a service, signal flow will always be at the center of what you are doing. Once you understand how audio moves through the console, everything else begins to make sense.

PART 3: THE LIVE MIX

5. Practical Mixing & Live Operation

Introduction

Up to this point, we have focused on understanding the tools. You have learned how to start up the system properly, how to navigate the console, and how signal flows through the X32. All of that knowledge is essential, but it leads to a bigger question: what do we actually do with all of it during a live service?

This chapter is where everything begins to come together. Practical mixing is the art of taking what you know about the console and applying it in real time to support what is happening on stage. It is not just about making things louder or quieter. It is about creating clarity, balance, and an environment where the message can be heard and the music can be experienced without distraction. In a church setting, that responsibility carries weight. The goal is not to draw attention to the mix, but to remove barriers so people can engage fully in the service.

As we move through this chapter, we will break down the core building blocks of a good mix. You will learn how to establish proper gain structure, shape sound using EQ, control dynamics with compression, and add space using effects like reverb and delay. We will also look at how to approach different elements of a service, such as speech versus music, and how to respond when things do not go as planned. Each section is designed to build on the previous one, so take your time and allow the concepts to connect.

It is important to understand that mixing can feel overwhelming at first. There are many controls, many variables, and everything is happening at once. That is normal. The key is to avoid trying to do everything at the same time. Focus on one concept, practice it, and then move on to the next. Start with gain and balance before worrying about EQ. Get comfortable with EQ before adding compression. Layer your skills gradually, just like building a mix.

Remember, no one becomes a great audio engineer overnight. This is a skill that develops through consistent practice and intentional listening. Give yourself permission to learn, make mistakes, and improve over time. If you stay patient and keep your focus on serving the room rather than impressing it, you will grow into a confident and capable engineer.

5.1. The Audio Engineer's Role During a Service

Before we begin talking about faders, EQ, and compression, it is important to understand the role you are stepping into when you sit behind the console. The audio engineer is not just operating equipment: You are actively supporting everything that happens on stage and helping communicate it clearly to the room. Every decision you make has an impact on how people experience the service.

In a church environment, the goal of the audio engineer is not to be noticed. A great mix is one that allows the congregation to focus on the message and the music without distraction. When the sound is clear, balanced, and natural, most people will never think about the audio at all. On the other hand, when something is off, it becomes immediately obvious. Because of this, your role is less about showcasing technical skill and more about removing barriers.

One of the most important aspects of this role is **awareness**. You are not just listening to individual instruments or voices. You are listening to the *entire room*. This includes the band, the pastor, the congregation, and even the acoustics of the space itself. What sounds good in your headphones or at the console may not translate the same way throughout the room. It is your responsibility to constantly evaluate what people are actually hearing and make adjustments as needed.

Communication is another key part of serving effectively. The audio engineer works closely with worship leaders, musicians, and speakers. This may involve simple things like confirming microphone assignments, helping a vocalist feel comfortable with their monitor mix, or making sure a speaker's microphone is ready before they walk on stage. Clear and calm communication helps prevent problems before they happen and builds trust within the team.

It is also important to approach this role with the right mindset. *There will be moments when things do not go as planned*. A microphone may stop working, a musician may play louder than expected, or feedback may occur at the worst possible time. In those moments, your job is to remain calm, think clearly, and respond quickly. Panic does not solve problems, but a steady and focused approach often does.

Finally, remember that consistency matters. Showing up prepared, following proper procedures, and paying attention to detail all contribute to a reliable and trustworthy audio experience week after week. People may not always notice when things go right, but they will quickly notice when they do not. By taking your role seriously and approaching it with a servant's mindset, you help create an environment where everything else in the service can succeed.

5.2. Gain Structure & Building a Mix from Scratch

A good mix does not start with EQ or effects. It starts with proper gain structure. If the foundation is not solid, everything built on top of it will be harder than it needs to be. Taking the time to set levels correctly at the beginning will make the rest of the mixing process smoother, more predictable, and more consistent.

Gain structure simply refers to how strong the signal is at each stage of the signal path. The goal is to have a signal that is strong enough to be clear and usable, but not so strong that it distorts or causes problems later in the chain. On the X32, this begins at the preamp. Setting the input gain correctly ensures that the console is receiving a healthy signal before any processing is applied.

Setting Input Gain

When setting gain, start by having the person speak or play at the level they will use during the service. Avoid setting levels based on a quiet test, as this will lead to problems later. Watch the input meter on the channel and aim for a strong signal that stays safely below clipping. As a general guideline, you want consistent signal with occasional peaks, but never hitting the red. This gives you enough headroom to handle unexpected increases in volume. In *most* cases, you should aim for the channel's input level to be somewhere between -20 and -12 dB, with -18 dB being a good target to strive for.

Normal input levels around -18 dB with peaks around -10 dB allow plenty of headroom before clipping occurs

Building the Mix

Once your input gains are set, the next step is to build a basic mix using the faders. A helpful approach is to start with all channel faders down and then bring them up one at a time. Begin with the most important element, which is typically the lead vocal or speaking microphone. Set that at a comfortable level for the room, and then gradually bring in other elements around it. Think of this as building a foundation rather than trying to fix problems.

It is also important to understand the relationship between the input gain and the channel fader. The gain controls how much signal enters the console, while the fader controls how much of that signal is sent to the mix. If your gain is set too low, you may find yourself pushing faders higher than necessary. If your gain is too high, small fader movements can have a big impact and make the mix harder to control. Proper gain staging allows your faders to operate in a comfortable and usable range.

For reasons discussed in earlier chapters, most of your faders (with some very few exceptions) should be able to live comfortably at or close to unity (0 dB). When building the mix, if you find a channel that wants to be well below or much higher than unity (lower than -10 dB or higher than 5 dB), you should see that as a major red flag and revisit your input gain level. If all channels are gained so that the input levels are all close to the -18 dB target, the faders should all produce similar output volumes at unity. A notably quite or overly loud channel is indicative of that channel's level being incorrectly gained.

As another note, if, when building the mix, you find that *all* or *most* of the channels are too loud (or quiet) when the faders are at unity, this indicates an issue with the system's master output volume. You should investigate the levels of the main bus and master output matrices and make any necessary adjustments to the volume of the PA as a whole in order to be able to mix with the channel faders at or near unity.

Poor gain structure causes faders to be inconsistent and far from unity

Proper gain structure results in faders that all live very close to "home" (unity)

At this stage, keep things simple. Focus on balance, not perfection. You are aiming for a clear and natural relationship between all elements. If something feels too loud or too quiet, adjust the fader rather than reaching for EQ or compression right away. Additionally, adjusting the balance of the mix my making adjustments to the input gain should rarely be done – only after exhausting all other options within the signal chain. Remember: many issues that seem like tone problems are actually level problems.

As you continue to build the mix, listen to how elements interact with each other. Instruments should support, not compete. Vocals should sit clearly on top without sounding disconnected. The goal is not to make everything equally loud, but to create a balanced and intentional sound.

Active Mixing

One of the most common habits for new engineers is to build a mix at the beginning of a set, decide that it sounds good, and then leave it alone. This is known as *passive mixing*. While it may feel safe, it almost always leads to a mix that slowly becomes unbalanced as the service progresses.

Live audio is constantly changing. Vocalists may sing louder in one song and softer in another. Different vocalists might be leading different songs and, so, different singers need to be louder (or quieter) at different times. A guitarist may switch parts, a keyboard player may change patches, or a speaker may vary their intensity as they talk. Even the energy in the room can affect how things are perceived. Because of this, a mix that sounded right five minutes ago may not be – and often isn't - right anymore. Passive mixing ignores these changes and allows small imbalances to grow into noticeable problems.

Active mixing, on the other hand, means staying engaged with the mix from start to finish. It involves making small, intentional adjustments in real time to maintain clarity and balance. This does not mean constantly moving every fader or overcorrecting every detail. Instead, it means paying attention to what is happening and responding when something shifts.

A good way to think about active mixing is that you are guiding the mix, not setting it on autopilot. If a vocal starts to get buried, you bring it forward. If an instrument becomes too dominant, you pull it back. If the energy of the song builds, you support that movement in a controlled way. If a song is softer and more intimate (or has a quieter "breakdown" in it), you adjust effects to enhance that intimate energy. These adjustments are often subtle, but they make a significant difference in how the mix feels to the listener.

It is also important to avoid the opposite extreme. Active mixing does not mean *overmixing*. Constant, unnecessary movement can be just as distracting as doing nothing at all. The goal is not to draw attention to your adjustments, but to maintain a consistent and natural sound as things change.

As an audio engineer, your job does not end once the mix is built. That is where your job begins. Stay engaged, keep listening, and be willing to make adjustments as needed. A great mix is not something you set once. It is something you maintain.

Conclusion

Starting with good gain structure and a simple, well-balanced mix that you actively monitor will set you up for success in the next steps. Once this foundation is in place, you will be in a much better position to use EQ, compression, and effects in a way that enhances the mix rather than trying to fix it.

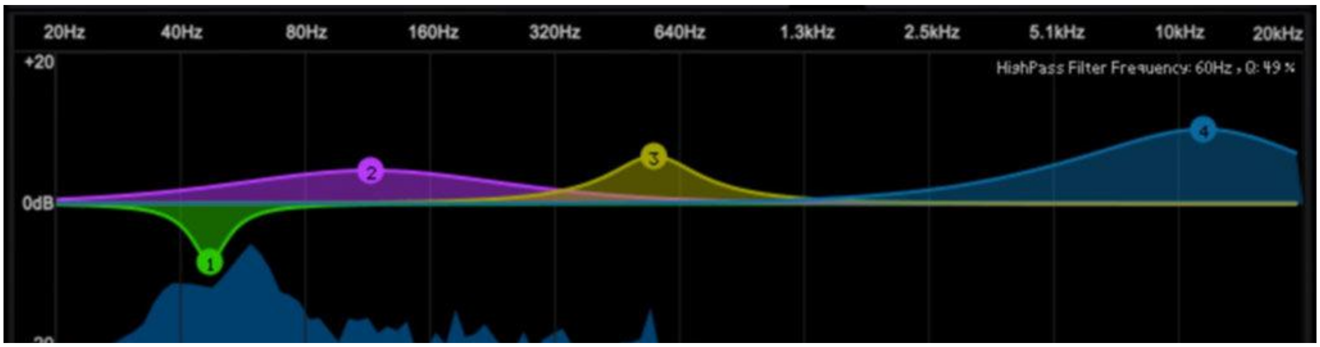
5.3. Fundamentals of EQ for Live Sound

Once a solid mix has been established through proper gain structure and balance, the next step is shaping that mix so each element can be heard clearly. This is where EQ becomes an important tool. Used correctly, EQ helps remove problem areas, create space between instruments, and improve overall clarity. Used incorrectly, it can quickly make a mix sound unnatural or harsh. The goal is not to dramatically change how something sounds, but to help it sit better within the mix.

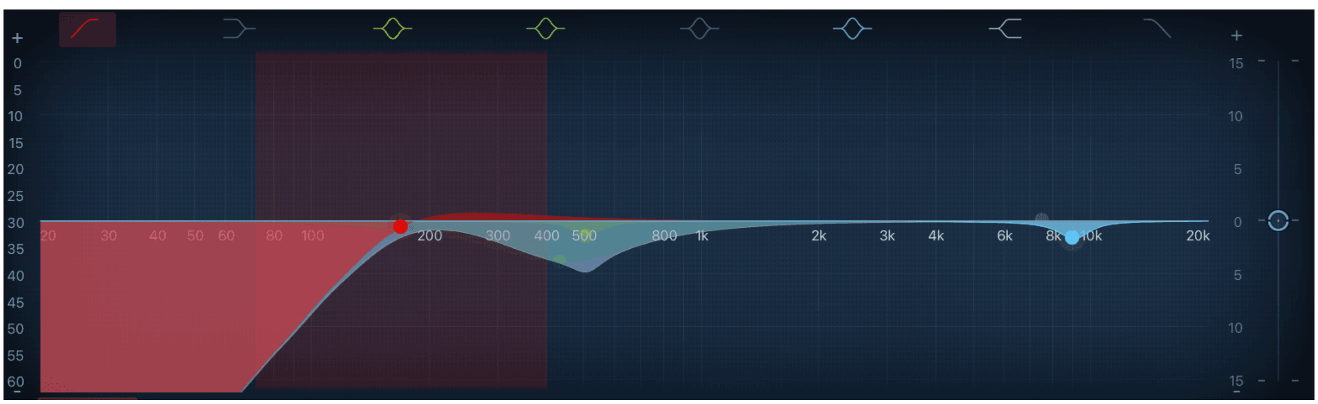
At its core, EQ allows you to adjust different frequency ranges within a signal. Low frequencies carry weight and fullness, mid frequencies contain most of the detail and character, and high frequencies add clarity and presence. Every instrument and voice occupies a range within this spectrum, and when too many elements compete in the same space, the result is often a muddy or cluttered mix. EQ helps separate these elements so each one has its own place.

Subtractive vs. Additive EQ

A helpful principle to follow is to favor *subtractive* EQ over *additive* EQ. Instead of immediately boosting frequencies to make something stand out, start by removing what is unnecessary. Cutting problematic frequencies often results in a more natural and controlled sound. For example, reducing low-frequency buildup can clean up muddiness, while gently cutting harsh mid or high frequencies can make a vocal easier to listen to over time.



Additive EQ – beneficial in some situations, but can easily result in "runaway" frequencies



Subtractive EQ – usually preferred over additive; results in a more natural tone and more control

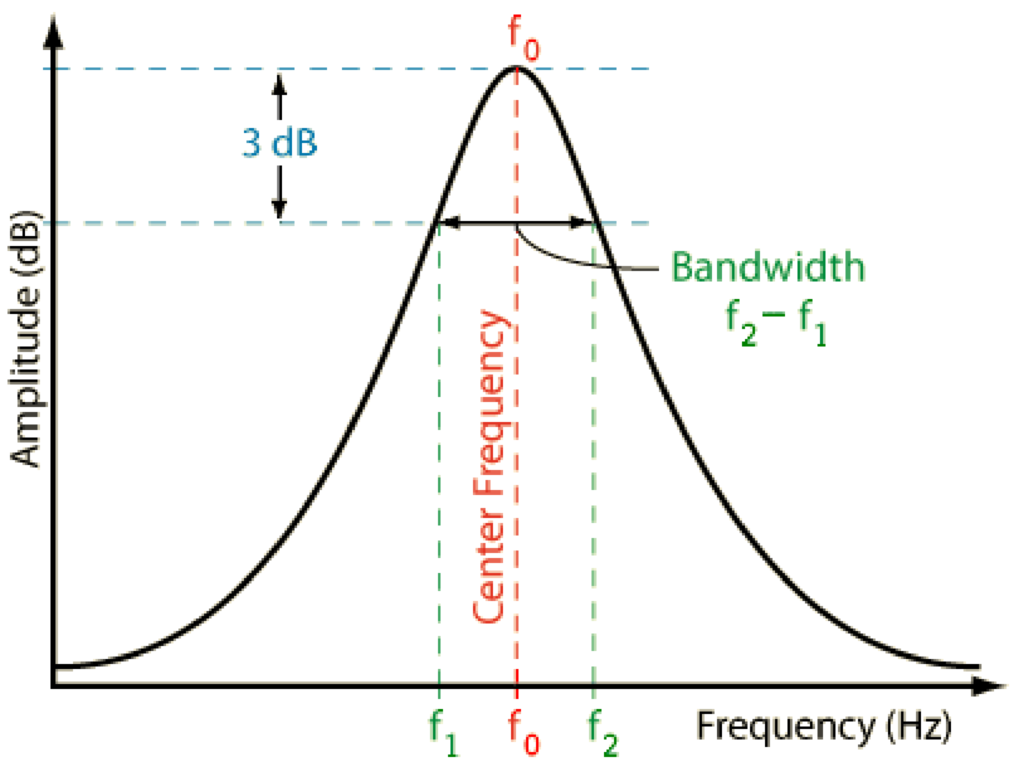
EQ Tools

When working with EQ, it is important to understand the basic controls available to you. While different consoles may present them in slightly different ways, the core tools remain the same. Learning what these controls do will help you make more intentional and precise adjustments instead of guessing or overcorrecting.

One of the primary controls is frequency selection. This determines where in the frequency spectrum you are making a change. Whether you are adjusting low, mid, or high frequencies, you are choosing the specific area you want to affect. Being able to identify problem areas by ear takes practice, but over time you will begin to recognize common frequency ranges, such as muddiness in the low-mids or harshness in the upper-mids.

Another key control is gain, which determines how much you are boosting or cutting at the selected frequency. Positive gain increases that frequency range, while negative gain reduces it. As a general rule, smaller adjustments tend to produce more natural results. Large boosts can quickly make a signal sound artificial, while large cuts can make it feel thin or lifeless. Subtle changes are usually more effective and easier to manage within a full mix.

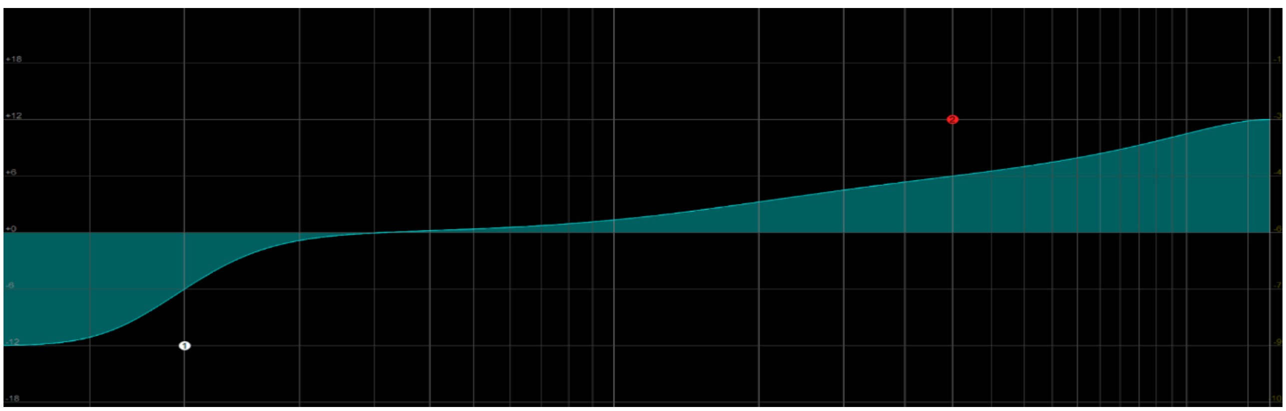
The Q, or bandwidth, control determines how wide or narrow your adjustment is. A wide Q affects a broader range of frequencies and is often used for gentle shaping. A narrow Q targets a very specific area and is useful for addressing problem frequencies, such as ringing or feedback. Understanding how to adjust Q allows you to be more surgical when needed, while still making smooth, musical changes when appropriate.



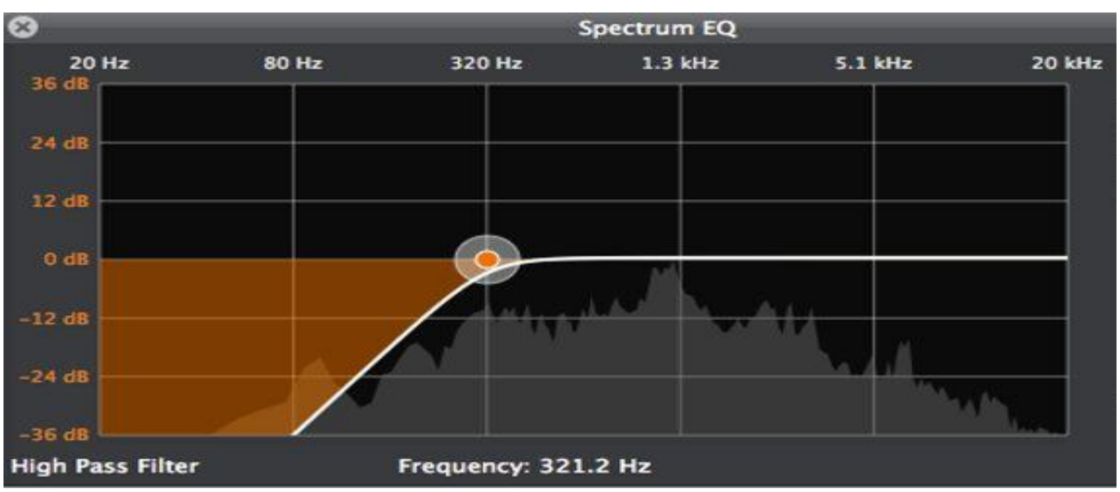
Bandwith ("Q") describes the width of the adjustment; higher Q values result in a wider adjustment

You may also encounter different EQ shapes or filter types. These include options such as bell curves, shelving filters, and high-pass or low-pass filters. A *bell curve* is commonly used for boosting or cutting a specific frequency range. Shelving filters raise or lower everything above or below a certain point, which can be useful for broad tonal adjustments.

Filters remove frequencies beyond a selected cutoff, helping to clean up unwanted low-end rumble or excessive high-frequency noise. One of the most common and effective tools is the *high-pass filter*. This allows you to remove low frequencies that are not needed for a particular source. Many inputs, especially vocals and guitars, do not require deep low-end content, and leaving it in can contribute to a muddy mix. Applying a high-pass filter where appropriate helps create space for instruments that actually belong in that range, such as bass and kick drum. We also use its twin, the *low-pass filter* to remove high frequencies when a channel doesn't use frequencies on the upper end on the spectrum. A classic example of this is using a low-pass filter on a kick drum or bass to prevent unwanted high frequencies from bleeding through.



High-Pass Filter (Left) and High Shelf Filter (Right)



RTA (background) shows result of using high-pass filter at 320 Hz

Shelf filters can be used to increase or, as in this image, decrease the entire high (or low) end of the spectrum without completely cutting it like a pass filter would

As you begin working with these tools, focus on understanding what each control is doing rather than relying on presets or habits. Take the time to experiment and listen carefully to how each adjustment affects the sound. The more familiar you become with these controls, the more confidently you will be able to shape your mix in a way that serves the overall goal of clarity and balance.

EQ Philosophy

It is important to make EQ decisions in the context of the full mix, not in isolation. A channel may sound full and rich on its own, but when combined with everything else, it may take up too much space. Always listen to how changes affect the overall balance rather than focusing only on the individual source. The goal is not to make each channel sound perfect by itself, but to make everything work well together.

As you begin working with EQ, make small and intentional adjustments. Large, aggressive changes are rarely necessary and can introduce new problems. Subtle changes often produce the most natural results. Trust your ears, take your time, and resist the urge to over-process.

EQ is a powerful tool, but it is only effective when built on a strong foundation. If gain structure and balance are not in place, EQ will not fix the problem. When used thoughtfully, however, it allows you to move from a basic mix to one that is clear, defined, and easy to listen to.

5.4. Dynamics Processing

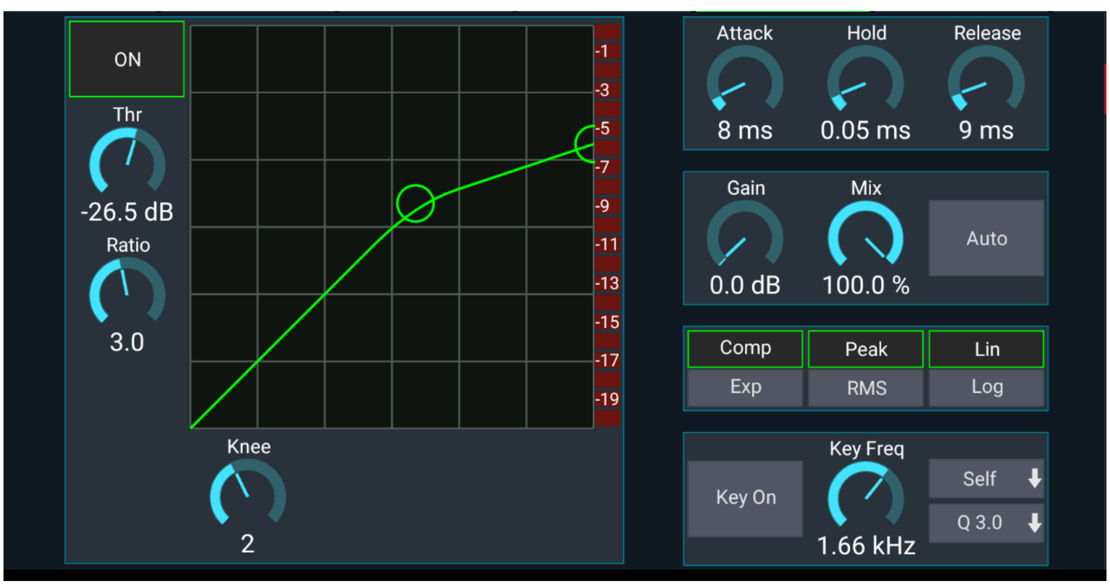
Once your mix is balanced and shaped with EQ, the next step is learning how to manage dynamics. Dynamics refer to the difference between the quietest and loudest parts of a signal. In a live environment, these variations can be significant. A vocalist may sing softly during one moment and then become much louder in the next. Without some level of control, these changes can make a mix feel inconsistent and difficult to follow.

Compression

Compression is a tool used to control these variations by reducing the difference between loud and quiet signals. When a signal exceeds a certain level, the compressor reduces its volume according to the settings you choose. This helps keep things more even and allows important elements, like vocals, to stay present in the mix without constantly adjusting the fader.

There are several key controls that determine how a compressor behaves. The *threshold* sets the level at which the compressor begins to act. Signals below this level are unaffected, while signals above it are reduced. The *ratio* determines how strongly the signal is reduced once it crosses the threshold. A higher ratio results in more noticeable compression, while a lower ratio provides a more gentle effect.

Attack and release control how quickly the compressor responds. The *attack* determines how fast the compressor engages after the signal exceeds the threshold. A slower attack allows the initial impact of a sound to pass through before compression begins, which can help maintain clarity and presence. The *release* determines how quickly the compressor stops reducing the signal after it falls back below the threshold. If the release is too fast or too slow, it can cause the sound to feel unnatural, so it is important to listen carefully and adjust as needed.



The built-in compressor on the X32

In a live church setting, compression is most commonly used on vocals, bass, and speaking microphones. The goal is not to eliminate all dynamic variation, but to make it more manageable. A well-compressed vocal will remain clear and consistent without sounding overly processed. If the compression is too aggressive, the signal can sound flat or unnatural, and it may even introduce unwanted artifacts.

As with EQ, it is important to use compression with restraint. It should support the mix, not dominate it. If you find yourself relying heavily on compression to control a signal, it may be a sign that your gain structure or balance needs attention. Start with simple, moderate settings and make small adjustments while listening in the context of the full mix.

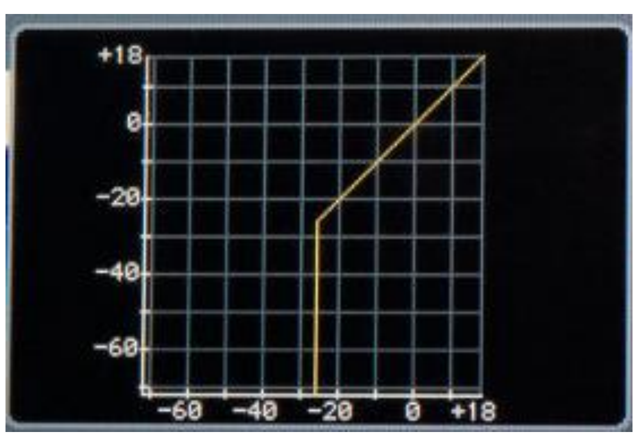
Compression is a powerful tool when used correctly. It helps create stability and consistency, allowing the listener to stay focused on the message and the music. When applied with care, it can make a mix feel more polished and controlled without drawing attention to itself.

Noise Gates

In addition to compression, another useful dynamics tool is the *noise gate*. While a compressor controls signals that are too loud, a gate controls signals that are too quiet. Its purpose is to reduce or eliminate unwanted noise when a source is not actively being used.

A noise gate works by setting a threshold level. When the signal falls below that level, the gate closes and reduces or mutes the output. When the signal rises above the threshold, the gate opens and allows the sound to pass through normally. This can be especially helpful for sources like drums, where you may want to isolate a specific drum without picking up as much bleed from other instruments on stage.

Gates can also be useful on microphones that are not in constant use. For example, if a vocal mic is not being sung into, a gate can help reduce background noise, stage bleed, or low-level feedback. However, this must be used carefully. If the threshold is set too high, the gate may not open properly when the person begins speaking or singing, causing the beginning of words to be cut off.

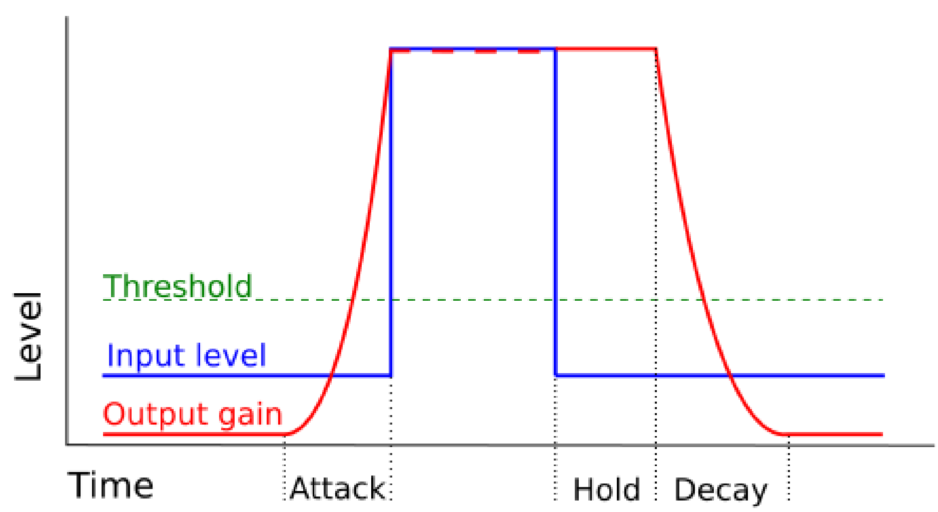


Noise gate example – This gate will only open above -28 dB

Like compression, noise gates include controls that affect how they behave. Attack determines how quickly the gate opens once the signal crosses the threshold. A fast attack allows the sound to come through immediately, while a slower attack can soften the initial transient. Release controls how quickly the gate closes after the signal drops below the threshold. If the release is too fast, the sound may cut off abruptly. If it is too slow, unwanted noise may still come through.

It is important to use gates with intention and restraint. They can be very effective when used to clean up specific sources, but they should not be used as a substitute for proper mic placement or good gain structure. In many cases, especially with vocals and speaking microphones, it is better to rely on careful mixing rather than aggressive gating.

When used appropriately, noise gates can help keep a mix clean and focused. They reduce distractions and allow the important elements to stand out more clearly. As with all processing, the goal is not to make the effect obvious, but to support the overall clarity and consistency of the mix.



Visualization of noise gate settings

5.5. Effects and Sonic Real Estate

Once your mix is balanced, shaped with EQ, and controlled with dynamics, the next step is learning how to manage what we call *sonic real estate*. Every sound in your mix occupies a place, not just in frequency, but in perceived space. Some elements feel close and upfront, while others feel distant and behind. Some things sound like they're over here, but others sound like they're way over there.

Reverb, delay, and panning are the primary tools that allow you to intentionally shape that space and decide where each element sits.

Reverb

Reverb is what gives a sense of environment. It simulates how sound reflects in a room and helps determine how "far away" something feels. A completely dry signal can feel unnatural or disconnected, while a small amount of reverb can help it sit comfortably within the mix. When thinking in terms of sonic real estate, adding reverb is like moving a sound slightly farther back in the room. The more reverb you apply, the more distant that element becomes. This is why lead vocals are usually kept more forward, with just enough reverb to feel natural, while supporting elements can be placed slightly farther back.

Delay

Delay also plays a role in shaping space, but in a more controlled and intentional way. Instead of creating a wash of reflections like reverb, delay creates distinct repeats that can add depth without pushing a sound too far back. A subtle delay can help a vocal feel fuller and more present while still maintaining clarity. In terms of sonic real estate, delay can help an element occupy more space without necessarily moving it out of the spotlight.

A technique you may hear or experiment with is a *delay throw*. This is when delay is applied to a specific word or phrase rather than staying on continuously. Instead of having delay active the entire time, the engineer briefly sends a vocal or instrument to the delay effect at a key moment, allowing that word or phrase to echo and then fade away. This can add emphasis, create space, and enhance musical transitions without cluttering the mix. In terms of sonic real estate, a delay throw allows you to momentarily expand a sound outward without permanently moving it back in the mix. It is a subtle but powerful tool when used sparingly, especially during vocal lines where you want to highlight a lyric without sacrificing clarity.

Panning

Panning is another tool that helps you manage sonic real estate by placing sounds from left to right in the stereo field. Instead of everything coming from the center, panning allows you to spread elements across the mix so they each have their own space. This can make a mix feel wider, more open, and easier to listen to. For example, two guitars playing similar parts can compete with each other if they are both centered, but by placing one slightly to the left and the other to the right, you create separation and clarity.

In a live church setting, panning should be used with intention and moderation. While it can add width and definition, it is important to remember that not everyone in the room hears the mix the same way. Someone sitting on the far left side of the room may hear very little of what is panned hard to the right, and vice versa. Because of this, it is usually best to avoid extreme panning for critical elements like lead vocals or speaking microphones, however, tasteful and appropriate panning of these elements is crucial in order to generate a full and clear mix.

When used thoughtfully, panning helps organize the mix and reduce crowding in the center. It works alongside EQ and level balance to give each element its own place. The goal is not to create a dramatic stereo effect, but to support clarity and make the mix feel natural and well-spaced.

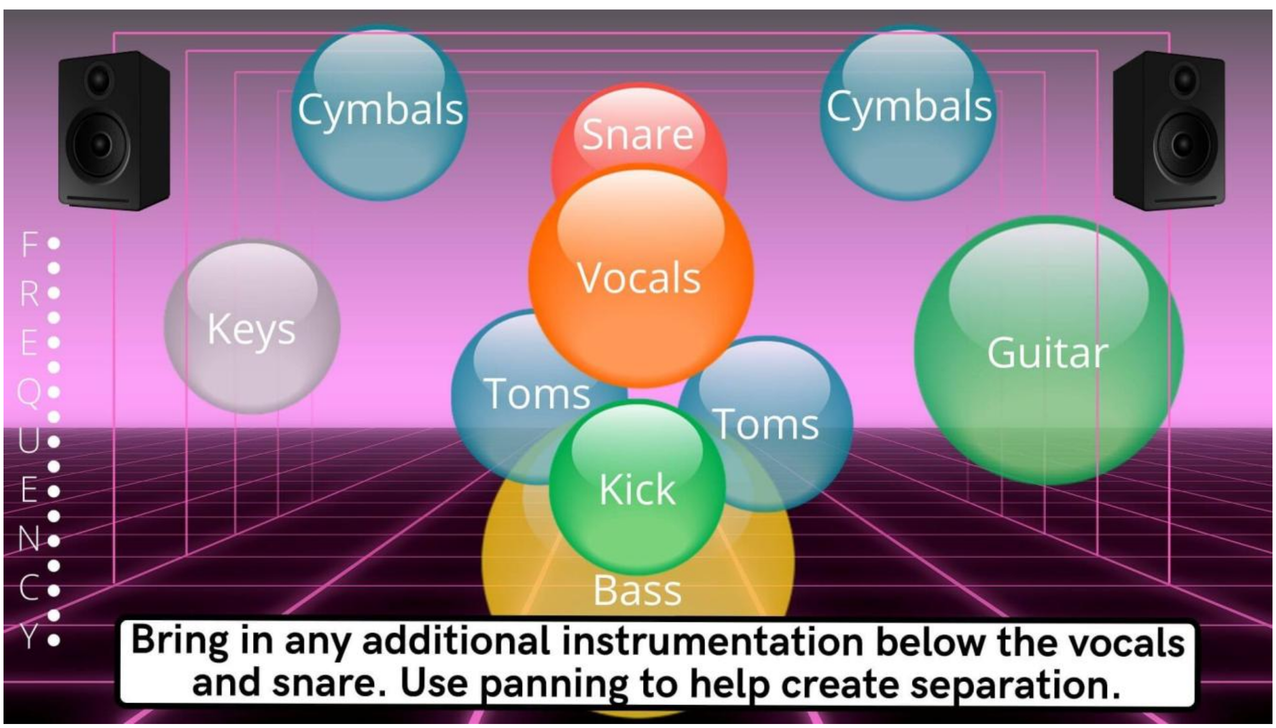
Approaching Sonic Space

A helpful approach is to think of your mix as a three-dimensional space. You are not only deciding how loud something is, but also where it sits from front to back and from left to right. If everything is treated the same way, the mix can feel crowded and flat. By using reverb, delay, and panning wisely, you create separation and depth, allowing each element to have its own place.

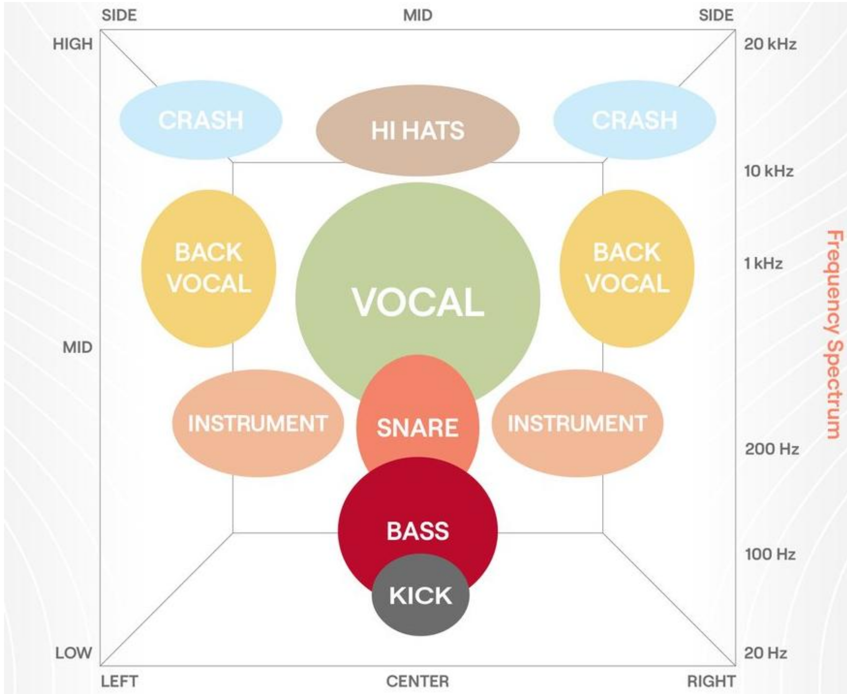
As with all processing, restraint is key. Too much reverb can quickly fill up your available space and make the mix feel distant and unclear. Too much delay can become distracting and pull attention away from what matters most. Extreme panning will make your mix feel unnatural and over-processed. If your mix begins to lose clarity, effects are often the first place to check.

It is also important to build your mix without effects first. Establish balance and clarity, then begin adding reverb and delay to enhance what is already working. Lastly, use panning to widen the stereo field and add clarity. Effects should not be used to fix problems, but to refine and support a solid foundation.

When you approach mixing with an understanding of sonic real estate, reverb, delay, and panning become more than just effects. They become tools for placing each element exactly where it belongs, creating a mix that feels open, intentional, and easy to follow.



Visualization of the stereo image



Stereo image showing how frequency affects perceived height in the sound space

5.6. Mixing for Different Elements

Not every part of a service should be mixed the same way. One of the most important skills you can develop as an audio engineer is learning how to adjust your approach depending on what is happening on stage. The way you mix a full band is very different from how you mix a speaking pastor, and understanding that difference is key to maintaining clarity and consistency throughout the service.

Speech

When it comes to speech, clarity is the highest priority. The goal is simple: every word should be easy to understand without effort. This means keeping the speaking microphone clear, present, and free from distractions. Effects like reverb and delay should be used very sparingly, if at all, since they can reduce intelligibility. Dynamics should be controlled enough to keep the level consistent, but not so much that the voice sounds unnatural. In this context, a clean and straightforward mix is always the best approach.

Music

Music, on the other hand, allows for more creativity and depth. A full band mix involves balancing multiple elements that all contribute to the overall sound. The lead vocal should still remain the primary focus, but the supporting instruments help build energy and emotion. This is where tools like EQ, compression, panning, and effects can be used more freely to shape the mix and create a sense of movement and space. Even so, the goal is still clarity. Each instrument should have its place and should not compete with the others.

It is also important to understand how roles shift within a song. A guitar part may be subtle during a verse and then become more prominent in a chorus. Background vocals may come forward at certain moments and then step back. Active mixing plays a big role here, as you adjust levels to support what is happening musically. The mix should follow the flow of the song rather than staying static.

Transition Moments

Transitions between different parts of the service require attention as well. Moving from music to speech, or from one speaker to another, should feel smooth and natural. This may involve adjusting levels, removing effects, or resetting your focus to match the new priority. Being prepared for these transitions helps avoid distractions and keeps the service moving seamlessly.

Conclusion

Ultimately, mixing for different elements comes down to understanding what matters most in each moment. Whether it is a single voice or a full band, your job is to support that focus and make it as clear and effective as possible. By adapting your approach to fit the situation, you create a mix that serves the entire service rather than treating every moment the same.

5.7. Troubleshooting During a Live Service

No matter how well you prepare, things will go wrong at some point. A microphone may stop working, feedback may suddenly appear, or something in the mix may not sound the way you expect. These moments are a normal part of live audio, and how you respond to them is just as important as how you build your mix.

The first priority in any troubleshooting situation is to remain calm. Panic leads to rushed decisions, and rushed decisions often make the problem worse. Take a breath, listen carefully, and identify what is actually happening before making adjustments. In many cases, a quick and simple change is all that is needed, but only if you understand the problem clearly.

When an issue arises, start with the basics. Check signal flow. Is the microphone on? Is the correct channel selected? Is the fader up? Has something been muted or routed incorrectly? Many problems can be traced back to a simple oversight. Developing the habit of checking these fundamentals first will save time and prevent unnecessary frustration.

Feedback

Feedback is one of the most common challenges during a live service. Feedback is a loop that occurs when sound from a speaker is picked up by a microphone, amplified, and sent back through the speaker again. This cycle repeats very quickly, causing certain frequencies to build up and become much louder than everything else. The result is the high-pitched ringing or squealing sound that is commonly associated with live audio systems.

Feedback usually happens when a microphone is too close to a speaker, when the volume is too high, or when certain frequencies are being emphasized in the system. Because it builds on itself, it can escalate quickly if not addressed. Understanding that feedback is simply a repeating loop helps you respond more effectively by identifying the source and breaking that loop.

When it occurs, resist the urge to make large or random adjustments. Instead, quickly identify the source and reduce it in a controlled way. This may involve lowering a fader, adjusting a monitor level, or making a small EQ cut in the problem frequency. The goal is to resolve the issue without disrupting the rest of the mix.

Feedback Prevention

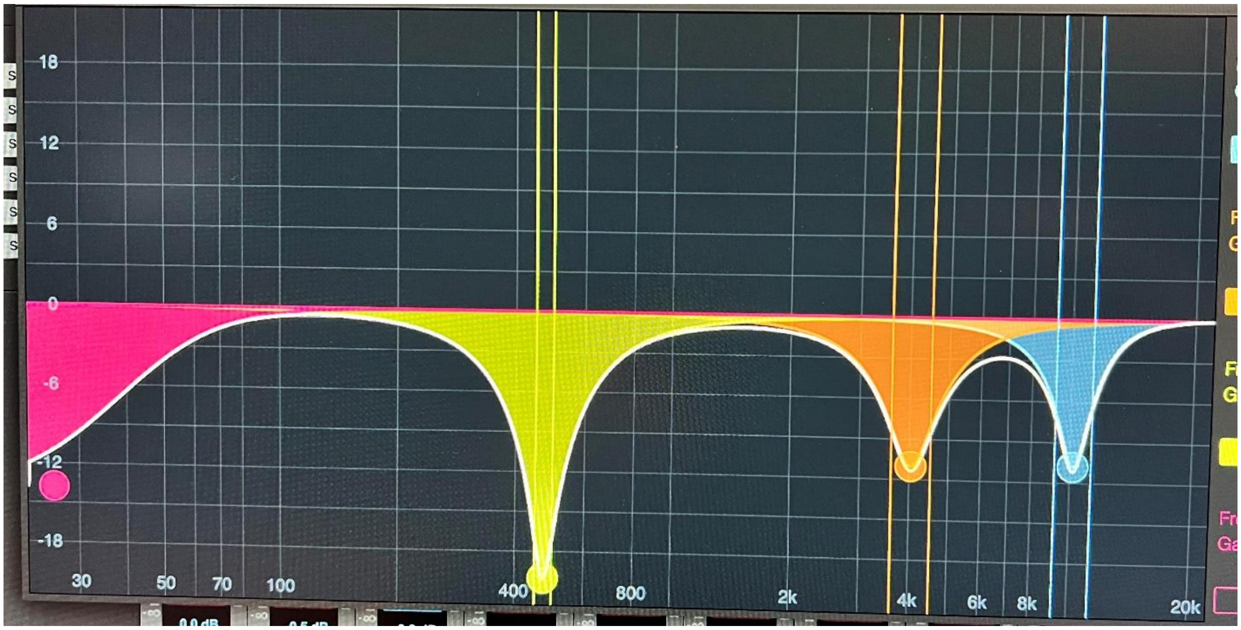
As the old saying goes, an ounce of prevention is worth a pound of medicine. This is certainly the case when it comes to feedback. Experiencing feedback during a service can be abrupt and disruptive and would definitely cause the sound system to fail at one of its primary goals: *remain unnoticed*.

Feedback mostly occurs at very fine, very specific frequencies. Often the frequency range causing feedback is as small as 10-20 Hz. We prevent feedback by ringing out these frequencies from our speakers.

Ringing out speakers (or monitors) is the process of identifying and reducing the frequencies that are most likely to cause feedback before the service begins. This is typically done during soundcheck by gradually raising the level of a microphone until feedback just begins to occur. When you hear that ringing frequency, you can identify it using the spectrum analyzer and use EQ to make a small, narrow cut at that specific frequency, then continue raising the level and repeating the process as needed.

The goal is not to eliminate all feedback completely, but to increase the system's gain before feedback so you have more usable headroom during the service. This is especially important for stage monitors, where microphones are often pointed toward speakers and more prone to feedback.

It is important to make small, precise adjustments rather than large cuts. Over-adjusting with the EQ can negatively affect the overall sound quality and make the system feel thin or unnatural. When done correctly, ringing out helps create a more stable system, allowing you to mix with confidence and reduce the likelihood of feedback during live operation.



Feedback correction (extreme example)

Other Thoughts on Troubleshooting

It is important to know when not to overreact. Not every small imperfection needs immediate correction. If a minor issue will pass quickly or is barely noticeable to the congregation, it may be better to leave it alone rather than risk creating a larger distraction. Good judgment comes with experience, and learning to prioritize what truly matters is part of that process.

Preparation plays a major role in effective troubleshooting. Knowing your console, understanding your signal flow, and being familiar with your system all contribute to faster and more confident responses. Over time, you will begin to recognize common issues and address them more efficiently.

Ultimately, troubleshooting is about maintaining control in unpredictable situations. By staying calm, thinking clearly, and focusing on simple solutions, you can handle problems in a way that keeps the service moving smoothly and minimizes distractions.

5.8. Developing Your Ear

All of the tools and techniques in this chapter are important, but they are only as effective as your ability to listen. Developing your ear is one of the most valuable skills you can invest in as an audio engineer. It is what allows you to move beyond simply operating the console and begin making intentional, informed decisions.

Listening well means more than just hearing sound. It means learning to recognize balance, clarity, and tone. It involves asking questions as you mix. Can I clearly understand the vocal? Are any instruments overpowering others? Does anything sound muddy or harsh? Over time, these questions become instinctive, and your ability to identify issues will improve.

One of the best ways to develop your ear is through focused practice. Instead of trying to listen to everything at once, choose one element to pay attention to. Spend time listening specifically to the vocal, then shift your focus to the low end, then to the overall balance. This kind of intentional listening helps train you to recognize details that might otherwise go unnoticed.

It is also helpful to listen to well-produced music outside of a live setting. Pay attention to how different elements are balanced and how space is created within the mix. While a live environment presents different challenges, these references can help you develop a sense of what a clear and balanced mix should sound like.

Another important part of developing your ear is consistency. The more time you spend mixing, the more familiar you will become with your system and your room. What once felt overwhelming will begin to feel natural. Patterns will emerge, and your confidence will grow as you learn what works and what does not.

A great way to test yourself is to actively listen to the mix on weeks when you are *not* operating the console. Try and pick out things that need adjusting. Think about specific things you notice within the mix. Is it balanced? Does that vocal sound like it’s got a bit too much high end? Did you hear the volume level change when they keyboard switched to a different patch? See if you can spot whenever the engineer makes a small, nuanced change in the level of an element or effect.

Finally, be patient with yourself. This skill takes time to develop, and progress may feel slow at times. That is normal. Stay engaged, keep listening, and continue learning from each experience. As your ear improves, so will your ability to create mixes that are clear, balanced, and effective.

6. Broadcast Mixing

Introduction

Up to this point, our focus has been on mixing for the room. Every decision we have discussed has been centered around what the congregation hears in the building. In this chapter, we shift that focus to a different audience—those who are not physically present but are still fully participating in the service through a broadcast.

Broadcast mixing requires a different way of thinking. In the room, sound naturally exists before it ever reaches the speakers. The band is on stage, voices carry through the air, and the room itself contributes to what people hear. The sound system simply supports and reinforces what is already happening. In a broadcast, none of that exists. The listener hears only what you send them. There is no natural acoustic environment to fill in the gaps, which means every element must be built intentionally.

Because of this, a mix that sounds great in the room does not always translate well to a livestream, radio feed, or assisted listening system. Elements that feel balanced in the room may feel distant or unclear in headphones. A mix that feels full in person may sound dry or empty in a broadcast. This is why we create a separate broadcast mix rather than relying entirely on the house mix.

As you work through this chapter, it is important to adopt the right mindset. You are now mixing for someone who cannot feel the energy of the space and cannot rely on natural sound. Your job is to create a clear, balanced, and engaging experience using only what comes through the speakers or headphones. When done well, the broadcast feels natural and immersive. When done poorly, it can feel disconnected or difficult to follow.

Do not let this feel overwhelming. The same foundational principles still apply. You will still build a mix using gain, balance, EQ, and dynamics. The difference is in how you apply those tools. As with everything else, take it one step at a time and focus on listening carefully. With practice, this way of mixing will become just as natural as mixing for the room.

6.1. Understanding the Broadcast Signal Flow

Before we begin shaping a broadcast mix, it is important to understand how the signal is routed within our system. This connects directly back to what we learned earlier about signal flow. If you understand where the signal is coming from and where it is going, you will be able to make more confident and intentional decisions.

In our setup, the broadcast mix is created using a dedicated stereo bus on the console (the “Broadcast Bus”). Each input channel is sent to this bus using a post-fader send. This means that the adjustments you make to the main mix, such as fader movements, will also affect the broadcast mix. In many ways, this allows the broadcast to follow along with what is happening at front of house, providing a consistent foundation.

However, these sends are still independently adjustable. This is an important detail. While the broadcast mix is built from the same sources as the house mix, it is not locked to it. You can increase or decrease how much of each channel is sent to the broadcast bus without affecting what is heard in the room. For example, you may choose to send more reverb, more room microphones, or bring certain elements forward to better suit a broadcast listener.

Once the broadcast mix is created on the stereo bus, it is then routed to multiple outputs using matrices. Each matrix serves a specific purpose. One feeds the online livestream, one feeds the radio broadcast, and another feeds the assisted listening system. While they all originate from the same broadcast mix, each output may be adjusted slightly or processed differently to meet the needs of its audience.

Understanding this flow is critical. It allows you to see that the broadcast mix is not a completely separate system, but rather a carefully controlled extension of your main mix. You are building on the work you are already doing, while making intentional adjustments to ensure that the final result translates well to listeners outside the room.

6.2. FOH vs. Broadcast: Key Differences

One of the most important concepts to understand is that front of house and broadcast mixing serve two very different purposes. In the room, the sound system is working alongside natural acoustics. The congregation hears the stage directly, and the speakers simply reinforce and balance what is already present. In a broadcast, that natural sound does not exist. The listener hears only what is sent through the mix.

Because of this, elements that feel balanced in the room may not translate well to a broadcast. Vocals are a common example. In the room, the natural voice of the singer or speaker often fills in some of the clarity. In a broadcast, that natural reinforcement is gone, which means vocals often need to be more forward and more consistent in level. The same is true for instruments. What feels full in the room may feel thin or disconnected in headphones.

Another key difference is consistency. In the room, small fluctuations in level are often less noticeable because of the environment. In a broadcast, those same changes can feel much more dramatic. This means the broadcast mix often requires tighter control and more attention to detail.

It is important to recognize that a good FOH mix does not automatically result in a good broadcast mix. While the two are connected, they require different priorities. Learning to hear those differences and adjust accordingly is a major step in becoming a well-rounded audio engineer.

6.3. Building and Shaping the Broadcast Mix

The broadcast mix begins with your front of house mix as a foundation. Because the sends to the broadcast bus are post-fader, any adjustments you make at FOH will carry over. This is helpful, as it keeps both mixes moving together and reduces the need to build something entirely from scratch.

However, this is only the starting point. From there, you will shape the broadcast mix using the individual sends from each channel. This is where you begin making intentional decisions based on what the broadcast listener needs to hear.

A common adjustment is bringing vocals slightly forward in the broadcast mix. Since the listener does not have the benefit of hearing the room, vocals must be clear and present at all times. You may also choose to increase reverb or add more room microphones to create a sense of space. Without this, the mix can feel dry and disconnected.

This process requires active mixing. Just like at front of house, the broadcast mix should not be set once and left alone. As the service progresses, you will need to make small adjustments to maintain balance and clarity. If a vocal becomes less present, you bring it forward. If an instrument becomes too dominant, you adjust accordingly. The broadcast mix should follow the same intentional movement as your FOH mix, while still being tailored to its own environment.

Think of this as a shared mix with independent refinement. You are not starting over, but you are also not locked in. You are taking what works in the room and shaping it into something that translates well to listeners outside of it.

6.4. Space and Dynamics in Broadcast Mixing

When mixing for broadcast, the concept of sonic real estate becomes even more important. Since the listener is not physically present, you must create a sense of space entirely within the mix itself. Without careful attention, the result can feel flat or lifeless.

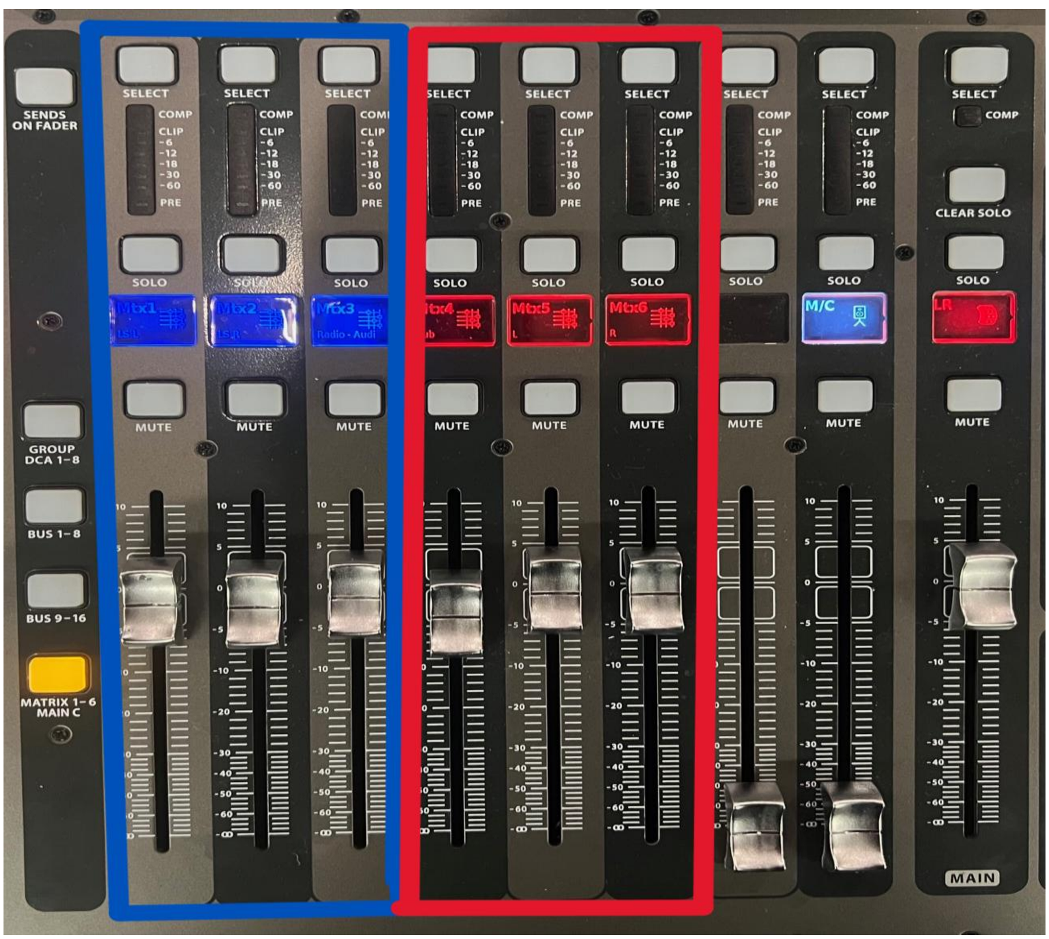
Reverb and ambience play a key role in this. Adding a bit more reverb than you would use in the room can help recreate a sense of environment. Room microphones can also contribute to this, allowing the listener to feel connected to what is happening rather than isolated from it. The goal is not to overwhelm the mix with effects, but to create a natural sense of depth.

At the same time, dynamics must be managed more tightly than in a live setting. Large swings in volume can be uncomfortable for someone listening through headphones or a home speaker system. This is where compression becomes especially useful. A well-controlled mix will feel steady and easy to listen to, even as the energy of the service changes.

Balancing space and clarity is key. Too little ambience can make the mix feel sterile, while too much can reduce intelligibility. Too much dynamic variation can feel inconsistent, while too much compression can make the mix feel flat. The goal is to find a balance that feels natural, controlled, and engaging.

6.5. Matrix Outputs and their Purpose

By now you should be familiar with the three matrices that control the PA system in the room: the Sub Matrix and the L and R Main Matrices. We also use matrices to send our broadcast mix to its different destinations. Each of these matrix outputs serves a specific audience, and understanding their purpose will help you make better decisions.



Matrix array – Livestream, radio, and assisted listening matrices in blue; Sub and L/R matrices in red

The *livestream matrices* is typically the most complete version of the broadcast mix. It should feel full, balanced, and engaging, as it represents the primary way people experience the service remotely. This is where your attention to detail in space, balance, and dynamics is most noticeable.

The *radio matrix* places a higher priority on clarity and consistency. Since listeners may be in a variety of environments, the mix should be easy to understand and free from distractions. This may mean slightly reducing effects or ensuring that speech is especially clear and present.

The *assisted listening matrix* is designed to provide a clean and intelligible signal for those who need it. In this case, simplicity is often best. A clear, direct mix without excessive processing will be the most effective.

While all of these outputs originate from the same broadcast bus, small adjustments at the matrix level can help tailor each one to its specific purpose.

6.6. Monitoring the Broadcast Mix

One of the most important habits you can develop is actively monitoring the broadcast mix. It is not enough to assume that what you are sending sounds correct. You need to listen to it directly.

This is most often done using headphones connected to the console. Listening this way allows you to hear the mix in a way that is much closer to what the broadcast audience is experiencing. It can quickly reveal issues that may not be obvious in the room, such as vocals being too low, too much low end, or a lack of clarity.

Whenever possible, it is also helpful to check the actual stream. Even a brief check can confirm that everything is translating as expected. This step helps catch issues before they become noticeable to the audience.

Avoid relying solely on your FOH perspective. The room can mask or compensate for things that will stand out in a broadcast. By intentionally monitoring the broadcast mix, you ensure that you are making decisions based on what the listener is actually hearing.

6.7. Common Broadcast Mixing Mistakes

There are several common mistakes that can negatively affect a broadcast mix. One of the most frequent is creating a mix that is too dry. Without enough ambience or space, the mix can feel flat and disconnected.

Another common issue is vocals being too low. Since the listener does not have the benefit of hearing the room, vocals must be clearly present at all times. If they are buried in the mix, the entire experience suffers.

Low-end balance is another area to watch carefully. Too much low frequency content can make the mix feel muddy, especially on smaller speakers or headphones. At the same time, too little can make the mix feel thin. Finding the right balance is key.

Ignoring room microphones or ambience can also lead to a less engaging mix. Without some sense of environment, the broadcast can feel isolated from what is happening in the room.

Finally, one of the biggest mistakes is treating the broadcast mix as passive. Just like FOH, it requires attention and adjustment throughout the service. A mix that is set once and left alone will rarely stay balanced.

6.8. Final Thoughts: Mixing for the Unseen Audience

Broadcast mixing requires a shift in perspective. You are no longer mixing for a room full of people, but for individuals listening through speakers, headphones, or devices in completely different environments. They rely entirely on what you provide.

This is both a challenge and an opportunity. While you lose the natural support of the room, you gain the ability to shape the entire listening experience. Every element, every balance decision, and every adjustment contributes directly to what the listener hears.

Approach this role with the same mindset we have emphasized throughout this manual. Stay engaged, listen carefully, and make intentional decisions. Do not be discouraged if it feels different at first. With time and practice, you will begin to recognize what works and develop confidence in your approach.

When done well, a broadcast mix can feel just as engaging and meaningful as being in the room. By staying focused on clarity, balance, and purpose, you help create an experience that allows people to fully participate, no matter where they are.

7. Monitor Mixing & Stage Sound

7.1. Understanding Monitor Systems & FOH Differences

Before you can build effective monitor mixes, it is important to understand how they differ from what you are doing at front of house. While both involve mixing audio, the purpose and approach are not the same.

A monitor mix is what the performers hear on stage. Instead of one unified mix for the entire room, you are often creating multiple mixes, each tailored to a specific person or group. These mixes are typically sent through monitor busses and routed to either floor wedges or in-ear monitors. Each bus functions as its own independent mix, allowing you to customize what each musician hears.

There are two primary types of monitor systems. Floor *wedges* are speakers placed on the stage and aimed toward the performers. They are simple and effective, but they introduce additional sound into the room and increase the risk of feedback. *In-ear monitors*, on the other hand, deliver sound directly to the performer through earphones. They provide more isolation and control, but they require more careful setup and attention to detail.

The key difference between FOH and monitor mixing comes down to perspective. At FOH, you are creating one mix for the entire congregation. On stage, you are creating multiple mixes based on individual needs. What sounds balanced to one musician may not work for another. Because of this, monitor mixing is inherently more subjective.

It is also important to recognize that stage sound directly affects FOH. Excessive volume from wedges can spill into microphones and muddy the overall mix. A well-controlled stage leads to a cleaner and more manageable front of house mix. Understanding this relationship will help you make better decisions as you build monitor mixes.

7.2. Building a Monitor Mix

When building a monitor mix, the goal is not to create a perfect or polished sound. The goal is to give each performer what they need to perform confidently. This usually starts with the most important element: their own voice or instrument.

Begin by asking what the performer needs to hear first. In most cases, this will be their own vocal or instrument. Set that at a comfortable level, then begin adding other elements one at a time. Keep the mix simple and focused. It is better to have a few clear elements than a crowded mix where nothing stands out.

One of the most important concepts in monitor mixing is *gain before feedback*. Because monitors are often positioned close to microphones, they are more prone to feedback than the main system. As you increase volume on stage, you are also increasing the risk of creating a feedback loop.

To manage this, pay close attention to microphone placement and monitor positioning. Directional microphones should be aimed away from wedges whenever possible. Keep monitor levels as low as possible while still meeting the performer's needs. The more volume you add, the less control you have.

Ringing out monitors can also help create stability. By identifying and reducing frequencies that are prone to feedback, you increase the usable headroom of the system. However, this should be done carefully. Small, precise adjustments are far more effective than aggressive cuts.

Ultimately, building a monitor mix is about balance and control. Give the performer what they need, but avoid adding unnecessary volume or complexity. A clean and simple mix will always perform better than one that is pushed too far.

7.3. Common Monitor Mixing Mistakes

There are several common mistakes that can make monitor mixing more difficult than it needs to be. One of the most frequent is using too much volume on stage. When everything is loud, clarity is lost, feedback becomes more likely, and the overall mix suffers. More volume does not solve problems. In most cases, it creates them.

Another common issue is overloading the mix with too many elements. It can be tempting to give a performer everything that is happening on stage, but this often results in a cluttered and confusing mix. Most musicians only need a few key elements to perform well. Keeping the mix simple helps them focus and reduces unnecessary noise.

Ignoring feedback risks is another mistake. If monitors are pushed too hard or positioned poorly, feedback will eventually occur. Waiting until it becomes a problem during the service is not the right approach. Being proactive with levels, placement, and EQ will prevent many of these issues before they start.

Finally, trying to mix monitors the same way as FOH can lead to frustration. Monitor mixes are not about creating a polished sound for an audience. They are about meeting individual needs. What sounds "good" to you may not be what the performer needs. Learning to shift your perspective is key.

7.4. Serving the Stage

Monitor mixing is, at its core, an act of service. Your goal is not to create something impressive, but to support the people on stage so they can do what they have been called to do. When a musician can hear clearly and comfortably, they perform with greater confidence and consistency.

A well-managed stage leads to better outcomes everywhere else. Musicians who are not struggling to hear themselves will sing and play more naturally. This, in turn, makes your job at front of house easier. Everything is connected.

Approach monitor mixing with the same mindset we have emphasized throughout this manual. Stay attentive, keep things simple, and make intentional decisions. You are there to support, not to control. When you do this well, it often goes unnoticed—and that is a good thing.

By serving the stage effectively, you help create an environment where both the musicians and the congregation can engage fully in the service.

PART 4: SUNDAY WORKFLOW

8. Service Workflow & Sunday Readiness

Introduction

By this point, you have learned how sound works, how the system is structured, how to operate the console, and how to build a mix. This chapter focuses on something just as important: how all of that comes together during an actual service.

A strong understanding of audio concepts is only part of what makes an effective engineer. Consistency, preparation, and awareness are what allow those concepts to be applied successfully week after week. The goal of this chapter is to help you develop a clear and repeatable approach so that you are not reacting to problems as they happen, but preparing for them in advance.

In a live service environment, there is very little margin for error. Moments are happening in real time, and there are no second chances. Because of this, the best engineers are not the ones who make the most adjustments, but the ones who are the most prepared. When you know what to expect and have a plan in place, you are able to stay focused and make better decisions.

As you read through this chapter, keep in mind that the goal is not perfection. The goal is consistency. Small, repeatable habits will have a greater impact over time than occasional moments of excellence. By approaching each service with intentionality and discipline, you create an environment that is reliable, distraction-free, and supportive of everything happening on stage.

8.1. Pre-Service Preparation

A successful service begins long before the first note is played or the first word is spoken. Preparation is what allows everything else to run smoothly. Taking the time to get things ready ahead of time will prevent many of the problems that might otherwise arise during the service.

Start with personal readiness. Arrive early enough to give yourself time to think clearly and work without rushing. A calm and focused mindset is just as important as any technical preparation. When you are rushed, it becomes much easier to overlook small details that can turn into larger issues later.

Next, ensure that the system itself is ready. Confirm that the console is functioning properly, that the correct scene is loaded, and that all routing is as expected. Take a moment to look over the stage and verify that microphones, cables, and instruments are in place and properly connected. Small checks at this stage can save significant time later.

It is also important to review the flow of the service. Knowing what is coming allows you to anticipate needs rather than react to them. Be aware of who will be speaking, what music will be played, and any transitions that may require special attention.

As part of your preparation, work through a consistent checklist. Verify that all microphones are tested and producing clean signal. Confirm that playback devices are functioning properly. Check that monitor mixes are active and that the broadcast mix is working as expected. Developing a routine ensures that nothing is forgotten, even on busy or stressful days.

Preparation is not about doing something new each time. It is about doing the same essential things consistently. Over time, this routine will become second nature and will give you confidence that you are ready for whatever the service may bring.

8.2. Soundcheck and Rehearsal Flow

Once preparation is complete, the next step is soundcheck and rehearsal. This is your opportunity to build a strong foundation before the service begins. A well-executed soundcheck will make everything that follows much easier.

Soundcheck First

Before rehearsal begins, you should do a proper soundcheck. This is important so that if any adjustments need to be made or any issues come up, you don’t have to interrupt rehearsal to address them.

Begin with a line check to ensure that all inputs are working properly. This involves confirming that each microphone and instrument is producing signal and is routed correctly. This step is simple, but it is critical. It ensures that there are no surprises later.

After confirming signal, move into soundcheck. This is where you begin building your mix. Start with gain structure, then establish a basic balance using faders. As you work, keep things simple. You are not aiming for a perfect mix at this stage, but rather a solid starting point. This soundcheck should transition naturally into the rehearsal time.

Rehearsal

Monitor mixes should be addressed early during rehearsal. Musicians need to be comfortable on stage in order to perform well. Taking the time to get their mixes right now will reduce distractions later and allow them to focus on what they are doing.

Pay attention during rehearsal and take note of any potential issues. This could include problem frequencies, inconsistent levels, or transitions that may require extra attention. Identifying these ahead of time allows you to address them before they become distractions during the service.

Soundcheck is not just about adjusting levels. It is about preparing yourself to handle what is coming. The more intentional you are during rehearsal, the more confident you will be when the service begins. Rehearsal is a great opportunity for you to familiarize yourself with the set list so you can know when relevant elements are going to happen withing songs. For example, paying attention during rehearsal will allow you to anticipate "down" parts of songs so you can be prepared and ready to push more reverb or delay. Rehearsal gives you the opportunity to recognize, for example, that a different vocalist is leading the second song. Then, when service time comes, you know ahead of time that you'll need to push that singer's channel during that song. You can only learn these things by actively participating in the rehearsal process.

8.3. Mixing During the Service

When the service begins, your role shifts from preparation to execution. At this point, your focus should be on maintaining clarity, consistency, and awareness from start to finish.

Stay engaged at all times. It can be tempting to relax once a mix is established, but live audio requires constant attention. Continue listening actively and make small adjustments as needed. Remember the principle of active mixing. The mix should follow what is happening on stage, not remain static.

Prioritize what matters most in each moment. During speech, clarity is the goal. During music, balance and energy become more important. Being able to shift your focus depending on what is happening will help keep the mix aligned with the purpose of each part of the service.

Transitions require special attention. Moving from music to speaking, or from speaking to video playback, should feel smooth and intentional. Avoid abrupt changes in volume or tone that can distract from the moment. Preparing for these transitions ahead of time will help you execute them more cleanly.

Throughout the service, remain calm and focused. Not every moment will be perfect, and that is okay. The goal is to maintain control and keep the overall experience consistent. Small adjustments made at the right time will often have the greatest impact.

8.4. Post-Service Responsibilities

Once the service has ended, there are still important responsibilities to complete. Taking care of these tasks ensures that the system remains reliable and ready for the next use.

Begin by following proper shutdown procedures, as outlined earlier. This protects the equipment and prevents unnecessary wear or damage. Never rush this process, even if you are tired or ready to leave.

If necessary, reset the console to a known starting point. This may involve saving changes, reloading a scene, or clearing temporary adjustments made during the service. Leaving the console in a clean and predictable state helps the next engineer start with confidence.

Take a moment to reflect on the service. Consider what went well and what could be improved. If any issues occurred, make note of them so they can be addressed before the next service. Continuous improvement comes from paying attention to these details over time.

Finally, treat the equipment with care. Return microphones, cables, and other gear to their proper place. A well-maintained system is easier to operate and more reliable in the long run.

8.5. Consistency and Long-Term Growth

Becoming a strong audio engineer is not about a single service. It is about showing up consistently and improving over time. The habits you develop from week to week will shape your effectiveness far more than any individual moment.

Consistency begins with routine. Following the same preparation steps, approaching each service with the same mindset, and paying attention to the same details will create a stable and reliable process. Over time, this consistency reduces mistakes and builds confidence.

Growth comes from reflection and intentional learning. Each service provides an opportunity to improve. By paying attention to what works and what does not, you begin to develop a deeper understanding of both the system and the environment you are working in.

It is also important to remain patient. Progress may feel slow at times, but steady improvement over time leads to meaningful results. Focus on doing the small things well, and trust that those efforts will add up. In the end, excellence in this role is built on faithfulness in the details. By approaching each service with care, consistency, and a willingness to grow, you contribute to an environment that is dependable, distraction-free, and centered on what matters most.

PART 5: ADVANCED TOPICS

9. Virtual Playback

Introduction

Virtual playback is one of the most practical and valuable tools you can use as an audio engineer. It allows you to record a rehearsal and then play it back through the console as if the band were still on stage. Instead of being limited to real-time decisions, you now have the ability to slow down, listen carefully, and make adjustments without pressure.

This is especially useful in a church environment. During rehearsal, your attention is often divided between multiple responsibilities such as building the mix, communicating with the team, and preparing for the service. Virtual playback gives you a second opportunity to evaluate what you heard and improve it before the service begins.

It is also one of the best tools for training. New engineers can practice mixing without the stress of a live service, and experienced engineers can refine their skills in a controlled environment. Because the same material can be played repeatedly, it creates a consistent and reliable learning experience.

9.1. Virtual Playback Signal Flow

To use virtual playback effectively, you need to understand what is happening inside the console when it is enabled. Under normal conditions, the X32 console receives input from the stage box or the inputs on the back of the mixer. Microphones, instruments, and playback devices are all routed into the console through those physical inputs. When you switch to virtual playback, you are replacing those live inputs with audio coming from your computer via USB.

From the console's perspective, nothing else changes. Each channel still behaves as if it is receiving a live signal. This means you can use EQ, compression, gates, effects, faders, and sends exactly the same way you would during a service. However, there is one very important exception on the X32: gain (preamp) settings do not behave the same way during virtual playback. When you are in virtual playback mode, the console is no longer controlling the physical preamps on the stage box. Because of this:

- Any gain changes you make during virtual playback will not carry over when you switch back to live inputs
- The console will revert to the original preamp gain settings that were in place before playback

This is a known limitation of the X32 and can be confusing if you are not aware of it. All other adjustments, like EQ, compression, fader levels, effects, and sends, will remain as you set them. Only gain must be re-adjusted once you return to live inputs. Because of this, it is best to treat gain during virtual playback as a reference rather than a final adjustment. If you find that a channel needs more or less gain, make a note of it and apply that change once you are back on live inputs.

9.2. Recording a Rehearsal for Playback

To use virtual playback, you must first capture a recording. In our setup, this is done using a computer running REAPER, a digital audio workstation (DAW) that allows us to record and play back multitrack audio. The basic setup is:

- The X32 is connected to the computer via USB
- The console sends individual channels (multitrack audio) to REAPER
- REAPER records each channel to its own track

To set up REAPER for recording:

- Open REAPER on the computer
- Open the Virtual Playback Template file, which is already set up with the correct settings
- Save a new copy of the REAPER project (to prevent overwriting the template file)
- Ensure signal is present (you should see meters moving when audio is coming in)

Recording:

- Press the record button in REAPER
- Allow the rehearsal to play through naturally
- When finished, press stop and save the project

It is important to treat this recording like a real service. Make sure gain levels are set properly and that all channels are functioning correctly. A clean recording will make your playback session much more useful.

To playback your recorded session:

- To play back the recording, simply press play in REAPER
- The audio will be sent back to the X32 via USB, ready for virtual playback

9.3. Engaging Virtual Playback on the X32

Switching the console to virtual playback mode involves changing the input routing from the stage box to the USB return from your computer.

Steps to Enable Virtual Playback:

- Press the Routing button on the X32
- Navigate to the Inputs tab
- Toggle input preset from “Record” to “Playback”

Next, verify the virtual playback signal.

- Start playback in REAPER
- Confirm that channel meters on the X32 are active
- Bring up a fader and verify that audio is passing through the system

To return to live inputs:

- Go back to Routing → Inputs
- Toggle the input preset back to “Record”

This step is critical. Always double-check that you are back on live inputs before the service begins.

9.4. Applications of Virtual Playback

Once virtual playback is running, you can begin using it to improve your mix. This is where the real value of this tool becomes clear.

Start by treating the playback as if it were a live service. Bring up your mix and listen carefully to the overall balance. Focus first on the most important elements, such as vocals and speech, and ensure they are clear and present. From there, evaluate how instruments are sitting in the mix. Because you are not under time pressure, you can take a more intentional approach. Listen for specific issues:

- Vocals need to be more forward
- Effects need to be adjusted
- The mix needs more space or ambience
- Certain elements are too loud or too quiet

Take the time to refine these details. Unlike a live rehearsal, you can pause, rewind, and repeat sections as needed.

It is important to remain intentional and avoid overmixing. Not every detail needs to be perfect. Focus on the changes that have the greatest impact on clarity and balance.

9.5. Training and Practice with Virtual Playback

Virtual playback is not only a tool for refining mixes, but also for developing skill. For new engineers, it provides a safe environment to learn the console. You can practice building mixes, adjusting EQ, and working with dynamics without the pressure of a live service. Because the same material can be replayed, it allows for consistent learning and improvement.

For more experienced engineers, it provides an opportunity to focus on specific areas of growth. This might include improving active mixing, refining transitions, or developing a better sense of balance. To get the most out of practice sessions, approach them with a clear goal. For example:

- Focus on improving vocal clarity
- Practice balancing a full band
- Work on using EQ more effectively

By focusing on one area at a time, you will see more consistent improvement.

9.6. Limitations of Virtual Playback

While virtual playback is extremely useful, it is important to understand its limitations. The most obvious limitation is that playback does not fully replicate a live environment. There is no real-time interaction from musicians, and stage volume is not present in the same way. Because of this, some adjustments may need to be revisited once the band is live.

It is also easy to overmix during playback. Without the pressure of a live setting, you may be tempted to make too many adjustments or focus on details that are not noticeable in a real service. Keep your decisions practical and focused on what actually improves the mix.

Finally, remember that virtual playback is a supplement, not a replacement. It is a tool to help you prepare and improve, but it cannot replace the experience of mixing live.

9.7. Practicing with Purpose

Virtual playback creates an opportunity to grow in a way that is not always possible during a live service. It allows you to slow down, listen carefully, and make intentional improvements. The key is to use this tool with purpose. Set goals, stay focused, and approach each session as an opportunity to improve. Over time, these small efforts will lead to meaningful progress.

By incorporating virtual playback into your workflow, you create a cycle of preparation, evaluation, and refinement. This not only improves your mixes, but also builds confidence and consistency. When used well, virtual playback becomes an essential part of your development as an audio engineer.

10. System Tuning & Room Awareness

Introduction

As an audio engineer, it is easy to focus primarily on the console. Faders, EQ, compression, and effects are all powerful tools, and they play an important role in shaping the mix. However, the console is only one part of the system. The room itself is just as important.

System tuning is the process of understanding how your sound system interacts with the room and making adjustments so that audio is clear, consistent, and evenly distributed. The goal is not to create a perfect system, but to create one that works well within the environment you have.

Every room is different. Size, shape, materials, and layout all affect how sound behaves. Because of this, there is no single setting or approach that works everywhere. Learning to recognize how your specific room responds to sound will help you make better decisions and avoid unnecessary adjustments at the console. In this chapter, we will focus on practical awareness. While you will likely never be responsible for tuning the PA or the room yourself, learning about these concepts will increase your understanding of how the nuances of the sound system interact with each other, and how each part of the process can affect the quality of your sound in different ways.

It’s important to recognize that system tuning is an *extremely advanced topic*. Professional engineers spend years learning how to properly tune a sound system. This chapter will not attempt to teach you all the details involved in system tuning. Rather, it is written to be a brief introduction to the concept. The specifics of *how* to professionally tune a speaker system would be a full, separate manual in its own right.

Tuning the PA and the room is not, and will never be, a task that is expected from members of the production team – so don’t stress if you don’t fully understand. Just do your best to grasp the big picture.

10.1. Understanding the Room

As you learned early on in this handbook, sound does not simply travel from a speaker to a listener. It reflects off walls, ceilings, and floors. It builds up in some areas and cancels out in others. These interactions shape what people actually hear in the room. In more reflective spaces, sound can bounce around and create a sense of reverberation. This can make speech harder to understand and cause the mix to feel less defined. In more absorbent spaces, sound may feel clearer but less full. Most rooms fall somewhere in between.

Certain areas of a room tend to cause more problems than others. Corners can build up low frequencies, making the mix feel boomy. Hard surfaces like walls and ceilings can reflect high frequencies, creating harshness or echo. These are not things you can completely eliminate, but they are things you can learn to recognize.

The key is awareness. As you listen, begin to notice how the room is affecting what you hear. Is the mix clear in one area but muddy in another? Does speech become harder to understand in certain seats? These observations will guide your decisions and help you work with the room rather than against it.

Speaker Placement and Coverage

One of the most important factors in system performance is speaker placement. Where speakers are positioned and how they are aimed has a significant impact on what people hear. The goal is to direct sound toward the audience and minimize how much sound is hitting walls, ceilings, and other reflective surfaces. When speakers are aimed correctly, more of the sound reaches the listener directly, which improves clarity and reduces unwanted reflections.

It is also important to consider coverage. Ideally, sound should be distributed evenly throughout the room. If some areas are much louder than others, it becomes difficult to create a mix that works for everyone. Overlapping speaker coverage can also cause problems, as the same sound arriving from multiple sources at slightly different times can reduce clarity.

While you may not always be able to change speaker placement, understanding its impact will help you recognize why certain issues occur. In many cases, problems that seem like mix issues are actually system or placement issues.

10.2. Notes on System Tuning

What EQ CAN Fix…and What it Can't

EQ is one of the most commonly used tools in audio, but it is also one of the most commonly misunderstood. While EQ can be very effective, it has limitations. EQ can help shape tone and address minor imbalances. If something sounds slightly harsh or muddy, a small adjustment can improve clarity. It can also be used to reduce frequencies that are prone to feedback. However, EQ cannot fix problems caused by the room or the system. If a speaker is poorly placed or the room is highly reflective, no amount of EQ will fully correct the issue. Attempting to fix these problems with EQ often leads to over-processing and a mix that sounds unnatural.

The goal is to use EQ as a tool, not a solution for everything. Make small, intentional adjustments and focus on improving clarity rather than trying to solve larger system issues at the console.

Ringing Out the System

Ringing out the system is the process of identifying and reducing frequencies that are most likely to cause feedback. This is typically done during setup or rehearsal by gradually increasing volume until feedback begins, then using EQ to make a small cut at that frequency. This process increases your system's gain before feedback, giving you more usable headroom during a service. It is especially important for monitors, but it can also be applied to the main system.

When ringing out the system, it is important to make small, precise adjustments. Large or excessive cuts can negatively affect the overall sound quality. The goal is to reduce problem frequencies without changing the natural tone of the system. Ringing out is not about eliminating feedback entirely. It is about creating a more stable system that allows you to operate with confidence.

10.3. Listening to the Room

One of the most valuable things you can do as an engineer is step away from the console and listen from different areas of the room. What you hear at front of house is not always what others are hearing. Walking the room during rehearsal allows you to identify areas where the mix may be too loud, too quiet, or unclear. You may notice that certain frequencies are more pronounced in some areas than others. These observations can help guide your adjustments.

It is important to balance what you hear in the room with what you hear at the console. You cannot perfectly optimize the mix for every seat, but you can aim for consistency and clarity across the majority of the space. Developing this habit will improve your awareness and help you make more informed decisions.

Stage Volume and its Impact

Stage volume plays a major role in overall sound quality. When sound from the stage is too loud, it spills into microphones and mixes with the system output. This can make the mix feel muddy and difficult to control. Lower stage volume leads to a cleaner and more controlled mix. When the sound system is the primary source of what the audience hears, you have greater ability to shape and balance the mix.

This does not mean that stage sound should be eliminated, but it should be managed carefully. Encouraging reasonable monitor levels and being aware of instrument volume will help maintain clarity.

10.4. Final Thoughts

Common Mistakes

A common mistake is trying to fix every issue with EQ. As discussed earlier, many problems originate from the room or speaker placement and cannot be solved at the console. Another mistake is ignoring the room entirely. If you only mix from one position and never listen from other areas, you may miss important issues. Overcomplicating the system can also create unnecessary challenges. Simple, well-understood setups are often more effective than complex ones. Finally, failing to listen carefully is one of the biggest obstacles to improvement. Awareness and intentional listening are key to successful system tuning.

Working With the Room

You cannot change the room, but you can learn to work with it. By understanding how sound behaves and making thoughtful adjustments, you can create a system that supports clarity and consistency.

Focus on simple, effective solutions. Pay attention to what you hear. Over time, your awareness will grow, and your ability to work within your environment will improve.

11. Troubleshooting: A Deep Dive

Introduction

No matter how well a system is designed or how carefully it is prepared, problems will occur. This is a normal part of live audio. Equipment can fail, settings can be changed unintentionally, cables can wear out, and unexpected situations can arise at any time. The presence of problems does not mean something has gone wrong in your preparation, it simply means you are working in a live environment.

The goal of troubleshooting is not to eliminate every possible issue, but to respond to problems quickly, calmly, and effectively when they happen. A strong troubleshooting approach allows you to maintain control of the situation and minimize distractions for both the congregation and the team on stage.

This chapter is designed to give you a clear and repeatable process. Instead of reacting randomly or guessing at solutions, you will learn how to identify problems, trace their source, and apply simple, effective fixes. Over time, this process will become second nature and will significantly increase your confidence during a service.

11.1. The Troubleshooting Mindset

The most important part of troubleshooting is not technical, it is mental. When something goes wrong, your first reaction will often determine how the situation unfolds. It is easy to panic when audio suddenly stops working or when an unexpected noise appears in the system. However, panic leads to rushed decisions, and rushed decisions often make the problem worse. The first step is always to remain calm. Take a breath, focus your attention, and begin to think clearly about what is happening.

Avoid the temptation to make multiple adjustments at once. Rapidly changing faders, toggling mutes, or adjusting EQ without understanding the problem can create additional issues and make it harder to identify the original cause. Instead, make one change at a time and observe the result. This controlled approach allows you to learn from each step and prevents confusion.

It is also important to listen carefully before acting. Identify the type of problem you are dealing with. Is there no sound at all? Is the sound distorted? Is there feedback? Each type of problem has a different cause and requires a different approach. Developing a calm, methodical mindset will make you far more effective than simply knowing more technical information. Confidence comes from process, not guesswork.

11.2. Signal Flow & Troubleshooting

Signal flow is your most powerful tool when troubleshooting. Every sound in your system follows a path from its source to its destination. When something is not working correctly, the problem exists somewhere along that path. The key is to follow the signal step by step and determine where it is being interrupted or altered. A typical signal path might look like this:

> *Source → Microphone/Instrument → Cable → Stage Box → Console Input → Processing (EQ, compression, etc.) → Fader → Bus/Output → Amplifier/Speaker or Broadcast*

When troubleshooting, start at one end of the chain and move through it systematically. Ask yourself clear questions at each stage:

- Is the source producing sound?
- Is the signal reaching the console (check the channel meter)?
- Is the channel configured correctly (not muted, fader up)?
- Is the channel configured correctly (not muted, fader up)?
- Is the output system functioning properly?

By breaking the problem down into smaller steps, you can quickly isolate where the issue is occurring. This prevents unnecessary adjustments and allows you to focus on the actual cause, turning troubleshooting from a guessing game into a logical process.

11.3. Problem: No Sound

One of the most common and urgent problems is having no sound. When this happens, it is important to resist the urge to panic and instead follow a simple, structured checklist.

- Start with the source. If it is a microphone, confirm that it is turned on and properly connected. If it is a wireless microphone, check the battery and ensure it is powered on and synced correctly. If it is an instrument, confirm that it is producing sound.
- Next, check the console input. Look at the channel meter to see if signal is present. If there is no signal, the problem is likely before the console—such as a cable, connection, or stage box issue.
- If signal is present at the meter, move to the channel controls. Ensure that the channel is not muted and that the fader is raised to an appropriate level. Verify that the channel is assigned to the correct bus or main output.
- From there, check the output stage. Confirm that the main mix or bus is active and that the output system (amplifiers, speakers, or broadcast feed) is functioning properly.

By working through these steps in order, you can quickly identify where the signal is being lost and address the issue without unnecessary adjustments.

11.4. Problem: Distortion/Clipping

Distortion is typically the result of signal levels that are too high. When a signal exceeds the system's capacity, it becomes clipped, resulting in a harsh and unpleasant sound. The most common cause of distortion is excessive gain at the input stage. If the preamp is set too high, the signal can overload before it even reaches the rest of the system. This type of distortion cannot be fixed later in the signal chain, it must be corrected at the source.

To troubleshoot distortion, begin by identifying where it is occurring. Listen carefully to determine if the distortion is constant or only happens at certain moments. Check the input meter for signs of clipping. If the signal is too hot, reduce the gain until it is within a safe range.

It is also important to check other stages of the signal chain. Compression settings, effects, or output levels can also contribute to distortion if pushed too far. However, input gain is usually the first place to check.

The key principle is to fix distortion at the point where it originates. Reducing levels later in the chain will not remove distortion that has already been introduced.

11.5. Problem: Feedback

Feedback is one of the most noticeable and disruptive issues in live audio. It occurs when sound from a speaker is picked up by a microphone and re-amplified, creating a loop that quickly builds in volume. When feedback occurs, your response should be quick but controlled. The first step is to identify the source. This is often a specific microphone or monitor mix. Lowering the fader of the affected channel or reducing the monitor level will usually stop the feedback immediately.

Once the feedback is under control, consider why it occurred. Was the microphone too close to a speaker? Was the monitor level too high? Was there a frequency that needed to be reduced? If necessary, make a small and precise EQ adjustment to reduce the problem frequency. Avoid making large or aggressive changes, as this can negatively affect the overall sound.

Feedback is best managed proactively. Proper microphone placement, reasonable monitor levels, and careful EQ adjustments will reduce the likelihood of it occurring during a service.

11.6. Problem: Intermittent Signal Issues

Intermittent issues can be some of the most frustrating problems to troubleshoot because they do not occur consistently. A microphone may cut in and out, or a signal may drop unexpectedly. These problems are often caused by physical issues such as loose cables, damaged connectors, or faulty equipment. Wireless systems can also introduce intermittent issues due to interference or weak signal.

To troubleshoot, begin by isolating the problem. If possible, swap out components one at a time. Replace the cable, try a different microphone, or use a different input channel. This process helps identify which part of the system is causing the issue.

Pay attention to patterns. Does the problem occur when something is moved? Does it happen at certain times? These clues can help narrow down the cause. Unfortunately, many intermittent issues will only be solved by painful trial-and-error.

11.7. Problem: Broadcast Issues

Broadcast issues can be less obvious because they are not always heard in the room. This makes it important to monitor the broadcast feed directly.

If there is no audio on the stream or broadcast, begin by checking the broadcast bus. Confirm that signal is present and that channels are being sent correctly. Then follow the signal path to the matrix and output. If levels are too low or inconsistent, adjust the mix to ensure that key elements, especially vocals, are clearly present. Compression may also help maintain consistency.

It is also important to verify that the signal is reaching the final destination. This may involve checking streaming software, encoders, or external devices. Regularly listening to the broadcast feed will help you identify and resolve issues before they affect the audience.

11.8. Final Thoughts

Quick Decision Making

During a live service, not every issue needs to be fixed immediately. Some problems are minor and may not be noticeable to the audience. In these cases, it may be better to leave them alone rather than risk creating a larger distraction. Prioritize issues that affect clarity and overall experience. For example, a microphone that is not working or a loud feedback event should be addressed immediately. A small tonal imbalance may be less urgent.

Making good decisions under pressure requires awareness and judgment. As you gain experience, you will become better at recognizing which issues need immediate attention and which can wait. The goal is to maintain a smooth and distraction-free experience, even when challenges arise.

Building Confidence Through Experience

Troubleshooting is a skill that improves with experience. The more situations you encounter, the more familiar they become. Over time, you will begin to recognize patterns and respond more quickly. Each problem you solve adds to your understanding of the system. Take note of common issues and how they were resolved. This knowledge will help you handle similar situations in the future.

Do not be discouraged by mistakes. They are a natural part of the learning process. What matters is how you respond and what you learn from each experience. Confidence comes from repetition, awareness, and a willingness to grow.

Calm, Clear, and Controlled

Effective troubleshooting is not about reacting quickly—it is about responding thoughtfully. By staying calm, following a clear process, and making intentional adjustments, you can resolve problems efficiently and maintain control of the situation. Trust your training and rely on your understanding of signal flow. When something goes wrong, slow down, think clearly, and work through the problem step by step.

A calm, clear, and controlled approach will not only help you solve issues more effectively, but will also create a sense of stability for everyone involved.