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Sound Reinforcement

Jamie Rio and Chris Buono

HOUSE OF WORSHIP SOUND REINFORCEMENT

Jamie Rio and Chris Buono

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House of Worship Sound Reinforcement

Jamie Rio and Chris Buono

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To Joe and Florence for bringing me into this amazing world.

—Jamie Rio

To a decade of life, love, and happiness with my one and only. May we live within our love for each other forever and longer.

—Chris Buono

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This book would not have been written if it were not for my relationship with an all-knowing, all-caring Supreme Being. My friend, God, has directed my life for as long as I can remember and there is no way I could write a book on house of worship sound without Him. As for my earthly companions with whom I share this life, I thank my friends and family who have cared about me, supported me, and shared in my deep passion for what I do. Thanks to my editor at *F.O.H.*, Bill Evans, who offered me my first job as a freelance writer and later a columnist. Bill has not only influenced me as a writer, but has been a friend from the beginning of our professional relationship. Thanks to Chris Buono, my co-author, for being my mentor and motivator through this project. Finally, thanks to all those who have influenced my interest in music and the invigorating world of live sound.

CHRIS BUONO

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About the Authors

Jamie Rio's first foray into the music industry was almost three decades ago when he recorded and toured throughout Brazil with his band, BackStreets. It was during those years that Jamie became enthralled with the process of recording his band's music and the live sound production of the tour. These newfound obsessions led to his working as a recording engineer in a variety of studios in the Los Angeles area while continuing to record music and tour with bands and duos of his own design. As Jamie's expertise in sound reinforcement developed, he found himself focusing on technical-reference journalism and on starting his live sound company, RioDeluxe Audio. Since the creation of RioDeluxe Audio, Jamie has been behind the console for more than 3,000 live events. Jamie also currently writes the "Sound Sanctuary" column for *F.O.H.* magazine. In addition to contributing articles to various other industry magazines, Jamie continues to thrive as a live sound technician and installer.

Chris Buono's fascination with sound began as a Catholic elementary school student; during more-than-regular attendance at Mass, his wandering adolescent mind became enthralled with the actual sound of it all. It was during the frequent visits to these halls of heavenly resonance that he developed his obsession with the grandiose effects of reverberation and echo. This who, what, how, and wow of all-things-audio developed into a life-long commitment to the production and manipulation of sound through Chris's work as an overachieving multimedia guitar madman. Since those days, Chris has played in various parts of the globe with the best of the best; served as a professor for five years at the esteemed Berklee College of Music, where he managed the world's most comprehensive guitar effects lab; became an in-demand contributing writer for *Guitar Player*, *Guitar One*, *Just Jazz Guitar*, *Mel Bay's Guitar Sessions*, and *Guitar Teacher* magazines; and was lucky enough to run into a guy named Mark Garvey, who took a chance and let Chris author the officially endorsed *M-Audio Guide for the Recording Guitarist* for Cengage Learning. Today, Chris continues his journey on myriad major and independent releases and countless video appearances as a clinician, most notably for Truefire. After nearly 30 years spent working with sounds, Chris retains the same boyish wonder first instilled in his psyche while he sat quietly in the pew, taking it all in.

Contents

Introduction

An unsung hero, one often unbeknownst to the congregation, is the blessed soul who ensures that the uplifting sounds of worship are heard in all their glory throughout the house of worship. These individuals are the ones who manage every aspect of the sound system within a worship place. From the rigors of setting up for each and every event in the house to manning that seemingly complex console of lights and buttons to maintaining the health of every last piece of equipment, these talented people have their hands, heart, and ears committed to the job. It is through their tireless dedication to the art of sound reinforcement that your worship experience is realized.

The book you hold in your hands contains the knowledge these humble individuals possess—and what *you* need to know to join their ranks. In the chapters to come you will find everything you need to know to start your own journey to becoming a functional sound technician in the house-of-worship realm and beyond. Although the concepts in this book are complex, they were purposely written about in a non-technical manner, enabling those with absolutely no prior knowledge of sound to benefit from its contents. The idea set forth by the authors was to avoid attempting to convey intimidating physics and formulae and instead to bring forth real-world information for the many real-world situations you will encounter. Whether you are just beginning your journey into live sound, you've been mixing at your own house of worship for a period of time, or you have years of experience already logged in, you will surely find useful information, tricks of the trade, and sound advice in this book that will serve you for years to come.

WHO ARE YOU?

You are first and foremost intrigued by the wonder of sound. You often find yourself pondering how and why the preacher's voice resonates throughout your house of worship the way it does or what all those controls on your sound technician's mixing console do. You are the one staying after services, asking questions or raising your hand to help set up the stage area whenever the opportunity arises. You gladly hang around the mixing board during outreach events to see if the tech needs you to adjust a speaker cabinet or retrieve a defective microphone on stage. You want in on the action. Bad.

By choosing to read this book, you have elected to walk the path of the sound technician—a person whose effectiveness is measured in how well he or she can avoid being noticed. The truth is, the congregation and the worship team notice the sound technician only when some audible aspect of the service goes wrong. This makes you humble and satisfied with helping the greater good, even when there's not much reward in return.

If you're just starting out, you're most likely a volunteer trying to contribute to the betterment of your house of worship. If you have some prior experience, you may be interested in trying to take your talents to other houses of worship and perhaps start receiving some compensation for your efforts. You may even want to invest in some sound equipment of your own and start a side business. Whatever your situation, at the core, you have a thirst for knowledge and a desire to ensure that everyone's worship experience is enhanced with the most glorious sound possible.

WHO ARE WE?

The authors of this book share many passions, but none more than a deep admiration for the deliverance of superior live sound. We are zealots. Whether it be rock, blues, jazz, reggae, swing, or any other style of music, we sincerely want it to sound the best it possibly can. We are two individuals who have spent incalculable hours listening to music projected in myriad venues. From concert halls to amphitheaters, auditoriums, gymnasiums, and, of course, worship houses, we've subjected our ears to a lifetime of vibrations that resonate throughout our psyches every moment of every day. More than that, we are purveyors of knowledge, constantly educating our readers on subjects that span the worlds of musical performance, technology, and sound reproduction. We are committed to bringing forth information in a manner everyone can understand so anyone who has the desire to immerse their lives in the glory of sound can do so. We are honored by your interest in what we have to say and we sincerely hope you realize the same joy we have both experienced working with this wonderful thing called sound.

HOW THIS BOOK IS ORGANIZED

House of Worship Sound Reinforcement consists of 10 chapters followed by a glossary of terms. Following is a breakdown of the chapters you will be studying in your journey to becoming house of worship sound technician. Starting with the basic foundations of sound, you will progress into learning how the sound in your house of worship is captured with microphones and transduced into electricity. From there, you will explore the wonders of the mixing console, where all the audio you're capturing is sent, processed, and mixed together. Next up will be a thorough examination of how sound is projected not only to the congregation but also back at the worship team through the speaker system. After delving into the world of digital processors, you willl learn about how to create an optimal environment for projecting sound in your worship space, which includes properly setting up your system. Finally, you will discover what it takes to maintain your system as well as how to readjust and/or create a new system for outreach events.

- **Chapter 1, "Sound and Sound Advice."** In this chapter, you take your first steps by learning what sound actually *is*. Through real-world analogies, you will learn about the basic concepts such as frequency and decibels as well as delve into deeper subjects such as overtones. From there, you start to connect these concepts with the everyday sounds heard in your house of worship, such as those made by the preacher, the choir, and the worship band. You also learn about how the walls that surround you affect sound.

- **Chapter 2, "Microphones."** Sound technicians are only as good as their gear. This chapter introduces one of the most important pieces: the microphone! Here you will discover the various microphone types, their construction, and how all these variables affect how the mic captures sound. From polar patterns to phantom power, we go deep. You name it: troubleshooting, common uses, accessories, and the ever-important subject of mic placement—it's all there.

- **Chapter 3, "Mixers and the Art of Mixing."** To the uninitiated, a mixing console can seem like a complex collection of lights, switches, sliders, and wires. It is this chapter's job to clearly explain what all those lights, switches, sliders, and wires actually *do*. After starting with the input channel strip and all of its components, you will also learn about many other essential features of a mixing board, including auxiliary sends, submix groups, and the mighty master section. In addition, throughout your reading, you will discover a plethora of tricks and tips to help you best control your console so you can deliver superior mixes to your congregation.

- **Chapter 4, "Speakers."** Speakers must accurately reproduce whatever is fed to them, so it's important you become familiar with every aspect of them. Here you will explore the inner workings of a speaker and how it functions, and delve into aspects not commonly discussed such as 2-way and 3-way configurations and frequency distribution. You'll even look into how the construction material and shape of the actual cabinets play into how the speaker performs and learn about components that work in conjunction with your speakers such as external crossovers, power amplifiers, and speaker-management processors.

- **Chapter 5, "Monitors."** While the congregation has their ears tuned into the FOH enclosure system, the stage participants have their ears tuned to the subject of this chapter: monitors. Although monitor wedges, as they are aptly called, are indeed speakers and have many properties that are similar to those of their FOH counterparts, there are the matters of placement, equalization, feedback, and systems to consider. You will also discover a popular alternative known as in-ear monitors.

- **Chapter 6, "Outboard Gear."** Just as fascinating as the mixing console are the separate audio processing units that work in conjunction with it known as outboard gear. From more-advanced means of equalization to processors dedicated to dynamic control to special effects—it's all here and more. You will be guided through the controls, setup, and application of these essential peripherals, as well as explore the concepts behind their design.

- **Chapter 7, "Creating a Sound Environment."** Have you ever just sat back and looked at the interior of your house of worship? You will now. This chapter examines every wall, carpet, drape, and ceiling panel in your house. Applying knowledge from previous chapters, especially with respect to the basic fundamentals of sound, you will learn about how sound waves travel within your worship-space walls, how they are absorbed, how they are reflected and much, much more. No matter what sonic dilemmas your house has—they all do—you'll discover a solution that ensures your house sounds heavenly.

- **Chapter 8, "Putting It All Together."** In this chapter, you will do as the title suggests: Put it all together. Tying up everything you've learned to this point, you'll be instructed as to how to literally plug it all in, set it all up, and turn it all on—but not before you discover how to organize it all. Additionally, you'll examine new topics such as tiered risers, podiums, stage platforms, and more.

- **Chapter 9, "Healing the Sick and the Tired…Gear, That Is."** In the midst of all this glorious gear is one sad fact: This stuff breaks. But don't worry; in these pages, you'll learn about the tools needed not only to perform some minor repairs yourself, but to test and maintain your gear to best prevent equipment failures. You'll be introduced to the time-honored tradition of troubleshooting through a lengthy guided explanation of how to pinpoint a common yet nerve-wracking problem. Adding to this collection of vital information are pages dedicated to grounding—a concept of paramount importance to the success and safety of your sound system.

- **Chapter 10, "Outreach."** At this point, you've digested the knowledge needed to set you off on the right foot to becoming a functional sound engineer inside your house of worship. But what about *outside* your house's four walls? This chapter contains the ins and outs of conceptualizing and organizing an outreach event from a sound reinforcement point of view. You'll learn about the additional gear required, the benefits (and drawbacks) of renting versus purchasing said gear, the pros and cons of possible outreach locations, and more. Upon completion of this chapter, you'll be more than ready to play a key role in the success of your house of worship's ever-so-important outreach events.

A NOTE ABOUT DIGITAL MIXERS

Digital-mixing consoles have become more and more common in live-sound applications and will continue to so. While this book does not explore the intricacies and nuances of digital boards and digital mixing as a whole, the knowledge you will acquire throughout your reading will properly prepare you for when you encounter one. With a solid foundation and thorough understanding of sound systems—specifically, analog mixing consoles—you will find it easier to make the transition to digital mixing. In fact, you may never fully appreciate the depth of digital consoles without any prior knowledge of their analog predecessors. We urge you to spend time working with a traditional analog-based mixing console in your formative years as a sound technician, but at the same time encourage to have an eye toward the future, as we may live to see the complete dissolution of these brilliant analog machines.

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CHAPTER 1

Sound and Sound Advice

Your job as a sound technician/mixer for your house of worship ultimately comes down to one principal factor: being a good listener. Of course, that's not your only job, but it is the most fundamental element. And what is it you are listening to? Well, sound, of course; but don't let the simplicity of that answer sway you from the matter at hand. In this chapter you will quickly realize that the phenomenon we call "sound" has several foundational components of which you will need to be aware. Understanding the mechanics of sound will greatly improve how you hear sound, and that's what we're going to dive into in this first chapter. This chapter starts with the basic principles of sound—pitch and volume—and then helps you associate these principles with the sounds you will be dealing with in your house of worship. From the preacher to the choir to the walls around you, this chapter will get you off on the right foot (er, ear).

What Is Sound?

The phenomenon of sound is unique to our planet (well, as far as we know, anyway). That is, without an atmosphere, there is no sound. It is in this atmosphere that sound is created, dispersed, and eventually heard by way of moving molecules. If that's a new concept for you, don't worry—all will soon be clear.

Let's start by taking a look at the basics of sound, examining what it is, how we hear it, and how we measure it. It cannot be stressed enough how important it is to have a strong understanding of the foundations of sound. It will not only provide you with the ability to mix live sound, it will allow you do so more accurately and efficiently.

VIBRATIONS

To best describe any abstract form of matter such as a sound wave, it's a good idea to compare it with something in the material world you probably know well—say, a cricket. Imagine that right now there is a cricket outside your window vibrating its legs (or wings) as fast as it can. As a result, vibrations move through the air in waves, producing sound (see Figure 1.1). This is the genesis of any sound. As a matter of fact, if an object—whatever it may be—doesn't vibrate, it cannot make sound.

Figure 1.1
As the cricket rubs its legs together, it creates vibrations that travel through the air. It is these vibrations that are heard by whomever is within range of the sound waves.

These waves have length (hence the term wavelength) measured in feet and inches (sometimes meters). As these sound waves reach your ear, your eardrum also vibrates, and it is those vibrations that are recognized by your brain. Your brain, then, working at the speed of light (not sound), sorts through all the "files" of vibrations that have come through your ear in the past. If the vibration is recognized, you instantly know what you are hearing (or have just heard).

Luckily for us, our ears and brains are capable of sorting and storing millions upon millions of sound references. This includes all the nuances that make up an individual's voice or a musician's playing style. Just think of all the voices (speaking or singing) you already immediately recognize. How about all the sounds you can accurately identify—your friend's car, your dog's bark, etc. Conversely, if your brain doesn't recognize the vibrations, you may think to yourself, "What is that sound?" Once you identify whatever mysterious sound you are hearing, it, too, will become part of your internal sound library.

All these wonderful waves of sound vibrate at various speeds or cycles per second, which brings us to the next topic at hand: hertz.

HERTZ

With the concept of the aforementioned vibrations in mind, consider this: Sound causes air molecules to fluctuate in ripples or waves. To best illustrate this, imagine dropping a stone in a calm pond. The ripples move away from the stone at an even rate of speed (see Figure 1.2). In terms of sound waves, the number of fluctuations per second is what determines the pitch (highness or lowness of sound), which is more commonly referred to as frequency. Those fluctuations can be thought of as cycles per second (cps). So, the more cycles per second, the higher the frequency, or vice versa.

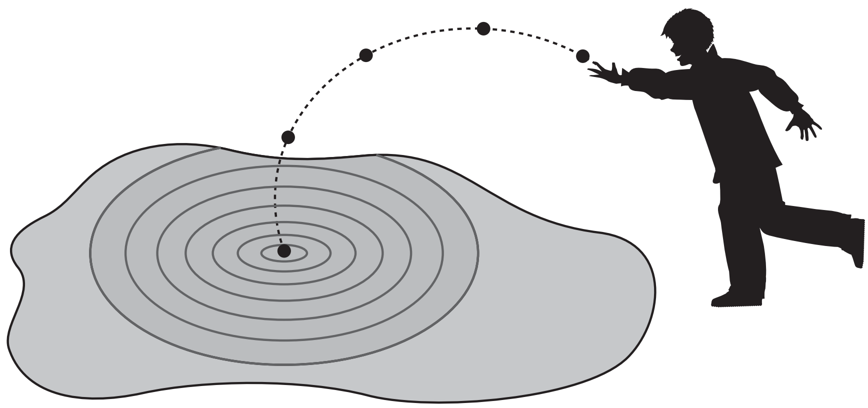


Figure 1.2
Sound waves move through the air much like waves ripple in a calm pond after an object such as a stone has been dropped into it. Most importantly, the time interval between the waves should be constant in order to maintain pitch.

Perfect human hearing begins at 20cps and caps at 20,000cps. Cycles per second are more commonly known as Hertz (Hz), which is a term derived from the surname of the man who brought forth this knowledge: 19th century physicist Heinrich Hertz. For example, 20cps = 20Hz (Hertz). When you reach 1,000 hertz, or 1 kilohertz, it will most likely be abbreviated as kHz (for example, 20,000cps = 20kHz).

WHAT'S THE FREQUENCY HEINRICH?

The term frequency is commonly used to mean the same thing as pitch. So why use the term frequency at all? First, remember this: Pitch refers to the highness or lowness of the sound. Low pitches are deep sounds, such as those made by a tuba or a note from the low register (left side of the keyboard) on your house of worship's organ. At the other end of the spectrum are high pitches—sounds made by flutes or female voices. Here's where frequency comes in: Sound is comprised of vibrations that travel in cycles per second, and the frequency with which those cycles occur determines the pitch. The vibrations of low-pitched sounds have longer wavelengths and fewer (less frequent) cycles per second; with high frequencies, it's the opposite.

Now, what about the volume?

DECIBELS AND THE DECIBEL METER

Again using the material world to help describe the abstract world, we will describe yet another common scenario in everyday life: a dog barking. The sound that comes from a dog as it audibly makes its presence known to your lawn specialist or mail carrier has a multitude of the aforementioned frequencies coming in waves at a variety of hertz measurements. These waves are pushing through the air, creating varied levels of force or pressure. We can measure this sound-pressure level (SPL) with another unit of measurement called decibels (dB). Depending on the source, the SPL can vary from subtle to intense. An average dog's bark can hit your ears at a range of 75–80dB. If you are wondering exactly how loud that is, 0dB is the point where the threshold of hearing begins, and 68 to 72dB would be the average level of two people conversing.

To determine the SPL or decibel level of a given sound such as our barking dog, you would use a decibel meter like the one pictured in Figure 1.3. With a decibel meter, you can measure the SPL of anything that produces a sound. For instance, you can stand on the sidewalk near fairly heavy traffic, hold up a decibel meter, and get a reading of about 90dB. Did you know unamplified classical orchestras can measure close to 100dB of SPL from the front row? You will if you get down to your local park for a free concert with a decibel meter!

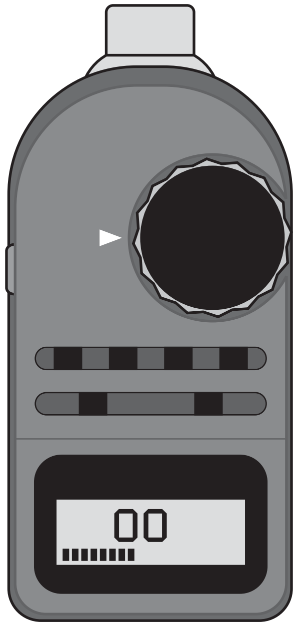


Figure 1.3
A decibel meter is a compact, battery-powered device that measures the SPL of whatever it hears. This gives you a way to accurately assess the volume levels of whatever sound you need to keep an eye (ear, that is) on.

As you can see, the sounds of everyday life come in a large variety of decibel levels. When you walk into your house of worship to mix sound, there is already an average decibel level from the congregation. This background decibel level can vary greatly from congregation to congregation. But whatever the average decibel level is, you must learn to push the sound-reinforcement system above the background sound without hurting the congregation. With that responsibility in the mix, always carry a decibel meter to your gigs (worship or otherwise). If you do not own a decibel meter, buy one or get your church to buy you one—these are invaluable tools. Many of the worship houses you will mix at have very tight decibel limits, with 90–95dB being the average. To some people's ears, 90dB is not very loud; most people can yell a lot louder than 90dB.

While it's necessary to cover all the possible decibel ranges you might encounter, here's a small chart of some decibel milestones (see Table 1.1). While decibel levels are mostly common sense when it comes to a worship service, no one in the congregation wants to leave church on a Sunday with their ears ringing. And just as important, nobody wants to miss out on the word of God that is being delivered.

Table 1.1 Decibel Milestones

Decibel Ranges	Hearing Markers
0dB	Threshold of hearing
10dB	Breathing
40dB	Soft whisper (six feet)
65dB	Average conversation (three feet)
90dB	Heavy street traffic (five feet)
100dB	Loud orchestra (10 feet)
120dB	Loud rock band (10 feet)
130dB	Pain threshold
140dB	Aircraft carrier flight deck
150dB	Eardrum ruptures (ouch)

Frequency Ranges

You may already have a good understanding of frequencies and how sound waves act and react in different environments, but until you can fairly accurately identify 1kHz, or 630Hz, or 5kHz, or many of the numerous frequencies between 20Hz and 20kHz, you will have difficulty controlling the sound in your worship house. The goal is to become an excellent front-of-house (FOH) mixer; in order to do that, your ears must recognize all frequencies to some extent. The better you develop this ability, the better you will be able to mix, and the better everything will sound in your house of worship. For example, suppose you hear a rumble through your house sound system during a worship song. With your newly tuned ears, you would be able to identify this rumble as mud in the kick drum at 125Hz. With that knowledge in hand, you would simply attenuate (reduce) that frequency; the rumble vanishes! Another example could be that the lead vocalist sounds harsh or shrill; you estimate the problem lies at 2kHz. Again, with a twist of a knob, you would decrease that frequency to smooth out the vocal sound.

THE HUMAN POTENTIAL

The range of human hearing is documented as 20Hz–20kHz. That means any frequencies lower than 20Hz will not be heard, as your ears won't recognize them as sound. On the other hand, 20kHz is the topmost range of human hearing, but in actuality it's higher than most people can hear. The fact is, as we age, most of us lose some of our ability to hear upper frequencies, and that's normal. If you want to gauge your range of hearing, consider this: Say you've suffered some high-frequency loss and you currently only hear up to about 15kHz. Does that mean you might have problems with regard to your mixing efforts? In a word, no. If your hearing drops to 10kHz or below, however, you should become concerned. That said, if you work or intend to work in sound production, you should have your hearing checked regularly (every year or two).

WORSHIP INSTRUMENT AND VOCAL FREQUENCIES

Now that we have some idea of how sound and hearing work, let's look at what instruments are producing which frequencies. Modern worship bands usually consist of the following:

- Guitar (acoustic or electric)
- Drums
- Bass
- Keyboards

Of course, we can't forget about perhaps the most important instrument of all: the human voice!

If you look over Table 1.2, you will see how the different frequencies can affect every instrument that is used in a typical house of worship.

Table 1.2 The effect of frequencies on various instruments

Frequencies in Hertz	Effect on Instrument Sounds
100 to 160Hz	Increase to add warmth to bass guitar. Decrease to remove boominess from kick drum (bass overload) and bass guitar. Decrease to remove flabbiness from electric guitar.

(continued)

Frequencies in Hertz	Effect on Instrument Sounds
200Hz	Increase to add fullness to guitar, snare drum.
	Increase to add fullness to vocals.
	Decrease to decrease muddiness in vocals, guitars, keyboards, drums.
400Hz	Increase to add punch to kick drum, bass guitar.
	Decrease to remove "boxy" sound of floor toms.
630Hz	Increase to add clarity to vocals.
800Hz	Increase to add clarity to bass guitar.
	Increase to add punch to snare.
	Decrease to remove "boxy" sound of guitars.
1.6kHz	Increase to add presence to vocals.
	Increase to add snap to snare drum.
2kHz	Decrease to remove harshness from vocals.
3kHz	Increase to add attack to electric/acoustic guitars.
	Increase to add attack to piano and keyboards.
	Increase to add hardness to vocals.
5kHz	Increase to add attack to kick drum.
	Increase to add brightness to vocals.
	Increase to add brightness to piano, guitars, keyboards.

Frequencies in Hertz	Effect on Instrument Sounds
6.3kHz	Increase to add brightness to vocals.
	Increase to add snap to cymbals.
	Increase to add sharpness to guitars, piano, keyboards.
	Decrease to remove "s" sound from vocals.
10kHz	Increase to add sparkle to vocals.
	Increase to add sparkle to cymbals.
	Decrease to remove "s" sound from vocals.

Frequency designations may be a bit overwhelming, but don't worry. As you experiment with these frequencies, you will hear what we're talking about.

It's important to note that adjustments to frequencies are made throughout a service by way of an equalizer (see the following sidebar). The process of increasing or decreasing a specific frequency or range of frequencies is called—you guessed it—equalization (also explained in the sidebar). We'll cover equalization much more in Chapter 6, "Outboard Gear," but for now, the point is this: A little equalization can go a long way. That said, you will not spend time with frequencies below 80Hz because most instruments in worship houses will not even produce frequencies below that. Looking closely at Table 1.2, you will also notice that not one instrument except for the synthesizer travels as high as 10kHz. You may be asking yourself, "Why do I need a sound reinforcement system that will reproduce frequencies to 20kHz?" That question requires a two-part answer. Here's the first half: Most good sound systems reproduce frequencies from 40Hz to about 16kHz, tops. A controversial statement, but true nonetheless.

Did Someone Say Equalizer?

Equalization is simply the process of adjusting frequencies. It's done with a piece of gear aptly called an equalizer or, more commonly, an EQ. The most common equalizer you will use in your house of worship is the graphic EQ like the one seen in Figure 1.4. Frequency controls are presented in 31 bands (think of a band as a physical slice of the frequency spectrum). Each band from 20Hz to 20kHz will have a slider that allows you to increase or decrease the frequency of the chosen band. Mixing consoles always feature some sort of equalizer similar to what is displayed in Figure 1.5, though the depth of control varies from one unit to the next. Regardless, these are present at every input channel as adjustment knobs. The knobs affect the frequencies in a wider context than the graphic EQ, but they are designed to increase or decrease frequencies nonetheless.

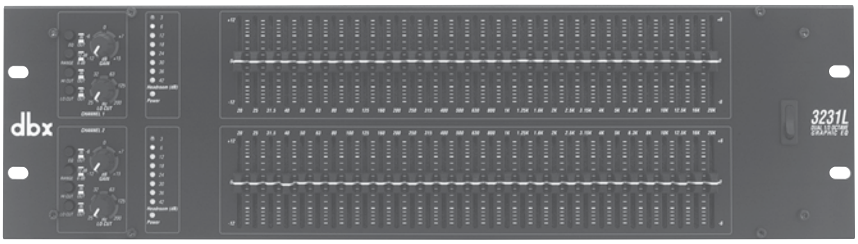


Figure 1.4
When it comes to using any type of EQ not found on your mixing console, you will most likely use a graphic equalizer like this classic dbx 3231. Take note: This is an example of a piece of outboard gear, which simply means it's a standalone piece not built into your mixer. (Image courtesy of dbx Professional Products.)

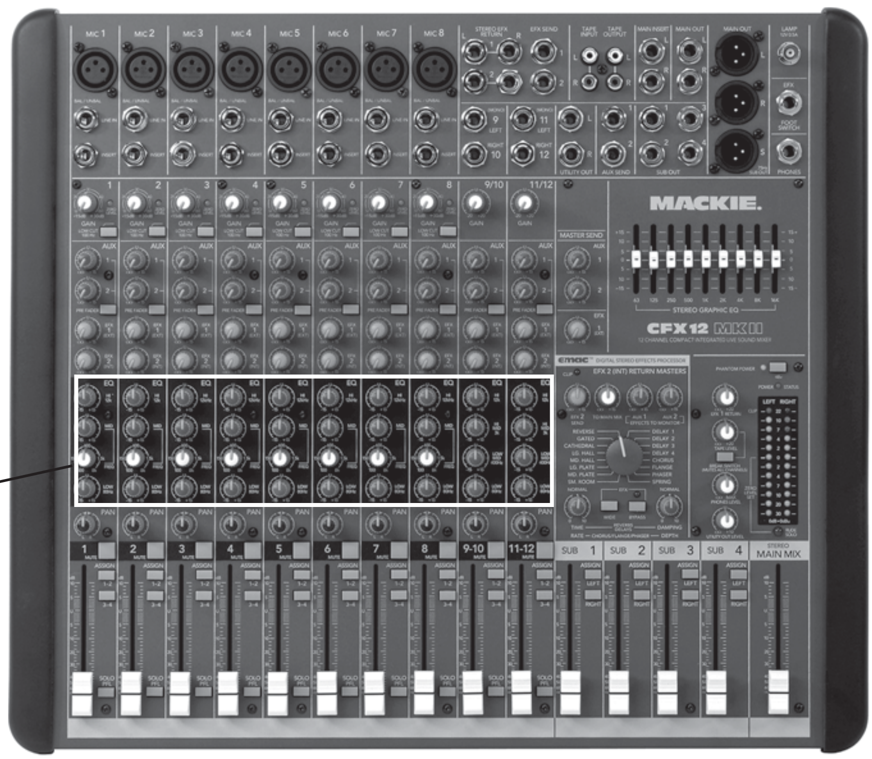


Figure 1.5
Most mixing boards feature some form of equalization controls, as seen here on this Mackie CFX12 mkII board. While these more condensed sets of controls might not be as thorough as the ones featured on outboard EQs, more times than not they will do the job just fine. (Image courtesy of Mackie.)

Hopefully, Table 1.2 will help you put some punch in your kick drum or make your choir sound like angels. While memorizing the information contained in the preceding table would serve you well, if that's not possible, no problem; just make some notes on 3×5 cards and be sure to keep them with you. Or better yet, keep this book close at hand!

FEELIN' LOW

On the low side of the frequency spectrum, we have pipe organs and synthesizers. Sadly, you won't run into as many pipe organs as you will synthesizers. But many of these types of electronic keyboards are capable of playing notes all the way down to A0 (that is a very low "A" note measuring in at 27.8Hz). Notes in this frequency range are much lower than any that a typical worship band would produce, and not too many sound systems are capable of reproducing that pitch. If you happened to be in a house of worship and these exceedingly low notes were being accurately reproduced, you would feel them more than hear them.

OVERTONES

Sound waves emanating from a musical instrument or a voice are a blend of many different frequencies. For example, let's take the A string on a guitar. That string has a fundamental (basic or foundational) frequency of 440Hz, but that's only the beginning. The guitar string sound is a composite of the basic 440Hz plus all additional overtones. Overtones are the frequencies that are over and above the fundamental 440Hz string tone. These overtones, commonly referred to as harmonics or partials, can be much higher than the original frequency of their fundamental.

It's these overtones that allow our ears to differentiate between various instruments or voices. How these overtones are accentuated allows you to differentiate the Pope's voice from the President's. A singer can sing the exact same pitch at the same volume as a saxophone, but will sound very different because of the harmonics inherent in each tone. Overtones make the singer and sax have their own unique tonal color, which is known as timbre. Many musical instruments can produce overtones that exceed the limits of the human ear (20Hz to 20kHz). Take, for instance, the aforementioned cricket; it's probably producing additional overtones that sail past the limits of anyone's hearing.

Now the second half of the answer to the earlier question of why you need a sound system that has a range up to 20kHz is this: to avoid the dull, muffled, lifeless sound that comes from not reproducing all the overtones that the human ear can hear. Even a sound as seemingly simple as a speaking voice has overtones up to 7 or 8kHz. It goes without saying, of all the instruments, voices, and sounds that we mix in our various houses of worship, the spoken word is at the top of the list.

The Acoustical Space

At this point you have a basic understanding of:

- Vibrations and sound waves
- Frequencies and Hz
- Decibels and SPL
- Overtones (a.k.a. harmonics or partials)

Now we need to talk a bit about the environment in which your sound-reinforcement system lives. We will only touch on some basic aspects of your particular house of worship at this point, as we will dive into acoustic treatments later in Chapter 7, "Creating a Sound Environment." However, it is important that you have an idea of what is happening to the sound inside your particular house of worship right from the start.

LIVE OR DEAD?

The insides of most buildings are made up of one or more of the following materials:

- Wood
- Brick
- Plaster
- Drywall or sheet rock
- Glass
- Concrete
- Linoleum
- Carpeting

Sound waves interact and react with all these surfaces differently. Let's say you have speakers hanging above the stage of your church and 100 feet in front of those speakers is the back wall of the building. The wall happens to be constructed of concrete block with a plaster coating—in other words, an extremely hard surface. When the sound waves from the speakers hit that wall, a few things happen.

- Some of the sound waves are reflected back into the room.
- The wall absorbs some of the waves.
- The wall lets some of the waves pass through it.

If your house of worship has walls, floors, or ceilings made of the aforementioned materials, you have a problem. We have all been in a high-school gym; if you clap your hands or stomp your feet in a gym, you will hear high- and mid-frequency sound waves bouncing all over the place like a ping-pong ball on four cups of espresso. These sound waves are traveling through the air at 1,130 feet per second (assuming it's 68 degrees and you're at sea level). The speed of sound is independent of frequency. The fact of the matter is, sound is moving pretty darn fast. When it bounces off the hard boundaries, the reflected sound is called reverberation or reverb (see the following note). If you experience this reverb effect in your house of worship or any building, it is referred to as a live room. How live your church is varies with the combination of hard surfaces. The opposite of a live room is obviously a dead room. This describes a room in which there are very few reflected sound waves. Take note: An exceedingly live worship room can wreak havoc on the sound of the choir or worship band, or more importantly, the intelligibility of the preacher, priest, rabbi, etc.

GOT REVERB?

Let's take a closer look at how reverb works. You first hear the sound from your worship house speakers. An instant later, some of that original sound bounces or reverberates off the back wall of your church and into your ears again. In another instant, a bit more of the original sound ricochets off the floor and ceiling and once again into your ears. As more sound leaves your worship house speakers, more sound bounces into your ears and, if the inside surfaces of your house are hard or mostly hard, your ears get overwhelmed with a wash of frequencies. This, ladies and gentlemen, is the acoustic property known as reverb. Reverb in small doses in many instances is pleasing to the ear. The world around us is filled with reflected sounds, and our ears recognize limited reverb as normal. But when there is too much reflected sound, our hearing is overdosed, and we have difficulty identifying frequencies.

KNOW YOUR SPACE

Listen to your worship space. A moderately live room can enhance sound by providing natural reverb, which makes music and voices sound fuller—and that, to most people's ears, is better. But there is a fine line here. Use your ears and you will have a better understanding of your worship space and how to work with it.

Of course, using your ears may not be quite enough—you may want to use your feet as well. Walk around your house of worship during a service and listen to how the choir or worship band (if you have one) sounds from different locations. If you have a balcony, get up there and listen. While you're at it, listen from under the balcony. Listen everywhere in the worship room that you can. Ask yourself questions like, "Can I hear every word the preacher is preaching?" "Do I understand the words the choir is singing?" or "Can I distinguish the piano from the guitar in the worship band?" Whether you are mixing a service or not, get your ears tuned up to the entire space.

NEW GIGS ALWAYS MEAN NEW DIGS

If you are a hired gun (that is, you are employed by various houses of worship to mix) and you get a new house of worship as a client, be sure to visit that church before you mix a service. Make an appointment with the pastor or a staff person so you can give the worship space a once-over. Throw a CD or an iPod up on the system and listen to the room. Getting into a worship house and listening prior to your first gig there can give you a great deal of insight into your new work space. It is definitely worth whatever time and energy it takes, and it will make you look like a real pro to the people who are hiring you.

So far we've covered a lot of definitions and characteristics of sound. If anything is setting in, let this be at the forefront: Mixing live worship sound is less about mathematics and more about hearing. With the basic knowledge laid out for you in this first chapter, your ears should start to become more in tune with the sounds of your house of worship and the world around you. Whatever you do to improve your hearing skills will pay off tenfold. This also includes listening closely to the sound in your house of worship. Tuning into and understanding what the environments in which you work are made of and how those materials affect sound will increase your ability to do your job. If you can hear a problem, there is a good chance you can fix it. So, listen, listen, listen; and, well, continue reading!

CHAPTER 2

Microphones

The microphone represents the beginning of the audio signal path you project to your congregation, making it an indispensable weapon in your arsenal. In this chapter, you will discover what a microphone is, how it operates, and the different types you are most likely to find in your house of worship. You will learn how to identify microphones by their polar patterns and what type of mic to use for specific applications. You will also explore techniques for miking your various house-of-worship subjects—the preacher, the choir, and the worship band. Finally, you'll explore the many tools involved in the miking process as well as learn about the most important type of microphone in any house of worship: the wireless mic.

Microphones Defined

Simply put, a microphone "hears" a voice or instrument through its diaphragm (see the upcoming note) and sends what it hears to the mixing console (explained in detail in Chapter 3, "Mixers and the Art of Mixing"). The term "hear" refers to a mic's ability to accurately change acoustic energy into electric energy (more on that in the next section). In order to get the most out of your mics, it's a good idea to take the time to get to know them from top to bottom—so, let's do just that.

WHAT THEY ARE

As stated earlier, a mic turns the sound it hears—known to us as acoustic energy—into electrical energy by way of a process called transduction. This creates an audio signal. Transduction is done by—you guessed it—a transducer, which is simply a device that changes one form of energy into another. A mic is a type of transducer you'll use in your house of worship on a daily basis along with another type of transducer: speakers. (In the case of a speaker, however, that's a transformation of electrical energy back into acoustic energy.) After a mic picks up the signal, the transduced signal then travels through a microphone cable into the house mixing board and ultimately out through the speakers (see Figure 2.1). Although there are hundreds of makes and models of microphones, they all do the same job—albeit in a variety of ways.

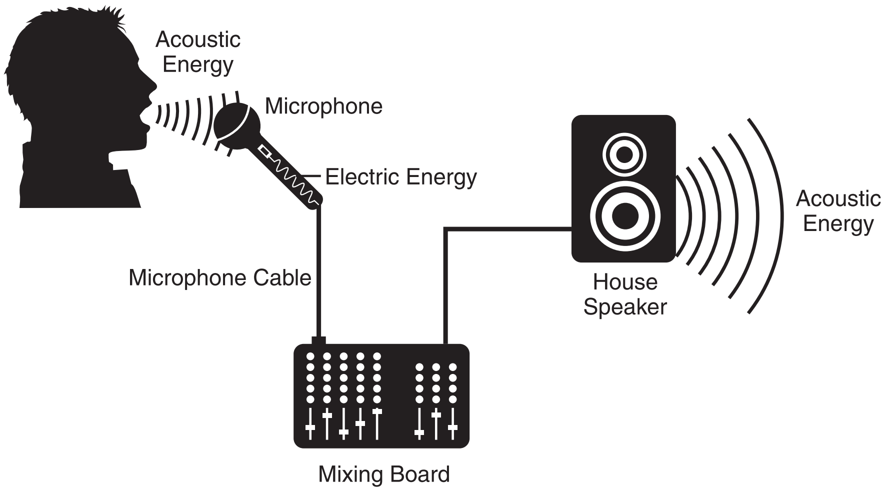


Figure 2.1
A common example of acoustic energy you'll deal with in any house of worship will be the human voice. From the mouth to the mic is where the first half of transduction takes place. After the signal travels through the mic cable and mixing board, it's transduced once more back to acoustical energy by the speakers.

DIA- WHAT?

While microphones come in various forms, they all have one thing in common: a diaphragm. The diaphragm is the component that makes microphones do what they do by vibrating in response to the sound that it is subjected to. That vibration is what creates the electricity that travels to the mixer and beyond. For instance, in a dynamic microphone's cartridge is a voice coil attached to the diaphragm as seen in Figure 2.2. When the diaphragm is stimulated into movement by incoming sound waves, the voice coil, which sits between two magnets, moves—thus creating an electrical charge in the cartridge. A condenser microphone also has a cartridge with a diaphragm at the center of it, but the difference is that the condenser mic diaphragm (see Figure 2.3) is already charged with electricity via phantom power. The diaphragm is mounted just above a conductive ceramic plate separated by a small air gap. When stimulated, it vibrates and moves back and forth toward the ceramic plate.

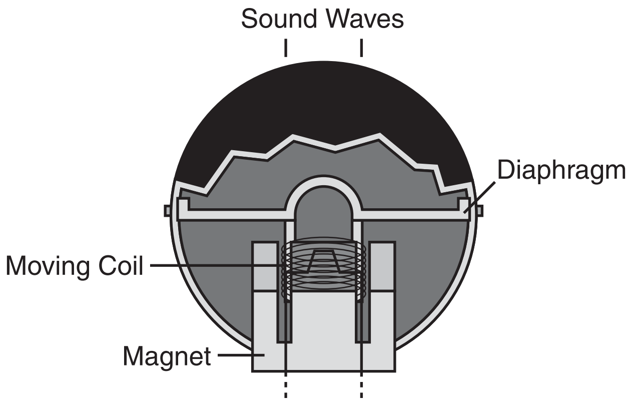


Figure 2.2
As you can see in this cross section of a dynamic mic, the voice coil sits between two magnets and, when stimulated by incoming sound waves hitting the diaphragm, generates an electrical charge. This is the transduction process, which is the very same process of transduction employed by electric guitar pickups!

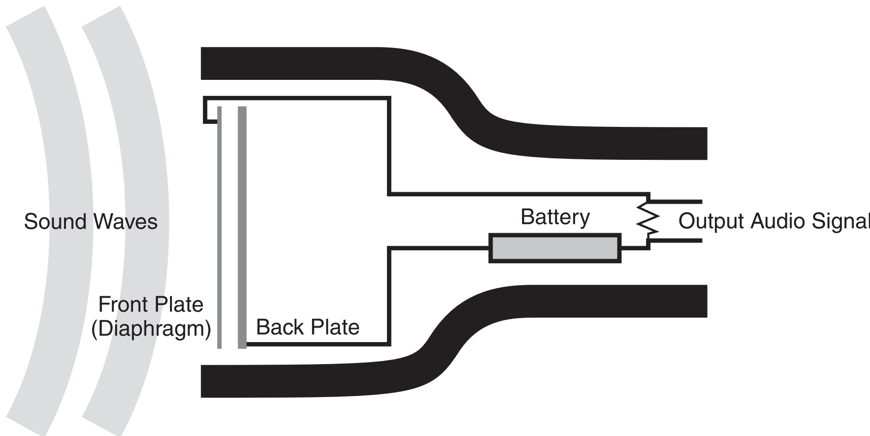


Figure 2.3
Looking at this cross section of a condenser mic, you can see the diaphragm and the conductive ceramic back plate are wired. This matrix is powered by phantom power, which travels from the mixing board, where it is generated, to the mic through cables.

TYPES OF MICROPHONES

Microphones come in a variety of shapes, sizes, colors, and so on, but the two most important characteristics of any microphone are its type (how it hears and translates signal) and its polar pattern (its directional sensitivity). Let's start with the common types of mics you'll encounter in your house of worship:

- ▶ Dynamic
- ▶ Condenser
- ▶ Electret condenser

Dynamic Mics

The most common mic is the aforementioned dynamic microphone. In fact, it's safe to say that if you have one of type of mic in your house of worship, it's a dynamic. This mic design is extremely durable. It can take a lot of physical abuse and that's, well, a godsend. As mentioned, dynamic mics work like this: Sound energy (your voice, your pastor's voice, or any other sound) hits the diaphragm, making it vibrate. This vibration creates a small amount of electrical energy that travels through the mic cable to the mixing console and on to the house speakers.

This transduction of sound to electricity is really quite simple; it's when you have to decide where to position your mics in order to get the best sound out of an instrument or person that your job becomes more complex. Be sure to carefully read the section "Microphone Techniques" later in this chapter for information and advice on the mystical subject of mic placement. With a high-quality dynamic mic such as a Shure SM 58, you will not only get the durability mentioned earlier, but you can also expect a very natural vocal sound.

Condenser Mics

The next most common microphone is the condenser mic. As mentioned, like its dynamic brethren, this mic is also equipped with a diaphragm. The difference is it needs to be charged with electricity to operate. Power between 9 and 48 volts is required to fire up your condenser mic. Although most of these mics have compartments for a battery (refer to Figure 2.3), typically the power will come from your mixing board through the phantom power supply circuitry via the mic cable to the condenser mic.

Condenser mics are generally better than dynamic mics at picking up sounds from a distance. Therefore, you'll find these mics used as choir mics, podium mics, and/or drum overhead mics. We also like condensers for solo vocals and acoustic guitars because of their natural, smooth sound. Condenser mics tend to have a hotter (louder) signal because they are all equipped with an amplifier to boost the signal. Compared to a dynamic mic, the condenser mic will output a few more decibels as a result. This attribute will come especially in handy when you need to mic softer-spoken speakers and singers.

The downside of condenser mics is their durability. Unlike a dynamic mic, you really don't want to drop a condenser mic or knock it around. Also, if your house of worship is in a climate relative to the jungles of Brazil, the moisture from the humidity will wreak havoc on the internal components.

Electret Condenser Mics

The main differences between an electret condenser mic and the previously described type is the size (the electret mic is much smaller) and that the diaphragm in the electret mic is constantly charged. With regard to the latter, the manufacturer gives the diaphragm a static charge that lasts indefinitely, which allows an electret to function *sans* phantom power. Now only the internal amplifier needs voltage to operate. Need an example? The compact lavalier (or lapel) microphone (see Figure 2.4) that your preacher has clipped to his or her shirt or worn around his or her neck is almost always an electret condenser.

Figure 2.4
Unlike its larger cousin, the full-sized condenser microphone, this pint-sized electret condenser mic holds a constant charge, meaning it doesn't require phantom power.

While there are certainly more microphone types out there, the three types introduced here—dynamic, condenser, and electret condenser—are the most common. Other types are rare in worship sound use.

GOING THE EXTRA MILE

To add to your job's complexity, you will have to decide what type of mic to use in any given scenario—but your house of worship's limited microphone arsenal may already decide that for you. Unless you're a hired gun who travels from one house of worship to another (in which case you will probably carry a few of your favorite mics with you for all the worship mixing gigs you do), you will ultimately use the microphones that your house owns. Most of you reading this book are most likely volunteering at your house of worship and will not go out and purchase a half dozen high-end microphones. But if you bring just one high-quality mic to your next worship event, you will probably become the hero of your church. Whether you go to a church, temple, mosque, ashram, or whatever, God needs heroes!

MICROPHONE POLAR PATTERNS

A microphone's polar pattern (also known as its pick-up pattern) is its sensitivity to incoming sounds at certain angles as compared to its primary axis point. The physical placement of the mic in relation to its subject and the polar pattern employed determines how it will perform. Put simply, it's how a mic hears its subjects. Some mics hear incoming sounds from only one area (unidirectional) of the mic while others hear from all areas (omnidirectional). This, along with the type of mic, really defines a microphone.

Once you put some time into working with microphones, you will come to identify mics by their type and polar pattern and be able to quickly assess what mic to use by these specs alone. Some polar patterns to be aware of are listed here. As you read through these descriptions, take note: When looking at any polar pattern on paper, try to imagine the pattern surrounding the mic three dimensionally since all patterns exist in the three dimensional world.

- **Cardioid.** Named after its heart-shaped pattern, as seen in Figure 2.5, the unidirectional cardioid pattern is the most common microphone polar pattern. Mics set to a cardioid pattern primarily pick up sounds that come straight into them and reject sound from both the sides and rear of the mic. The front of the pattern is the primary axis, which, in the case of a cardioid mic, is the point where the microphone hears most accurately and naturally. Consequently, the rejection of sounds entering the rear of the mic helps reduce sensitivity to feedback, a subject all its own, to be discussed later in the book. It's important to note that off-axis sounds (those not coming straight into the microphone) are not completely cut off, but do tend to take on a different sound or tonal color. Therefore, when you adjust the EQ on a mic with a cardioid pattern, you must place the mic directly in front of your subject.

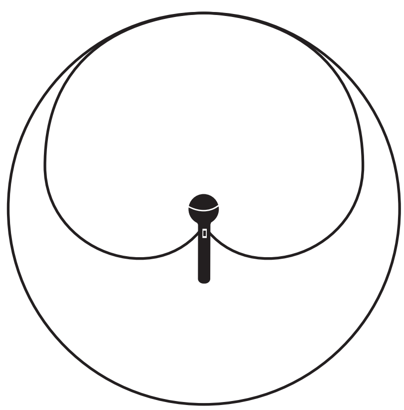


Figure 2.5
Here we see a cardioid microphone polar pattern. The middle circle represents the mic's capsule, and the heart-shaped image represents the sensitivity zone, where the mic can hear its subject.

- ► **Omnidirectional.** The prefix "omni" offers a hint as to this mic pattern—it hears sounds equally from all directions (see Figure 2.6). Although it's not exactly equal, for the purposes of our studies, "equal" describes the pattern well. Occasionally mics used for choirs are omnidirectional, and lavalier microphones are almost always omnis. Whatever the application, they can sound very natural with a rich bass frequency response. Most houses of worship shy away from using mics with an omni pattern because they may have less in the way of feedback-rejection qualities than the cardioid pattern.

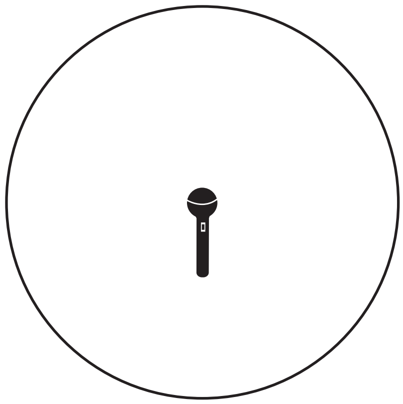


Figure 2.6
The all-hearing omnidirectional pattern allows a mic to pick up sound from all angles.

- ► **Supercardioid.** This super-unidirectional pattern hears sound coming straight into the primary axis point, but in a much tighter manner than the cardioid pattern (see Figure 2.7). The result is even more off-axis (side) sound rejection. Microphones with this pattern are used when there is a desire to really limit the pickup of sound from the sides of the mic.

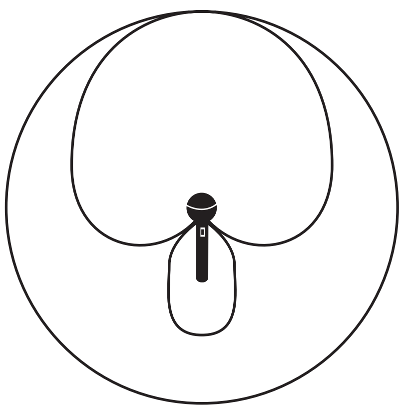


Figure 2.7
This supercardioid pattern enables a mic to isolate its sensitivity zone more so than the standard cardioid pattern.

Just like there are other microphones to work with, there are other pickup patterns—but it's unlikely you'll come across any of them in your worship sound journey.

SOME TECH TERMS TO KNOW

Throughout your work as a house sound technician, you'll encounter myriad situations where you'll need to troubleshoot problems. In order to effectively do so, you must be able to identify them. Here are some important microphone-related terms that you will absolutely need to be aware of to be on top of your game. While we will delve deeper into each of these subjects later in the book, it's never a bad idea to get a head start on learning something new!

- **Feedback.** When a sound entering a microphone travels out to the house speakers or stage monitor speakers at a decibel level that is loud enough to be heard by the same microphone, it may result in that sound creating an indefinite loop. This cycling of sound can cause a sharp squeal, a low howl, or slowly build up into a deafening ring. Not all frequencies entering the microphone cycle though. Usually, a specific frequency or band of frequencies that resonates with the stage or worship house itself will be the culprit. The simplest way to reduce feedback is to lower the level of the microphones, but we will be looking into a variety of other feedback remedies later in the book.

- F**requency response.** This refers to a microphone's ability to reproduce the frequency content of the sound it hears. Manufacturers of microphones (and speakers) sometimes design their products to reproduce audio with a flat frequency response, meaning the unit will not color the sound in any way and will therefore give the listener the most accurate representation possible. We all want our preacher, choir, and worship band to sound the same through the sound system as they do naturally (without a sound system), right? Some microphones, however—as well as speaker enclosures—have their own characteristic frequency response. With regard to a microphone, this can be both an advantage and a disadvantage. Using a mic with an exaggerated bass or low-end frequency response can give your preacher more presence and power during the delivery of his or her sermon or, conversely, give a preacher with strong vocal cords too much "umpf," so to speak. Using a microphone with a slightly inflated high-end or high-frequency response can make your choir sparkle and sound clearer or, on the other hand, make them sound thin and almost be painful to the congregation's ear.

- **Proximity effect.** As your sound source (especially vocals) gets closer in proximity to a cardioid or supercardioid microphone, the low frequencies increase. This proximity effect can be useful once you understand it and can control it, but it can also cause boominess and make vocals unintelligible. Note: This effect doesn't occur with omnidirectional mics.

MICROPHONE USES

In general, the mic pattern will be a secondary consideration to the type of mic you use. Table 2.1 is a collection of common scenarios and suggestions for what type of mic should be used in those scenarios. Also included are some less-than-standard instruments. Even though every house of worship is physically different, the basic use of microphones stays the same from house to house. That said, lots of different scenarios—including electronics, mixer settings, and EQ—can and will affect how a mic sounds. So keep that in mind when implementing these suggestions; the results will vary from room to room, and you may find yourself, after some experimentation, going with another choice.

Table 2.1 What Mic for What Occasion?

Voices and Instruments	Recommended Microphones
Preacher at a podium or pulpit	Condenser/cardioid pattern
Preacher with loud voice at podium	Dynamic/cardioid pattern
Roving (non-stationary) preacher	Electret condenser lav/omni pattern
Choir (10 or more singers)	Condenser/cardioid pattern
Individual singers	Dynamic/cardioid pattern
Electric guitar amplifier	Dynamic/cardioid pattern
Acoustic guitar	Condenser/cardioid pattern
Electric bass amplifier	Dynamic/cardioid pattern
Acoustic bass guitar	Condenser/cardioid pattern
Kick (bass) drum	Dynamic/cardioid pattern
Snare drum	Dynamic/cardioid pattern
Tom drum	Dynamic/cardioid pattern

(continued)

Voices and Instruments	Recommended Microphones
Hi hat	Condenser/super-cardioid
Overhead drum	Condenser/cardioid pattern
Acoustic piano	Condenser/cardioid pattern
Saxophone, trumpet (brass)	Dynamic/cardioid pattern
Flute	Condenser/cardioid pattern
Cello	Condenser/cardioid pattern
Violin	Condenser/cardioid pattern

Although you can stick with these suggestions, as you gain more experience with the use of different mic types, feel free to mix and match mics and patterns and listen to the difference. Experimentation and exploration are key in this field!

Microphone Techniques

Selecting the right microphone for a specific task is only half the battle. The positioning of the mic can be as important as the type of mic you choose. In this section, some tried-and-true mic techniques are explained. Once you feel like you have a handle on these concepts, as always, try some of your own.

MIKING YOUR PREACHER

Wouldn't you know it? The most important person in a house of worship, the preacher, is the most difficult to mic. (Note: The word "preacher" is used to describe male or female pastors, rabbis, priest, etc.) Many preachers deliver their message across a large dynamic range—approximately 80–105dB—which can wreak havoc with your mix. If there's too much power in the delivery, then intelligibility might be lost. On the other hand, a quiet line or two for drama's sake, and the words might not be heard. If this describes your preacher, you will have to ride the channel fader (explained in the next chapter) during the sermon. A good approach is to have your preacher stationary at a podium or pulpit with a mic attached a podium stand. Many preachers, however, want to move around while delivering the message—this is where a handheld wired or wireless mic, discussed at the close of this chapter, comes in handy. A lavalier mic (see Figure 2.8), if in a wireless configuration, will not only allow movement but will also let your preacher use his or her hands to accentuate what he or she is saying. In the end, whatever microphone is used, a powerful or gentle preacher needs a good sound man to help with the delivery.

Figure 2.8
Lavalier mic positioning ultimately depends on what's most comfortable for the preacher. Once that's established, it's your job to accommodate its placement with any needed volume or EQ adjustments.

MIKING YOUR CHOIR

Oddly, compared to a preacher, choirs are much easier to mic. To mic the choir, your house of worship may have condenser/cardioid mics already installed, which may be hanging from the ceiling. If this describes your house, the height at which they are set is probably adjustable. If risers are used, set the mic or mics for the first row—about two feet above the choir's heads, and about two feet in front of them (see Figure 2.9). If there are more than one row of mics and more than two rows of choir members, try setting up the second row of mics two feet above the heads of the choir and two feet in front of them as well.

Should your house have a worship band with a few singers in place of a choir, or if you have singing musicians, try miking each voice individually. A dynamic mic with a cardioid pattern is generally a good choice, but use a condenser mic if you need to give a softer singer a decibel boost. Whichever you choose, have your singers sing straight into the microphones—ideally no more than an inch or two from the screen.

Figure 2.9
Here you see a fairly large choir on multi-step risers with four mics on stands set up about two feet above the first row of singers' heads.

MIKING INSTRUMENTS

A human voice can have a very complex set of overtones and that complexity can change from singer to singer or even song to song. Likewise, instruments can be just as complex with their own array of overtones. That said, instruments can be more consistent in their overall tonality, which could be attributed to the fact that instruments are designed with a particular sound in mind. All things considered, miking an instrument can be very different from miking a voice. In this section you'll find some useful information about how to mic various instruments you are likely to encounter in your house of worship.

Fretted Instruments

Although miking guitar amps can be a matter of taste for many sound techs, a good place to start is with a dynamic/cardioid microphone right up to the grill cloth, a few inches just left or right of the speaker's center (see Figure 2.10). If you can't see the speaker(s), try looking in the back of the amp to see inside. If the amp has a closed back, not allowing you to look in from the rear, use a flashlight and point it into the grill.

Figure 2.10
Placing a mic right up against a sound source, like this dynamic mic is to the grill cloth of this Fender amp, is called close miking. The mic seen here is the venerable Shure SM-57—a staple for anyone who mikes guitar amps for any reason.

For acoustic guitars without any electronics to amplify the guitar (called pickups), use a condenser/cardioid set in front of the sound hole on a slight angle as seen in Figure 2.11. Be sure to set your mic far enough away so as not to interfere with the guitar player's strumming hand. If for some reason this miking technique produces low-end feedback in the 100–200Hz range, try placing the mic closer to and slightly below the fingerboard pointing toward the sound hole.

Figure 2.11
Miking acoustic guitars with a condenser mic on the sound hole at an angle is the way to go if you're looking for that instrument's true tone. Make sure when miking any musician on any instrument that the mic is not in a position where it can be hit during the performance.

When it comes to electric bass guitars, it's normal to use a direct line or direct box (see the "Direct Boxes" section later in this chapter), which sends the signal straight to the mixing console. If for some reason you need to mic a bass guitar amp, use the same technique as for an electric guitar amp: a dynamic/cardioid microphone set back a couple inches from the grill cloth and a few inches off axis from the center of the bass speaker.

Drums

With regard to drums, you're most likely miking a drum kit, so you should always allot yourself enough time to do the job right. Following are some general suggestions for miking your house of worship's drum kit. (See Figures 2.12–2.16 as you read). Though the steps in the list appear in a particular order, you might find yourself miking each drum using a different sequence—and that's just fine. As stated several times before, be sure to experiment with different microphones and placement configurations for the best overall results.

1. **Kick drum.** Kick drums have several variables to consider when it comes to the front of the drum. If your drummer's kick has a drum head on the front of the drum and it has a hole in it, place your dynamic/cardioid mic just inside the hole (see Figure 2.12). Should you have a drum head with no hole, place your mic just in front of the head without making contact. If there's no cover, place your mic on a pillow or piece of foam rubber on the bottom of the drum.

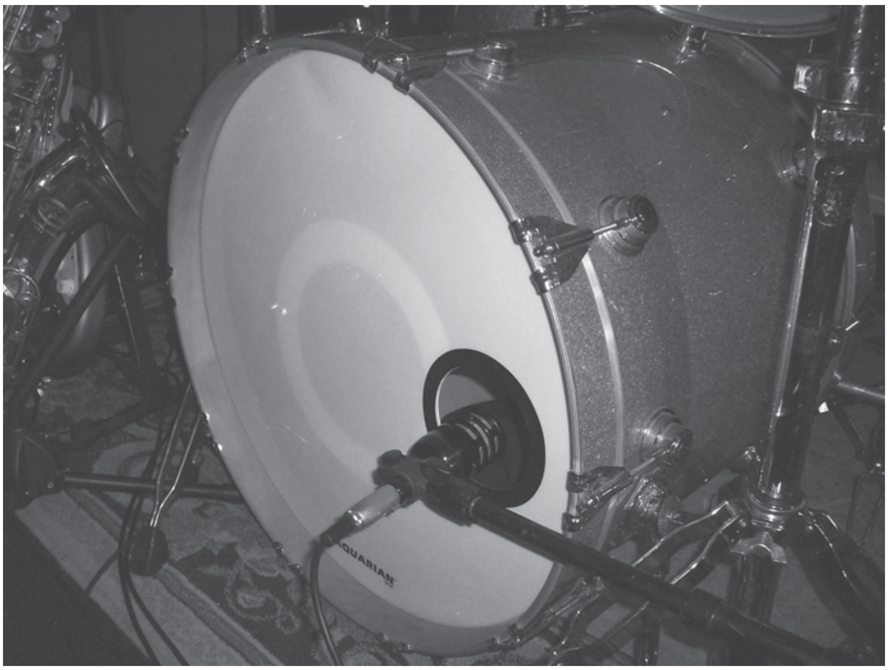


Figure 2.12
One way to mic a kick drum is to place a mic just inside the hole in the drum-head cover on the lower right.

2. **Snare drum.** Place a dynamic/cardioid mic just above the rim of the top head of the snare (see Figure 2.13), but be sure to keep this mic out of the way your drummer. While this goes for all mics, it's especially important here because this is the drum your drummer will hit the most.

Figure 2.13
When miking a full drum kit, place the mics where they will be able to best pick up the sound, but make sure the drummer isn't likely to make contact with them while playing.

3. **Hi hat.** Try using a condenser/supercardioid mic on the hi hat, placing it at the outer edge, pointing up at the bottom cymbal (see Figure 2.14).

Figure 2.14
Hi hats can be miked from the bottom cymbal (pictured) or the top cymbal.

4. **Tom-toms.** With regard to the upper and lower (floor) tom drums, place each dynamic/cardioid mic needed at the outer edge of each drum (see Figure 2.15).

Figure 2.15
In order to mic drums such as upper and lower toms, you'll need mic stands that can clip onto the drums, as shown here.

5. **Cymbals.** Place two condenser/cardioid microphones on mic stands on each side of the kit so the mics are 12 to 18 inches above the highest cymbal (see Figure 2.16).

Figure 2.16
This miking scheme is often referred to as overhead miking. When your mic collection (or budget to purchase mics) is small and you need to mic a full kit, quite often an overhead setup, along with a mic on the kick, could do the trick.

Other Instruments

Depending on your house of worship, or if you venture out and become a freelance sound tech for other worship houses, you may encounter some more esoteric instruments to mic from time to time. Following are some suggestions for rarely seen—but beautiful-sounding—worship-house instruments that may need your attention someday. While these suggestions will be good reference for your initial microphone setups, remember to experiment with your own configurations. Every room is different and requires its own setup parameters.

- **Piano.** Occasionally, you will need to mic an acoustic piano. For this frequency-rich instrument, try using two condenser mics, both set to a cardioid pattern, with one at either end of the harp (also referred to as the soundboard) and placed about six inches in. Point your mics toward the center of the harp but not directly at each other. Be warned: Properly miking an acoustic piano will take a fair amount of trial and error, patience, and keen hearing—but the sonic rewards are well worth the effort.

- **Wind instruments.** With brass instruments such as trumpets and trombones, and with reed instruments such as saxophones, try a dynamic mic set to a cardioid pattern on a stand close to the bell (opening) of the instrument. If you run into a flute, use a condenser/cardioid combo and set it on a stand near the player's mouth.

- **Bowed instruments.** For cellos, violins, and violas, use a condenser/cardioid combo with the mics on stands set four to six inches back from the sound holes.

Microphone Connection Tools

No matter what type of mic you choose to use and where you place it, each and every one of them needs a cable to take the signal they hear to the mixing console (the wireless microphone being a notable exception). In this section we'll take a look at the primary cable type used for connecting microphones while exploring some additional hardware options for making your connection more efficient and functional.

BALANCED XLR CABLES

Once sound enters a microphone, the now transformed energy starts its journey to our ears. To make that journey, the signal will need to travel through a specific cable—called an XLR cable—that's connected from the bottom of the mic to the mixing console (or whatever is the first piece of gear in your signal chain after the mic). An XLR cable is easy to spot; it has three prongs at one end (known as the male end for connecting to a mixer or snake end) and three holes (known as the female end) at the other (see Figure 2.17). Internally, these cables are composed of two wires of equal length, which are individually coated with a plastic material and wrapped in a third wire that acts as a shield surrounding the two wires. Together these three wires are encased in a coating of rubberized plastic (most common). The shield wire in an XLR cable is grounded; it is this component that protects the cable from electromagnetic noise. With this in mind, it's not a good idea to have your XLR cables running across your electric AC cables or vice versa because crossing these wires can introduce unwanted noise into the signal path.

Figure 2.17
Pictured are the jacks for an XLR cable with the female end on the left side and the male end on the right.

Going back to the two wires encased in the shield, these are considered to be balanced. Understanding this term will help immensely when you're assembling a sound system that you need to be noise free (i.e., with no buzzing or humming). The term balanced is used to describe the relationship between the signals carried on the innermost pair of wires within an XLR cable. These two wires come in a positive and negative configuration, with the negative being designed for noise reduction. Because the receiving device (usually a mixing console) is immune to the noise from the negative wire, it's not heard by you and your congregation—and that's a very good thing.

Another advantage to a balanced XLR cable is its ability to run long distances without significant loss to high-end frequencies (known as capacitance). This is also a very good thing since it's not uncommon for microphones to be far away from the mixing console, requiring you to run long cable lengths.

SNAKES

Because we're talking about cables, let's address the fact that you're going to run a *lot* of them—and depending on how far your console is from the stage, it could create quite a mess. Don't fret; a handy piece of hardware called a snake will clean that up and help organize things immensely. A snake (see Figure 2.18) combines a group of balanced XLR cables into one organized, rubberized plastic casing. At one end you have a stage box with numerous XLR plugs in it (although there can also be several male XLR jacks, called returns, as seen in Figure 2.18), and at the other you have an equal number of independent male XLR jacks called a pigtail. The cables encased in the snake can carry audio signals to and from the mixing console without the hassle of multiple cables strewn all around the floor, making a snake, well, another one of those godsends.

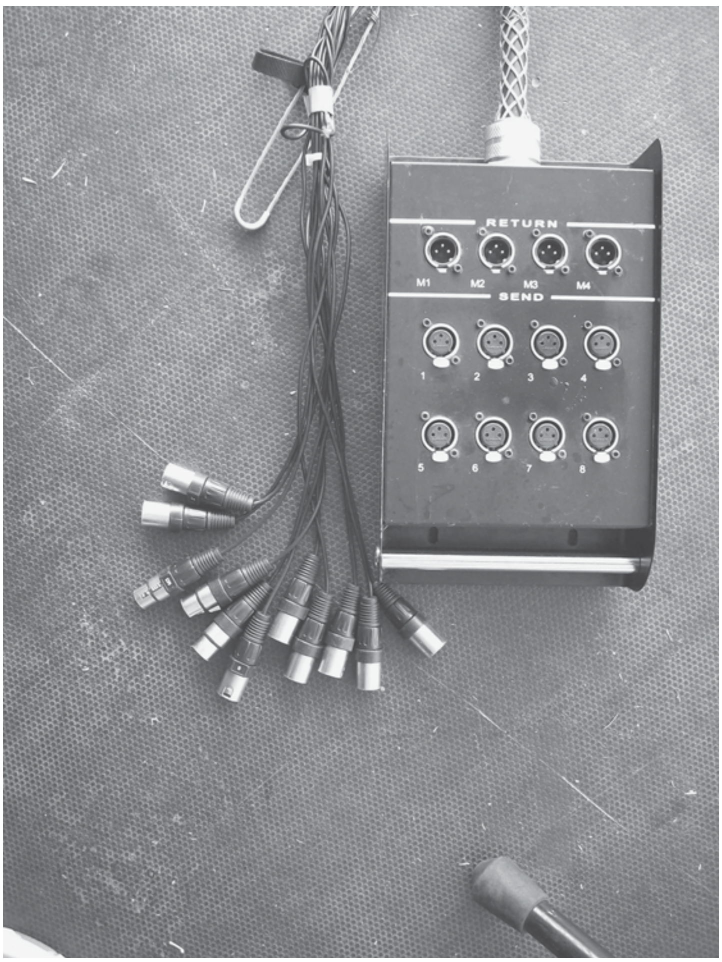


Figure 2.18
The two main components of a snake are the pigtail (seen on the left), which makes the connection to the mixing console, and the stage box, which is where you connect to the stage mics using XLR cables.

DIRECT BOXES

Another essential tool to have in your toolbox is a direct box or DI (short for direct input or injection). This pint-sized box (seen in Figures 2.19 and 2.20) converts an unbalanced output signal from a source carried on a 1/4-inch cable to a balanced signal and sends it off to your console via an XLR cable. This not only helps organize your cabling by making all your cables coming to the board XLRs, but more importantly, it ensures the signal is noise free. Some examples of instruments with unbalanced outputs that require DIs are as follows:

- Electric keyboards
- Synthesizers
- Bass guitar
- Acoustic guitar with its own pickup/pre-amp system
- Computer or sampler

Additionally, a direct box will have a second 1/4-inch phone jack, allowing an instrument cable to be sent to an instrument amplifier for on-stage monitoring (see Figure 2.19) as well as other controls for changing the polarity or lifting the ground (see Figure 2.20).

Figure 2.19
This side of the DI has 1/4-inch jacks for plugging in your unbalanced instrument and sending that same signal to an amplifier for monitoring.

Figure 2.20
From this side of the DI, the signal is sent out to the mixing console as a balanced signal on an XLR cable.

While a passive (non-powered) direct box uses an internal transformer to convert the input signal from unbalanced to a balanced output, many *active* direct boxes use an electronic circuit to convert the input signal. An active DI may serve you better as they tend to send a cleaner, more powerful signal, and they're easier to deal with because you don't have to remember to put batteries in them! Direct boxes may also include the following controls in the form of a switch or a button:

- **Pad.** If an incoming signal is too hot (loud) for your system to handle, it may not be usable by the time it reaches your house mixer. The pad function is provided as a means to reduce the input signal level to one that is more suitable for your mixer so there's no harsh-sounding distortion.

- **Ground.** The ground button can be engaged to reduce noise; sometimes it works and sometimes it doesn't. The inconsistency lies in the sad fact that older buildings (including *many* houses of worship) generally have poor grounding—although, depending on the direct box, the circuitry may be sufficient enough to overcome ground buzz and noise.

Wireless Microphones

Until now, this chapter has discussed a system that almost exclusively works on cables. But what if there were a way for the preacher or lead vocalist to free themselves from the constraints of wires? Or a way for you to eliminate some wires in the spaghetti-like mess that will inevitably come to be? Well there is—wireless microphones!

WHAT THEY ARE

A wireless mic system is made up of a microphone, a transmitter (which can sometimes be in the mic itself), and a receiver. The transmitter sends the incoming signal off to the receiver, which is connected to your mixing console by an XLR cable making this a balanced connection. Much like a radio receiver tunes into a radio station, the wireless mic receiver needs to be tuned to the wireless mic's transmission frequency. Wireless mics can come in two configurations:

- **Handheld wireless mics.** These incorporate the transmitter in the body of the mic along with room for the battery (or batteries) that are needed to power the transmitter (see Figure 2.21).

Figure 2.21
With a wireless mic system, your preacher or vocalist is free to move, making his or her sermon or performance more dynamic.

- W**ireless lavalier microphones.** In this very popular type, a thin cable connects a small lavalier mic (refer to Figure 2.4) to a pocket-sized belt pack that houses the transmitter and batteries. This case is referred to as a body pack and can be clipped to a belt or waistband or shoved into a pocket.

HOW THEY TRANSMIT

Wireless microphones transmit their sounds through frequencies that are regulated and controlled by the Federal Communications Commission (commonly known as the FCC—the same guys who regulate commercial radio frequencies and content). It's important to note these frequencies are in the megahertz range (commonly abbreviated as MHz), which is millions of cycles per second and far above what our ears can hear. This range is known as the radio frequency (RF) spectrum. The signals transmitted by wireless mics are in the following frequency MHz ranges:

- **VHF (very high frequencies).** VHF wireless microphone systems use frequencies from two distinct ranges. Inexpensive systems can transmit in the range of 49.81–49.90MHz. Now, 49,000,000 hertz may seem like a very large number, but in this spectrum, it's actually on the low side and subject to interference (see the next section). For this reason, this band of frequencies is not very desirable, which brings us to the next range of VHF: The higher 150–216MHz range transmitters are less susceptible to RF and other noise. Even still, some TV stations transmit at these frequencies, meaning your preacher's wireless system might pick up a soap opera and blast it through the house of worship during a service.

- **UHF (ultra high frequencies).** The more expensive, but more stable, ultra high frequency systems range from 400–470MHz and 900–950MHz. The FCC allows more power to be used at these higher frequencies, making it less likely any man-made noise will creep into a UHF system. Another important feature of UHF systems is that more than one UHF wireless microphone can be used at a time because the broadcast frequencies are separated.

CAUTION: INTERFERENCE

Unlike microphones, the wireless system you use in your house of worship can be affected by a variety of outside sources. Interference can come from such simple things as fluorescent lights, auto ignitions, dimmer switches, and sometimes the cab driver talking on the radio as he or she drives by. If your house already has a wireless system and you are experiencing interference problems during your services, you may want to consider a new system. If you're in the market for a new system, take a close look at what you're already using. A UHF system will usually experience less interference than a VHF system for the aforementioned reasons. But the location of your house and various construction materials (especially steel framing) can create anomalies that interfere with the broadcasting capabilities of certain makes and models of wireless systems. With that in mind, see if you can arrange a demo from your local supplier or, better yet, rent a system for a while to decide what system to go with. You really won't know if you have a problem with your system until you set it up in your house of worship and give it a once (or twice) over.

DIVERSITY SYSTEMS

A diversity or true diversity system will sport two or more antennas on the receiver. Circuitry in the receiver will automatically look for the strongest signal from the wireless transmitter. If one antenna has a weak signal, the receiver will switch to the other antenna, which is almost always stronger. This switch process is seamless (and thankfully noiseless) and keeps the microphone signal from dropping out.

We have covered a lot of microphone information in this chapter, which makes a good deal of sense since microphones represent the beginning of the signal path. From our mics, whether traveling through cables or the air, the signal then travels to the mixing console, covered next.

CHAPTER 3

Mixers and the Art of Mixing

Mixing consoles (also referred to as mixing boards, a mixing desk, or just mixers) are the control center of your sound system (see Figure 3.1). Simply put, they receive audio signals from microphones, instruments, and pre-recorded sound sources, and re-route those signals to user-defined outputs. Along the way there will usually be some level adjustment and equalization, not to mention signal processing. In the end, all mixing consoles, no matter how simple or complex, basically do the same thing. At first glance, a mixer with all those knobs, buttons, faders, and lights can be overwhelming, making the prospect of mixing a worship service somewhat intimidating. This chapter's primary purpose is dispel any apprehensions you may have by clearly defining what a mixer is and how to use it to best suit your needs as a worship house sound technician.

Figure 3.1
Mixing boards, though somewhat intimidating, are a hardware organizational tool you cannot do without in your sound reinforcement system. The Mackie Onyx 24.4 shown here is jam-packed with standard mixing console components that will be discussed throughout this chapter.

THE MACKIE FACTOR

All mixing boards are not created equal, but they do share many of the same standard features—although they might be labeled differently. While there is a chance your particular house of worship's board may not be outfitted with some of the features discussed in this chapter, it's more likely it will. That said, items throughout this chapter will be described referencing the same mixing board: a Mackie Onyx series 24.4 (refer to Figure 3.1). Mackie produces exceptional mixing consoles at all levels and can be found in myriad applications all over the world. More than likely, the board you have in your house or worship is a Mackie!

Input Channel Strip

You learned in the last chapter how microphones, along with mic cables and other items, capture acoustic energy (sound), convert it into electrical energy, and deliver those signals to the mixing console. The signal enters the board via an XLR or 1/4-inch cable, which is plugged into a series of vertical strips called input channels (see Figure 3.2). It is these strips of knobs, buttons, lights, and a lone fader that make up the primary matrix of your mixing console and where you do the most basic sound tweaking such as adjusting volume levels, EQing (explained later in the chapter), and signal routing (also explained later) to name a few. This section takes a step-by-step, detailed tour of the input channel strip starting from the top mic/line input so you can get a firm grasp of the essential components contained within them. In order to best understand the concepts, it's necessary to jump around to different sections as you progress down the channel strip. Note: It's important to fully grasp these controls in order to digest the other parts of your mixing desk, so be sure to read this section carefully. Come to think of it—read it twice!

BACK-DOOR ENTRANCE

On medium-sized to larger mixing boards such as the Mackie Onyx 24.4 referenced throughout this chapter, the connecting XLR or 1/4-inch cables are actually inserted at the rear of the board as seen in Figure 3.3. In addition to these connections, there will also be jacks for speakers, outboard effects, and other peripheral items. This not only affords more space for controls on the console's surface, it also gets any dangling cables out of your way.

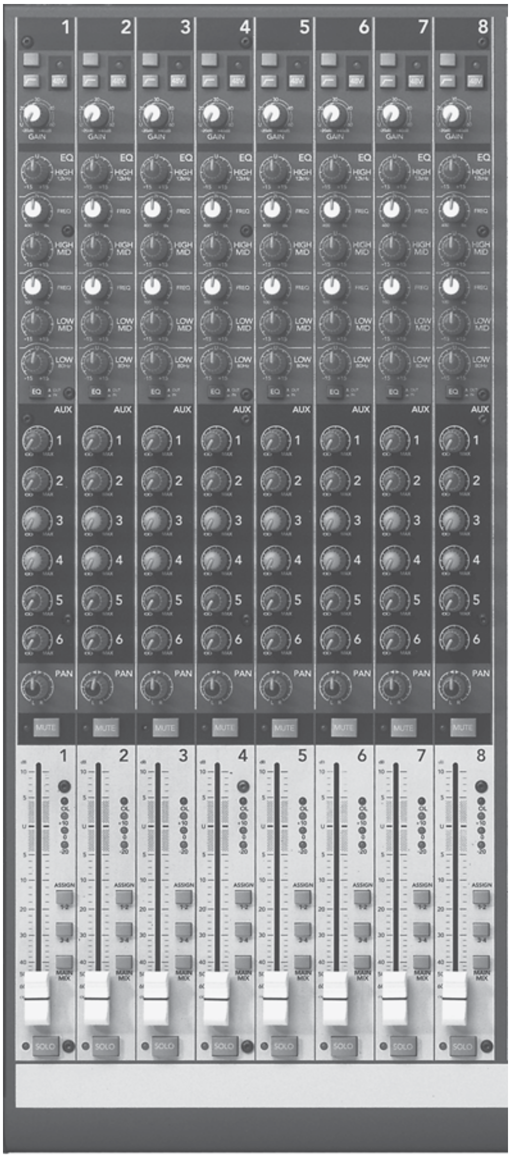


Figure 3.2
A mixing console's primary control matrix is its channel strips. Aside from some feature variations, channel strips usually follow the same order of controls from top to bottom: input signal section, EQ section, auxiliary section, pan control, mute and solo buttons, and fader section.

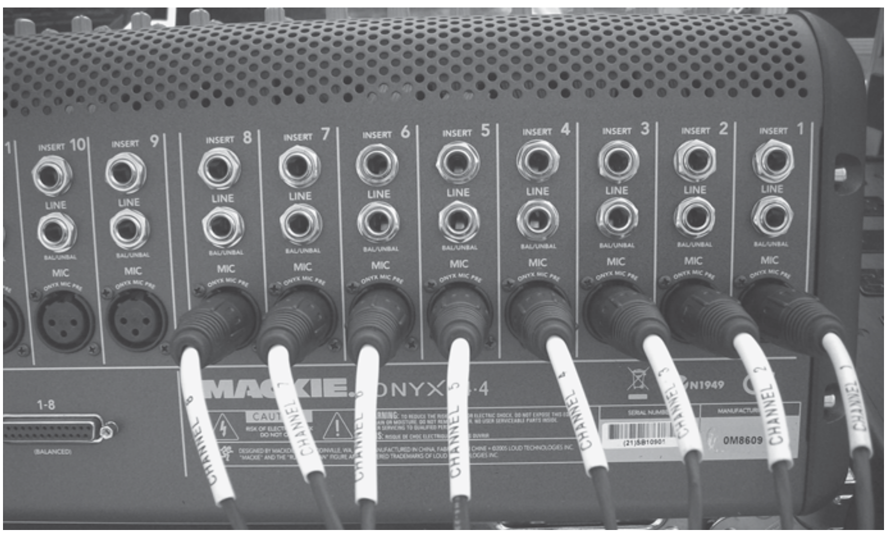


Figure 3.3
The rear of many mid- to large-scale mixing desks will be where most of your cabled connections reside, including the 1/4-inch jacks for inserts (explained later in the chapter) and line-level sources, as well as XLR jacks for mic cables. Note how the mic cables are labeled to identify which channel each one is connected to—good move!

MIC/LINE INPUT

The first stop your incoming signal makes when entering the console is the mic/line input section (see Figure 3.4) of the input channel strip. As stated, the signals can be sent to the board via two cable types: XLR or 1/4-inch. Signals that travel through these cables come in two signal strengths:

- **Mic level.** These signals almost always travel through an XLR cable that plugs into a 3-pin XLR mic input jack, which is a balanced, low-impedance input. (Refer to Chapter 2, "Microphones," for a review of what balanced means; impedance is discussed in the Chapter 4, "Speakers.")The low-level signal that a microphone produces must be boosted by a pre-amplifier. Don't worry; your mixer has pre-amps built into all mic input channels to compensate for the diminished signal strength.

- **Line level.** The line-level input signal enters your board via a 1/4-inch phone jack located below the XLR input. Line-level signals are hotter and generally come from a CD player, iPod, keyboard, or basically any device that produces a signal that doesn't require nearly as much pre-amplification as mic and/or instrument level sources.

Most boards (hopefully including the one in your worship house) have both a mic and line-level input on each channel to accommodate these signal strengths. That said, you must avoid plugging a microphone and a line-level device into the same channel simultaneously. Depending on who manufactured your board, the simultaneous use of a mic input and a line input will either result in the channel strip being muted (producing no sound) or the mic input superseding the line input. With regard to the latter, your board may have a mic/line switch near the gain knob, allowing you to toggle between which source (mic or line-level) is routed into the channel strip.

There are several standard controls in the mic/line input section that you'll be working with constantly (see Figure 3.4). These include the following:

- **Gain knob.** The gain knob adjusts the amount of boost that the channel strip's pre-amp applies to the incoming signal. As you adjust the gain for your individual channels, you are setting up the gain structure for your system (more on that soon). If you crank up this knob too much, you will get distortion or clipping—a harsh sounding transformation of your incoming signal that will be very unpleasant for anyone to hear. Conversely, if you have it set too low, your signal will be too weak and you may experience an audible hum or hiss in the output. The idea here is to get the signal level as strong as possible without clipping. Your house board should have a light emitting diode (LED) next to or near the gain knob that will flash red when you have overloaded the channel's preamp. While many engineers agree that the best way to go about setting proper levels is making use of a channel strip's incoming signal LEDs in conjunction with the PFL function (discussed later in the chapter), you should also make a conscious effort to also use your ears to make these adjustments in order to train them.

- **Pad button.** Should the signal be too strong for your input channel, you can use the pad button. If your mixer has one, it will be by the gain knob and can be used to attenuate the input level by as much as a couple dozen decibels! Keep in mind that when boosting or cutting levels, you will always need to consider the signal-to-noise ratio (see the following note).

SIGNAL-TO-NOISE RATIO DEFINED

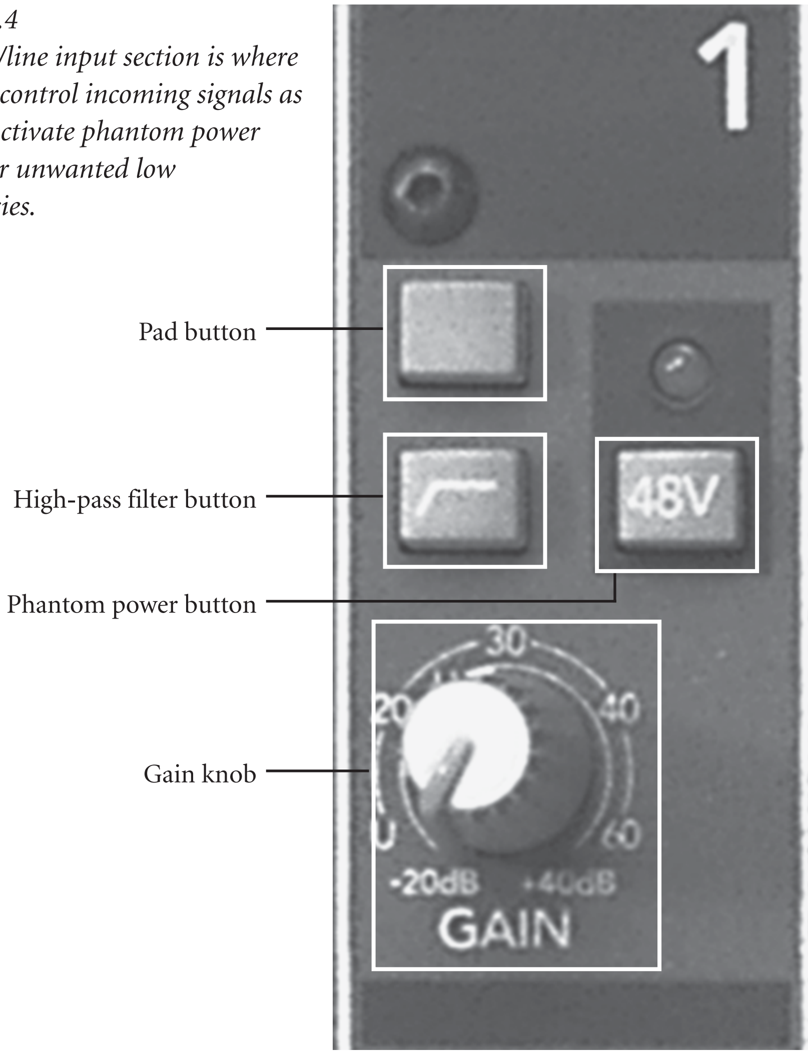
Whenever any piece of audio gear is turned on, it produces a certain amount of low-level noise (usually in the form of a hiss). This is measured by the signal-to-noise ratio (usually abbreviated as either SNR or S/N). Luckily, you can hear this only when the unit is set low enough that the operating noise overpowers the intended output. So when padding a mic, be sure to listen to see whether you've attenuated the mic to a point where the operating noise becomes an issue. If so, you'll need to lower the source producing the signal the mic is hearing. Conversely, when boosting, there's a possibility you can bring out noise that was not apparent previous to your tweak. This can be remedied through outboard gear processing, which you will learn about in Chapter 6, "Outboard Gear."

- **High-pass filter button.** A high-pass filter is used to cut low frequencies or, said another way, to allow high frequencies to pass (hence the name). Usually, the low frequencies you cut are anything below 80 or 100Hz. The high-pass filter control is a quick fix for reducing or removing boominess from an instrument or bass frequency anomalies that may be introduced into an unrelated microphone. For example, say a vocal microphone is on a mic stand on your house stage. The worship band's bass guitar is vibrating the stand at 75Hz and causing boominess through the entire house sound system. You engage your HPF and cut 80Hz and below, making the problem disappear.

BOOST/CUT

The terms boost and cut are used often in the language of sound reinforcement. It's not uncommon to hear a tech use these terms, see them employed in any documentation pertaining to sound gear with regard to increasing (boosting) or decreasing (cutting) the overall volume of main speakers or monitor speakers, or see them used to describe adding or subtracting certain frequencies.

Figure 3.4
The mic/line input section is where you can control incoming signals as well as activate phantom power and filter unwanted low frequencies.



EQUALIZATION

Continuing down the channel strip from the mic/line input section is the equalization (commonly abbreviated as EQ) section (see Figure 3.5). Similar to the HPF controls, these are tone controls for boosting or cutting various frequencies within each channel. A basic console will give you bass and treble controls, while a more professional board like the Onyx 24.4 will offer low, mid-, and high-frequency controls. Generally, EQ is necessary to improve deficiencies in various instruments or voices, make the spoken word more intelligible, and overcome anomalies in a room's natural acoustics. Very often, an instrument that sounds wrong can be made to sound right with proper EQ. But before you reach for those knobs, consider this:

- If an instrument sounds odd, inspect the instrument or the microphone you are using on the instrument before you try to EQ it at the board.
- It's better to remove unwanted frequencies than to add frequencies. The concept of "less is more" works well here. Overly boosting frequencies can have a negative effect on your worship house's speakers and your congregation's listening experience.

Figure 3.5
The EQ section of a mixing board enables you to fine-tune a sound by cutting or boosting various frequency ranges.

Basic Operation

When an EQ knob is in the straight up or 0 position—also known as flat—you affect no frequencies whatsoever. A turn to the right will boost the frequency, and a turn to the left will cut the frequency. With a sweepable EQ system, you will have two knobs (see Figure 3.6)—one to determine which frequencies you will be adjusting and another to boost or cut the selected frequency. Note: Boosting frequencies can cause feedback, so be careful while you tweak.

Figure 3.6
This is a channel strip EQ section from a Mackie 1604 VLZ3. Notice the mid-range frequency control has two knobs, making it a sweepable EQ. The bottom knob gives the option to select several mid-range frequencies, and the top knob acts as the cut or boost control for the chosen frequency.

DETENT

When turning some knobs or moving a fader on your mixing console, you may feel a point of extra resistance. Don't worry—nothing is stuck. It's called a detent or click-stop, and it's used to mark a position of importance such as a neutral setting for an EQ knob or the unity gain setting for a volume fader.

Making Adjustments

If your board has two or three knobs with set frequencies—say, low, mid, and high—simply start by turning the knobs and closely listening to whatever sound you're trying to shape. Let's take a typical application like adjusting the EQ on a snare drum. If the drum is boomy, cut the low end and listen. If you want some additional snap, then boost the mids slightly. Or there could be a ringing overtone that could be remedied by cutting the mids. If you're looking for a little more crispiness in the drum, boost the highs. These are just a few possibilities you can try.

When it comes to sweepable EQ knobs, your options are expanded. Here's another scenario: Say you have an electric guitar that has a bad hollow sound somewhere in the mid range. Start by substantially boosting the level knob of your sweepable mids. Now rotate the frequency selector knob through the range of frequencies until the offending sound is heard clearly. When you have done this successfully, you have identified the offending frequency, and you can cut it with the level knob.

FADERS

Skipping down to the main track control section shown in Figure 3.7, you can see the main slider, called a fader, which serves as your input channel's output volume control. If there's one component that defines a mixing board, this is it. As you work to blend the incoming signals to sound even with respect to volume, it is the fader that you will constantly be riding throughout the service.

Figure 3.7
Faders enable you to easily adjust volume levels up or down on the fly.

After you have set up your mics and adjusted the levels in the mic/line input section of your channel strip, you should set the fader to 0. Take note: Before raising any faders, it's a good habit to make sure the master fader (see "The Master Section" later in the chapter) is all the way down. This prevents any overly loud audio from being mistakenly sent through the system. This setting is known as the nominal or unity gain position, and is the optimal spot to start adding or subtracting the signal level that is being sent to the master section.

IMPORTANT FADER INFORMATION

An input channel fader does not affect the level of the incoming signal, so if you have any distortion problems, either adjust at the gain knob in the mic/input section or lower the source itself.

THE THREE AMIGOS

Before moving onto the auxiliary section, there are three controls on your channel strip that seem simple enough—but are indispensable tools nonetheless (see Figure 3.8). They are as follows:

- **Panoramic knob.** More commonly known as the pan knob, this control in a stereo sound system sends the input channel signal to the right or left master output. When the knob is turned all the way to either side, it's known as hard panning. By setting the pan knob dead center (usually indicated by a detent), the input signal is equally sent to the right and left master out. It's up to you to decide where each incoming signal needs to sit in the stereo spectrum so your entire mix doesn't sound crowded or one-sided. If your board has submix groups (explained later in the chapter), the pan knob, in conjunction with the sub-buttons, can send the audio signal to any of the different subgroup buses.

- **Mute button.** Simply put, a mute button turns the sound off from within the channel through which it's flowing without having to bring down the fader or gain knob. You can mute more than one channel at any given time; some boards may even enable you to mute all channels with one master button. You should always mute a channel when setting up or changing out mics because plugging them in and out makes a loud popping noise that's not only harmful to your congregation's ears but could blow a speaker in your system. Mute buttons can come in handy when the preacher is preaching and you need to mute the instrument and vocal mics when they're not in use to ensure no unwanted noise will be heard. Just don't forget to un-mute the worship band when it's time for them to play! If you have a sudden burst of feedback in a specific channel, the mute button can be your saving grace.

- **Solo button.** Also labeled PFL (pre-fader listen), this control is used to monitor the input channel signal via headphones as well as send the incoming signal to the channel strip LEDs (see the "Meters" section later in this chapter), allowing you to visually check if the gain is at the optimal setting. Being able to listen to an individual channel without interrupting the mix is very handy for troubleshooting, especially during a service.

Figure 3.8
The pan and mute controls (as well as the solo button seen in Figure 3.7) look deceivingly simple compared to the more elaborate EQ and aux send sections, but they serve vital purposes nonetheless.

AUX SENDS

Traveling back up the channel strip, right above the pan and mute controls, are the auxiliary send knobs (see Figure 3.9)—sometimes labeled monitor or effects knobs. These knobs let you simultaneously send the channel signal to another destination such as stage monitors, effects processors, and/or recording devices from corresponding auxiliary outputs (most likely in the form of a 1/4-inch jack). Your board may have sets of two, four, six, eight or more aux sends. With the aux knobs, you can set up a separate mix for your performers, preacher, or choir by adjusting the aux send level control of each channel, which are fed to the aux master and out of the mixing console to the monitors.

Figure 3.9
With aux sends and their corresponding controls, it's possible to set up mixes for your performers to hear that are independent of what the congregation hears.

Your aux sends may be internally wired as either pre- or post-fader, or your console may have buttons that enable you to choose. For auxiliary mixes (ones not heard by the congregation) that you don't want your channel-strip faders to affect, a pre-fader aux is used—on-stage monitor mixes are the most common application. This way, you can mix voices and instruments through your main speakers without changing your monitor mix and vice versa. Pre-fader aux sends are usually pre-EQ and pre-mute button. If you have an aux signal that you want faders to have control over, you engage the post-fader switch. Effects sent through a post-fader aux send, such as reverb, are a good example of this. You can bring up a solo vocalist in your main house mix and increase the reverb level by simply pushing up the vocalist channel fader. In this case, the channel fader directly influences the post-fader aux send. Pre or post signals all eventually travel through a master aux bus, where the ultimate aux level is controlled. You'll examine the master section later in this chapter.

SUBMIX GROUPS

A great organization tool offered by some boards, including our Mackie Onyx 24.4, is submix groups. Located in the master section of the board (see Figure 3.10), submix groups enable you to send multiple input signals from a group of channel strips to a single channel for convenient global adjustments. Should your console have submix groups, consider yourself blessed, as they make your job much easier.

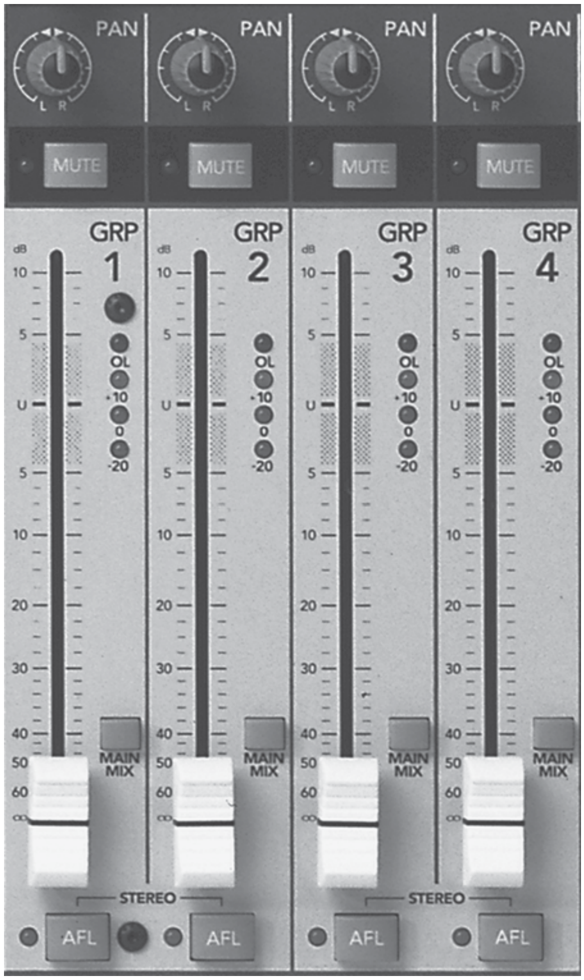


Figure 3.10
Submix group channel strips afford you global volume control over user-defined (that's you) groups of input channel strips.

The way it works is this: Input signals are sent to various submix group faders by way of the channel assignment buttons and a pan knob associated with each respective channel through which they enter the mixing board. Looking back at Figure 3.7, you can see a group of buttons next to the channel fader marked 1–2 and 3–4. Pressing the 1–2 button on any input channel sends the signal to subgroup 1 and 2. (These are the channel strips marked GRP 1 and GRP 2 in Figure 3.10.) From there, you use the pan knob to choose which one of the two subgroup strips will be used—a turn to the left sends the signal to subgroup 1 (GRP 1), while a turn to the right sends the signal to subgroup 2 (GRP 2). Adjusting the pan knob to the center detented position sends the signal to both submix groups' 1 and 2 channel strips.

Submix groups can be a very handy tool during your live mixing efforts. For example, say you've balanced four choir microphones to perfection and you want to be able to adjust their overall level during various songs without disturbing that balance. No problem—just send those four mic channels to a submix via their respective subgroup assign button, adjust your pan knob accordingly, and then tweak the single submix fader when needed. The same can be done with instruments, drums, or background vocals, to name a few. Also, consider yourself blessed once again if your board has submix mutes (also seen in Figure 3.10), as they make it much easier to mute the submix groups than doing so via each individual channel.

AFL BUTTON

Similar to the PFL button is the AFL (after-fader listen) button. You can find this button in the master stereo and aux master bus sections (discussed later in the chapter), as well as in the subgroup section (refer to Figure 3.10). When the AFL button is engaged, it routes a post-fader signal to your headphones and LEDs, enabling you to discretely listen, and visually monitors your stage monitor mixes, aux effects, or main outs, among other things. Unlike the PFL, the AFL enables you to listen to not just one channel, but all the channel-strip signals. You can also listen to all the aux sends; because you will be listening to those channels after the fader, you will be able to accurately hear channel EQ and level.

DO I REALLY NEED HEADPHONES?

There have been a few hints throughout this chapter about having a set of headphones. Those of you who don't have them might be wondering whether you really need them. In a word: absolutely. Headphones are an important part of your arsenal. They enable you and you alone to hear various instruments, voices, monitors, and overall mixes without disrupting a service or bothering the congregation. The better the headphones, the more accurately you will be able to hear audio signals sent through the solo or PFL and AFL buttons.

And Don't Forget…

Now that you have a handle on the ins and outs of an input channel strip—pun intended—let's move on to some other important features of a mixing console that the controller (you) needs to know. It would be impossible in one book to cover every option that could possibly be included on any given desk, but with your newfound channel-strip smarts combined with the information to come, you should be able to walk into your house of worship and understand the basics of whatever board is there. In addition to reading this book, however, it's imperative that you read the console's user manual and set aside some time to explore the board thoroughly. The better you know your gear, the more effective you'll be as a house engineer.

THE MASTER SECTION

Just as a mixing console is the control center of your house sound system, the output section of a mixer (see Figure 3.11) is the control center of your console. Think of this section as the final frontier for all incoming signals before they leave the console. From the aux send master and stereo returns to the submix section, all signals will pass through to the master output fader and out of the board onto their respective destinations (house speakers, recording devices, etc.). In addition to signal control, the output section can feature a talk-back mic, a connection for a lamp so you can shed some light on the console when needed, and master output meters.

Figure 3.11
The master section of any mixing board, including this Mackie Onyx 24.4, allows you total control over all the final destinations of your incoming signals.

BUSES

In order to fully grasp the output section's functions, you need to understand what a bus is. A bus is a signal path composed of multiple signals that have been routed (assigned) to a singular, common destination, which from there travel as one combined signal. Confused? Think of it as being like getting kids to school on a school bus. The kids (incoming signals) are routed to the bus stop by their parents (channel strips), where they are picked up and then delivered (or bused) as a combined group to their school (main output).

In mixing-board terms, a simple example is how the auxiliary sends matrix works. All your input channels have an aux 1 send knob that, when engaged, routes each channel's incoming signal to the aux 1 master bus. From the master aux 1 bus knob, you can send (assign) those collected signals to a single output such as a monitor mix, an effects processor, or maybe a recording device while controlling the overall level of the mix.

All the level controls (knobs or faders) seen in the master section are bus controls. The signals from individual channels that are sent to master subgroup faders are thought of as being bused as well. Think about it: The subgroup faders combine all signals sent to them and send them off on a single path.

STEREO VERSUS MONO

A stereo sound system is composed of two speakers—left and right—usually in a symmetrical configuration to approximate how we hear naturally. To deliver true stereo sound, your mixer must be equipped with a pair of output jacks to connect to the left and right speakers (speakers are the subject of the next chapter). Commonly labeled as main outs or main output, these connections are made from the rear of the board, usually via 1/4-inch jacks with cables designed specifically for connecting to speakers. The fader labeled main mix (refer to Figure 3.11) controls the volume for this final stereo output. The aforementioned pan controls in each input channel strip or subgroup strip allow you to determine where in the stereo spectrum you want to send an audio signal—left speaker, right speaker, or both. In a mono sound system, there is one channel of audio coming out of your board, affording you no options for the reproduction of your sound. So no matter how many speakers are hooked up, they will all receive the entire mix at all times.

There is no doubt that a worship band will sound better in stereo. It is easier to hear all the instruments, and it is more acoustically pleasing to the ear if they are spread out over the stereo field. However, the spoken word sounds better in mono—and we all know the spoken word is king in the house-of-worship realm. So be sure to put the preacher, priest, rabbi, etc. at the top of the list sonically, and the choir, band, or solo musicians second. That said, it's possible to put the orator's speaking voice out to the congregation where it can be intelligibly heard in stereo.

METERS

As signal comes into a channel, very often you will have meters (most commonly a vertical row of LEDs) to give you a visual idea of the input signal strength. For example, in the master section, there will be a pair of meters, which visually monitor the output strength of the signal (see Figure 3.12). From bottom to top, these meter lights will usually start with green lights to indicate a signal is present followed by yellow lights to display a usable signal strength and then red indicating too much signal.

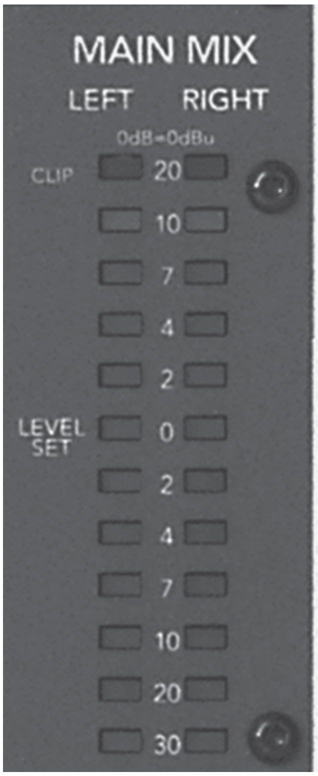


Figure 3.12
Meters are your visual indicator as to how strong an incoming or outgoing signal (as is the case with Main Mix meter) is so you can monitor and control any possible distortion.

To get a good idea of how your input/output meters operate, begin by running a tone from one of the following:

- A tone CD
- A tone generator
- A keyboard into an input channel

To do this, start by doing the following (refer to Figure 3.7):

1. Turn down the channel fader and the master fader that buses audio to your main speakers.

2. Plug your tone generator into any channel strip.

3. Turn the gain control in the mic/line input section all the way down.

4. Plug a set of headphones into the master section.

5. Flip on your PFL (or solo) button.

6. Slowly turn up the gain control and watch the meter in your channel strip.

As the lights move into the red, you should hear the tone signal become distorted. Turn the gain knob down until the lights leave the red and the distortion disappears. (That being said, some incoming signal sources may produce sudden increases in volume that will consequently overload the channel anyway. If you know in advance which ones may do just that, be sure to leave a little bit of what's called "headroom" by raising levels so they're not right up against the red.) You can do the same test at the output of the master section—only you use the AFL listening switch. Just crank up your output faders, master aux buses, or subgroup faders, and follow the same procedure for the PFL. Getting an idea of how your house-mixing console's meters work will help with setting up your mixes.

INSERTS

When an input signal enters your mixer through an input channel strip, you can route it elsewhere for processing via an insert. Inserts enable you to, say, send a lead-vocal signal to a device that will help greatly improve the overall sound of that vocal, which is then sent back to the channel and off to the ears of the congregation. An insert is a single 1/4-inch jack usually found near the line-level jack of your channel input, as seen on the backside of the Mackie Onyx 24.4 (refer to Figure 3.3). Its function is to divert an incoming signal to an external processor (external processors, also known as outboard gear, are discussed in great detail in Chapter 6) and then back to the channel using only one jack. To do so, a bi-directional tip-ring-sleeve (TRS) cable, also known as a type of Y cable (shown in Figure 3.13), is needed. The end with the single jack plugs into the insert jack of your board while the remaining two ends respectively plug into the input and output of your processor. These two ends should be labeled as send, which is wired to the tip of the single end, and return, which is wired to the ring of the single end. What you're doing is sending the incoming signal from your board to the input of the processor and then returning the signal back from the output of the processor to the board.

Figure 3.13
To utilize your channel inserts, you'll need a bi-directional Y cable, which allows you to send and receive an audio signal through a single 1/4-inch jack.

Basic Setups

Now that you've spent some time exploring how and where a signal travels through a mixing console and what the essential components are, let's run through a few mixing scenarios. After all, this chapter is called "Mixers and the Art of Mixing"—so let's mix.

THE PREACHER

In the first scenario, let's set up one microphone for the individual delivering the word—i.e., the preacher. This microphone may be a stationary mic at a pulpit or podium, a wired microphone that the preacher is holding as he or she moves around the stage, or different types of wireless mics. Whichever mic you choose or is preferred by the preacher, the main goal is for your preacher to sound natural and intelligible.

Before you do anything, make sure you check the preacher's mic before the service—trust me, it's a very good idea to do this. After the mic checks out OK, do the following:

1. Make sure you have the EQ set flat. That is, zero out the EQ controls so they're not cutting or boosting any frequencies.

2. Have someone on stage—preferably the preacher—speak into the mic.

3. Bring up the gain until your meters are flashing yellow or your signal light is on. Unless your preacher specifically asks to hear his or her voice in the monitors, cut any aux sends from the preacher's channel input.

4. Listen to the voice being picked up by the mic and ask yourself some simple questions. Does it sound natural? Speaking voice fundamentals lie in the 160–250Hz range, but overtones can sail up to 8kHz. Does

the voice sound boomy? Try cutting the low frequencies (use the high-pass filter). If the voice is harsh, try cutting the mids a bit or, if you have sweepable mids, reduce 2kHz to 4kHz. You may need to boost the mids a little to improve clarity—or, again with sweepable mids, try boosting to 600 or 800Hz.

The suggestion for multiple EQ settings comes down to one simple fact: All voices are different. You will have to use your new-found skills and, ultimately, your ears to get the most clarity and intelligibility out of your preacher. When experimenting, keep the following in mind:

- 1.25–4kHz are important frequencies for intelligibility, so boost or cut here sparingly.
- 5–8kHz can add clarity to the preacher's message.

THE CHOIR

For this next scenario, let's assume you have a 12-person choir positioned in two rows with the back row on a riser. For this group of singers, you will use two to four condenser mics (or dynamic mics if you don't have condensers) set up similar to what's shown in Figure 2.9 in Chapter 2. Do the following:

1. Set your EQ flat.
2. Bring up the gain until you see (or hear) a good strong signal. At this point, really use your ears; you want your choir to blend and sound like one body of voices. Listen to how all the voices come across.
3. Before attempting any EQ fixes, try repositioning the mics and stands.
4. When you feel you are picking up all the voices as equally as possible, address the equalization. Beware: Choirs can be a bit tricky. First make certain there is no boom or bass flab present. Start by engaging your high-pass filter and listening to those lower frequencies (100–160Hz).
5. With the low end of the choir tightened up, you can move to the 315–500Hz range. This area is important to the overall voice quality. Boost or cut in this frequency area to achieve a good blend.
6. On up the frequency scale, you arrive at the 630Hz–1kHz range. Here you can define the naturalness of your choir voices. Be cautious about boosting the 1kHz, as this can cause a bit of a nasal quality in the voices and is a frequency that feedback seems to love. As an alternative, maybe boost 630Hz a bit to put the choir in front of the band.

It's important to note that the instruments that back up your choir can make a world of difference in your mix. A choir supported by just an organ, piano, or synth keyboard will be easier to mix than one that is backed by drums, electric bass, guitar, and keyboards. When working with a full backup band for your choir, try slightly boosting the overall choir mix between 2kHz and 4kHz and cutting the instruments at the same frequency. This gives space for the choir harmonics to sail over the band. Just as with the preacher, a slight boost at 5kHz to 8kHz can add clarity to the choir if needed; conversely a cut at that range can remedy a brittle-sounding output. Finally, 10kHz to 16kHz can add air to the choir, but generally try not to boost or cut at these frequencies.

THE WORSHIP BAND

The greatest complexity in your mixing duties lies in your worship band. The main goal with all worship bands is to first control their decibel levels. The most common—not to mention biggest—complaints in houses of worship are that the bands are too loud. As mentioned earlier, most houses have decibel limits with regard to worship music. So before you start mixing your worship band, talk to the members of the band about the needs of the congregation. If you work and worship at the same house, take the opportunity to develop relationships with the band members, as this will make communication easier. Musicians tend to be a bit, shall we say, sensitive, and may resist direction from a relative stranger. Along with developing skills as a sound tech, try to sharpen your church diplomacy skills too. Even though you are not a preacher or musician, in many ways you are the director of the service.

Drums

Your drums will generally be the loudest instrument and can be the cause of the dreaded boom and rumble that will muddy your mix. Should your kick drum have too much low end, push that high-pass button. If the kick is thin, boost a little in those 80–125Hz frequencies. Try engaging the high-pass (80Hz) and then boost 80Hz. It would seem that these two actions would cancel each other out, but the effect is often a tightening of the kick-drum sound. The 160–250Hz range can also be a home for boominess. Sometimes your house of worship stage can increase the boom in this frequency area. Is your stage wood? Is it hollow? Is it carpeted? Look into what you are working with. A slight boost at this range can inadvertently increase your bass guitar fullness. Boosting in the 315–500Hz frequency bands will give you added punch in your bass and drums. But as always, a little goes a long way when boosting frequencies. As you move into the 600Hz–1kHz range, you will affect everything from keys to guitars to bass and drum harmonics. You can get a nice snap out of your snare drum and a slap out of your bass in this range. Climbing to the 5–8kHz frequencies, you can accentuate the clarity or the cymbals and hi hat by boosting or create distance and transparency by cutting.

Boosting 10kHz to 16kHz can add air and brightness to the overall mix, but there may also be sound-system hiss and noise up here waiting for you, so you might actually need to cut.

Guitar, Keys, and Horns

From 1–4kHz is the frequency range to start working on guitar, keys, and/or horns. Above all, it's a good idea to look for specific frequencies you may need to cut out of these instruments as they are prone to cause listening fatigue or hurt the ears of your congregation. For instance, cutting 2kHz can smooth out a harsh electric guitar. Conversely, boosting at certain ranges, such as 4kHz, to your keys can add presence. So listen closely to the relationships between the instruments here.

Hopefully, at this point, you will find mixers and mixing less intimidating. Mastering the nuances of mixing itself can be time consuming, but it all starts with understanding the basics of your house mixing board.

CHAPTER 4

Speakers

At the beginning of the signal flow is the microphone, which, as you know, is a transducer that changes acoustic energy (sound) into electrical energy. Multiple instances of these transduced signals travel into a mixing console, where you have total control over them, including the ability to mix the sounds together. As stated in previous chapters, this mixed (usually stereo) signal is finally sent out to the congregation to hear by way of speakers (see Figure 4.1).

Figure 4.1
It is the speakers that finally deliver the spoken word, music, and whatever else is happening in your house of worship's service to the congregation.

Chapter 2, "Microphones," mentions that speakers, like mics, are transducers, but that they work in the opposite manner—converting electrical energy into acoustic energy that can be heard by our ears. This may lead you to believe that speakers are an arbitrary piece of gear. In fact, speakers are every bit as important—one of the most important things, actually—to your sound system as mics and other equipment. Speakers must be able to accurately reproduce your preacher's message, your choir's singing, and your band's playing, or the congregation will not be able to fully realize their weekly worship experience. Not only is it your job to prevent that from happening, it's also your job to enhance the congregation's experience by providing the best delivery of worship audio you and your sound system are capable of. In order to answer this supreme calling, you must know every component of your system—and that absolutely includes the speakers.

Like microphones and mixing consoles, there are many variables that determine how a speaker will perform. At the same time, with all the components of your worship house system, you are obliged to use whatever is available. This chapter's purpose is to provide you with enough information to develop a basic understanding of speakers and speaker systems so that you can take control of whatever system you encounter and get right to work. While there are laws of physics that preside over how speakers operate, don't worry—this chapter will not bog you down with complex formulae. It will instead provide you with the essential information to correctly assess your house speakers so you can, for example, properly match any amplifiers to any speaker system so you don't blow them up! And if something were to happen—and it does—you will gain some much-needed insight should you need to go shopping for a new speaker system.

WHAT'S IN A NAME?

Speaker is a generic term, which may cause some confusion as you read this chapter. To make sure we're all on the same page, the primary internal component—universally recognized as the speaker—will be referred to as a driver and/or transducer (shown in Figure 4.2), the related components will be called their respective names (e.g., crossover, binding posts, etc.), while the boxes that house the speakers will be called cabinets and/or enclosures (shown in Figure 4.3). When referring to the entire package (see Figure 4.4), the term loudspeaker will be used.

Figure 4.2
A loudspeaker's driver, which will be discussed in detail as the chapter progresses, is the mysterious circular black silhouette seen behind the enclosure grille.

Figure 4.3
The actual box that houses the driver(s) is called the enclosure.

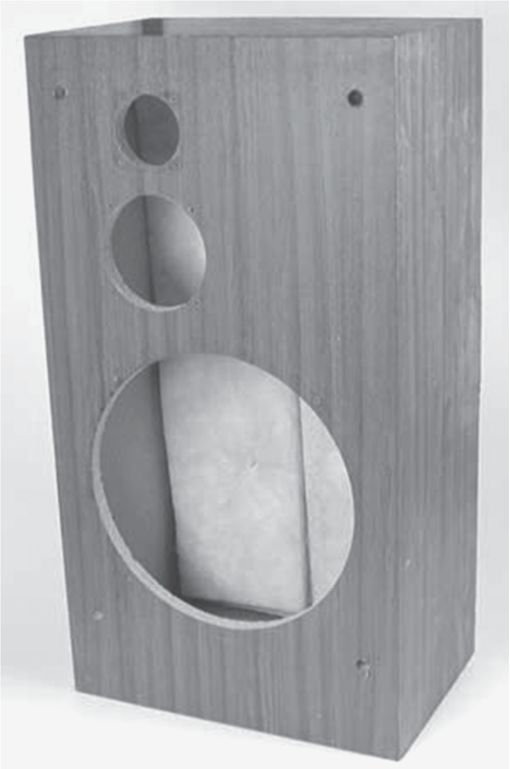


Figure 4.4
A loudspeaker is made up of an enclosure loaded with drivers, as well as various other components depending on the type of loudspeaker. (Image courtesy of Electro-Voice.)

Loudspeaker Components

It's no secret loudspeakers have a tough job. After an incoming signal is transduced into electrical energy and sent from the mixer, it's the driver's job to transduce that signal back into sound by moving a cone and/or other mechanical devices. To do so, a driver has to undergo a pretty difficult process. It's the enclosure's job to house the driver, but don't let that fool you into thinking the enclosure is not as important and as carefully designed as the driver(s). Enclosures not only secure the drivers so there's no rattling, but more importantly they prevent any sound waves from the rear of the driver from canceling out the waves emanating from the front. To get the best performance from, to maintain, and possibly to add to or replace your house's loudspeaker system, you'll need to know some basic information about these two primary components.

THE DRIVER

The driver, usually unseen by the congregation aside from a dark, round circle lurking behind the enclosure grille, is the spherical apparatus within the enclosure that does the lion's share of the work. Figure 4.5 shows the various components that make up a driver and how they all come together to produce sound. Notice how the anatomy is not unlike our friend, the dynamic microphone, in reverse.

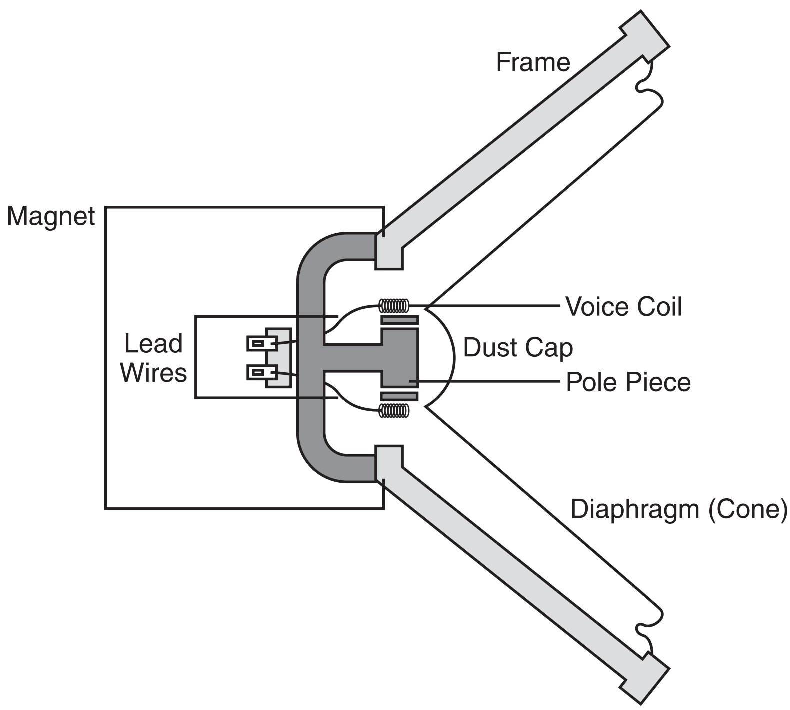


Figure 4.5
Taking an inside look (literally) at a loudspeaker's driver reveals that it's simply a dynamic microphone in reverse.

Here's how it works: A lightweight, cylindrical voice coil composed of fine copper wire is mounted so that it can move freely within a magnetic field. The magnetic field is provided by the massive permanent magnet at the base of the driver (this is what makes the driver heavy), which has a pole piece protruding from the center that the voice coil surrounds. The speaker cone or diaphragm (usually made of paper, but could also be metal, plastic, or a composite material like fiberglass) is attached to the voice coil and also to the outer frame (sometimes called the basket) and has a dark center circle called the dust cap. To get this whole thing working, an electrical current is sent to the voice coil via the lead wires, making it electromagnetic. Once this happens, the voice coil and the magnet interact with each other through mechanical forces, causing the coil to move back and forth, thus moving the connected diaphragm. This, ladies and gentlemen, produces sound, or rather reproduces the sound coming through the connected speaker cables.

Driver Size

Let's focus on the size of the driver, as this is the primary factor that determines what range of frequencies can to be heard. The basic rule of thumb is this: Large drivers handle low frequencies, medium-sized drivers handle mid-range frequencies, and small drivers handle high frequencies. This should make sense considering what you learned in Chapter 1, "Sound and Sound Advice." For instance, knowing that low-sounding pitches are large waveforms, it should come as no surprise that there's a need for large drivers to *drive* them.

There are generally four types of drivers, and they are classified by their size. The sizes mentioned here refer to the diameter of the diaphragm that is connected to the basket. They are from large to small:

- **Subwoofer.** Usually coming in at 18 inches, subs as they're often called are used to reproduce frequencies lower than 100Hz. These are covered in more detail later in this chapter.

- **Woofer.** Three inches smaller at 15 inches in diameter, woofers are also considered to be large speakers and are often used to handle the job of a subwoofer when one is not included in the system.

- **Mid-range.** These speakers can come in 8- 10- and 12-inch sizes and are used for handling the middle-range frequencies, hence the name.

- **Tweeter.** These little transducers come in at sizes of 1–4 inches in diameter and are for reproducing high frequencies. In larger-scale sound systems such as ones found in many houses of worship, these drivers are connected to a horn, which is sometimes referred to as a compression horn.

Of course, these are general descriptions; there are myriad loudspeaker system configurations that can have one or more of these speakers serving the role of another. For example, in many systems you can find 15-inch drivers taking on the role the sub, woofer, and mid-range speakers all at once, with only the addition of a tweeter horn for highs. Regardless, the information here will surely provide a solid foundation for your understanding of the basic properties of dynamic loudspeakers and the frequency output potential of their drivers.

Impedance

Voltage (pressure) is what drives the transducers in a loudspeaker enclosure. Without the required voltage, the system will perform poorly, cause a short, or experience a complete breakdown. As the electrical signal travels to the speaker, it will inevitably encounter some resistance, and therefore diminish the signal's voltage. This resistance is called impedance and is measured in ohms, named after Georg Ohms, creator of the equation used to calculate it. Ohms are represented by the Omega symbol from the Greek alphabet Ω and determine that power in watts (explained in the next section) is equal to the voltage squared divided by the resistance of the load. Whatever size your worship house's drivers (and enclosures) are, you *must* pay close attention to the impedance ratings when connecting them to whatever will be feeding them current. To give a simple explanation of what impedance actually is without some degree of complexity presenting itself is nearly impossible—so read this section carefully.

Simply put, impedance is the measurement of a circuit's opposition or resistance to an electrical current. That means the greater the ohms, the more resistance and the fewer volts available. All loudspeakers (in this case called the load) have input impedance, while all mixing consoles and/or power amplifiers (in this case called the source) have output impedance. If these impedance ratings are not properly matched, it can wreak havoc on your system so much as to permanently damage the components.

Say that your power amp achieves its highest output and efficiency levels at 4 ohms and your speaker enclosures (or combination of enclosures) also have a rating of 4 ohms. This matching of ohms between the power amp and speaker enclosures ensures that both pieces will work at maximum efficiency. Should your amp deliver top performance at 4 ohms but your speakers have a resistance of another value—8 ohms, for instance—your amp/speaker system will work, but not at its best. In the case of mismatched impedances, it works (partially) only when the load exceeds the source, *not* the other way around.

If you have an opportunity to look at the back of your house speaker cabinets, you will see at least two jacks. One jack receives the signal coming in from your power amp, and the other jack sends the same signal to an additional enclosure, allowing you to increase the power amp's wattage output. When utilizing this feature, you will again need to pay close attention to impedance. For instance, plugging from one 8 ohm cabinet to another 8 ohm cabinet connects them in what is called parallel and as a result will halve the two 8 ohm enclosures to a 4 ohm amp load. As stated previously, even though your amplifier realizes its full power at 4 ohms, that doesn't mean it only operates at 4 ohms; you just can't operate the amp at *less than* a 4 ohm load. The amp can actually run with an 8 ohm load, but only realizing about half its power or with a 16 ohm load realizing about a quarter its power. As for lower loads, 2 ohms was previously thought of as dangerous to a system, but is now becoming a more common rating. This is in contrast to 0 ohms, which will cause a short circuit and damage to your system. (Ohms measure resistance to the electrical current, and zero resistance is the very definition of a short.)

Putting this idea into action, let's go through another impedance-matching scenario, this time with watts considered. The enclosures in your house of worship are most likely rated at 8 ohms (they may also be 4 ohms or 16 ohms—the latter being a rare occurrence). Suppose you have a 2,000-watt stereo amplifier in line after the mixing console and it realizes 1,000 watts per channel (side) at 4 ohms (this is common, as amplifier manufacturers tend to design their amps to realize their maximum operating power at 4 ohms, which seems to be just the right amount of ohms to offer enough resistance to adequately protect the amplifier). With this amp, you can effectively power a total of four 8-ohm enclosures (two per side) by setting them up in a parallel configuration, therefore halving their load to 4 ohms per side.

Watts

As seen in one of the previous scenarios, when dealing with impedance, you will also need to focus on a related electrical concept called watts. A watt is a unit of measurement for electrical power dissipation that is derived from multiplying voltage by amperes. Power amps are rated in the number of watts they produce, and speaker enclosures are rated in the number of watts they can handle.

As with impedance, when connecting loudspeakers to their receiving source, you must pay close attention to the wattage. For instance, if you plug a power amplifier with a high watt output into a speaker enclosure with a considerably lower wattage-handling capacity, you can severely damage the components in the enclosure.

Because you're probably working in a house of worship that has a system already installed, knowing the wattage of your power amps and the watts that your enclosures can handle will be helpful in determining whether your system is set up for optimum performance. Of course, if you are about to install a new system or upgrade an existing one in your church, understanding ohms and watts is essential. If either occasion were to arise, consider this: When it comes to power, more is better. You want to choose a power amp or set of amps that delivers more power than you need—not a lot more, which can lead to damage, but enough to provide a strong signal to properly drive the loudspeaker system. A good number to keep in mind is 10 to 20 percent more than your loudspeakers are rated for. Let's say your two 8-ohm enclosures can handle 400 watts each; matching those cabinets with a stereo power amp that develops 500 watts a side at 8 ohms or 1,000 watts at 4 ohms would be good. Pushing 100 watts into a cabinet that can handle 400 watts, however, can cause distortion and that can damage the speakers and the amplifier. The distortion comes from attempting to get too many decibels out of the cabinets without enough power to do so.

THE ENCLOSURE

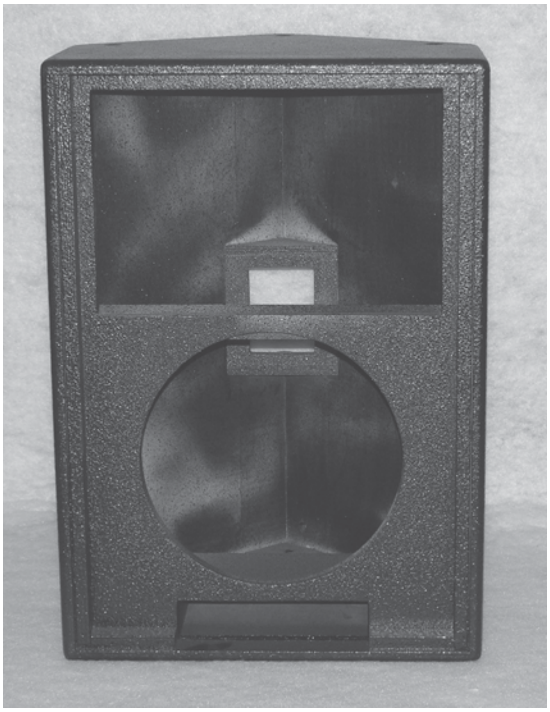
Loudspeaker enclosures serve a dual purpose: providing a physical frame in which to mount drivers and electronics and acting as a container to isolate sound waves emanating from the rear of the driver so they can't interfere with the (intended) ones being projected from the front. The most desirable characteristic of a good enclosure is that it doesn't alter the sound of the speakers installed in it. Overall, speaker enclosures are designed to reproduce sound, not produce sound. Key factors that determine the performance of the enclosure are the shape and the material used to construct it; let's look at both.

Shapes

The shape of an enclosure plays a key role in how sound will be dispersed from it. Rectangular shaped cabinets (refer to Figure 4.3) are what most people think loudspeakers look like. While this design is relatively easy to build and has been the norm for years, rectangular boxes can have resonance problems because they are designed with parallel walls. This means they can produce unwanted sound anomalies that can be heard by your congregation. If you were to open this type of cabinet, you would most likely find foam or some sort of sound-dampening material lining the interior to counteract this problem. This is not to say rectangular cabinets are a failed design; in fact, narrow, deep rectangular boxes are often used because of their favorable bass response.

Enclosures with non-parallel walls, such as triangular or trapezoidal boxes (see Figure 4.6), tend to be more effective. For instance, they have fewer problems with internal reflections of sound waves than rectangular boxes.

Figure 4.6
Enclosures that feature non-parallel wall designs, as seen in this empty ISP HDM 112 cabinet, are not susceptible to internal reflections like the more commonly seen rectangular design. (Image courtesy of ISP Technologies.)



Construction Materials

Not only can different shapes affect the sound of your enclosures, the materials they are built out of can be very important when it comes to reproducing sound waves. Modern speaker enclosures are made of either wood or plastic. The more rigid and dense a cabinet is, the less it will vibrate. And the less enclosure vibration, the more pure the components will sound. Along with rigidity and density, weight must be considered during the design and manufacturing of speaker cabinets. Following is a list of materials and their properties, providing you with some insight as to why certain materials are used and why some cabinets may cost more than others.

- **Particle board and MDF.** Used in less-expensive cabinets, these materials have a dense, hard quality that is desirable. That said, they are actually heavier and weaker than other woods and wood composites. All together, these characteristics make for a less desirable material.

- **Plywood.** This is hands down the strongest, lightest, and most popular material for enclosure construction. Marine plywood is light and strong and has great resistance to moisture as a result of the glues used to make the plywood. Birch- or Baltic-birch–based plywood is dense and strong, making it the material used in most professional speaker enclosures. This type of plywood has all the qualities cabinet builders look for: It's strong, high density, rigid, relatively lightweight, and reasonably priced.

- **Injection-molded plastic.** This has become popular for smaller enclosures because the plastics used by manufacturers today can be produced with very accurate specifications. Because it's a man-made material, it's possible for each cabinet to have identical weight and density measurements. That said, there are some limitations with regard to the size specifications to which these enclosures can be built.

Speaker cabinets often incorporate internal bracing to improve rigidity along with strong glues, screws, and nails to hold them together. Enclosures are stained, painted, covered in carpet, or finished with a variety of rugged spray-on texture coats; the finish used can also add to the density and rigidity of a cabinet.

Types

The following list gives some examples of different speaker enclosure types. These can be designed based on any of the aforementioned speaker shapes.

- **Flat-panel closed box.** Also known as acoustic suspension, this is the simplest enclosure design. It features holes cut out to the size of the cone drivers and horns on a single sheet of wood (usually 5/8 to 3/4 inch thick). The components are mounted on this sheet of wood and installed in the cabinet. If this is the case in your house of worship, make sure you inspect the enclosure to make sure it's properly sealed to prevent any problems, including internal wave cancellations.

- **Reflex.** Also referred to as bass reflex or vented cabinets, these enclosures incorporate vents or ports to allow low-frequency energy from the rear of the speaker transducer to exit the cabinet along with the sound waves from the front of the transducer. This type of enclosure is known for a punchier low end than its counterparts.

- **Passive radiator.** This enclosure creates additional low frequencies through a drone speaker. The drone or passive speaker is not wired to an amplifier but instead moves in conjunction with changing pressure in the speaker enclosure. (Don't worry if that sounds a bit complex; these enclosures are rare in houses of worship.)

Configuration

When talking about loudspeakers, the term configuration refers to the number of drivers the enclosure features and how they're arranged. Every major manufacturer produces multiple enclosure configurations to service myriad applications. With regard to your house of worship, this is a good thing—the more configuration choices, the more options to fill your needs. To keep things simple, let's explore the two most popular designs: 2-way and 3-way.

2-WAY

As you sit (or stand) in your house of worship listening to a sermon, chances are your ears are receiving sound reproduction through a pair of 2-way speaker enclosures (see Figure 4.7). Typically less expensive than the 3-way configuration, the 2-way setup is by far the most widely used. The term 2-way refers to a loudspeaker enclosure that employs two components to reproduce frequencies: a low-frequency driver and a high-frequency horn. If you have 2-way enclosures in your house of worship, they probably consist of one cone transducer and one horn driver. The idea behind this type of cabinet is that the low-frequency cone transducer reproduces the low and mid frequencies and the high-frequency horn driver handles the high-mid and high frequencies. Typically, these cabinets will have one 12- or 15-inch transducer or a pair of 10-inch drivers (this configuration works great for vocals). As for the horn, there are a variety of designs, but most horns use a high-compression driver to amplify their frequencies. Overall they control the dispersion patterns of high-mid and high frequencies.

Figure 4.7
Loudspeakers with two types of drivers in the same enclosure, such as this Electro-Voice Eliminator, are known as 2-way loudspeakers. (Image courtesy of Electro-Voice.)

To distribute the incoming frequencies to the proper drivers, an internal component called a passive crossover is used. This type of crossover is referred to as passive because it does its job automatically, without the need of an external power source. While that's convenient for the end user (that's you), passive crossovers allow no control over their operation. The crossover frequencies have been preset by the manufacturer and generally cannot be adjusted. Other types of crossovers, which are discussed later in this chapter, offer more flexibility.

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2 MEANS TWO, RIGHT?

One would think the term 2-way denotes two drivers in the enclosure, right? Nope. The terminology does not refer to the total number of drivers in the enclosure, just the number of types utilized. Therefore, there can be one or more low-frequency drivers and/or one or more high-frequency horn drivers in a 2-way–configured enclosure.

3-WAY

Where 2-way enclosures incorporate a low-frequency and high-frequency driver, 3-way enclosures (see Figure 4.8) are loaded with an additional mid-frequency driver. Similar to the concept stated in the previous note, the term 3-way denotes how many types of transducers are utilized to reproduce incoming frequencies. Typically, a 3-way enclosure will have a 12- or 15-inch low-frequency woofer, a 6- or 8-inch mid-range cone driver (speaker), and a high-frequency horn (see the upcoming note). Alternatively, a mid-range driver attached to a horn can reproduce mid-range frequencies. The premise behind a 3-way system is that by separating the frequency spectrums across three driver types instead of two, the listener experiences a higher level of fidelity. But this seemingly superior reproduction doesn't come without a price, as the internal crossovers incorporated in a 3-way system are more complicated than a 2-way crossover, and 3-way enclosures are typically larger and—you guessed it—heavier, the latter being an important consideration if you are contemplating flying (hanging) your enclosures from the ceiling or walls.

Figure 4.8
Stepping it up a notch are loudspeakers with three types of drivers in the same enclosure, such as this ISP Technologies Tripower 900 enclosure. This is known a 3-way enclosure. (Image courtesy of ISP Technologies.)

THE HORN

If you've ever looked at a loudspeaker and noticed a deep, cavernous port that reminds you of a sci-fi black hole and wondered what it is—it's a horn! Horns are employed because of their unique ability to control the direction of sound waves, and are a common feature in many enclosure designs. The development, construction, and tuning of high- and mid-frequency horns has become a major endeavor for many speaker-enclosure manufacturers' research and development departments. The first horns (and some still today) were made of metal, but it was later revealed that metals have resonance characteristics that interfere with natural-sounding frequency reproduction. As a result, plastics and fiberglass were then used to create horns. No matter how perfect the horn's design, it still needs a well-designed and constructed driver to sound good.

PHASE

As you listen to the audio signal (sound waves) from your house of worship's loudspeaker system, there's a specific time relationship between the loudspeakers and your ears that you need to be aware of. That relationship is referred to as the phase relationship of the signals and is represented in degrees, with one sound-wave cycle equaling 360 degrees. In Figure 4.9, you see two sound waves arriving at their destination (the listener's ears) at the same time, making the waves in-phase with each other. Acoustically, phase can be an issue when sound from multiple loudspeakers reaches the same seats from varying distances. For instance, if two sound waves reach your ears 180 degrees apart, they are perfectly out of phase and could cause cancellations within the sound at varying degrees. Phase-related problems can also occur when two microphones pick up the same audio signal from different distances and the audio signals are combined in the mixing console.

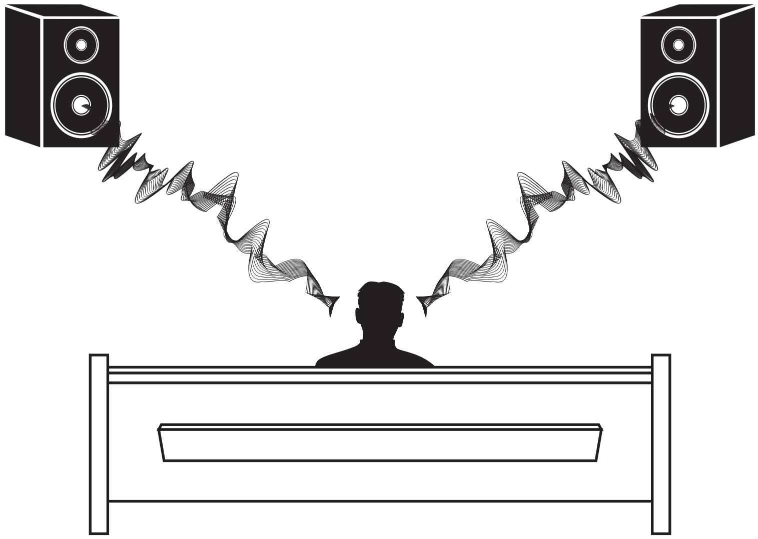


Figure 4.9
When sound waves from multiple loudspeakers reach our ears at the same time, they are said to be in phase with each other. This enables optimal sound reproduction and allows whatever is emanating from the enclosure to be heard at its highest fidelity.

With this in mind, you can see that phase is vital to the effectiveness of your house sound system. When walking around your house of worship during a service, you will find areas where everything sounds great. These are known as the sweet spots in your house—where sound waves reach your ears in phase. Don't be alarmed if all the sound in your house doesn't come across with at the same quality as your sweet spots—if they did, they wouldn't be so sweet! The human ear does an exceptional job at compensating for phase inconsistencies, so unless there is a total cancellation of sound, your congregation should be fine. Your worship space may not be able to be 100 percent in phase, but with some adjustments you can improve almost any space. Here are some remedies if you need to make some adjustments:

- Turning the speaker cabinet toward the congregation can eliminate phase inconsistencies.
- Multiple enclosures set in various places around the sanctuary can remove cancellation spots.
- If you have a subwoofer (covered later in this chapter) in your system, experimenting with its placement can help remedy low-end peaks and valleys. How the cabinet is placed in relation to walls, the floor, and the ceiling can affect these.

PHASE VERSUS POLARITY

Your mixing console may have a phase button at the top of each channel strip. That button, however, actually reverses polarity, *not* phase. This is just one of the many instances in which the concepts of polarity and phase are confused. Polarity has to do with electricity, whereas phase involves time. Older microphones can be wired opposite to that of modern mics. The phase button reverses this wiring difference so that the polarity problem is corrected; it has nothing to do with phasing. It's possible that your worship house speakers could be inadvertently wired in reverse polarity. If so, you will have poor low-frequency response, bad imaging, and diminished sound quality. This problem is unfortunately often described as being out of phase, but these are two very different phenomena.

Frequency Distribution

To get the most out of 2-way and 3-way loudspeaker enclosures, there are several options available to cater to specific ranges of frequency bands. This allows you more control over your house system and provides superior sound distribution, making the congregation's worship experience that much more fulfilling.

CROSSOVERS

Previously there was mention of a passive crossover, which is an internal component that divides and distributes incoming frequencies to the appropriate driver types in a 2-way or 3-way loudspeaker. With regard to a 2-way speaker system, the crossover is used to separate the audio spectrum into two bands—highs and lows. In a 3-way system, your crossover will split the audio into three bands—highs, mids, and lows.

Let's take a look at another type of crossover: the external active or electronic crossover (see Figure 4.10). This type of crossover is inserted in the signal chain between the mixing console (right after any external EQ) and the power amplifiers. Like all components that are referred to as active, this type of crossover needs to be provided with power. In order to use an external crossover, the enclosure has to be designed with that intention.

Figure 4.10
Outboard crossovers like this dbx 234XL offer control that passive crossovers do not, making them invaluable to your system's collection of processors. (Image courtesy of dbx Professional Products.)

Most of you reading this book already have a system installed in your house of worship. To see if you can use an external crossover, check out the back of your speaker enclosures. Do they have separate input jacks for low, low-mid, mid, and/or high frequencies? If so, your enclosure is external-crossover ready. (Keep in mind that some speaker enclosures are not designed for external crossovers, so you may see a more simplified connection system for plugging one or two cables.) External crossovers have adjustable crossover points as well as level (volume) controls for the different bands of frequencies. In a 2-way crossover, you can adjust the crossover point to 1kHz, making all frequencies below 1,000Hz travel to the low-frequency amplifier and all frequencies above 1kHz powered by the high frequency amp. With level controls for both the low and high frequencies, you can turn up the lows and highs independently. So although external crossovers add another layer of expense and complexity to your system, in return you get increased efficiency, flexibility, and control over the audio signal.

SPEAKER-MANAGEMENT PROCESSOR

External crossovers have been available for many years and are widely used, but to obtain maximum control over your speaker system, a speaker-management device would be the next step up. A speaker-management processor (see Figure 4.11) is a digital crossover that packs in several features that make your sound system operate more smoothly. Similar to the aforementioned external crossovers, this device plugs into the left and right outputs of your mixer and allows you to control input and output levels and equalize all your speakers after the crossover points. Additionally, they can limit the output signal to protect your speakers as well as providing a time-alignment feature, which basically controls your drivers so they're all firing at exactly the same time. This is a great feature because it eliminates most internal phasing issues you may run into. Also, these processors sport specific controls for sub-woofers (see the next section).

Figure 4.11
Speaker-management processors, such as the Yamaha SP2060, are in essence a more advanced external crossover. (Image courtesy of Yamaha Technology.)

SUBS

Sub-woofers (usually referred to simply as subs) are speaker enclosures dedicated to reproducing only low frequencies—usually100Hz and below. If you've ever been to a large-scale production—e.g., a concert—and literally felt the thump from a drummer's kick drum, then you've experienced the power of a sub. Because larger speakers produce low frequencies more easily, it's no surprise these drivers can range from 15 to 18 inches in diameter.

But while subs are great for, say, delivering your house band's bass guitar, you probably don't want your congregation to feel like Sunday service is a rock show—unless of course you have a rock 'n' roll house of worship! The truth is, many houses don't have subs-woofers in their system and really don't need them. If you work with a system that does, sub-woofers must be used sparingly—and you'll need to understand what this type of speaker enclosure will do to the overall sound of your house system.

If you've never heard sub-woofers at work, check out some local houses of worship that have subs installed and listen to what that extended low end sounds like. If there are plans to purchase a new system or improve an existing one, you will have to decide whether subs will serve the congregation. Another factor to keep in mind when deciding: You will need at least one dedicated power amp to power those behemoths.

FOH

The speaker enclosures discussed throughout this chapter are known as front-of-house or FOH loudspeakers. This term aptly describes these enclosures because your congregation is listening to the loudspeakers that are right in front of them.

Non-Powered Systems

Non-powered loudspeaker enclosures are standard in house-of-worship sound systems. Just like the name implies, these enclosures receive their power from an outside source; they do not require for you to plug them in. The source from which they derive their power is an external power amplifier, which is the subject of the next section.

As stated many times in this book, it's important to understand the properties behind each component in a house sound system to best deliver the word, music, or Sunday announcements to the congregation, the power amp is one of your most important components. Think about it: A power amp's sole purpose is to send electrical current at the proper signal strength to the enclosure via speaker cabling (discussed later in this chapter). If that's not working or configured properly, your system will be at a standstill. So read carefully.

POWER AMPLIFIERS

By definition, a power amplifier (see Figure 4.12) increases or amplifies the amplitude (volume) of a signal (in the form of voltage or current) that passes through it. Looking at Figure 4.12, you can see that power amps are fairly simple, with the front panel usually sporting only two knobs (if it is a stereo amp): the gain knobs. The term gain refers to the relationship of the input to the output signal, which is called the transfer function; gain adjusts the magnitude of that property. If you have the owner's manual for your amplifier, it may suggest where to set these knobs for the best amplification results. If not, try cranking them wide open and control the volume from the master fader on your mixing console.

Figure 4.12
This deceivingly simple-looking piece of gear is the muscle behind your entire sound system. Power amps such as this QSC PL325 amp receive the main outs from your mixing console, considerably amplify the signal, and then feed that boosted signal to the house loudspeaker system. (Image courtesy of QSC Audio Products.)

Do not let the simplicity of the front panel throw you. There's more to a power amp than two knobs and few LEDs. Following are some other details about power amps you'll need to be aware of when dealing with them.

- **RMS and peak values.** As mentioned, all power amps are rated in watts, which tells you how much power the amp unit will deliver to a specific load (rated in ohms, as discussed in the section titled "Impedance"). Watts are rated in root mean squared (RMS) and peak levels, with RMS indicating the average sustained power output that your amplifier will produce and peak power indicating the amp's wattage limit for brief periods of time. Both are numbers you need to pay close attention to; understanding what the maximum power levels of your amplifiers are and knowing the limits of the power-handling capabilities of your speakers will help you operate your sound system properly, ensuring its longevity.

- **THD.** Another value you should pay attention to is the total harmonic distortion (THD) rating of your amp, which is a percentage. When a signal is passing through your power amp, a certain amount of content is added that slightly distorts the output as compared to when it entered the amp. This distortion slightly changes the incoming signal's tone, so you want this number to be low; 0.05 percent or less is ideal, although anything below 1 percent in considered inaudible.

- **Frequency response.** Like other sound components, power amps have a frequency response. This response measures the ability of the amp to reproduce the audio signal accurately. You will often hear the term clean when describing an amplifier; your power amp will ideally deliver a clean, uncolored, transparent signal to your speaker boxes. Generally, manufacturers list their amplifier specs in the owner's manual. These specs include THD input sensitivity, dynamic headroom, maximum distortion, and a laundry list of other specs. While it's good to know your particular amplifier's specifications, in the real world of sound, you will have to use your ears to listen for anything that doesn't sound right.

Where you place your power amps will have an impact on how they perform. Ideally, your house power amps should be positioned relatively close to your actual speaker cabinets. That's because speaker wire is unshielded, which makes the connection susceptible to noise. Therefore, the shorter the distance the wire has to travel from your amplifier to your speaker enclosure, the better. That signal can be delivered from a power amp a few different ways, but the most common is in stereo, covered next.

STEREO

A stereo amplifier is basically two separate mono amps in one package, with each amp (also called a channel in this context) referred to as a side. The two gain knobs shown in Figure 4.12 are for controlling each individual side—one for channel 1 or A and one for channel 2 or B. This basically means one amplifier is dedicated to powering the left speakers and the other to powering the right speakers.

The area between the two speakers is called the stereo field. Back at your mixing console, you can use your pan control to place the audio signal from the individual input channels into the stereo field (see Chapter 3, “Mixers and the Art of Mixing,” for a review of mixing-console concepts).

There are five traditional adjustments made in the stereo field, known as simply five-position. They are as follows:

- Center position or dead center
- Hard left
- Hard right
- Left of center
- Right of center

Hard-left and hard-right settings mean you have adjusted the pan knobs all the way clockwise (right) and/or counterclockwise (left), sending the audio signal completely to the left or right speaker. The left-of- and right-of-center settings are slight adjustments of the pan knob off the detent, while the dead-center setting involves no adjustment from the detent. Even though your worship house (typically) does not have speakers in the center, center left, or center right, through the phenomenon of the stereo field, you will hear the signal in these locations when those settings are used.

The advantage of stereo is that musical instruments and voices can be spread out across the stereo field and consequently can be heard more clearly—but with one caveat: The stereo field can be heard properly in only a rather limited area. Certain seats (somewhere toward the center) in your house of worship will be perfectly positioned for stereo listening. These seats are the sweet spots, where patrons can experience the full stereo spectrum. It’s important to remember, however, that when it comes to the clarity and intelligibility of the human voice, mono (single channel) is superior—especially when that mono signal is projected from a single speaker or array of enclosures that’s positioned in the center of your worship space.

MONO

As the name implies, a mono signal gives you one simple choice as to where your audio can emanate from. Each speaker in your sound system will receive and produce the same signal. Your house stereo mixing console may have a separate mono output with a corresponding mono level knob or fader. If you choose to utilize a mono output and use your house console in mono mode, you will need to adjust the frequencies with a single graphic EQ or equalizing device.

Setting up your worship sound system in mono has some distinct advantages, the most important one being that everyone in the congregation will hear the same audio signal regardless of where they are sitting. Every seat in the house will be an ideal listening spot for the sermon.

Learning to mix your worship band in mono makes you use your ears in a more discerning manner, as you can't spread individual sources such as vocals and instruments out as in a stereo system. Because mono forces you to stack all your entire mix into one output, it will train you to hear where there is a buildup of frequencies between the various sources. Once you hear these frequencies, you will have to EQ out the congestion.

BRIDGED

The word bridged refers to combining two sides of your stereo amplifier to function as one. Putting a power amp in a bridged state offers you the advantage of getting twice the power or more. To take this a step further, bridged power amps can be combined into a stereo system—one bridged amp for the left side and one bridged amp for the right.

Suppose, for example, that you have an amp producing 400 watts per side at 8 ohms. In bridged mode, that same amp can produce around 800 watts at 8 ohms. You will need to refer to your specific amps owner's manual to get your exact specifications, but this is a safe approximation. Often, your amplifier will safely operate only at higher ohm loads (8 ohms) in a bridged configuration. Lower ohm loads such as 4 ohms or 2 ohms can damage the amplifier, speakers, or both—again, consult your owner's manual.

You will most likely be required to utilize output connectors specifically meant for bridged-mode connections (see Figure 4.13). In some cases, the power amp may also have a bridge switch to change the internal electronics. Once the necessary adjustments have been made, the bridged amplifier will process the audio signal equally from both sides as if it were one mono amp. Note: Bridged speaker output connections are usually different from your amp in normal or stereo mode. As a result, you may need to use a different type of speaker cable (see the next section).

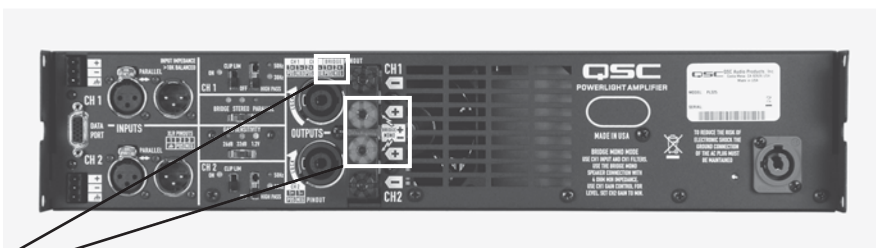


Figure 4.13
When utilizing bridged mode, you may need to use specific output connectors as seen on the rear of this QSC PL325. (Image courtesy of QSC Audio Products.)

Bridge connections

While the term mono and bridged are often use interchangeably, there is a difference. Mono means using the amplifier to facilitate one signal source to your speaker cabinets. Bridged means combining the left and right sides of the amplifier. A bridged signal will be mono, but a mono signal may not necessarily be bridged.

SPEAKER CABLE

Just as the XLR cable serves a simple yet crucial role in delivering the newly transduced energy from your mic to the mixing console, it is speaker cable that carries the audio signal (in the form of electricity) from your amplifiers to your speaker enclosures. Compared to the lower audio signal carried in a balanced XLR cable, speaker wire carriers a relatively high signal strength. Another major difference between the two is that the more-powerful audio signal carried across speaker wire is nearly impervious to the electromagnetic fields surrounding sound-system environments, whereas XLR cables are highly sensitive to these fields. Because of this high audio-signal strength, speaker cable will never be shielded.

Speaker-cable length plays an important role in enabling your house system to function properly. When possible, you want to keep the connecting speaker cables as short as possible between your power amps and speakers. There are formulae for determining speaker-cable length; Table 4.1 provides a starting point should you find yourself evaluating your worship house system's current setup or wiring additional loudspeakers.

Table 4.1 Determining Speaker-Cable Length

Load in Ohms	Speaker Cable	Length and Size
	1 to 100 Ft.	**Over 100 Ft.**
16 ohms	16 gauge	14 gauge
8 ohms	14 gauge	12 gauge
4 ohms	12 gauge	10 gauge

Speaker cable can add significantly to the overall impedance of the loudspeaker, and can cause your amplifiers to waste power while getting the signal to the speakers. The extra energy spent by your amplifier while trying to operate your speakers ends up creating extraneous heat. One way to remedy this problem is to use heavy-gauge wire, which carries audio signal with less resistance.

There are basically three types of speaker-wire connectors in use today. The banana plug speaker cable connector (see Figure 4.14) plugs into binding posts on a power amplifier when they are available. While 1/4-inch phone jacks (see Figure 4.14) are fairly common, these jacks and their corresponding plugs are not suitably rated to handle high power-amp–output levels. For this reason, Neutrik Speakon connectors (see Figure 4.15) have become the norm in pro audio sound systems, as this speaker wire connector has been specifically designed for this application. They offer a secure locking connection and can handle high-power amp current.

Figure 4.14
Two speaker cable connectors you're sure to work with are banana plugs (bottom) for plugging into binding posts and 1/4-inch jacks (top). (Image courtesy of the RapcoHorizon Company.)

Figure 4.15
Another common connector is the Neutrik Speakon. (Image courtesy of the RapcoHorizon Company.)

Self-Powered Speakers

Self-powered cabinets offer many advantages over their non-powered counterparts, making them very attractive (technically) and a flexible option for today's worship house. This is because powered speakers have the power amplifier, crossovers, and other signal-processing components mounted inside the speaker enclosures along with the drivers. Considering what you now know about loudspeakers systems, you can better appreciate the advantages of this, like having the internal power amps and components matched to the transducers for optimum performance. This saves you the trouble of having to think about ohms or speaker cable gauges or watts!

Speaking of speaker wires—they're not even used when connecting your house mixer and outboard gear to your powered speaker enclosures. All audio signals are sent via XLR cables. Overall, the self-contained nature of a powered enclosure makes for easier operation than non-powered systems. Let's take a look at two features that lie inside a self-powered enclosure: on-board crossovers and time-alignment processing.

ON-BOARD CROSSOVERS

As stated, self-powered speaker enclosures incorporate on-board electronic crossovers. But unlike outboard crossovers, their crossover points cannot be adjusted. The manufacturer of the enclosure will actually set up the crossover frequency points internally, making less work for you.

To work alongside the internal crossovers, separate power amplifiers are installed to drive each transducer. For instance, a 2-way powered speaker enclosure can carry one amplifier for the low and mid frequencies and another for the high frequencies. Because these amplifiers are separately fed by the on-board crossover, they operate in the most efficient manner possible. In the end, this means that right out of the box, you get a great-sounding speaker enclosure that doesn't require the tweak time that non-powered enclosures demand.

TIME-ALIGNMENT PROCESSING

Not only are power amps and crossovers stored internally in self-powered enclosures, but often there will be additional signal processing designed to maximize the performance of these enclosures. A very common and effective processor is the time-alignment circuit. You learned that all frequencies travel at the same speed under the same conditions—but the origin of audio signals from a speaker enclosure with two or more transducers is unique to each transducer. This means the sound exiting a high-frequency horn is leaving the enclosure at a slightly different time than the audio signal from the cone transducer of a woofer. What a time-alignment circuit does is cause the audio signal from all transducers to leave the speaker box at exactly the same time. This alignment of the audio signal allows your ears to hear all frequencies simultaneously, which is of course optimal.

CHAPTER 5

Monitors

In the last chapter, you were introduced to the loudspeaker—one of the most important components in your house of worship's sound system. The front-of-house (FOH) loudspeaker system is solely responsible for accurately reproducing and projecting whatever is happening in the service to the congregation.

While the FOH loudspeakers are indeed crucial, another set of loudspeakers called monitors that your congregation may never hear—or in some cases never see—are just as vital. Unlike the FOH speaker enclosures that your congregation is listening to, monitors are designed to be heard only by those who occupy the stage so they can hear themselves doing whatever it is they are doing—preaching, singing, playing an instrument, etc. This allows those individuals to, well, "monitor" their performance. This helps to bring the vocalists' and musicians' performance together, making the worship service that much better. With monitors, singers can accurately hear themselves and their fellow singers and thereby best maintain their intonation. If you have multiple musicians, monitors will enable them to clearly hear each other, allowing them to play more cohesively as a group. These factors alone make monitors an indispensable component of your house of worship's sound system.

Monitors, also known as wedges, are different in shape from FOH cabinets (see Figures 5.1), but they still come in 2- or 3-way transducer configurations. The purpose of the monitor's wedge shape is to make the speaker boxes less visible and at the same time more directional toward the listener. Placing your monitor cabinets in the optimum spot on your stage can take a bit of trial and error, but as difficult as that task may be, it is also one of the most important. Along with placement, proper EQing is vital to the effectiveness of monitors because it allows those on the stage to hear themselves in the most accurate manner. More importantly, effective equalization will eliminate the horrors of feedback that can easily come about in monitor systems.

Figure 5.1
Monitors such as this 3-way ISP HDM 112 are used so a worship service stage participant can hear a tailored mix of what's happening on stage, helping him or her to deliver an accurate performance.

Set-up

Setting up the monitors in your house of worship can be very different from setting up your FOH speaker system. For one thing, the loudspeaker enclosures in your house are most likely permanently mounted to the wall or hung from the ceiling. Also, your house of worship's sound system is tuned (EQed) to the room in which it operates. Once this setup and tuning is done, barring minor adjustments, it's basically a "set-it-and-forget-it" scenario. But because monitors are designed to serve the individual—or at most a small group—on your stage, they must be able to be moved at any given time depending on your needs. From week to week, or even service to service, the needs, desires, and even the individuals in the worship band or choir can and will change. For example, the early-morning service may have a different lineup than the later services or the group may have a special guest performer. These changes may include a change in how they want to hear their monitors. It could be as simple as a slight EQ adjustment, or it could be moving your monitors to a completely different location and/or making substantial tonality adjustments. Starting with physical-position changes, this section provides insight into how to set up your house of worship's monitors.

PLACEMENT

Monitor speakers should be set up in close proximity to the person or people they are serving (see Figure 5.2) so they can project to the person(s) who needs to hear them. This allows the monitor to operate at only the volume necessary, thus reducing feedback problems, interference with other sounds emanating from your house sound system, and other musicians' stage sound. It can also operate at a much lower volume than your FOH loudspeaker enclosures, which is an easy way to combat the seemingly inevitable feedback that can result from the use of monitors. Because house wedges are never permanently affixed to the floor, this gives you the option to move the monitors wherever they're needed. If you ask your performers their preference, you will likely find that many of your worship singers and musicians have an idea where they want their monitors positioned. They will also tell you how they want those monitor wedges to sound and how loud they should be. Depending on the complexity and versatility of your mixing console and monitor system, you may be able to accommodate everyone with an individual monitor mix (explained in the next section).

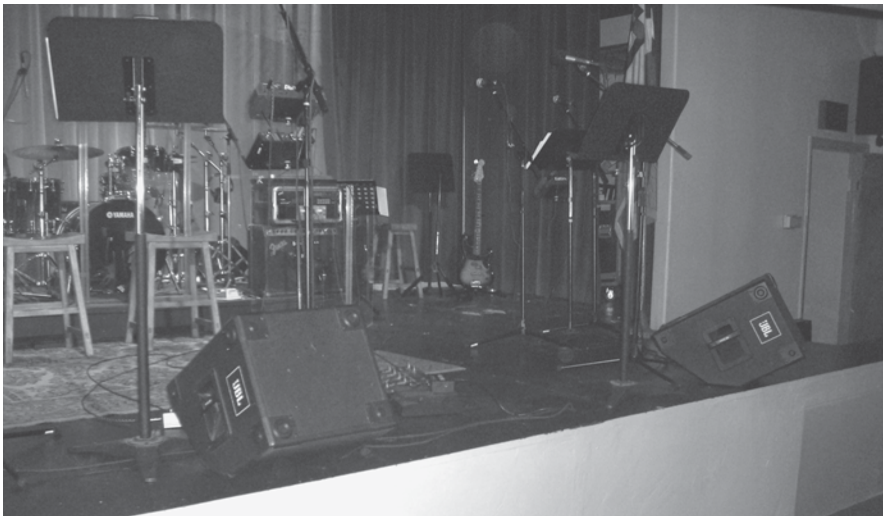


Figure 5.2
Because monitors need to be close to their listeners, it's a good idea to place the wedges in the center-most location such as between the mic stand and the music stand as shown here.

Whenever possible, it's a good idea to plan out where monitors need to be placed in advance. Taking a hint from secular live-sound reinforcement, try creating a stage-plot drawing, similar to the one shown in Figure 5.3. While these illustrations are more common in performances that are more concert-oriented, they will still prove useful in whatever type of monitoring scenarios you may encounter at your house of worship. For instance, a stage plot may save you time as you can use it when you begin setting up your monitors before the performers arrive. In Figure 5.3, the monitor configuration is what could be considered a standard layout and could be used as a template for your particular worship platform. (Although there are five monitor wedges shown, that's not to say you have to set up your plot the same way; this is strictly an example.) Take a look at your house's stage configuration and draw up a plot that addresses your typical needs. And be prepared to be flexible and able to implement changes on the fly.

Figure 5.3
Use a stage-plot illustration like the one shown here to set up a monitor scheme in advance; then all you need to do is tweak it when your performers arrive.

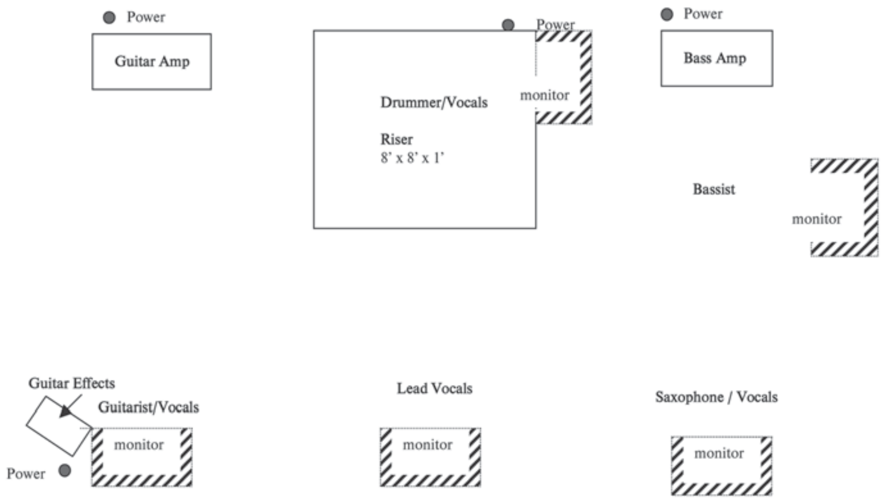
Artist Management/BMI Publisher
Planet Squared Inc.
Contact: Richard Soumakian -Manager
Tel: (714) 846-0650 Ext. 4
Fax: (714) 846-0096
Official Site: www.planetsquared.com
Myspace: www.myspace.com/qwiksand

qwiksand

Representation for Booking
Fast Lane Int'l (757)497-2669
info@fastlaneintl.com

Advance Contact
Chuy Vidales (714)234-5410
chuyvidales@yahoo.com

STAGE PLOT



SOUND / LIGHTING REQUIREMENTS:

1) MUSICIANS FOR QWIKSAND:
RAS (Guitarist & Vocals)
Trey Campbell (Lead Vocals)
Adrian Olmos (Saxophone & Vocals)
Dustin Ericksen (Bassist)
Chuy Vidales (Drummer)
2) MUSICAL DESCRIPTION:
Funk/Reggae/Rock
3) MAIN SPEAKERS:
We are looking for clear / crisp vocal reproduction over the normal vocal range and a general sound system that is used by a rock quintet with enough power for clean highs, mids with no-feedback, and tight low-end punch to cover the entire audience area.
4) STAGE MONITOR REQUIREMENTS:
One wedge stage floor monitor for each musician (5 total) with high and low speaker output using a single monitor mix.
5) LIGHTING:
Lighting should cover entire stage area.
6) POWER:
Please provide 110 AC power to stage. Power drops need to be behind Bass Amplifier, Drum Riser, Guitar

7) INPUT LIST

Ch.	Instrument	Mic.	FOH ins	MON ins
1)	Kick Drum	SM91A	gate	gate
2)	Snare Top	Beyer 201	gate	gate
3)	Snare Bottom	SM57		
4)	Hi Hat	SM81	compressor	
5)	Rack Tom 1	SM98A	gate	gate
6)	Rack Tom 2	SM98A	gate	gate
7)	Floor Tom	SM98A	gate	gate
8)	Ride	SM81		
9)	Overhead SR	414 KSM32		
10)	Overhead SL	414 KSM32		
11)	Bass D.I.or Mic	Audix D3		
12)	Keyboard D.I.			
13)	Guitar Mic	Shure SM57, or KSM32		
14)	SR Mic	Shure SM58		
15)	SC Mic	Shure SM58, or Wireless SM58		

WHAT ABOUT THE PREACHER?

One of the most important people in the service—the preacher—typically doesn't have monitors near him or her. That's because the preacher doesn't necessarily gain the same advantage from monitors as the band and singers. Because the preacher is on his or her own, there's no competition for the message. As a party of one, the preacher can usually clearly hear his or her own voice through the FOH loudspeaker system, therefore negating the need for any sort of monitor wedge.

MONITOR MIXES

In Chapter 3, "Mixers and the Art of Mixing," you learned that one of the functions of the auxiliary send controls is to route signal to monitors. In this capacity, each individual aux send acts like a channel strip fader, allowing you to create a monitor mix. The accumulated mix is then sent to the aux master bus, which then travels out of the mixing console to the monitors. The added bonus of employing the aux sends section for creating mixes is that each channel strip usually features multiple aux send controls, enabling you to configure more than one mix at a time. So the more aux sends you have at your disposal, the more monitor mixes you can potentially create. This helps immensely when you need to support various monitoring scenarios for your performers. Table 5.1 provides some common multi-monitor mix possibilities using a board with an aux sends section that features four auxiliary controls per strip.

Table 5.1 Common Multi-Monitor Mix Configurations

Aux Sends (Monitor Mixes)	Instruments and Vocals
Aux 1	Worship singers or choir, keyboards
Aux 2	Acoustic guitar, worship singers, keyboards
Aux 3	Keyboards, electric guitar, worship singers
Aux 4	Bass guitar, electric guitar, worship singers

In each of these four mixes, you have the ability to assign which instruments and/or vocals are to be routed to a particular monitor while easily controlling their volume levels by simply by turning up the corresponding auxiliary send knob. For example, in aux 1, you can dominate the mix with worship vocals and add in just enough keyboards to keep your singers on pitch with the rest of the group. In the aux 2 mix, the acoustic guitar player's instrument can be up front in the mix while the keyboards and singers are included for the player's reference. Of course, all these mixes can be changed to accommodate the myriad changes that may occur throughout the service. So even though you'll spend ample time setting up your monitor mixes before the service begins, be prepared to make adjustments.

EQUALIZATION

Considering the fact that monitor mixes are intended to be heard directly by the subject producing the sound—the most critical listener in human history—it should come as no surprise that you can potentially spend a lot time EQing those mixes. Face it: If you were the musician, singer, or preacher using a monitor, you'd want your voice or instrument to sound as natural as possible. Without good equalization, the whole purpose of monitors is defeated, as the subject will be distracted by the fact that his or her sound is not there and thus not perform up to par. And let's not forget that equalization is your best defense against feedback (see the next section). That said, be patient while configuring and tweaking each monitor mix—it could make or break the listener's performance and hearing!

As you adjust each individual channel strip's EQ, you will also be adjusting the EQ of the signal routed to your monitor mixes via the aux sends. This is because most aux sends are post EQ, which means the audio signal passes through the EQ section of a channel strip before it reaches the intended aux send. The potential problem is that the equalization that sounds good for the FOH speakers may not sound good for the monitors. To counteract this, you would ideally want a separate EQ on your console for each of your aux sends. Since that's not the case in many instances, the most common alternative to EQ those aux send signals is a graphic EQ (a 31-band unit is preferred), which can be inserted in the signal flow between the aux sends and the monitor cabinets. By using something as precise as a graphic EQ, you can provide the most natural and realistic sound in your monitor mixes. In a perfect world, each auxiliary send should have its own graphic EQ. If that's not within your house's budget but your system features a stereo graphic EQ, you can use one side for one monitor mix and the other side for another. Regardless of what type of EQ you use to tweak your monitor mixes—it could be both—when starting to make adjustments, be sure to reference Table 1.2 in Chapter 1, "Sound and Sound Advice," to see what frequencies affect what instruments.

Feedback

The annoying cycling of frequencies known as feedback will be an ever-present problem in your house sound system, especially within your monitor system. As with microphones, the feedback that can emanate from your house monitors may range from a sharp screech to a high-pitched squeal to an out-of-control howl. This makes sense because the feedback heard in the monitor system wouldn't exist if it weren't for the presence of a microphone.

Vocal microphones, whether for your singers or preacher, are the most likely causes of monitor feedback because the vocal mics will be pushed to the highest level of any mics on the platform. Feedback also commonly comes from acoustic instruments, such as an acoustic guitar, amplified by way of an internal pickup. Any acoustic wood instrument is subject to this sonic treachery as well. But as mentioned in Chapter 2, "Microphones," all feedback happens in a somewhat narrow frequency range. This makes removing feedback a relatively simple process—although it does take some ear training and practice. The graphic EQs mentioned in the previous section are also very effective combatants against feedback.

A SOLUTION TO THE PROBLEM

As with many other properties in sound reinforcement, there are few absolutes. Although there are a few different ways to detect and eliminate feedback, be prepared to use various methods at different times on the same system when necessary. Also, don't be surprised if you find yourself improvising solutions at times.

When entering the battlefield against monitor feedback, try this tried-and true method to winning the war: With you at the mixing console, have an assistant speak (or sing) into a vocal microphone placed on a mic stand in front of a monitor on the platform. Adjust the mic EQ through the microphone's channel strip and the monitor EQ via an outboard graphic or other type of EQ. Once you have a good, natural sound from the monitor, increase its level until your assistant is satisfied with the decibel level. If the monitor begins to emit feedback before you get to an appropriate level, note what frequency is feeding back (this is where your ears' ability to identify a particular frequency becomes invaluable) and remove or lessen that frequency with a graphic EQ.

This doesn't imply that it's impossible to boost the level of your monitor to a useful volume without creating feedback. If this is the case, have your assistant take the vocal microphone off the stand and wave the diaphragm end at the monitor. Your assistant shouldn't point and hold the mic in front of the monitor or stick the diaphragm of the mic into your monitor horn; just have him or her carefully wave it by the monitor. Should feedback erupt, use your ears and remove the offending frequency from the monitor in the same manner just mentioned. Another option is to use an assistant to try to coax feedback by slowly placing his or her hand in front of a mic's diaphragm, about one inch away, or to simply open his or her mouth in front of the mic to create a reflection toward the monitor. Again, if any feedback presents itself, attenuate accordingly.

If you don't have an assistant, you can test the monitors yourself, go back to your mixing console and make your adjustments, and then go back to the stage and test again. And if you don't have an equalizer to insert between the aux send and the monitors, you can simply create a natural monitor sound, and then decrease the overall level of your monitors to control feedback—although this method is the least desirable because you may have to reduce the level of the monitors to the point where they are no longer useful to your performers. Keep in mind, too, that sometimes, simply altering the monitor's placement can reduce whatever feedback is plaguing your system.

FEEDBACK SUPPRESSORS

Although using your ears and EQ are common ways to eliminate feedback in your monitor system, there are devices on the market that are designed to automatically identify and remove feedback. These devices hear the feedback, lock onto the offensive frequency, and then remove or decrease it from your mix, making your job a whole lot easier. Some notable manufacturers of feedback suppression units include the following:

- Yamaha
- dbx (see Figure 5.4)
- Behringer
- Peavey

Figure 5.4
Feedback-suppression units like this dbx AFS 224 can be used to more accurately and—better yet—more easily combat any feedback that may present itself in your monitoring system. (Photo courtesy of dbx Professional Products.)

There are also many graphic equalizers with LCD displays that allow you to see the various frequency bands in real time (as they are created). With these units, you can visually and audibly identify the feedback frequencies and adjust them as needed.

KEEP YOUR GEAR CLOSE AT HAND

You've been looking at a few outboard units for controlling and eliminating feedback. Although they serve a vital role, their intentions are compromised if you cannot easily get to them. That's why it's important to have this gear at your fingertips. If you are fortunate enough to have a worship house whose sound system is outfitted with graphic (or other) EQs and/or feedback suppressors for your monitors, they need to be close at hand. Make sure they are mounted in a rack (discussed in Chapter 8, "Putting It All Together") next to your mixing console, as this is the only way you can effectively make adjustments during a service if need be.

In-Ear Monitors

As essential as floor-based stage monitors may be, they come with one major caveat: They add additional sound into your house system because the sound they produce bleeds back into the stage microphones and out into the house again, as well as reflect off nearby walls and the ceiling into the audience's ears. Eliminating the monitors helps reduce the overall decibel level on your stage, and helps you provide a cleaner FOH mix for the congregation. If only there were a way your stage personnel could hear themselves with customized mixes without using monitor wedges….

Well, ask and ye shall receive: Enter in-ear monitors! Due to the excessive decibel levels in many houses of worship, in-ear monitoring systems have become a popular alternative to the monitor wedge. As you can imagine, in-ear monitors (see Figure 5.5) are worn in your ears, much like a set of earbuds that come with an Apple iPod.

Figure 5.5
In-ear monitors like these M-Audio IE-10s are a great substitute for floor-based monitor wedges and provide users with low-volume isolated listening, which helps protect their hearing.

In-ears allow users to adjust their own mix by way of an in-ear personal mixer (see Figure 5.6). These specialized in-ear mixers enable users to select the instruments and vocals they want to hear. Just as you would create a mix on a full-blown console, users can select the various components to be included in their mix and then simply adjust the volume of those instruments and vocals to taste. An added bonus is that this affords you more time to concentrate on other aspects of the system.

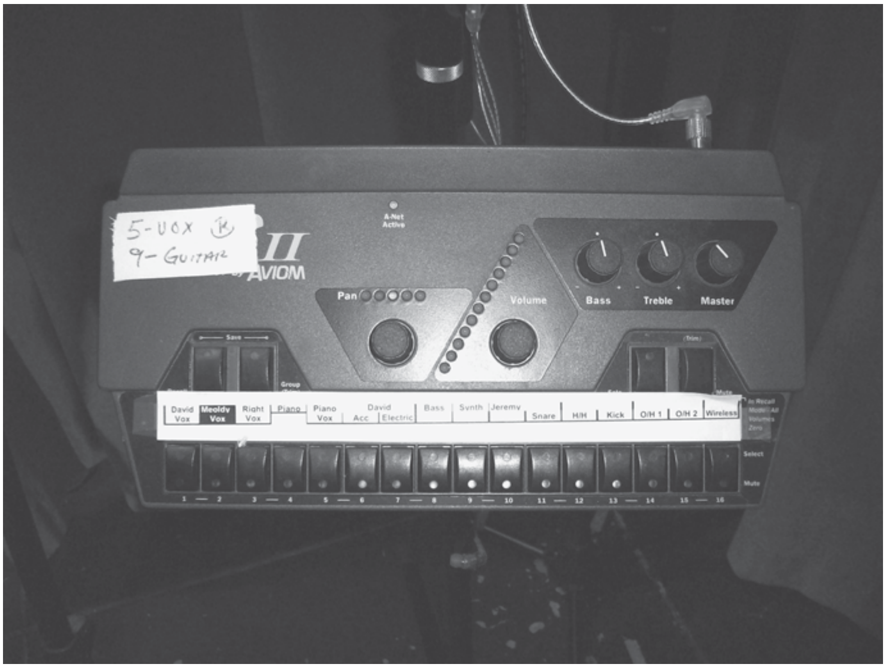


Figure 5.6
Augmenting the jubilation over in-ear monitors is their companion personal mixer, like this Aviom A-16II unit. This enables users to choose what components they want to hear and at what volume levels.

In-ear monitors can come in a wireless configuration. Unlike a wireless microphone, where the transmitter is incorporated in the body of the microphone and the receiver is elsewhere, the user of a wireless in-ear monitor carries the receiver in a belt pack. The advantage of using a wireless in-ear system is that it grants performers the freedom to move about the platform without sacrificing the monitor mix in any given position.

I KNOW ANOTHER WAY OUT!

In Chapter 3, you discovered the vast in-and-out configuration of a mixing console through careful examination of a Mackie Onyx series 24.4. Taking another look at the 24.4, you'll discover an additional output section called matrix outputs (see Figure 5.7). This supplementary output section collects audio signals from aux sends or subgroups and then sends those summed signals out in either a stereo or mono configuration (see Figure 5.8). This feature adds flexibility by providing a separate output section that can be specifically used for in-ear monitoring systems, making your I/O more organized. Another possible attribute to a console with a higher calling is direct outputs. These individual 1/4-inch outputs are located on the rear panel of a console (see Figure 5.9) and correspond to the channel strips they're aligned with, offering one output per channel. These output signals can be routed and then mixed through an individual's in-ear mixer, giving the user control of the monitor.

Figure 5.7
Many mid-sized to large-scale consoles, like the Mackie Onyx 24.4, feature a matrix output section, which affords you a secondary output section for routing signals to a personal in-ear monitoring mixer. (Photo courtesy of Mackie.)

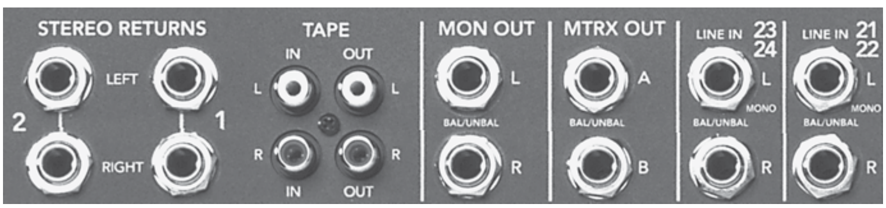


Figure 5.8
Matrix output connection ports (marked here as MTRX OUT) are located on the rear panel of the console. (Photo courtesy of Mackie.)

Figure 5.9
If your mixing board features channel-strip 1/4-inch direct outs, they are located on the rear panel of the board near the corresponding XLR and 1/4-inch line inputs, as shown on this Allen & Heath GL2400 console. (Photo courtesy of Allen & Heath USA.)

Powered Monitors Versus Passive Monitors

So far, the monitors covered throughout this chapter are ones that require the presence of a power amp in the chain in order to function. This type of monitor is known as passive. Conversely, monitors that house their own power amp are known as powered (see Figure 5.10) or self-powered.

Figure 5.10
Powered monitors like this FBT Maxx 6a feature their own internal power amplifier, which means you can cross off impedance matching from your to-do list. (Photo courtesy of FBT USA, Inc.)

Powered monitors provide several advantages over passive monitors:

- Just as you must pay close attention to the ohm load on the power amp when connecting speakers to your system, you must do so with passive monitors. This is not the case with powered monitors, however. This eliminates one of the many crucial aspects of your system you must pay attention to or face equipment damage.

- Powered monitors have their own on-board volume control and sometimes EQ (see Figure 5.11). These features enable you or the performer to adjust the level of each individual monitor. They also enable you to EQ individual monitor mixes to help overcome tone and feedback issues. The latter is especially valuable if your house system offers no other means by which to EQ your monitor mixes.

- Powered monitor mixes are limited only by the number of aux sends on your console. In contrast, passive monitors are limited by the number of aux sends as well as the number of power amplifiers (one mix per mono amp, two per stereo).

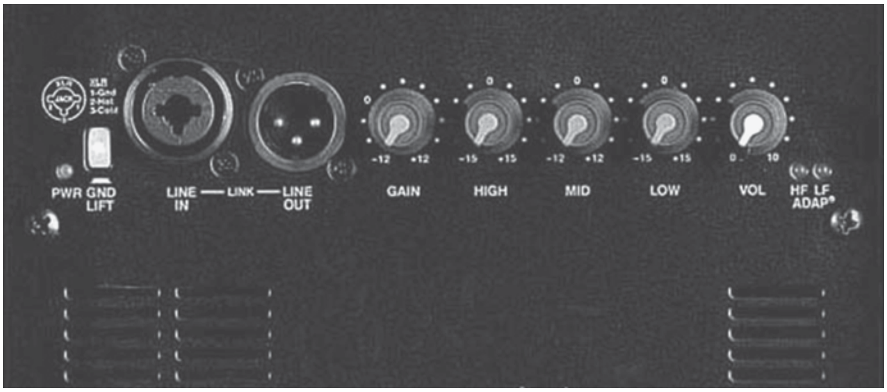


Figure 5.11
Another handy feature of powered monitors that makes your job that much easier is on-board volume control and EQ, which enable users to tailor the sound they're hearing right from the enclosure. (Photo courtesy of FBT USA, Inc.)

Of course, you will likely be limited to using the equipment that is available at your particular house of worship. As you plan for the future, however, you should strongly consider adding some powered monitors to your arsenal. You can easily add powered monitors to a passive-monitor system, as long as you have enough aux sends.

CHAPTER 6

Outboard Gear

So far you've learned about how to capture and amplify sound in your house of worship. This includes knowledge of fundamental theories of sound, the introduction to various components and their related concepts; and a handful of setup procedures. While the core apparatus of any sound system is microphones, a mixing board, power amplifiers, and speaker enclosures, there are many peripheral components that will be in the signal chain—especially in large-scale productions. These components, commonly categorized as outboard gear, are separate pieces of hardware designed for processing audio (some examples have already been introduced in previous chapters such as graphic EQs, speaker-management systems, and feedback suppressors). That these units are not physically incorporated into the mixing board is not meant to insinuate that they are any less important to your system and that they are never found in the designs of consoles and other staple equipment. With regard to the former, it's quite the contrary. After you harness the power of the units discussed in this chapter, you'll quickly realize your outboard gear is just as vital to effectively capturing and projecting sound in your house of worship.

While this chapter will expand on some outboard gear already presented in previous chapters, its focus is on the introduction of essentials like compression, noise gates, and effects processors. It's important to grasp the concepts contained in these next few pages because this equipment will help you immensely in controlling and/or enhancing the sound of your house audio system and worship team (including your preacher). Be aware that these units will require ample experimentation on your part, but rest assured that the time will be well-spent and a lot of fun!

EQ

Equalizers—mentioned in several previous chapters—are without a doubt at the top of the list in the must-have-that-piece-of-outboard-gear category. These outboard frequency processors are the tools you will use to balance the frequency spectrum of your worship space. Whether it is an overall "tuning" of the room or the elimination of a specific frequency causing unwanted hiss or feedback, equalizers—especially graphic EQs—are of paramount importance to your system. You could even go so far as to say that EQ is everything! How well you can use an EQ to identify and control individual and ranges of frequencies is a major factor in defining the level of your audio abilities and therefore your effectiveness as your house of worship's sound technician.

TO BE CLEAR…

Since we are talking about separate hardware units, we will not make any reference here to the equalization incorporated in your mixing console, just to the standalone equalizers within the signal path between the mixing console and the FOH loudspeaker enclosures.

TYPES

Should your house of worship have an equalizer, it most likely will be a graphic EQ. Typically, graphic EQs come in two types:

- **15-band.** This EQ divides the frequency spectrum into 15 center points (also known as a 2/3 octave EQ).

- **31-band.** This EQ divides the frequency spectrum into 31 center points (known as a one-third octave graphic EQ).

Each band of the EQ is a specific frequency point that features a linear, vertical slider (similar to a fader found on a mixing console) that allows you to boost (raise) or cut (lower or attenuate) the volume of that selected frequency band (see Figure 6.1). A 15-band EQ like one seen in Figure 6.1 can help you adjust general tonal characteristics of your FOH speaker boxes, while a 31-band EQ (see Figure 6.2) will be used for more precise tasks such as room tuning (see the upcoming note) and feedback control. A governing body called the International Standards Organization (ISO) establishes these frequency points, ensuring that all graphic EQs—15-band or 31-band—will have the same frequency points. Table 6.1 shows the exact ISO frequency center points for a 31-band and 15-band graphic equalizer.

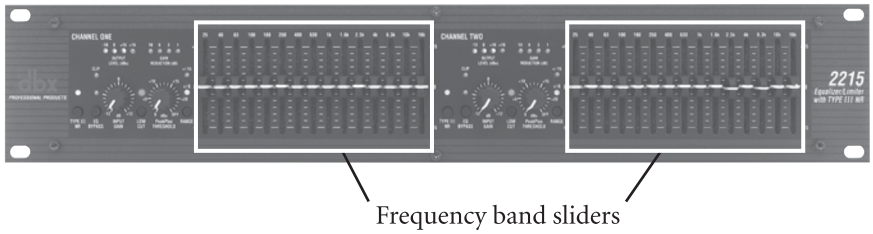


Figure 6.1
To adjust specific frequencies, move vertical sliders, like the ones seen on this 15-band, dual-channel dbx 2215, up (boost) or down (cut). (Image courtesy of dbx Professional Products.)

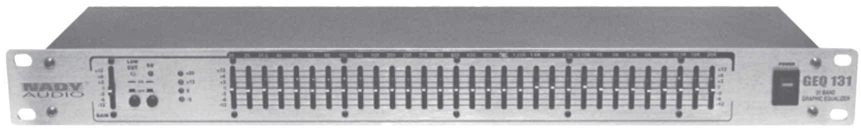


Figure 6.2
To make more precise frequency adjustments, a 31-band graphic EQ, such as this Nady GEQ 131, is used. (Image courtesy of Nady Audio.)

Table 6.1 ISO Frequency Center Points for a 31-Band and 15-Band Graphic Equalizer

31-Band Hz	31-Band Hz (continued)	15-Band Hz
20Hz	800Hz	25Hz
25Hz	1kHz	40Hz
31.5Hz	1.25kHz	63Hz
40Hz	1.6kHz	100Hz
50Hz	2kHz	160Hz
63Hz	2.5kHz	250Hz
80Hz	3.15kHz	400Hz
100Hz	4kHz	630Hz
125Hz	4kHz	1kHz
160Hz	6.3kHz	1.6kHz
200Hz	8kHz	2.5kHz
250Hz	10kHz	4kHz
315Hz	12.5kHz	6.3kHz
400Hz	16kHz	10kHz
500Hz	20kHz	16kHz
630Hz		

TUNING THE ROOM?

Up to this point in the book, the phrase "room tuning" has been mentioned in conjunction with equalization. To be clear, you're really tuning your loudspeaker system to sound better *in* your room. Since your preacher is at the top of the chain with respect to audio, you want the intelligibility of the human voice to come first. Consequently, tuning your system to accommodate your preacher may also work well for your worship singers and choir.

DIGITAL CONTROL

If your house budget allows for some extra special pieces, go for a digitally controlled unit like the Mackie EQ seen in Figure 6.3. With advanced features seen only on digital pieces—such as advanced algorithm development for more accurate representation of EQ settings, preset locations for saving EQ settings (up to 99), and LED display ladders—units like this make your job that much easier. Besides, it looks really cool!

Figure 6.3
Some graphic EQs—such as this four-channel 30-band Mackie Quad EQ—feature LED display ladders for a more-detailed picture of the EQ's effect on the output signal. (Image courtesy of Mackie.)

Another type of equalizer is the parametric EQ (see Figure 6.4). Like all equalizers, the parametric EQ gives you control over the sonic characteristics of your house loudspeaker system. But unlike graphic EQs that have their frequency bands fixed by the ISO, a parametric EQ allows you to sweep through the frequency spectrum so you can accurately select specific frequencies to boost or cut. To enable this, parametric EQs feature a control called the Q or bandwidth (the latter is seen in the Nady PEQ-5B 5-band parametric equalizer in Figure 6.4). The Q control allows you to focus on the broadness or narrowness of a selected frequency. Higher Q settings zero in on selected frequencies and their higher and lower adjacent frequencies, while lower Q settings affect a broader range of selected and adjacent frequency points. The downside to parametric EQs is that, compared to the graphic equalizer, they're a bit more difficult to use, making them scarce in house-of-worship sound systems. But, as always, you will be using the EQ device that is available in your particular house of worship and you need to be aware of the possible options.

Figure 6.4
If you need to attenuate specific frequencies outside of the ISO center points, then you'll need a parametric EQ like this Nady PEQ-5B. (Image courtesy of Nady Audios.)

CONNECTING

While reading through the upcoming steps and taking a look at the connection system found on the back of our go-to mixing console, the Mackie Onyx 24.4, as well as a Mackie graphic EQ, you can see that inserting your equalizer into the signal path between the mixing console and the FOH loudspeaker enclosures is a simple process:

1. Connect the right and left main mix (balanced) outputs of your mixing console (see Figure 6.5) to the corresponding right and left inputs of the equalizer (see Figure 6.6).

2. Connect the signal from your right and left EQ outputs to your power amp's left and right inputs.

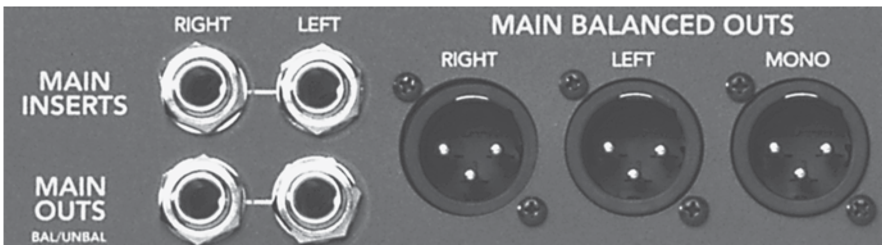


Figure 6.5
Taking a close look at the output section found on the rear of the Mackie Onyx 24.4 console, you see a choice for either balanced or unbalanced outputs from the main mix via XLR or 1/4-inch cabling.

Figure 6.6
Like the output choices seen on the Onyx console in Figure 6.5, EQs can also feature a choice of connection types.

These connections will be done with balanced XLR cables and/or TRS 1/4-inch cables. The location of your mixing console and power amps will dictate exactly how you make these connections. If your board and EQ unit are close to your power amps, you can connect directly from the EQ to the amps with balanced TRS 1/4-inch cables. If there's a significant distance between the EQ and the amps, you can use the returns of an audio snake (see Chapter 2, "Microphones," for a review) to take the signal back to the amplifiers. Using XLR cables, you will connect the one and two (left and right) returns from the snake to you power amp(s). However you make these connections, it's a good idea to label your connecting cables—left, right, in, out—as it makes for easier identification if (more like when) you have to troubleshoot a problem.

Compression/Limiting

If an equalizer is at the top of the must-have list, then a compressor/limiter (see Figure 6.7) is a close second. Although compressors and limiters are available as separate, standalone units, it's most beneficial for your house of worship to have these components bundled in one rack-mounted unit and to have as many as the budget will allow. Here's why: These two-in-one multi-purpose processors are designed to control the loudness and softness (also known as the dynamics of sound) of incoming audio signals. The compressor section, when used correctly, can dramatically improve the intelligibility of the preacher and/or vocalists and help control audio placement of band instruments in the overall FOH mix. The limiter section enables you to regulate erratic volume peaks in the signal by preventing unwanted high-volume sounds from traveling through your system, overdriving the mixing board, and ultimately distorting the FOH speakers. Take note: When these two components occupy the same device, as shown here, they generally operate independently of each other.

Figure 6.7
Compressor/limiters like this dbx 166XL are of paramount importance in your outboard arsenal because they control the overall dynamics of incoming signals as well as other sonic attributes. (Image courtesy of dbx Professional Products.)

ANATOMY OF A COMPRESSOR

To properly understand how to use a compressor/limiter, it's best to understand some basic concepts of compression. A compressor controls the dynamics (loudness or softness) of an audio signal by regulating the decibel (dB) level of incoming signals. There are five main elements to a compressor to be aware of. On the actual unit, they come in the form of knobs. They are as follows:

- **Threshold.** The most basic of all controls, this enables you to set the decibel level where the compressor starts to attenuate incoming signals. Lower settings (counterclockwise) instruct the compressor to start processing at low decibel levels, thus compressing more of the signal; high settings (clockwise) instruct the compressor to react to high decibel levels. Setting your threshold knob to a center position (usually denoted as 0) is a good place to begin to adjust the proper levels. Take note: The center point of this control is set by the manufacturer based on what they consider an average position, which might not be suitable for your needs. Be sure to experiment with this control to determine the best possible starting point for your environment.

- **Ratio.** This control, which sets the actual volume reduction, delivers the most obvious results. As indicated by the name, the reduction is set by the ratio of input signal to the output signal. For example, a setting of 2 will create a ratio of 2:1, which simply means for every 2dB of signal entering the compressor, 1dB will be heard at the output. Turning the knob clockwise, the next setting will most likely be a 4 (4:1), which will attenuate every 4dB of input down to 1dB of output. The higher the setting, the more compressed the output will be. Couple this with a high threshold setting and you can really squash the incoming audio.

- **Attack.** Operating in milliseconds (ms), this control sets the time it takes for the compressor to kick in. Fast attack settings (10 to 25ms) are good for processing vocals, especially when you want help eliminating sibilance (a process known as de-essing, used to control extreme S inflections in a vocalist's performance). Slower settings (50 to 100ms) allow more percussive sound's initial attack to pass through before any compression is applied, making for a punchier sound overall.

- **Release.** Also in milliseconds, the release control sets the time it takes for the compressor to stop processing the audio after it has dropped below the threshold. A quick release time (25 to 50ms) will cause the compressor to follow the signal closely so that rapid input changes will not be lost during compression. Slower release settings (50 to 125ms) smooth out the overall compression effect.

- **Gain.** Because the fundamental concept of a compressor is to attenuate volume levels, it's an added bonus when the unit features a gain knob (not all compressors have one). This control adjusts the compressed signal back to desired listening levels after processing as compared to the rest of the mix. If this control is not present, not to worry; it just means the compressor does it automatically. You just have more options when it's included.

Ideally, the un-initiated listener will never hear the compressor at work, as a well-adjusted compressor will do its work without detection. But making a compressor "well-adjusted" is no easy task. If you're not sure how to properly use a compressor, don't worry; many come with a function that allows you to put them on autopilot! Some units have an auto button that, when depressed, automatically sets ratio, attack, and release parameters. Other devices are completely automatic, with all functions controlled by onboard processors. dbx, Yamaha, Crest, Presonus, and Behringer, to name a few, manufacture semi-automatic or automatic compression products. But while it's true that getting positive results from a compressor will take some study, skill, and good ears, once you understand the controls of a compressor, you likely will want to make the adjustments yourself.

KNEE-JERK REACTIONS

Now that you have a basic understanding of a compressor's controls, there are two more related controls to explore: hard knee and soft knee. These settings adjust how the compressor engages an audio signal approaching the set threshold level. Hard knee means there's no processing of the incoming signal until it exceeds the threshold. On the other hand, soft knee sets the unit to apply a small amount of compression just before the threshold is reached and to continually increase compression through the threshold point and beyond. Soft-knee settings aid in making the incoming audio sound more natural, while hard-knee settings allow maximum loudness before compression. Your house compressor may have a hard knee/soft knee switch, enabling you to choose the mode. If not, it may be preset to operate in one mode or the other. As always, it's best to read the manual to discover such features.

COMMON USES OF COMPRESSION

Table 6.2 contains some settings to use as starting points when learning to apply compression into your mixes. Don't be afraid to tweak these suggestions from the onset and change your approach when circumstances change in your house. The key to be being an effective sound-reinforcement technician is to develop skills that enable you to adapt to changes on the fly.

Table 6.2 Compression Setting Suggestions

Instrument/Vocal Attack	Release	Ratio	Knee
Exuberant preacher	Fast medium	4:1 to 5:1	Soft
Vocals	Medium to fast medium	2:1 to 4:1	Soft
Acoustic guitar	Medium medium	3:1 to 4:1	Soft
Bass guitar	Fast fast	4:1 to 6:1	Hard
Electric guitar	Fast medium	4:1 or higher	Hard
Drums (snare, kick)	Fast fast	4:1	Hard
Brass horns	Fast fast	5:1 to 7:1	Hard

LET'S NOT FORGET THE LIMITER

A limiter is basically an extreme compressor that puts an impenetrable ceiling (at varying degrees, of course) on the overall incoming audio signal level. Just like it sounds, the signal hits a brick wall and will go no further. Limiters feature ratio settings from 8:1 to as high as 20:1, which provides the steep level reduction sometimes necessary to put a lid on any audio signal that passes the threshold. Limiting attack times are preset to be fast, while release times are set to be slow enough to create smooth, undetected results. You would be hard-pressed to find a standalone limiter, as this is almost always included in an outboard compressor.

CONNECTING

To use your compressor/limiter to affect the overall house mix, you will connect the right and left main mix outputs of your mixing console to the right and left inputs of your compressor/limiter. Should you also have an outboard EQ, you will connect the left and right outputs of your compressor to the left and right inputs of your equalizer. It's more likely, however, that you will want to compress only a specific voice and/or instrument. To do that, you will need to connect the compressor/limiter via the insert jack on the corresponding channel strip of the voice or instrument to be processed. (For a review on inserts, refer to Chapter 3, "Mixers and the Art of Mixing.") This connection requires a Y cable, which sends and receives the signal from the insert through a single TRS 1/4-inch jack, while the opposite end of the cable splits into one 1/4-inch jack for the input of the compressor and another for the output of the device (refer to Figure 3.13 in Chapter 3).

CHAIN OF EVENTS

As you've learned, you can connect multiple pieces of outboard gear in a line—main outs to compressor/limiter to EQ, and so on. This connection system is known as a signal chain, and is categorized as a series connection because all the units are connected from one to the next with no alternate route being created—just a straight line of signal, so to speak. There's another type of signal chain called parallel, where a split occurs at some point in the signal chain, making two or more paths and the signal run parallel to each other. Parallel connections may meet back up into a series connection at some point or stay parallel, thus creating multiple destination possibilities. As your audio skills evolve, you will come to more understand these concepts and find their advantages quite useful; for now, just be aware of the terminology and have fun tweaking those knobs no matter what kind of signal path you create.

Noise Gates

A noise gate is another type of dynamic processor, but this time it dramatically attenuates the audio signal passing through it when the signal drops below a preset threshold (see Figure 6.8). A common analogy used to describe a noise gate's function is a door closing; once the audio passes through, it drops below the threshold and the signal flow "door" shuts, preventing low-level operating hiss, unwanted noise, and any other audio anomalies from passing through the system out to the FOH loudspeakers to your congregation. A properly set noise gate will allow the audio signal from a specific instrument to pass through the gate to the mixing board unaltered and sound as if it's not even there, but there may be times when its own operating noise can be heard. If this is the case, it's a good idea to use a lower threshold setting to make it sound more natural.

Figure 6.8
Noise gates like this ISP ProRackG do exactly what its name suggests: They decimate underlying noise so it doesn't travel through your system out through the FOH enclosures for the congregation to hear. (Image courtesy of ISP Technologies.)

Noise gates can also have attack and release controls that function the same way they do on a compressor. Be sure to use your ears when adjusting these parameters to attain more natural-sounding results. If your noise gate doesn't have adjustable attack and release time controls, they will be automatic and preset by the manufacturer. Some noise gates also have a gain-reduction control in the form of a switch or knob. On most gates, you'll see two set levels of reduction—for example, 15dB and 60dB—but more advanced units will have a variable range control that enables you to dial in a value.

LET ME GET THIS STRAIGHT…
To be clear, thresholds on gates and compressor/limiters work opposite of each other. While they both deal with decibel levels, a noise gate affects audio signals that fall below a user-defined decibel threshold while a compressor affects signals that rise above a threshold.

COMMON USES

Generally, gates are not necessary for acoustic or worship music offered at a low decibel level. But if your house of worship rocks, then you'll need some noise gates for sure. Live microphones in a loud instrument setting are prime candidates for this type of dynamic processing. Here are some common applications:

- A gated kick drum mic will sound more natural and precise because the gate won't allow audio signal from other drums such as a snare or toms to enter the kick drum microphone. This also holds true for a gated snare drum mic.
- Electric guitar amplifiers are notorious noise makers, so when the guitar player is not playing, adjust the gate to remove all that excess buzzing.
- Worship vocal mics are rarely gated, but if your worship band is producing additional audio signal in the vocal microphones, a noise gate can eliminate this.

CONNECTING

The most common way to connect a noise gate is to use the insert jack on your channel strip, which enables you to gate one microphone at a time. Although it's possible to gate an entire mix, it's rarely done. In some cases, you may have a compressor/limiter with an onboard noise gate. If this is the case, you can compress, limit, and gate your entire mix by connecting the left and right outputs of your console to the left and right inputs of your outboard multi-task dynamic processing device.

RACK 'EM UP

Most outboard units are designed for mounting in a housing apparatus called a rack (see Figure 6.9). To ensure your piece will fit into a rack, a standard set dimensions exists. All rack-mountable outboard gear is 19 inches wide and has ears with holes that accommodate the mounting slots in the rack (see Figure 6.10). Rack heights are referred in units called spaces, which are 1 3/4-inch high. Although on rare occasions you will find a piece of outboard gear that is not rack-mountable, nearly all units marketed as such will fit into your rack just fine. If your house of worship doesn't currently own any outboard gear but is considering buying some or has a few pieces that are not mounted, it's highly recommended that you acquire a rack. Not only will it make your workspace much more organized, it will help protect the sizable investments involved in purchasing each piece.

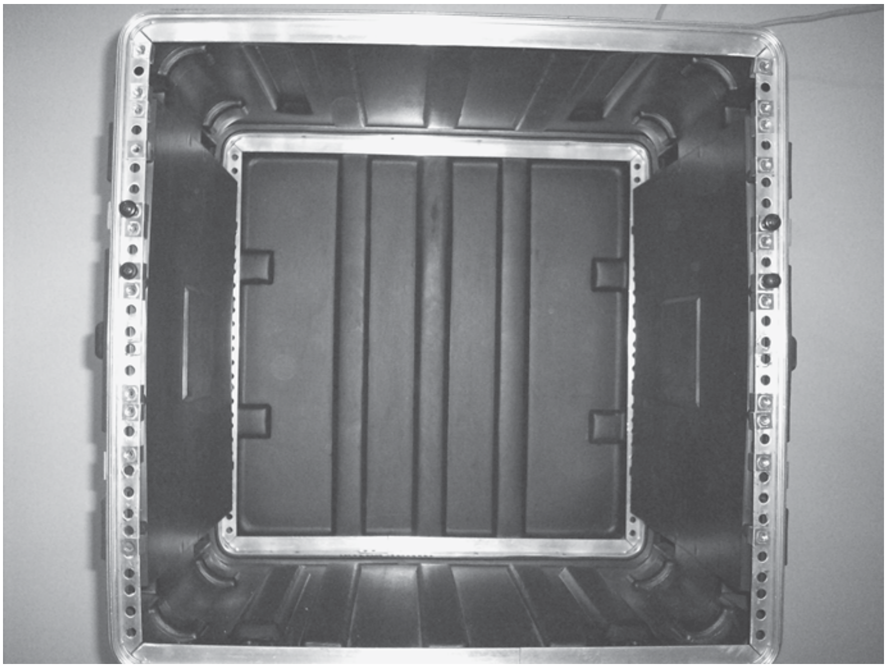


Figure 6.9
Rack cases are used to house outboard gear so you can easily access all your units in one convenient enclosure. This helps protect your house of worship's investments and makes all your outboard gear mobile if that becomes necessary.

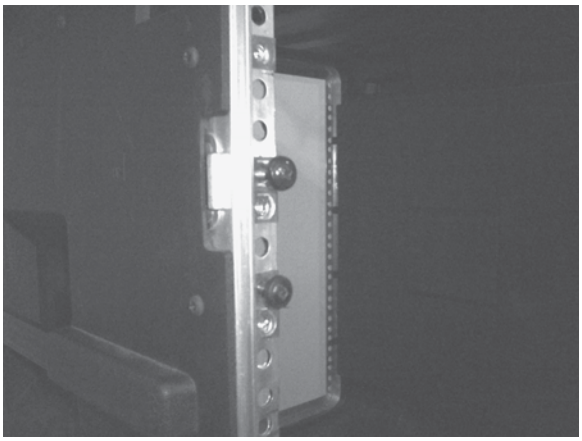


Figure 6.10
Most outboard gear dimensions are specifically designed to fit into a rack case—so much so that they could be described as being x *rack spaces in height. The mounting slots seen here are what the units are screwed into, suspending each piece in its place in the rack.*

POWER RANGER

When assembling your rack, you'll need to power all that outboard weaponry. Before you even think of it: *Do not rely on consumer-level surge protectors*. Instead, make sure you have a rack-mounted power conditioner like the one shown in Figure 6.11. These one-rack-space units not only provide multiple outlets (see Figure 6.12), they also offer much-needed features such as surge suppression, AC line noise filtering, and compensation for irregular voltages. What's more, a unit like the Furman PS-8R shown in Figure 6.11 can also be programmed to sequence system startup and shutdown. This is especially important in house-of-worship systems with pieces that must be turned on/off in a specific order. Overall, it's imperative that you correctly and safely power your system.

Figure 6.11
Providing your outboard gear—for that matter, all your gear—with pro-level surge protection and AC line filtration can be done with a rack mountable outboard unit like this Furman PS-8R II. (Image courtesy of Furman.)

Figure 6.12
In addition to the superior protection, power conditioners like the Furman PS-8R II provide conveniences such as multiple AC outlets (as seen here) and a power up/down sequencer. (Image courtesy of Furman.)

Effects Processors

Up to this point, the outboard processors we've discussed have been utility-type units. While they serve a variety of important needs in your house of worship system, the real fun lies within a category of outboard gear known as effects processors. The primary purpose of these units is to enhance the audio routed into them by applying special effects that make the outgoing audio sound fuller and more colorful. Effects processors can come as standalone units, such as the legendary reverb units from Lexicon, or all-in-one packages called multi-effects processors (see the upcoming note) like the venerable Yamaha SPX2000 (see Figure 6.13).

Figure 6.13
All-in-one rack mount effect processors like this Yamaha SPX2000 are most likely how you will be working with time-based and other spatial enhancing effects. (Image courtesy of Yamaha Technology.)

All-in-one rack mount effect processors like this Yamaha SPX2000 are most likely how you will be working with time-based and other spatial enhancing effects. (Image courtesy of Yamaha Technology.)

The most common effects used in worship houses are as follows:

- Reverb
- Delay
- Chorus

All three of these effects fall under a subcategory of outboard effects processors known as time-based effects. This moniker refers to the fact that these units' basic functioning properties are based on the manipulation of the incoming audio's position in time. Simply put, all three of these types of effects in one way or another repeat the incoming audio by user-defined parameters. The differences between them lie in how far apart the repetitions are and/or how those repetitions are treated.

Take note: Even though this section focuses only on the aforementioned time-based processors, it's of the utmost importance that you as a sound engineer seek out other types of effects processors and learn about their basic properties. You never know when you'll need to know how to properly adjust a pitch shifter or design a preset in a harmonizer!

MULTI-TASKING

When delving further into effects processors, it won't be long before you run into the mighty multi-effects processor. As their name implies, these powerful units feature more than one effect type under the hood. The concept makes sense, since many effects are derived from the same basic processing foundations. For instance, it's not uncommon to find a multi-effects unit that features all the aforementioned time-based effects along with others such as flanging and phasing. Depending on what type of services your house offers, it's likely your house already has at least one multi-effect unit tucked away in a rack somewhere in your system. However you get your hands on one, when the time arises, be sure to read the manual and really get to know the unit. Many engineers only scratch the surface of these over-achieving pieces of outboard genius by settling on some mediocre factory presets. If they only knew what they were missing!

REVERB

As mentioned in Chapter 1, "Sound and Sound Advice," reverb is naturally created when sound from your FOH loudspeaker cabinets almost instantaneously bounce off hard surfaces (walls, pews, support beams, etc.) in your worship space and create multiple reflections the human ear perceives as one encompassing, lush-sounding audio phenomenon. Not only can an effects processer artificially produce this magnificent sound, it also affords you control that's otherwise impossible. For instance, you have control over parameters such as how long the reverb will trail or the length of time between when the direct source stops and the reverberation begins. As with anything in audio, approach this effect with caution, as it is easy to overdo it. Too much reverb can wreak havoc during your house services by making the spoken word—the focal point of the event—difficult to decipher, and can even tempt feedback. But do try applying small amounts of reverb to help make voices and instruments sound as if they have more depth.

Another way to get familiar with digital reverb is to examine some basic parameters found on reverb units. You'll need to be familiar with these and whatever other controls are found on your particular unit to properly apply this effect to your house mix.

- **Space/room.** Since reverb is the persistence of sound in an enclosed space after the initial, original sound has ceased, it should come as no surprise that the most basic parameter would be your choice of room types. This parameter is often called space or room. Some common options are cathedral, concert hall, large hall, medium hall, and small hall. The larger the space, the longer the reverberation will last.

- **Decay.** The time it takes for the reflections to dissipate below human hearing (about 60dB) is known as the decay time. Many units enable you to modify the decay time, which is a handy feature when you're attempting to find a reverb that perfectly fits your worship space.

- **Mix.** The original sound source, with no reflections, is referred to as the dry signal, whereas any signal treated with reverb is referred to as a wet signal. The mix parameter gives you control over how much of the wet signal will be combined with the dry signal.

- **Pre-delay.** This adjusts the amount of time between the onset of the original signal and when the effects kick in. Always measured in milliseconds, pre-delay allows the original dry signal to leave your house FOH speakers almost instantly followed by reverb.

All this information is visible on the unit's rectangular LCD screen (see Figure 6.14), where you can also view all of the settings for choosing and tweaking your patch (see the upcoming note).

Figure 6.14
Effects processors are equipped with rectangular LCD screens that display information about the sound you have dialed up. What appears on the LCD screen differs depending on whether you're tweaking a parameter or you're simply viewing general information.

SAVING GRACE

After laboring over a sound with your digital effect processor that involved multiple tweaks within multiple parameters for something that you will use at most every event in your house, you will quickly realize that doing so over and over again is not very efficient. Luckily, you won't have to, as you can save your work just like you do on your home computer. Once you've programmed a sound you like on any one of your digital processors, you can save all the information pertaining to that sound to the unit's internal memory. This collection of parameter information is called a patch (or sometimes a preset) and can always be identified by the name you give it during the saving process as well as being placed in numerical order. You can make changes to your patch at any time or just delete it when you don't need it anymore. As an added bonus, many units enable you to back up or dump your patches to an external medium, further safeguarding your work.

DELAY

The delay effect is somewhat similar to reverb in that both replicate the original sound source—but the similarities stop there. Remember, reverb is sound bouncing off walls in an enclosed area, thus creating multiple reflections. That's what gives reverb its dreamy effect. In contrast, delay is more of a singular echoing effect. While both effects have to do with the re-creation of reflected sound waves, delay is all or part of the original signal re-created at a designated time. Delay repeats are measured in milliseconds, but delay does not incorporate space into its parameters; the effect deals strictly with time.

Let's examine the basis parameters found on a delay unit to familiarize you with how it works:

- **Delay time.** This parameter enables you to control the time between each repeat. Delay times have characteristic sound qualities to them. For instance, a short delay in the range of

50ms to 100ms would make whatever you're processing sound a bit like Elvis, as this was the delay time commonly used during the time of his early recordings.

- **Feedback.** This is where you adjust the number of repeats heard echoing after the original source has sounded. Generally, in house-of-worship settings, this parameter should be set low—say, two or three repeats.

- **Mix.** Like with reverb, this control adjusts the level of the effect as compared to the unaffected signal. In the case of delay, it adjusts how loud the repeats are compared to the original signal. Following suit with conservative feedback settings, a delay's mix setting in a house of worship should be low so as not to make the effect to prominent in the mix.

In most cases, delay will likely be allocated to your choir or worship singers and not be applied to your preacher at all.

CHORUS

The audio phenomenon we call chorus is achieved when multiple sound sources with similar tonal characteristics (timbre) come together to produce the same results and thereby sound as if they are one—but with a little extra. If you have a choir in your house of worship, then you've experienced real-life chorusing.

When singers in a choir sing together, they're not actually singing perfectly in tune with each other—that's actually impossible—nor are they singing perfectly in time with each other (another impossibility). Those slight variations in pitch and time produce a lush quality to the choir's overall sound that is nothing short of glorious. That shimmering sheen you hear is what we call chorusing. It's why we're so enamored of choirs and live orchestras with string sections.

This effect is exactly what an effects processor is trying to emulate when it's programmed to produce a chorus effect. For the processor to pull off such a feat, it must make a copy on the input signal (a form of delay processing), slightly modulate that copy, and then mix it in with the original dry signal it copied in the first place.

To fully grasp the chorusing effect, you need to understand the basic premise of the term modulate. Put simply, modulation means "change over time." With regard to audio, modulation refers to a particular changing parameter over time such as pitch (frequency) or volume (amplitude). The pitch of the copied signal is usually modulated by an LFO (low-frequency oscillator), which is a component that is designed to waver a specific parameter of a signal (in this case pitch) by a user-determined rate.

While there are several parameters that may be available to you on your processor, the bare-bones chorus controls are listed here:

- **Frequency/rate.** This parameter controls the speed or rate of the LFO. Slower rates (counterclockwise turn) make for richer-sounding chorus effects, while increasing this value (clockwise turn) makes the modulation repeat at shorter intervals and creates more underwater-like sounds.

- **Depth.** This parameter creates greater or lesser pitch changes. Modulation delay is a parameter that delays the chorus sound relative to the original sound. It is adjustable in milliseconds and should be used very judiciously.

- **Mix.** Like with delay, this control adjusts how much of the modulated, copied signal is heard along with the original dry signal.

Some other options might be a parameter that enables you to choose what type of wave form the LFO will use to modulate the copied signal. If you have this feature, try a sine wave first. This wave shape will sound smoother due to the overall curvature of the wave as compared to a square wave, which has more abrupt changes because the edges have hard angles.

DOUBLE DUTY

One of the harsh realities an engineer must face is the absence of gear. There can never be enough! If you have only one effects processor and that one makes your vocals sound great, you have to use that same effect for everything else. That's the sad truth for many of you—unless you have a two-engine effects processor! This is a stereo effects unit that enables you to operate the left and right channels (hence the two engines moniker) independently. With this type of a processor you can, for instance, use the left channel for reverb and the right channel for delay. You could even use a single channel as a mono effects processor if needed.

CONNECTING

Unlike the other outboard gear discussed already, your effects processor will connect and operate through a post fader aux send on your mixing console. Looking back at the Mackie Onyx 24.4 first seen in Chapter 3, "Mixers and the Art of Mixing," you will see six aux sends. Say you connect the output from aux send 6 via a 1/4-inch cable to the left (usually the default mono connection) input of a stereo effects processor. From there, return the left output of the effects processor through the board's aux return input using another 1/4-inch cable. The mixer's design is such that the aux return is routed to the left/right master output. Alternatively, you can return the effects processor signal to a channel strip via the line input if need be. Using a channel strip will give you more flexibility by allowing you to EQ the signal and send it to other aux sends.

This chapter has discussed a variety of ways to enhance audio signals in your house-of-worship system, but the actual space in which your worship services take place will ultimately determine the sound that exists within it.

CHAPTER 7

Creating a Sound Environment

The one thing that is as unique as our individual spiritual beliefs is the space in which we practice those beliefs. Every house of worship, especially ones with a sound system, has a definitive sound signature. The signature heard in each individual worship house is defined by the properties that make up the interior environment. That environment is composed of a combination of dimensions and the materials that are used to construct the interior boundaries (walls) of that building.

Buildings constructed of varying materials and architectural designs have drastic differences in the overall sound. For instance, a house of worship finished in drywall and carpet (see Figure 7.1) is going to handle sound completely differently from a church finished in marble and granite (see Figure 7.2). It is those variables that you will be exploring throughout this chapter.

Figure 7.1
Modern house of worship with a carpeted floor, drywall-covered walls, and cushioned seats.

Figure 7.2
Catholic church with marble floors, plaster walls, and wood pews.

These Four Walls (and More)

Because most congregations are not in the position to rebuild their house, nor are they planning to break ground to erect new construction with acoustics at the forefront of the design, it's important that you aware of what these existing edifices are made of. With this knowledge you can begin to understand how sound is dispersed, reflected, and/or absorbed in your particular house of worship. As a result, this information will enable you to strategically combat any negative sonic qualities your room may possess.

BUILDING MATERIALS

Houses of worship, as well as most other buildings of the same magnitude, are constructed from only a handful of building materials:

- Wood
- Reinforced concrete block
- Brick
- Steel
- Stone
- Poured concrete

It goes without saying that no house is built solely from one of these materials, nor does every building incorporate all of them. Also, the last two materials—stone and poured concrete—are used much less often than the other materials.

The interior of your worship house is finished using a combination of two or more of these materials:

- Plaster
- Drywall
- Wood
- Granite
- Marble
- Tile
- Linoleum
- Carpet

Now that you have your eyes open to what your house of worship is made of, let's examine what role these materials play in determining what the focal point—the worship space—sounds like.

THE WORSHIP SPACE

The most important room in any house of worship is the worship room, known in most denominations as the sanctuary. This interior space of your worship house is defined by the walls, floor, and ceiling of your worship space and should be considered as boundaries. These boundaries allow sound waves to do three things:

- Reflect (bounce)
- Be absorbed
- Pass through (see Figure 7.3)

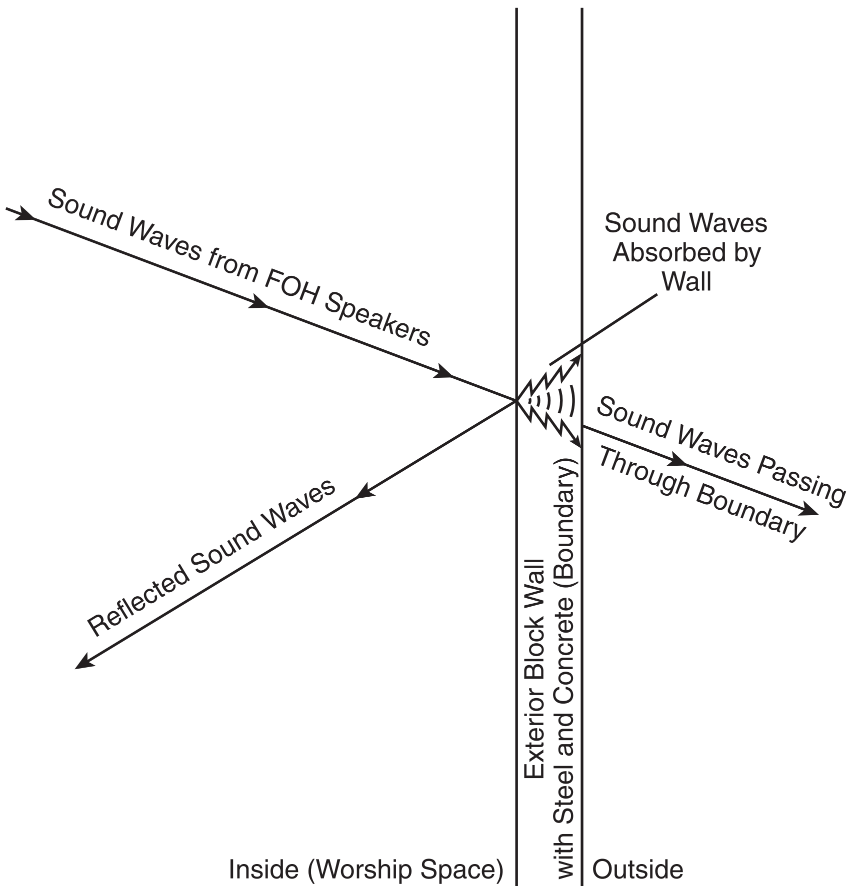


Figure 7.3
When sound waves hit any one of the boundaries in a worship space, they will do one of three things: reflect (bounce), be absorbed, or pass through.

Multiple reflections are what you need to be concerned with. You *do not* want an uncontrollable horde of sound waves reflected throughout the worship space. The overpopulation of reflected waves can create a significant amount of reverberation and, as stated in Chapter 1, "Sound and Sound Advice," too much reverb has a toxic effect on the overall sound of your worship space.

To have any chance at combating this potential audio dilemma, you must have an understanding of the dynamics and movement of sound waves so you can acoustically treat the walls, floor, and ceiling of your worship space. In order to do that, it's important to understand what material or combinations of materials make up your worship space and how they react to sound. Whether the walls, floor, and ceiling of your worship space are finished in concrete, brick, stone, plaster, wood, linoleum, tile, or carpet will have a varied effect on the interior sound. For example, hard, dense materials will reflect sound. Sound waves (energy) that are not reflected either get absorbed and converted to heat or pass through the boundary.

Let's take a look at some common construction scenarios, along with some suggestions as to how to deal with them. (The suggestions outlined here are explained in further detail later in the chapter.)

- If the walls of your house of worship are constructed out of concrete block (also know as cinder block), they are likely filled with concrete and rebar (cylindrical steel bars). These materials create a dense, reflective environment and must be treated with some sort of sound-absorbing acoustical material (you'll learn about many of the possible materials and absorption products commonly used in worship houses later in this chapter). Otherwise, the worship space will become too live, unable to support clear, distinct sound.

- The walls of your worship house may also be built of brick, which is another dense material. Brick can be painted; covered with plaster, stone, or marble; or left in a natural state (the same goes for concrete-block construction). Like concrete block, brick walls are too reflective; covering brick with sound-absorbing material will serve your environment best.

- Wood-framed worship houses will have their interiors finished in drywall, plastered over lath (a wood material designed to hold plaster), or finished in some sort of wood paneling. Wood-built buildings will have hollow walls, which creates an air space between the interior and exterior. Insulation that has been placed in that air space will help improve the interior sound quality by absorbing certain frequencies rather than allowing them to reflect back into the house of worship.

► Some large houses of worship are built around a steel framework, and the interiors are finished in some or all of the aforementioned materials, but more likely with wood or concrete block to create reinforced buildings. As with all dense reflective materials, they must be treated or covered with acoustical-frequency–absorbing materials in order to give you control of the sound in your worship space.

Knowing what your worship space is built of is important to understanding the effects the interior materials will have upon your sound system and audio signal.

TO ABSORB OR NOT TO ABSORB?

Table 7.1 displays various building materials, from the most dense and reflective to the most porous and absorptive. A common-sense scale from 1 to 100 (100 being the most absorbing) gives you sound-wave absorptions at different frequencies.

Table 7.1 Absorption Levels

Interior Material	Hertz (Hz) Absorption		
	160	**1k**	**4k**
Polished marble over brick	1	3	5
Concrete block	1	4	3
Brick wall (unpainted)	2	4	7
Interior plaster	2	6	4
Drywall (5/8-inch over 2×4 wood studs)	9	8	10
Plywood panels (1/8-inch over 2×4 wood studs)	20	10	8
Drapes (medium-weight cotton)	10	70	50
Drapes (heavy-weight cotton)	15	70	65
Sanctuary filled (full attendance)	50	85	80

This table should help you in starting to understand the sound signature of your individual worship house and its worship space. Be sure to look at the materials that are used in the interior of your worship space and, more importantly, to *listen* to the room's sound. While listening, ask yourself important questions like, Do the sound waves bounce from place to place like a rubber ball? (See the next section.) Does the room produce not a single reflection and therefore require an outboard effects processor to compensate for the deadness? Once you find the room's "sound signature," it's your job to cultivate it to sound the best it possibly can.

Natural By-Products

As you know, the primary room in your house of worship can be constructed with a variety of materials. You also know these rooms produce certain by-products (reflections) that appear at varying degrees as a result of the materials used. In this section, we'll take a look at the main by-products—reverberation and standing waves—and see how they can be controlled. It's important you develop this knowledge, as circumstances will change in your worship space. Indeed, these properties may appear even if you have not had to deal with them for a prolonged period of time. Also, if you will be mixing in other houses of worship, you will need to recognize these occurrences and act on them as needed.

REVERBERATION

In previous chapters, the concept of reverb—both natural and artificial—has been explored. In this section, you'll take a deeper look into the natural occurrence of this sonic event to best understand how to deal with it.

Once more, when sound waves hit the boundaries of your worship space, they are reflected, absorbed, or transmitted through the boundary. Reverberation is the by-product of the reflected sound waves. In order to best understand this abstract concept, think of the reflections as a small rubber ball, like the ones you can buy for 25¢ from a gumball machine. Those bouncing rubber balls seem to bounce endlessly once they hit their first surface. How hard you throw it is what determines how long it will continue to bounce. Similarly, reflected sound waves leave the FOH enclosures, hit the first boundaries in your worship space, and, depending on how hard and dense the boundaries are, will keep bouncing and bouncing. Now, if the boundaries are covered with sound-absorbing materials (draperies, carpeting, acoustic material, etc.), the reflections will act as if you threw your rubber ball at a pillow—there will be considerably less action, hence fewer reverberations.

Also, how sound waves reflect off the room boundaries depends on the angle at which they hit the boundaries. Going back once more to the rubber-ball analogy, if you throw it straight at the boundary wall, it will bounce straight back. But if you throw the ball at an angle, it will bounce back at an angle (see Figure 7.4). Ideally, this is what you want to have happen, as sound waves bouncing straight back at themselves can be more problematic than those reflecting at various angles (this will be explained more in the next section, "Standing Waves").

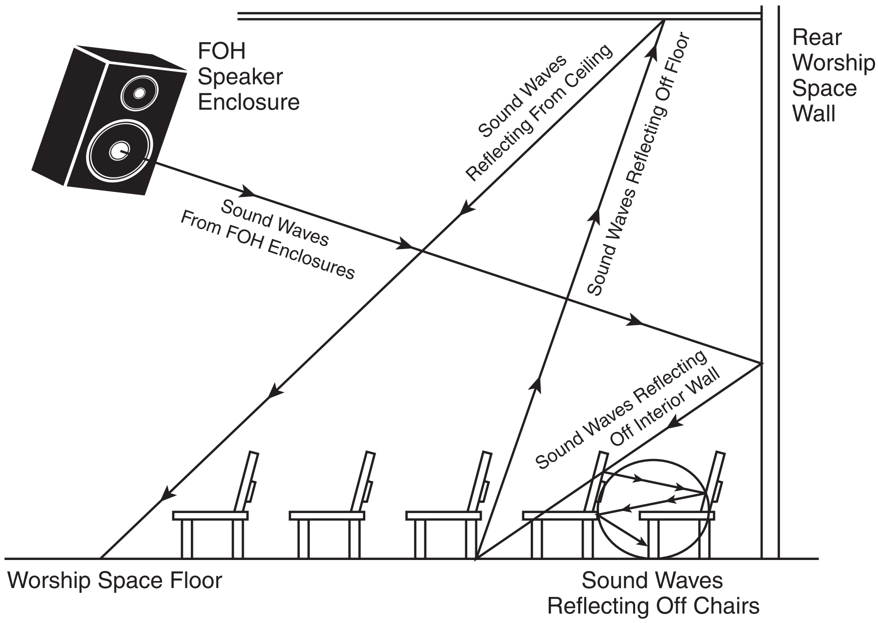


Figure 7.4
Multiple reflections, perceived in our ears as reverberation, occur after sound waves leave the FOH speaker enclosures and then travel on an angle toward the back wall of the sanctuary, only to bounce back into the room and continue to bounce off the floor, off the chairs, back toward the ceiling, and so on.

While the main boundaries of the worship space provide a playground for your reflections, there are a few variables to be noted that will have adverse effects on said reflections. Chairs, benches, and pews will act as obstacles that sound waves (unlike a rubber ball) will bend around (see Figure 7.5). People, on the other hand, act as barriers that sound waves not only bend around, but also absorb. Human beings absorb a great deal of mid-range and high frequencies, but even a fair amount of low frequencies as well. With this in mind, it should come as no surprise that if you have a full house on Sunday, those folks will have a significant effect on the acoustics of your worship space.

As you can see, many factors contribute to the unique way reverberation behaves in your worship house. Just so we keep things in perspective, not all reverb is to be considered negative. On the contrary, relatively short to moderate reverberation is perceived as musical, natural, and pleasant. More moderate reverberation can enhance some types of music, but will degrade the intelligibility of the spoken word. It's not until excessive reverberation levels have been reached that speech is difficult to understand and the dynamics of worship music are severely inhibited.

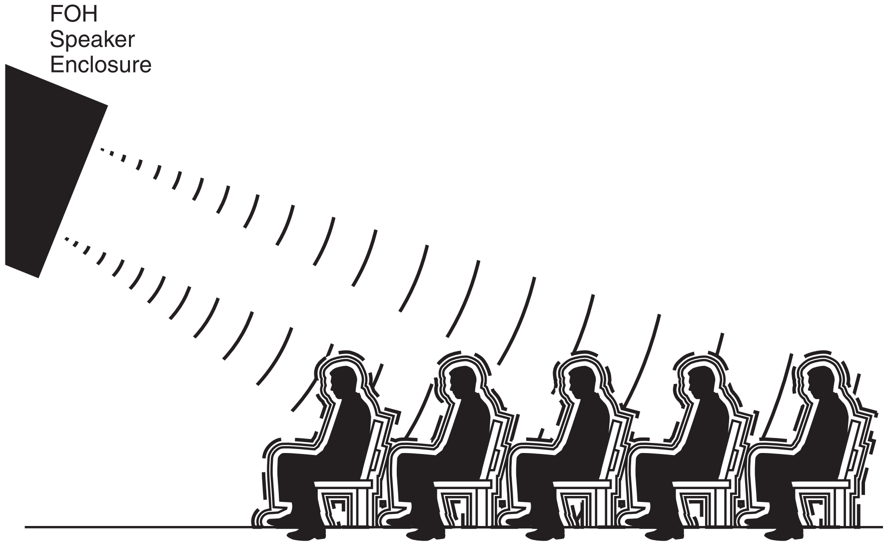


Figure 7.5
As much as sound waves bounce within the worship space, they also bend around obstacles such as seats and even the people in your congregation.

STANDING WAVES

When a sound wave of a particular frequency leaves your house speaker cabinets and strikes a dense boundary such as the rear wall of the worship space, that sound wave is reflected back into the room. Should this reflected sound wave collide with another sound wave of equal frequency from your house speakers, a summation of the two waves, called a standing wave, is created at the collision point. This phenomenon can cause a fluctuation in the volume of certain frequencies or, in extreme cases, the absence of sound. If you walk in the path of an existing standing wave pattern, you can hear the points where the sound is louder and where it is softer. Standing waves may cause, for example, a loss of the frequencies that best represent your preacher's voice or an over exaggeration of the mid-range volume of your guitar player's amplifier, skewing the volume and tone of what was intended to be heard. To further the effects of standing waves, this phenomenon can repeat itself along the sound-wave pattern. Therefore, standing waves must be eliminated or, better yet, prevented.

Like singular waves, standing waves have crests and troughs (see Figure 7.6). The difference in this case is that the crest and trough are by-products of two propagating waves traveling in opposite directions. The standing wave's crests, known as anti-nodes, are where the sound waves combine and reinforce one another, creating maximum sound pressure and a louder-than-intended signal. Opposite to the crest, the troughs of the standing wave combine and create a stationary pattern in the air of low sound pressure. These points are called nodes, and they are where sound is decreased as certain frequencies may cancel each other out and are no longer heard.

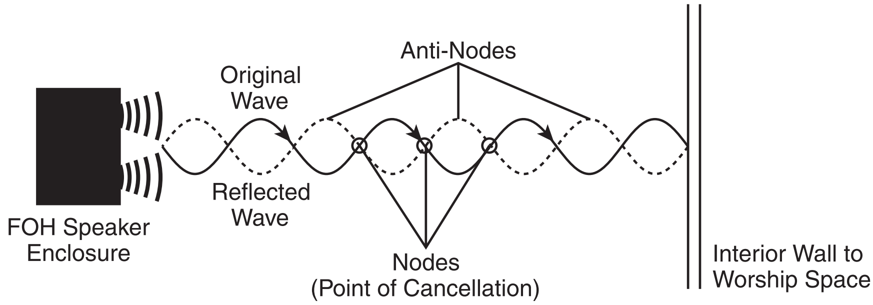


Figure 7.6
A sound wave that leaves the FOH speaker enclosure and bounces back from the opposite wall only to collide with itself creates what is known as a standing wave. With these waves are nodes and anti-nodes, which have undesirable audible effects that must be attended to.

Much the same as in the interior of a speaker enclosure, these waves can only be created in a worship space with parallel walls. Although a house of worship designed with acoustics in mind takes into account the resonance of a room and the possibility of generating standing waves, the reality is most existing houses of worship have parallel walls. So rather than rebuilding your worship house, let's look at how we can acoustically treat the interior in order to more easily control sound and thereby make your congregation's worship experience a fulfilling one.

Acoustic Environments

Ideally, when the congregation is worshiping together, the sound system should disappear. In other words, a sound system that is properly amplifying the preacher, the worship band, and singers does not exist in the consciousness of the congregation—it's simply a tool to facilitate worship. You, the sound technician, play an integral part in creating this dichotomy of purpose and transparency. Part of your job in doing so is answering one deceptively simple question: What is the best acoustical environment for a house of worship? Let's answer that question and at the same time examine some aspects that, well, may *not* make for the best acoustical environment.

BEST FOR SPEECH

Simply put, either you can understand your preacher or you can't. It goes without saying that the latter is unacceptable and that measures must be taken to prevent it from happening. But before we get into fixing problems, let's define the ideal worship space for the spoken word.

In examining worship spaces that provide an environment best suited for projecting the spoken word, you will generally find certain commonalities. Following is a list of materials to watch for and/or consider for your own house of worship's needs:

- **Carpeted floor.** A carpeted floor will improve reverb time as compared to a tile, wood, marble, or concrete floor.
- **Draperies.** Mounted over windows or along the wall space that is opposite the FOH speaker boxes, draperies will reduce the reflection of some high and mid frequencies back into the worship space.
- **Drywall.** Drywall over insulation will have better sound-absorbing qualities than brick, marble, wood, concrete, or plaster walls.

It's important to mention that upholstered walls or walls partially or totally covered with sound-absorbing material can also greatly improve sound-wave reflection; we'll explore the use and installation of these materials later in this chapter.

NOT THE BEST FOR SPEECH

Assuming your preacher has ample means of projecting himself or herself audibly—meaning, he or she is loud enough to be heard by the person in the last pew—there's only one real way to get in the way of the word. As stated several times before, excessive room reflections (natural reverb) will render the spoken word indecipherable. While there's no question that a certain amount of reverberation makes the human voice sound more natural and the spoken word more intelligible, there is no all-encompassing consensus on exactly how much reverberation is too much or what the ideal reverb time is.

If you are fortunate, the spoken word will already sound natural and clear in your particular house of worship. Generally, this would mean your house reverberation decays in less than one second. If you have a fairly large worship space, however, then this is most likely *not* the case, as these rooms tend to have lengthy decay times due to the abundance of hard surfaces throughout. Those long reverb times are what wreak havoc on the preacher's message. Conversely, there are houses out there that have almost no discernable reverb. In other words, they are nearly acoustically "dead." In this case, you will need to use your effects unit to add the most natural-sounding reverb you can. As you discovered in the Chapter 6, "Outboard Gear," not only can an effects processor manipulate the amount of reverb time, it can also adjust how much of the reverb effect you and your congregation will hear.

BEST FOR MUSIC AND SINGING VOICES

Vocalists and live music require longer reverberation times than the spoken word to sound their best. Typically, 1.5 to 2.5 seconds will do justice to the blending of singing voices and worship instruments. The optimal setup to naturally achieve these reverb times would be wood walls or paneling on the wall area perpendicular to the FOH speaker cabinets. In addition, high wood ceilings can provide increased reverb decay times. A combination of hard and acoustically absorbent materials can also improve the sound of music in your worship space. For example, draperies over wood paneling work well in combination. Another is carpeting, which can be used to offset the reflective effects of concrete block.

BEST OF BOTH WORLDS

All things considered, you could find having a worship space with a natural reverb time of one second or less to be more advantageous. This gives you the opportunity to add artificial reverb via an effects processor to afford you more control. In this scenario, you also need not have to worry about treating the room in order to prevent the word from being drowned out by excessive reverb times. Consider this: A room that has the perfect amount of reflection for your choir will most likely be a nightmare for projecting your preacher, as we discovered in a previous section. It follows that if you have control over how much reverb is added to your worship space, you can add more reverb time for your music and less for your preacher. In order to decrease reverberation in your house, you will have to address the areas in your space that are too reflective. So, let's examine just how to do that.

Fixing Acoustic Problems

By now you've realized this simple fact: More often than not, houses of worship have too much reverberation. These mighty marvels of human architecture tend to be too live to hear the preacher intelligibly, the worship band cohesively, and the singers and choir clearly. You've also discovered the best-sounding worship space is the one you have the most control over. With the ability to apply the desired amount of reverb, you can add a small amount of reverb to the preacher and then apply a completely different amalgamation of reverb parameters to your worship band or choir, thus enhancing their performances. This section examines how to evaluate the acoustic properties of your particular house and then lays out what to do to overcome its sonic shortcomings. Starting with simple, cost-effective options and going all the way up to small-scale construction projects, there's plenty that can be done to counteract problems that commonly exist in houses of worship.

BE A GOOD LISTENER

The first step to addressing any problems that may exist in your worship space is to find out where they are. A good start is to get into your worship space when it's empty. From there, fire up the sound system, put on some canned music (CD, iPod, etc.), and just listen to the room. Be sure to listen to the music from different locations, paying close attention to any standing waves that may exist. At the same time, listen for the sweet spots and try to determine why the music sounds so good in those locations. Remember, your worship space sounds markedly different when it's empty compared to when it is filled with people, but this is still a logical initiative nonetheless.

Once you've made your rounds around the room, turn off the music and clap your hands. What you're listening for now is reverb decay time. Once again, take a walk around your worship space and listen to the varying degrees around the room. If you have a balcony, make certain you visit there as well and put your hands together.

If you have the opportunity to perform these evaluations with an assistant, by all means do so. The old adage "two heads are better than one" (or, in this case, "four ears are better than two") goes a long way in this case. In addition to having another pair of ears handy, you can have your helper up on the stage talking through the preacher's microphone as you listen for standing waves and excessive room reflections that may emanate from the pulpit. After you finish these tasks, you will have a grasp of the acoustic properties of your worship space (when it's empty, anyway).

CALL FOR HELP

For some of you, making the call as to what needs to be done to acoustically treat your worship space might seem a little daunting. Do not fret, as there is an alternative. If your house has a good budget for sound, you can hire an acoustician. These highly skilled technicians will come to your worship house equipped with their ears, a variety of sound-measurement devices, and some serious math skills, and then proceed to evaluate the acoustical properties of your worship space. Upon completing their probe, they will make suggestions on everything from what soundproofing materials should be installed to how many enclosures you should employ to where those enclosures should be positioned. In the end, you will have everything you need to implement a plan to make your house of worship the best-sounding room in town.

SIMPLE SOLUTIONS

Now that you've got your ears around the acoustics of your worship space, you must determine how you can fix any inadequacies. While many of your actions will depend on how much money there is in your worship-house budget, there are a few freebies that will prove to be saving graces. A simple and very inexpensive fix is to fill your house of worship with people! Remember, human bodies are excellent sound-absorbing devices and effectively help diminish a wide range of audio frequencies. If you have upholstered chairs, you can also count those as built-in sound-absorbing materials.

Another fairly easy and potentially cost-free fix is to alter where your FOH enclosures are pointed. Look to see if your loudspeakers are pointing directly at the opposite wall from where they are positioned. Speaker cabinets sending their sound waves directly at an opposing flat surface (especially ones constructed of dense materials) can create focused reflections, also known as discrete echoes or slap back. These reflections sound like very quick repeats, much like what you hear on musical recordings from the '50s, especially in early rock 'n' roll and rockabilly records. (Got Elvis?) That type of setup can also create standing waves. To counteract these issues, the enclosures should be slightly angled in relation to the opposite walls (boundaries) of the sanctuary (see Figure 7.7). This helps disperse the sound as evenly as possible from the front to the back rows of the worship space.

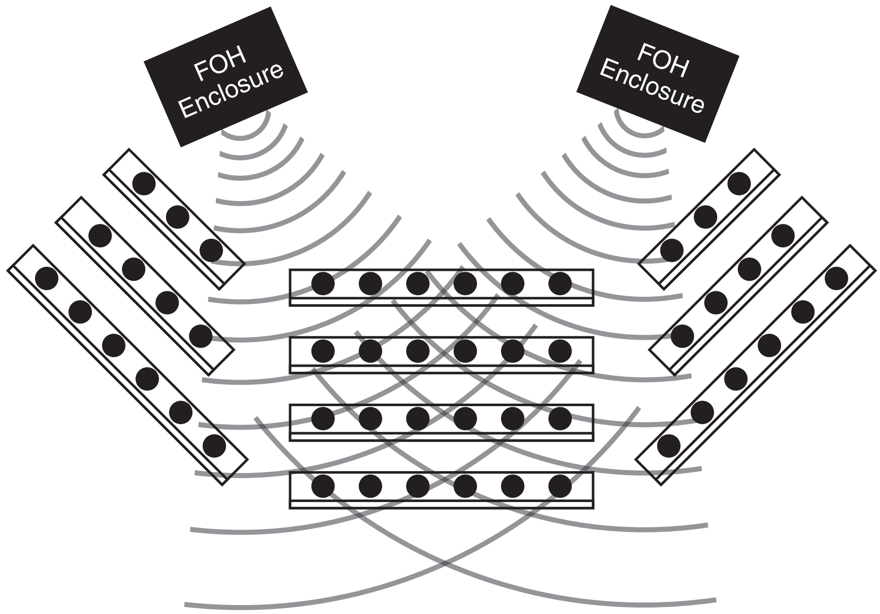


Figure 7.7
Mounted speaker enclosures that are positioned at an angle compared to the opposite walls help spread sound evenly throughout the worship space, not to mention counteract any offending reflective anomalies.

Utilizing materials such as carpeting and drapes can be simple fixes that, at the same time, enhance the look of your worship space. For instance, many older, more grandiose worship houses of larger dimensions have some sort of stone (marble, for instance) or wood floors. If this describes your house, consider throwing inexpensive carpet runners down the aisles, as carpeting on the floor can aid in controlling high-frequency reflections. As for drapes, they can do a lot to arrest reflecting sound waves. Whether you have windows in your worship space or not, heavy drapes hung over hard surfaces (brick, plaster, concrete walls, etc.) will trap a considerable amount of high and mid frequencies. If the draperies are mounted on retractable rails or can be electrically drawn, they can be opened or closed to change the acoustic properties and reverb times of your room.

PAINT

Did you know there are actually paints designed with specific acoustical properties? These acoustic paints have the ability to defuse and absorb high frequencies and can be helpful in controlling room reflections. In addition, many acoustic paints have insulation and adhesive qualities that can reduce vibrations and even improve climate control in your worship space. While there is much debate as to how effective these paints are, there's enough positive evidence to warrant your own investigation. If your house will consider a makeover of sorts, be sure everyone involved is aware that this type of paint leaves a thick texture on your walls.

ACOUSTIC PANELS

A common acoustic-specific add-on solution for treating your worship space is acoustic panels. Also known as noise-control panels, these products can be installed in various places such as mounted to hard surface walls (see Figure 7.8) or as substitute ceiling panels (see Figure 7.9). Typically, you can purchase these panels in 4×4 or 4×8 foot sizes that are two- or four-inches thick. However, many acoustic panel manufacturers, such as Auralex or GIK Acoustics, offer a large variety of sizes and shapes. Not only can acoustic panels be permanently mounted on the walls of your worship space, they can also be attached to rails for retractable applications (much like the draperies) or be hung from ceilings (see Figure 7.10).

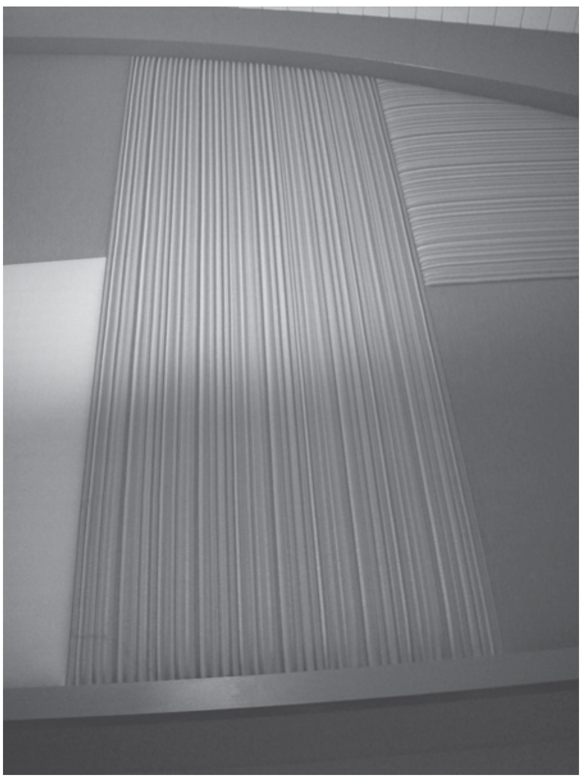


Figure 7.8
Acoustic panels here can be used in places where draperies are not needed and at the same time serve as a decorative attribute to the worship space.

Figure 7.9
Panels, known as acoustic or noise-control panels, are specifically designed for acoustic room treatment. (Image courtesy of Auralex Acoustics, Inc.)

Figure 7.10
Panels can be placed just about anywhere in your house of worship, including the ceiling, as seen here. Note the panels mounted the perimeter walls of the band area, which also includes a Plexiglas isolation fixture for the drummer for decibel reduction. (Image courtesy of Auralex Acoustics, Inc.)

As an alternative to hard-surface panels, acoustical panels made of foam (see Figure 7.11) can greatly improve control of your worship space sound as well. These geometrically diverse panels come in similar sizes to the noise control panels but have irregular surfaces. This type of panel generally is available in wedge, pyramid, and egg-crate designs (to mention a few). The varied surfaces are designed to diffuse, break up, trap, and absorb a wide range of frequencies. Many manufacturers even make acoustic foam products that fit well in corners and are excellent for controlling corner reflections (see Figure 7.12).

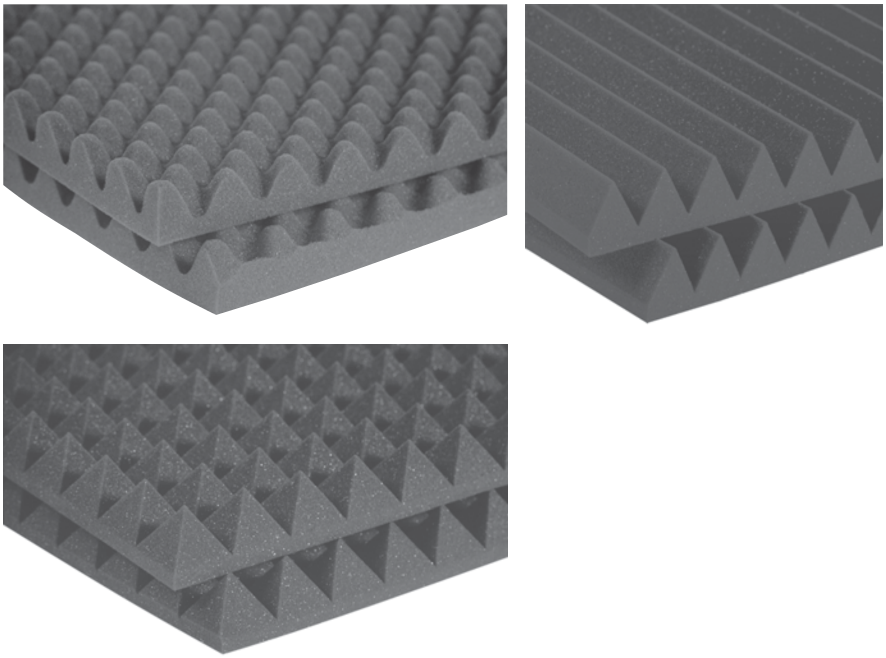


Figure 7.11
Acoustic room treatments can come in the form of foam panels as well. They come in a variety of shapes, including wedge-shaped and pyramid, with the most common being the egg-shaped panel. (Images courtesy of Auralex Acoustics, Inc.)

Figure 7.12
If you find yourself trying to cut foam panels to fit snugly into a corner, stop! Just get a corner piece! (Image courtesy of Auralex Acoustics, Inc.)

FALSE WALLS

If you have a large wall that needs to be acoustically treated, you will quickly realize the drawbacks of using acoustic panels:

- They only affect the area they cover, which requires you to use a lot of them.
- Their modern look may not work in a traditional worship house.

Another option is to consider building a false wall and filling it with acoustical insulation. Although some of you reading this book may not be carpenters or handy with a hammer, here's a summary of the steps to build such a wall:

1. Frame what would be very similar to a typical exterior wood wall (2×4 studs, 16-inches on center), but attached to your existing interior.
2. Use caulking glue between the 2×4 studs and the wall surface to prevent your wall from vibrating in any way.
3. Once you have framed your false wall over an existing wall, install your sound-proofing insulation (see Figure 7.13). This material is usually made of cotton fiber or rock wool, which will not only improve the sound characteristics of you worship space, but it's a great insulator that can improve the efficiency of your heating and cooling system!
4. Cover your new wall with regular or acoustical drywall. You can also use isolation clips, which keep the drywall from coming in contact with the wood studs. These clips will also improve the acoustical characteristics of your new construction project (see Figure 7.14).

Figure 7.13
Another method for sound control, especially in the case of isolation, is to install acoustically enhanced insulation like Mineral Fiber Insulation (MFI) from Auralex between the studs much the same way you would the "pink" stuff. (Image courtesy of Auralex Acoustics, Inc.)

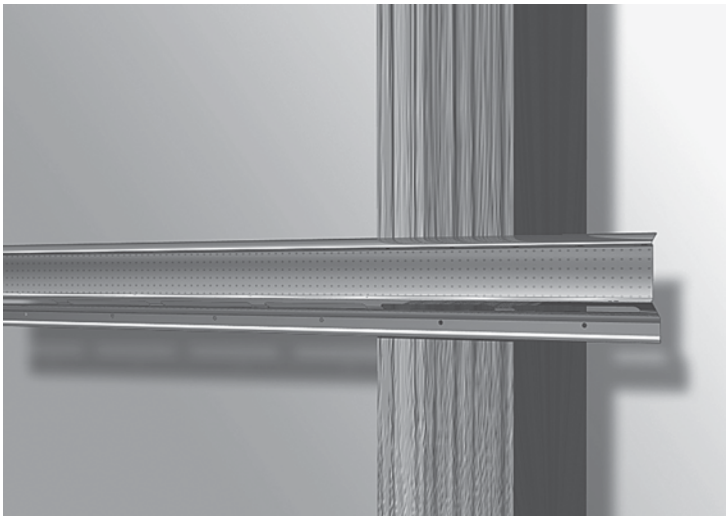


Figure 7.14
Isolation clips, like this Auralex RC-8 resilient channel, will help improve the sound-transmission characteristics of your walls by preventing the drywall from coming in contact with the studs. (Image courtesy of Auralex Acoustics, Inc.)

Building this type of wall works best over a brick, concrete-block, or poured-concrete surface. If you do not understand the construction terms just mentioned, you should not be the one building the wall; be sure to consult with a qualified, insured carpenter.

Regardless of what type of acoustic treatments you employ, aesthetics need to be considered. If your sound-control fixes look remotely strange to the congregation and/or the worship team, it won't really matter how improved your worship space sounds. If you are going to invest in a quality acoustical environment, be sure to make it look good, too.

Although this chapter has only scratched the surface of sound control and acoustic treatment, you now have a clear idea of how sound waves react in your house of worship. Couple this essential knowledge with the information found in the previous chapters, and you're ready to put it all together.

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CHAPTER 8

Putting It All Together

In the previous seven chapters, you were introduced to the core aspects of a sound system in a typical house of worship. In addition, you discovered many techniques essential to manning and maintaining that sound system. It's been a long road indeed, especially if the concept of sound reinforcement and its components are new to you. From the foundation of sound waves to the concept of impedance to the idea of why you might need a speaker-management system in your rack—hopefully, this book has served you well thus far, effectively conveying this foundational knowledge.

One way to find out is to put this knowledge to work by going into your worship space and starting to, well, put it all together. In this chapter, you will be guided in doing just that. It covers the connectivity and operations of a worship-house sound system, including tasks as simple as plugging in a microphone cable and as complex as gaining a deeper understanding of the flow of your audio signal. Hopefully, as you read this chapter, you will develop your own plan and rhythm for setting up sound for a worship service and you will be able to put that plan to use in getting your worship system operating smoothly and effectively.

Organizing

Like anything else in life, the more organized you are, the better off you will be. When it comes to effectively running something as complex as a worship-house sound system, that sentiment couldn't be more pertinent. In order to be organized, you have to know what that term implies in this case. To a house of worship sound technician, organization is knowing:

- Where the DIs are so you can patch in a special guest musician you were informed about 20 minutes before the service is about to begin

- Where the extra wireless microphone receivers are after you've watched your preacher drop, step on, and crush the one you just put on him/her

- Just where that pack of 9-volt batteries you bought at Costco is, as the batteries in the guitarist's tuner have just failed and now he/she is painfully out of tune

It's having all your tools—mic stands, extra monitor wedges, gaffers tape, etc.—close at hand that makes running your sound system a successful endeavor. Not only is it the physical location of these items, but also the manner in which your equipment (and platform area) is taken care of before and after a service. This section will bring to light many thoughts as to how to organize yourself so you most effectively and efficiently run your sound system, making the worship service a happy reality. Remember, the more the congregation knows not of what you do, the better you're doing it. So let's get organized!

LOCATIONS

Every sound system is made up of permanent or semi-permanent gear (mixing console, mounted speaker enclosures, drum riser, etc.) as well as myriad components that are easily—not to mention usually—moved around (microphones, mic stands, monitor wedges, cables, etc.).

When you walk into your worship space before a service, you expect the non-transient gear such as the power amp rack to be in the same place it has always been and the speaker cabinets to be aligned in the same exact position you painstakingly set them after a grueling bout with a naughty standing wave last month. These items should stay where you left them if for no other reason than because they're heavy or not easily accessible. Once these components are set, it's unlikely you will move them for any reason except for maintenance.

But what about the mic stands? And, while we're at it, where are the microphones themselves? Do you know where the XLR and 1/4-inch cables are? These are the items that are most likely going to be put away after a service for storage (and security) or simply moved to another location to accommodate the plethora of purposes they will serve in your worship house. That's why these types of items—the movable ones, that is—need to have a common place where they can be found without fail by you and your fellow technicians. Let's start with the cables.

IF IT AIN'T BROKE, DON'T FIX IT

Be aware that the suggestions in this chapter are just that. Your house of worship may already have its own system for the care and storage of its sound equipment. If your place of worship is already a pillar of organization—many of them pride themselves on it—then chances are you might be able to let it be. Instead of implementing changes just because you read about them in this book, incorporate only the ones that will improve your situation. Organized environments are usually created by people who like things the way they are. If you do in fact see that a change can be made to better serve the system, try to implement it, but be prepared to explain yourself.

CABLES

Sound systems incorporate an incredible amount of cabling, so keeping cables organized is of the utmost importance. Not only does it ensure that you'll be able to find them when you need them, but when stored properly, it also aids in their longevity. The ideal storage situation in any house of worship is a locked utility room, preferably near the stage. In this room would be all your cables and any type of short wiring (as well as microphones, mic stands, direct boxes, adapters, batteries, and any other peripherals when not in use). If this is not the scenario in your house, do not let that be an excuse for an unorganized mess, which is what even a small collection of cables can quickly become.

Regardless of what the scenario is, try to organize your cables by the criteria listed below.

1. Separate the cables by type: XLR, instrument, speaker, patch, etc.

2. Within those categories, further separate by size.

3. Tag the various lengths with colored tape or cable ties. For example, tag the 15-foot cables with red, the 25-foot cables with yellow, the 30-foot with blue, and the 50-foot with green (see Figure 8.1).

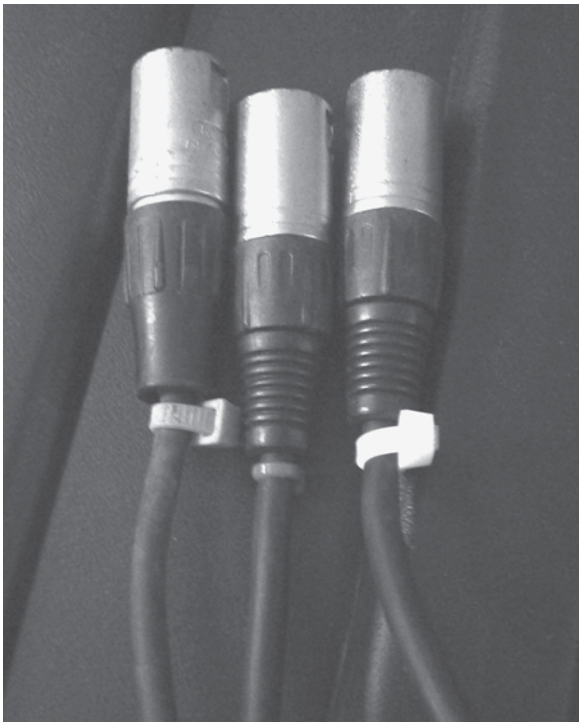


Figure 8.1
An easy way to identify cable lengths is by attaching color-coded tags, which correspond to the different sizes.

Once you have your cables sorted in this manner, hang them on hooks in groups in your utility closet. If you don't have a room dedicated for this purpose, try storing them another way that makes economical sense space-wise. For example, electrical extension cord spools (see Figure 8.2), which can be found at just about any hardware store, work great for keeping XLR cables conveniently and neatly stored without much fuss because XLR cables' ends connect one end to the other (female to male or male to female). The spools can then be hung on a wall near your stage for easy access. With regard to the short 1/4-inch patch cables (balanced or unbalanced) used for patching outboard gear, they should be kept near the mixing console (see Figure 8.3). When it comes to longer 1/4-inch instrument cables, generally musicians will have their own personal instrument cables, but it is a good idea to have a couple on hand for the musician who forgets his or hers one week. Cables such as these and, say, extra speaker cabling that you might have as a backup should be rolled individually and tied with Velcro cable ties to keep them orderly (see Figure 8.4).

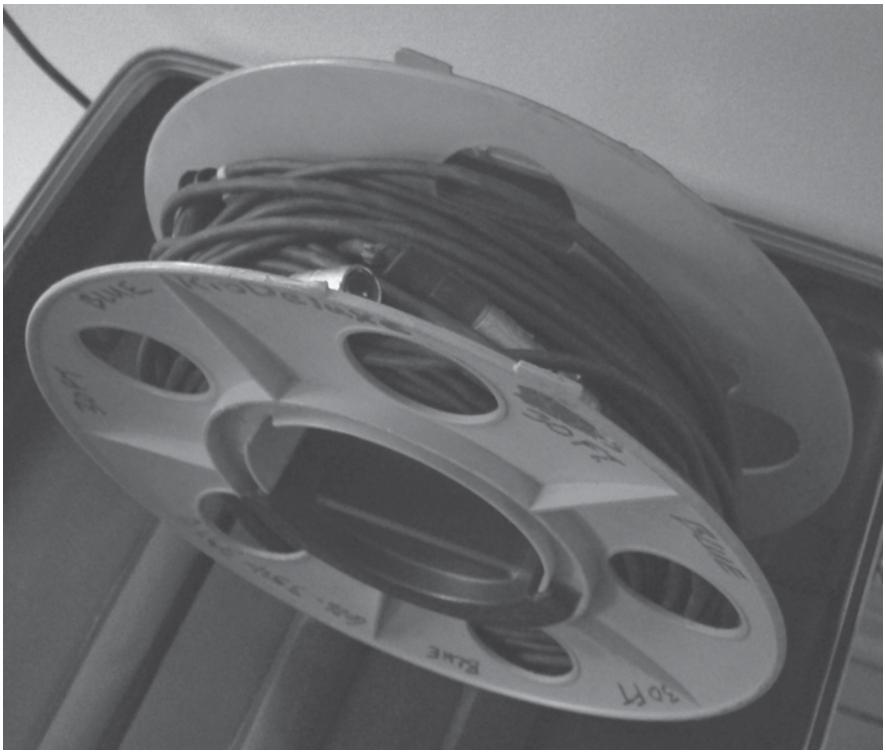


Figure 8.2
A great way to store those long XLR cables is with electrical extension cord spools. This prevents them from becoming tangled, knotted, or lost in some sort cabling mess that would inevitably appear if it were not for solutions such as this.

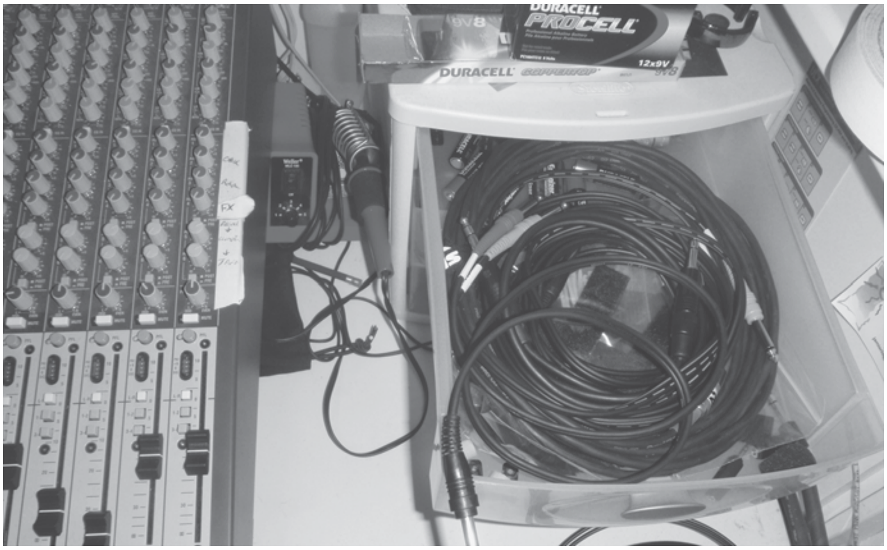


Figure 8.3
Short 1/4-inch phone-jack patch cables should be kept in a small box or compact storage unit near (or right next to) the console for easy access. Note that the packages of batteries are not too far away.

Figure 8.4
Another way to organize cables is to wrap them neatly and use a Velcro cable tie to keep them from unraveling, as seen here.

MICROPHONES AND RELATED ACCESSORIES

As durable as a microphone may seem to feel in your hands, the truth is—they're not. Microphones are often considered indestructible to laymen, and although ome mics are legendary for their sturdy design (the Shure SM-58 is considered the pillar of microphone construction), they need to be cared for and stored properly.

Mikes can be stored in a mic box (sometimes called a mic case) when not in use (see Figure 8.5). This is an especially good move if you have a collection of delicate condenser microphones, which usually carry a higher price tag than dynamic mics. These specifically designed cases have a foam interior with cylindrical holes cut in it so the microphones fit snug inside. This is by far the safest and most sensible way to store your mics as it will protect *and* organize your house of worship's valuable investments against myriad dangers.

Figure 8.5
A smart way to store and/or transport your microphones is by using a microphone box, like the one shown here. Notice the egg-shaped acoustic foam affixed to the top of the case for extra padding on the mic's capsule.

If there is no mic case in the picture in your particular house, an alternative would be leaving your mics on stage. This is certainly less time consuming than putting them away after a service, so if you are volunteering your time, this might seem like an attractive option. But microphones left on their stands on the worship stage have a better chance of being inadvertently knocked over and damaged by whoever may be walking by between services or throughout the week. That is not to say you should remove mics between every service should you have multiple Saturday or Sunday events; it's just best if you don't leave them up all week, waiting for the next service. Another unfortunate negative to leaving your microphones set up is they may not be there the next week when you come back—for example, it may be "borrowed" by someone who forgot to bring it back or just flat-out stolen. Either way, all this can be avoided with the purchase of mic case and a safe place to store it. Just get a case.

As for microphone stands, they can be left on stage between services, but should be folded and stored horizontally to make more room (see Figure 8.6). While it's much easier to store folded mic stands in a utility room, a medium-size closet will suffice. At the very least, choose a logical space on or near the platform for convenient access; you'll be grabbing them constantly.

Figure 8.6
Mic stands can be stored standing upright on stage, in a utility closet, or just folded up and lying down.

Peripheral items, including direct boxes, microphone clips, board tape, Sharpies, and cable adapters, should be by the mixing console. A good idea is to have your house of worship invest in some medium-sized storage cases (see Figure 8.7) for semi-delicate items such as these and keep a case near you at the console at all times. Wireless microphone systems should be stored in such a case so they will not be damaged. While we're on the subject of wireless systems, it's a good idea to keep some batteries close at hand and in that storage case. Besides making sure you have installed fresh batteries in wireless microphone packs before each service, it's still important to have fresh batteries in close proximity of the console or, better yet, in your pocket! There is nothing worse than losing the wireless signal during the sermon. Overall, the smaller the piece of gear, the easier it can be lost or mistakenly picked up. So to prevent loss of any kind, store these items in one sensible place.

Figure 8.7
For smaller items like mic clips, adapters, and/or smaller flashlights, try using small plastic storage cases and be sure to keep them near the console as you will often be grabbing for the items in them.

THE STAGE, DECK, OR PLATFORM

Whatever terminology is used by your house of worship with respect to the focal point for worship—altar, deck, platform, or stage (the latter being the most unlikely, since that term really refers to a place where events for entertainment take place)—it's important you keep it neat and orderly. When placing musical equipment on your platform, you need to juggle between what will sound best to the individual players and the congregation and making sure the equipment is not in the way of your preacher and singers.

One critical aspect to keep in mind is not to have anything on your deck underfoot. This means any worship participants should not be walking over sound-system paraphernalia such as cables or be susceptible to stepping on protruding appendages like mic-stand legs. Microphone cables must be tucked away and any and all excess cable should be coiled (see Figure 8.8). In addition to the safety issues, cables will do best out of sight, without excess weight put upon them from say, heavy guitar or bass amplifiers set on top of them. This goes for mic stands, guitar stands, and cymbal stands as well. The weight of these and other potentially heavy and/or sharp items can and most likely will cause damage to your cables.

Figure 8.8
When on stage, all excess mic cable should be coiled at the foot of the stand to help prevent any chance of stage participants tripping or tangled wire messes forming.

If for no other reason, your primary worship area needs to be clear to enable all service participants to be able to walk freely about the platform. This is especially pertinent if your house has a preacher who likes to move around during the sermon. While I'm on the subject of service participants: If you can help it, do not allow anyone to bring food or drink onto the main platform. While it's unlikely anyone will need to have a cheeseburger close at hand during a service, it's very likely during any sort of rehearsal that participants will be bringing along the dinner they are missing at home. It cannot be stressed enough the damage a spilled bottle of soda can do to your precious audio equipment. Even something as seemingly harmless as bottled water can wreak havoc on a speaker cone, not to mention creating an electrical shock risk.

Nothing a Little Tape Can't Fix

Gaffers tape (see Figure 8.9) is a sound technician's best friend for this one simple reason: It has a strong adhesive side that won't leave a residue on whatever it is attached to. This is a Godsend when you consider the number of items you will need to tape down temporarily and the fact that you can't afford to leave a sticky residue on many of the items you're taping down. If you are not familiar with this type of tape, it generally comes in 2-inch×60-yard rolls and is made of cloth or a cloth/vinyl blend. The tape has a high tensile strength and usually comes in a flat black color. Common uses for gaffers tape include the following:

- Taping down an XLR or instrument's cable to prevent stage participants from tripping
- Holding a mic stand to the platform
- Keeping the corner of the platform carpet from curling up
- Helping hold a drummer's carpet in place
- Putting an X on a bad piece of gear
- Taping a bunch of batteries together for storage
- Keeping the sheet music on the music stand

The fact of the matter is that this tape can do everything but heal the sick. Gaffers tape can be found at many hardware stores, along with its antithesis, duct tape. While duct tape has its place, it's certainly not on your platform.

Figure 8.9
Gaffers tape is the perfect solution when you need to tape something down without having to deal with the sticky residue often associated with using duct tape.

Making Connections

So, now that we have things organized—let's make sure we have things plugged in properly. Every connection within your house of worship sound system contributes to making that system come to life. Up to this point, this book has analyzed the concepts behind those connections to get you, the budding sound technician, off on the right foot in your house of worship. But somewhere along the line, you may have forgotten what goes where in this labyrinth of electricity and need a quick review. The following provides an overview of the signal flow starting from the original transduction of sound into electrical current at the microphone all the way to the reproduction of that captured sound heard at the speaker enclosure (see Figure 8.10). Within this list are simple instructions as to how to make those connections just in case you need them. Since much of this is culled from previous chapters, if you at any time stumble upon a concept that seems a bit hazy, be sure to use the book's table of contents as well as the index to find where that idea was first introduced and get up to speed.

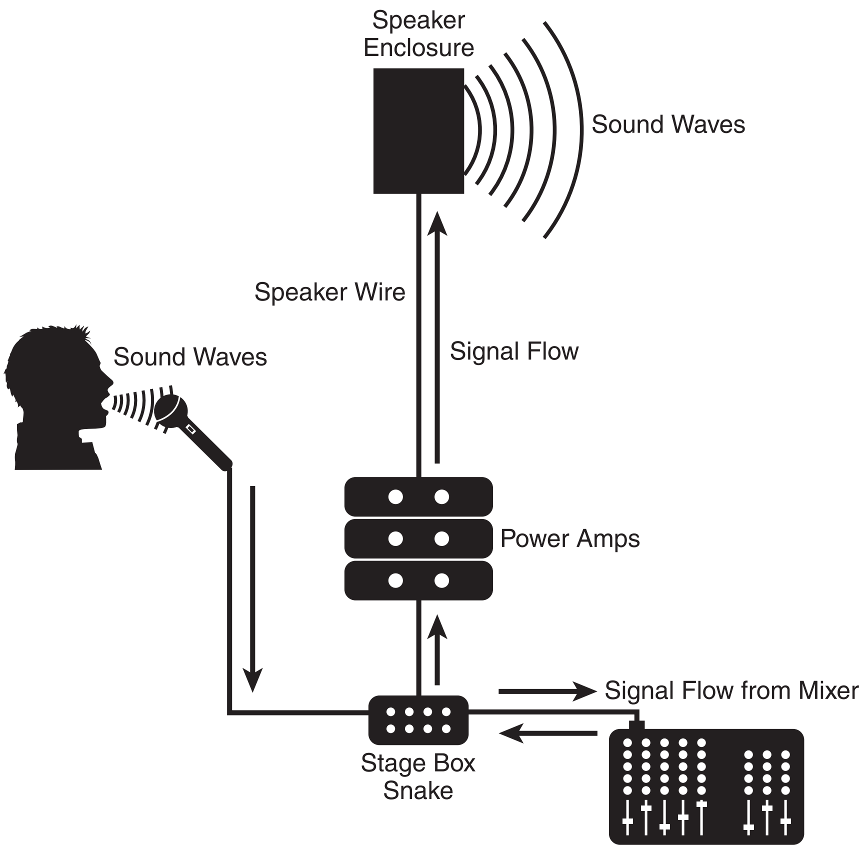


Figure 8.10
From the microphone to the FOH speakers and everything in between.

CABLE FACTS

From the microphone all the way to the speaker enclosures, the transmission of the audio signal is contingent on the fact that the cabling that connects this electronic matrix does its job. So it goes without saying that you need to make sure the cables that are used to make those connections are of the highest grade your house's budget will allow. Two key features to consider when shopping for your cabling is making sure (if you can afford them) they're oxygen-free cables and that the connectors are of high quality. With regard to the former, air over time oxidizes the copper in audio cables, making them ultimately transmit signal poorly. As a result, the more oxygen you can keep out, the better those cables will perform and the longer they will last. Having high-quality connectors ensures high-quality connections. Together, these two factors will help ensure your audio signal is transmitted properly and reliably.

FIRST STEPS

As stated, it all starts at the microphone. The sole purpose of a mic is to capture sound waves that are being produced by whatever source within its polar pattern and transduce the waves into electronic impulses that will travel through your system. That would not be possible if not for the XLR cables that connect the mic to the system. To set up this first stage, follow these steps:

1. Assuming all your mic stands are already set up and have the correct apparatus to secure whatever type of microphones will be placed on them, simply place the mic on the stand.

2. Plug the female end of the XLR cable (the one with the three holes) into the microphone. Don't try to force anything, as there is only one way to make this connection. If necessary, turn the connector end until it fits into the mic's male receptacle. You'll know it's the right fit when the two connect and you hear a small click.

3. Plug the other end (the male end) of the XLR cable directly into your mixing desk or into a stage snake box. If, after you make the connection, there's any leftover cable length, wrap the remaining cable so that the slack rests near the snake box, out of the way of your mic stands and stage participants.

OFF TO THE MIXER

The location of your console determines how this next stage of signal flow is set up. If you're mixing somewhere in close proximity to the platform (most likely the side), you will more often than not plug the XLR microphone cables directly into your mixing console input channels (located either on the rear of larger-scale mixing desks or on the face of the console at the top of each channel). If you're mixing from another location that is a significant distance from the platform, you should employ a snake box. Here's how to make that connection:

1. Strategically place the snake box on your platform out of the sight of the performers and congregation.

2. Plug the male end of the XLR cable into one of the snake's inputs. Be sure to note which mic goes into which input as you do this. For example, snake input 1 will be plugged into channel input 1, snake input 2 into channel input 2, and so on.

3. Plug the individual male ends of the ganged snake cable into the mixing console's corresponding channels.

BLENDING IT ALL TOGETHER

Once the signal is in the mixing console, it can flow only where you send it. Whether it flows through an aux send to a monitor mix, it's assigned to a bus group for group-level control, or it's processed through various pieces of outboard gear is up to you. As a house of worship sound technician, you are truly the master of any audio signal that enters your mixing console. At this point, what you learned about mixing boards in Chapter 3, "Mixers and the Art of Mixing," will be invaluable. (If at any time you feel that overwhelming feeling of confusion a mixing board can surely bring forth, take a deep breath, sit back, and read through Chapter 3 again.) Just as soon as the audio signal enters your mixing console, it wants out. No matter how many incoming signals you've directed into your mixer, they will all eventually exit the console through either a mono out, the main left and right outs, or any aux outs you are using.

IT'S OUTTA HERE

Signal can leave the mixing console in one of two ways:

- Through whatever main out configuration your board features to the FOH speaker enclosures.

- Through the auxiliary send's outputs to feed the stage monitors or to be directed to effects processors that will in turn feed the signal back to the board.

From the main outs, signal may flow directly to the power amplifiers, but it's more likely they will travel through one or more outboard signal processors such as a graphic EQ before being sent to the power amps. When connecting your main outs to their intended destination, use the cables that match that piece of hardware's outputs. For example, if your mixer main outs are balanced XLR jacks, then use XLR cables to connect from the console outputs to the EQ. If your mixer features multiple connection types but your EQ only features 1/4-inch TRS jacks, then TRS cables will have to be used for the connection. From the equalizer(s), the signal will travel back into the snake and return back to the stage box.

As for the auxiliary sends, regardless of how many aux sends your board has, they'll all be exiting through balanced XLR, TRS, or possibly mono 1/4-inch jacks. The audio signal flowing through the aux send outputs will either go to the stage monitors or to an effects processor. With regard to the latter, any signal going to the monitors should pass through an equalizer (preferably a graphic EQ) in order to create a suitable monitoring environment, which includes feedback prevention. In this scenario, the signal exits the board via the aux outputs, enters the equalizer, and then goes back into your snake on its way back to the stage box. From the snake, the signal will flow into the power amplifier(s) designated for the monitors and on through to the monitor wedge via speaker wire. Just like with the powered FOH enclosures, powered monitors will take their audio signal directly from the stage snake box returns, negating the need for a separate power amp. On the other hand, if the signal is sent to an effects processor that is not intended to feed the monitors but rather to help process audio within the console, it will be returned to the mixing console through an aux return or channel input.

POWER TO THE PEOPLE

Arriving back at the stage snake box, the signal will then flow through the appropriate cable to the power amplifier(s). The appropriate cable will be either a male XLR or a TRS. From there, the audio signal will leave the power amp via a speaker wire, enter the FOH speaker enclosures, and pass into the ears of your congregation. Take note: If your house of worship uses powered FOH speaker enclosures, the signal will be sent directly from the snake box return into the powered enclosure using an XLR or possibly a TRS cable.

Now that we have successfully followed the audio signal flow from a microphone to the FOH enclosures and the monitor wedges, our next challenge is determine where to put the worship instruments.

Captain Adapter(s)

What if, while making all these connections, there happens to be a mismatch in cable types? Never fear, just make an adapter appear (see Figure 8.11)! Whether you're transitioning from an XLR to a 1/4-inch TRS or summing a stereo output of some kind to a mono 1/4-inch jack or vice versa, there's an adapter for the job. To save yourself some time, look no further than the adapter masters: Hosa. Most medium-sized music retailers and/or electronic-supply stores such as Radio Shack carry a plethora of Hosa products to suit your needs. Or, just visit their Web site at www.hosatech.com.

Figure 8.11
Adapters such as the ones shown here can help transform just about any audio cable into the format you need whenever a connection-type mismatch occurs. (Images courtesy Hosa Technology, Inc.)

Dealing with On-Stage Instruments

Generally, worship music is presented in three different forms.

- The first and most basic form is a choir or a number of singers accompanied by an organ player or pianist. (While this type of worship music brings back memories of an elderly woman playing a large-scale organ while a small choir sings old classic praise and worship songs, most modern worship houses have replaced the organ and piano with an electronic keyboard to accompany the singers.)

- The second instrumentation group would base its music around acoustic instruments. An electronic keyboard could be present along with an acoustic guitar and other acoustic string instruments. There also might be an electric bass guitar in the ensemble as well.

- Lastly we have the rock 'n' roll version of worship music. A drum kit, electric bass, electric guitar, and keyboards would make up the nucleus of the band, but of course other instruments ranging from acoustic string instruments to brass to an accordion can be added to the group.

In the end, no matter what form your house of worship employs, all these instrument configurations exist for the sole purpose of supporting the singers. Whether you have a cantor, a half dozen singers, or a full choir, these people deliver the songs that the congregation will sing. That said, you have to place your singers and your instrumentation on the stage and in the mix so the singers are clearly supported and not overwhelmed by the accompaniment.

PLACEMENT

While there is obviously more than one way to set up a worship band, it's safe to say most worship groups will place their instruments in relatively the same way. For the most part, it's standard to place the instrument amplifiers around the drum kit, much like a rock band would (see Figure 8.12). Of course, the size of the platform has a lot to do with actual placement, but the location relationship (not distance) between the instruments stays basically the same. The drum kit will be centered toward the back of the stage and possibly on a riser. The electric bass amplifier will be positioned on one side of the drums—most bassists like their amp to be on whatever side the drummer's hi-hat is on—and the electric guitar amp on the opposite side.

Figure 8.12
More often than not, your worship band will set up much like a typical rock band would—drum kit in the center and instrument amplifiers on each side with the bass amp most likely on the hi-hat side. Notice the personal headphone mixers attached to the mic stands.

It has become increasingly common to place Plexiglas shields in front of the drum kit and sometimes the instrument amplifiers (see Figure 8.13). These shields dramatically reduce the decibel levels of the instruments and/or amplifiers they stand in front of, giving you, the sound technician, more control over the overall decibel level in your worship space.

Figure 8.13
When placing band instruments and amps on stage, make sure to leave enough room for Plexiglas shields like the ones seen here if your house of worship has them. These are an easy way to control high decibel levels and work especially well on drums and loud guitar amps.

If there are to be electronic keyboards in the band, they are usually set up more stage forward and off on one side in relation to the drum kit. The premise behind this setup is that the rhythms coming from the drums will be in the center of the projected music while the bass and guitar will be separated equally. As for the keyboard signal, it may be sent to an amplifier, but most likely it will be sent though the house speaker enclosures by way of a direct box connecting the instrument to the mixing console. Take note: Whether the guitar, bass, and drums are individually miked and amplified by the house sound system will depend on the size of the worship room and the desired volume of the instruments.

ELECTRICAL CONNECTIONS

Once you have your drum kit, keyboards, and instrument amplifiers set up where you want them on your stage, you have to plug them into an electrical source. As a general rule, the older a house of worship is, the more problematic the electrical will be. If you're lucky, you may be volunteering at a recently constructed worship house or your stage electrical system has been updated recently. Whatever your individual case is, there needs to be enough power to make your sound system and your electric instruments operate properly and safely. The next chapter addresses proper practices with regard to worship house electricity; in the meantime, here are a few facts to be aware of that may save someone from experiencing a shocking event while on your stage.

- For starters, properly grounded outlets are a *must* on your stage (see Figure 8.14). Improper grounding can cause noise such as hums and buzzes. More importantly, they can be extremely dangerous. A note of caution: The authors of this book implore you to not use a ground lift (see Figure 8.15) in an attempt to eliminate any hum or buzz problems. Contrary to what some people who have engaged in such a remedy may tell you, it is not wise and can lead to your musicians, singers, and possibly your preacher receiving a significant electrical shock.

- If you need to provide power in various places on the platform where there are no outlets nearby, consider using a stage electrical box (see Figure 8.16). Not to be confused with a stage snake box, a stage electrical box, alternatively known as a stage drop box, is basically an electrical power strip consisting of four or more outlets that's connected to a heavy-duty cable that plugs into a wall outlet. Be prepared: Many musicians will bring consumer-grade power strips and extension cords to port power from the wall outlet to their amplifiers and accessories. While these items may seemingly do the job, a good drop box will incorporate filters to improve the quality of the electric power and include a circuit breaker to protect the equipment plugged into it. The aforementioned filters remove electromagnetic interference (EMI) and radio frequency interference (RFI), which helps ensure the integrity of the electricity flowing through the drop box to your sound system and electric instruments.

Figure 8.14
You can rest a little easier when you see standard three-prong electrical outlets in your house of worship, as these are grounded and will help prevent the possibility of electrical shock.

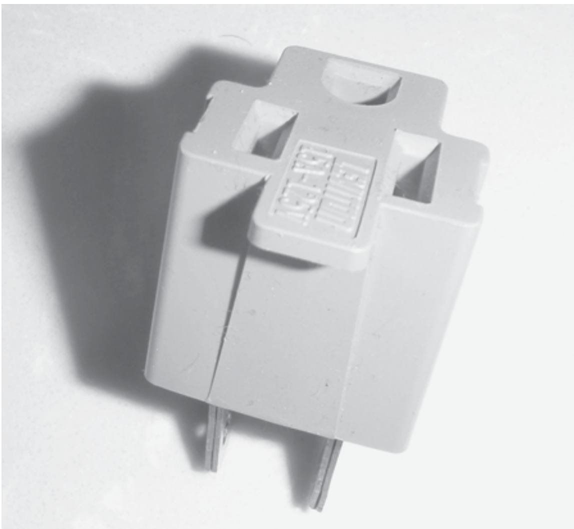


Figure 8.15
This three-hole female end to two-prong male end ground lift should never be used to try to eliminate noise on your system because it defeats the grounding in your system and leaves anyone near the apparatus plugged into it susceptible to shock.

Figure 8.16
If you need to provide outlets on your platform, use an electrical drop box like the one shown here. Not to be mistaken for a consumer-level power strip, these pro-grade extensions feature a circuit breaker, EMI and RFI filters, and heavy-duty cables, ensuring the best possible current transfer.

DIRECT BOXES AND LINE OUTS

It has been stated throughout this book that the microphone represents the beginning of the signal path, but remember: It's not the only device that initially captures the audio signal that eventually finds its way to the FOH speaker enclosures. In addition to microphones, direct boxes (see Chapter 2, "Microphones") are used in instances where it's either easier not to mic an instrument or, in some cases, a mic may not be necessary. Direct boxes, also known as DIs (short for direct input), can be used to receive an audio signal from an acoustic instrument with a pre-amp, an electronic keyboard, a synthesizer, a sequencer, or a computer via a 1/4-inch cable, which it then transforms into a balanced XLR output that is compatible with a snake box and/or the mixing console.

Another way to route audio to the mixing console sans microphone is by way of line outs. A line output—also known as a direct out—is an onboard XLR output that can deliver a direct audio signal to your snake box via an XLR cable. These connection ports are found on many modern bass amplifiers, guitar amps, and keyboard amps (see Figure 8.17) and may also feature an output level control. Some instrument amplifiers have 1/4-inch line outs as well and must be plugged into a DI box in order to have an XLR output available for your snake box.

Figure 8.17
Many modern instrument amplifiers feature a direct out in either XLR (seen here) or 1/4-inch configurations, eliminating the need for miking. This is most common in bass and keyboard amps, as these instruments are regularly patched into a DI so they may be plugged directly into a mixing console.

Singers and Preachers

Once you have the band instruments placed on your worship house platform and have wired everything up, thus creating the tangle of cables that makes sense only to you, it's time to consider where to position the singers and the preacher. As mentioned earlier, the purpose of all instrumental music is to support the singers. Where the singers are positioned on the platform relative to the band and preacher will play a role in achieving a fulfilling worship experience for all. As for the preacher, he or she will most likely speak with little to no accompanying sound, but his or her positioning will surely influence where the singers will be placed on stage. That all being said, you can see how each component of the worship team's positioning must be carefully considered, as one influences the other.

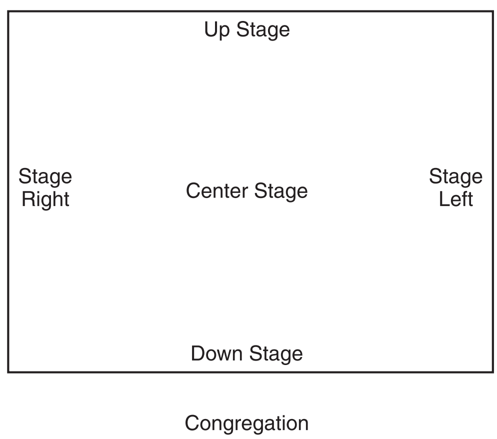
PLACEMENT

In general, it's important to have your performers' positions balanced on the platform. These locations, of course, will depend on the size of your stage and the size of your singing group or choir. Your preacher will most likely have a definite idea of where he or she wants to be positioned. If your preacher uses a podium (as many do), the podium will be the deciding factor as to what side of the stage the singers are on. For example, if the podium is center stage, then the singers/choir can be placed at stage left or right. Should the podium be set at stage right or stage left, the singers will be set up on the opposite side of the platform. In the end, there is no real standard as to where you should place your worship singers and your preacher, so you'll need to be prepared to make do according to your own individual circumstances.

Left is Right, Right?

All stage terms come from the vantage point of the person or persons on the platform (see Figure 8.18). If you're standing on the stage looking out at the worship space where the congregation would be, the area on *your* right side is known as *stage right* and the area on your left is *stage left*. The front of the stage is called *down stage* and the back is known as *up stage*. From the position of the mixing console looking out the stage, your vantage point—as well as the congregation's—is reversed. When communicating with people on the platform, you need to consider *their* vantage point and refer to directions as such. In other words your left side is stage right and your right side is stage left. Of course *center stage* will be in the middle of the platform, no matter where your vantage point is. Understanding these directions gives you and the people on stage a common line of communication wherever you are so there's no confusion.

Figure 8.18
It's standard protocol to refer to stage directions from the vantage point of the stage participants.



RISERS

Many houses of worship position the singers/choir on either a level or a tiered riser (see Figure 8.19). Prior to the advent of modern sound systems, singers were placed on a riser to help their voices project farther into the congregation. Positioning them on a tiered riser allowed one row to sing over the heads of the row in front of them so that all the participants were heard equally. Today, singers frequently use microphones and the power and versatility of a house system to amplify their voices. Although this may seem as if this would negate the need for risers, many worship houses still use them—and for good reason. Not only is there something to be said for the visual appeal of a choir standing on a tiered riser, if some singers are singing directly into the backs of other singers, much of their sound will be absorbed before making it to the mics. In this case, as in many others, think of your sound system as a means of sound reinforcement, a tool for enhancing the natural beauty of what is being created acoustically.

Figure 8.19
Previous to modern sound systems, the only way to ensure every member of the choir was heard equally was to use tiered risers.

PODIUMS

Any discussion that involves the preacher must include mention of the podium (see Figure 8.20) from which he or she preaches. Also known as pulpits, podiums serve an important purpose in a worship house. They have multiple uses, such as the following:

- They are used by the preacher as a focal point for the sermon.
- They are used to issue the announcements that are inevitably made every service.
- They also can be used as a position for prayer.

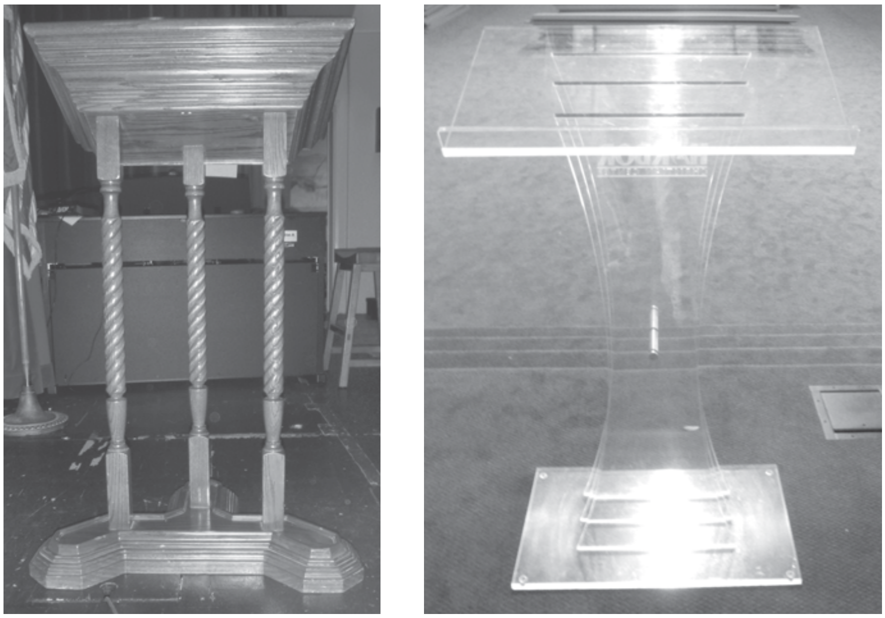


Figure 8.20
It's to your advantage that the podium be portable and not permanently attached to the floor. Note that in modern houses of worship, you may see a Plexiglas podium.

Podiums will either be permanently installed on the stage, or you will be able to move them around to suit your stage plot. Many podiums have a a short microphone stand mounted on them, saving you precious setup time (although with the increasing use of wireless lavaliere mics and, even more, headset mics, many preachers have no use for a microphone stand, so be prepared to mute the podium mic often).

As with the risers, pulpits are a combination of practicality and taste. In other words, they are not something that has to be there. In fact, some preachers prefer not to use a podium at all. Many preachers move from one side of the stage to another while delivering their message, and a podium just gets in the way of their delivery and animation. Once again, the presence of a pulpit means little to your room mix, but will influence placement of the worship team. You need to be concerned with the placement of your mixing console much more than whether or where the podiums and risers are on your stage.

Console Placement

A good sound technician's ears will learn to hear and, in some cases, adjust to their worship sound system from wherever the mixing board is located. Put another way, if your console is not in the acoustically preferred part of the worship space, you'll still be able to mix the music and vocals effectively. This is a good thing, because relocating a mixing console is not an easy task. But if you're lucky, you may find yourself in a scenario where you can indeed move the console—for instance, if your house of worship is planning to purchase a new console and you have the go-ahead to redesign the architecture of your house system. Since it's difficult to make suggestions as to where to place a mixing console without actually being in the worship space in question, let's start off with where not to place the board. From there, you'll look at some real-world realities related to console placement.

THE YOU-PROBABLY-DON'T-WANT-TO-DO-THAT LIST

Assuming you will be mixing from FOH, your mixing console must not:

- Be in the way of the congregation, either visually or physically impeding pedestrian traffic flow in and out of the worship space.
- Be so close to the stage that you can't get an accurate audio perspective of the house sound system.
- Be in a position where the audio snake is in the way of the congregation because of where the console is located.

It's imperative that these locations be avoided; otherwise, the mixing process will be a needlessly arduous task. Some other places to be wary of are as follows:

- **Under the balcony (see Figure 8.21).** This location can be a difficult place to mix from, especially if your back is against a hard wall. From this point of audio reference, it's impossible to determine what frequencies will be trapped in the area under the balcony. In addition, there will ultimately be some reflection of said frequencies off the wall behind you, which will muddy up and confuse what you hear.
- **Within the balcony area (see Figure 8.22).** Mixing your services from the balcony area can bring its own problems, such as not being able to effectively hear low frequencies as well as if you were mixing from the floor of the worship space. The exact frequencies and how dramatic the loss will vary from house to house. If you mix from the balcony area already, be certain to

make frequent visits down to the floor of your worship space during a service. This will give your ears a reference, enabling you to compare the audio signal from both places so you can make the proper adjustments when needed.

- **Enclosed areas.** In rare cases, the mixing console may be located in an enclosed room with either a window or video screen from which you can observe the stage (see Figure 8.23). If this describes your existing situation, you should have a pair of monitors in the room with you in order to mix the service properly. To make the best of this unique situation, listen and compare the worship space speaker enclosures and your personal room monitors and then adjust your monitor's tonality as closely as possible to the house sound system. While the two speaker systems will never sound identical, you can adjust your reference monitors to the point where the relationship between the two systems will allow you to mix accurately.

Figure 8.21
Notice the solid wall right behind the console. This makes this already-difficult mixing location under the balcony area even more challenging, as soundwaves coming from the FOH enclosures will reflect off the wall and blur your perception.

Figure 8.22
While it may seem like a good idea to mix from the balcony due to the bird's eye view, be aware that your low-frequency perception will be challenged and you'll need to make frequent visits to the ground floor for more accurate assessments.

Figure 8.23
If you find yourself in the not-so-common position of having your mixing console in an enclosed area, be sure to have a monitor handy and try your best to tweak its output to match the FOH enclosures.

REALITY CHECK

The reality is this: More often than not, your mixing console will reside on the left or right side of the worship room, so you need to be prepared to deal with this. Whether you are mixing in mono or stereo, a left or right console position should be fine, as your ears will ultimately adjust to whatever location you are mixing from. One thing to keep in mind is if you are close to the back wall of your worship space, it's a good idea to treat that wall with the proper acoustic material. Anything from curtains to sound panels to acoustic foam can be used to prevent sound waves from the FOH enclosures from reflecting off the wall back into your ears.

With that in mind, once you have your singers and worship band mixed well for the congregation, you should make it a habit to walk around the room (if you are allowed) during the service and listen carefully. As you do this, you may discover locations within your worship space that this book has deemed sweet spots. Try not to be too enamored by these spots. They are unique unto themselves. Moving just a few feet away from them will bring another reality to light: They're few and far between. That's why it's not a good idea to consider moving your console to one of those spots. Besides, if your sweet spot is located in row 5 of the center aisle or row 11, seats 1 and 2, your house manager is not likely to listen to your pleas to rip those seats out so you can plant a mixing console there. One thing you do have in your corner is the fact you may be able to change your speaker enclosure locations to better accommodate your console's location while at the same time widening or even multiplying those sweet spots and ultimately improving the sound or your house.

Speaker Placement

Your speaker enclosures will most likely be mounted from the wall or ceiling, stacked on the left and right sides of your stage, or they may be clustered above the stage (see Figures 8.24–8.26). Regardless of how and where they are situated, the placement of FOH speakers in your worship space is considered to be one of the most important factors in the overall sound quality and effectiveness of your system. Poor decisions in speaker enclosure placement can cause inadequate dispersion of sound and invite feedback. These remaining sections will discuss several concepts you should be aware of when considering speaker placement.

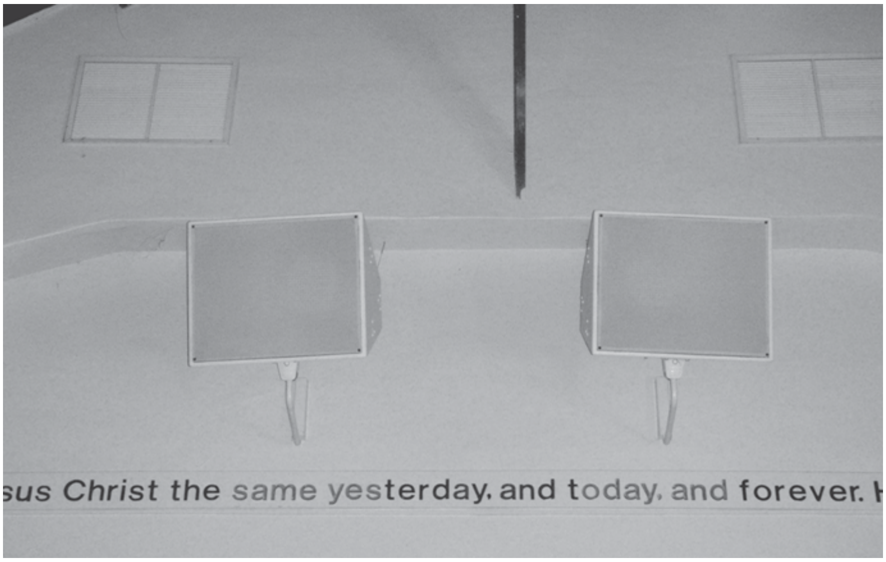


Figure 8.24
It's not uncommon to see enclosures mounted above the stage in pairs. Take notice of the angle at which both of these enclosures have been placed.

Figure 8.25
Elevated enclosures will often be mounted in corners such as this one. Note the chain attached to the top of the cabinet for extra security.

Figure 8.26
Enclosures can be stacked on top of one another and placed at the side of the staging area. Note the massive sub-woofer on the bottom of the pile.

GENERAL GUIDELINES

Although every worship house is unique, there are several guidelines that govern the placement of all speaker enclosures. These include the following:

- Never place your speaker boxes behind the microphones. Sound waves emanating from the enclosures will be picked up by the microphones and, as a result, will potentially produce massive feedback.
- Be certain to raise your enclosures so that the high-frequency horns are above the congregation's heads while they are standing.
- Never point the speaker enclosures directly at the vertical back wall of your worship space because the reflecting sound can cause standing waves.

Perhaps the most significant dividing line when it comes to placement of your enclosures is whether the speaker cabinets will be stacked or flown. (Flown or stacked speaker enclosures can be pointed at the congregation for better audio coverage.)

- **Flown.** This term refers to mounting or hanging the speaker boxes from the wall or ceiling. Enclosures that are designed to be flown will have fly points (see Figure 8.27) incorporated in their construction. These fly points are hooks that attach to the actual enclosure, where other hooks, bolts, or brackets can be attached for the purpose of suspending the enclosure.

- **Stacked.** This is where cabinets are placed on either side of the stage or adjacent floor. They may be single cabinets or ones stacked on top of each other. Sometimes a full-range enclosure will be stacked on a subwoofer. Overall, the term refers to any speaker enclosure or combination of enclosures that are not flown.

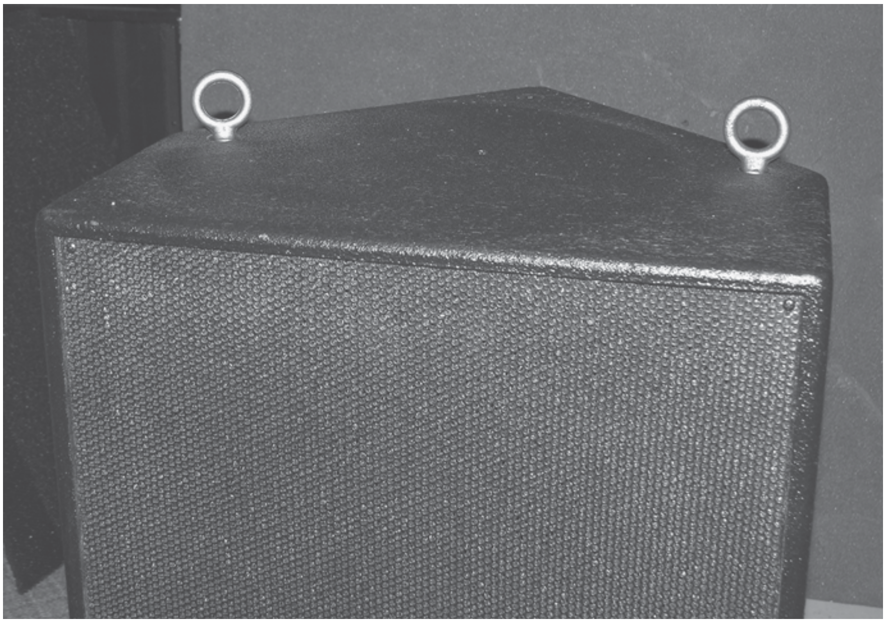


Figure 8.27
Fly points are installed during the construction of enclosures that are intended to be flown. (Image courtesy ISP Technologies.)

AIMING SPEAKERS

When it comes to placement, aiming stacked speakers tends to be easier than pointing flown boxes. This is mainly because the stacked enclosures are on the floor or placed on the stage and not permanently installed in their location. Flown cabinets, in contrast, are permanently secured and sometimes can be quite a distance above the floor or stage, making it difficult to easily get to them for placement adjustments.

In the end, however your enclosures are placed in your worship house, the aiming process will be the same. A laser (see Figure 8.28) is an excellent tool for determining where exactly your speaker enclosures will be directed. Following is a list of steps you can follow while using a laser to adjust the enclosure's position.

1. Place the tool on the top, side, or bottom of a speaker cabinet in order to see where the cabinet is pointed.

2. When positioning the enclosure, aim the enclosure so it projects into the congregation starting at the first row of seats and covering as many people as possible. (Keep in mind that aiming your speaker enclosures just above the heads of the first row when standing will help produce ample dispersion over the entire congregation.)

3. Since every room and every sound system are different, be prepared to experiment.

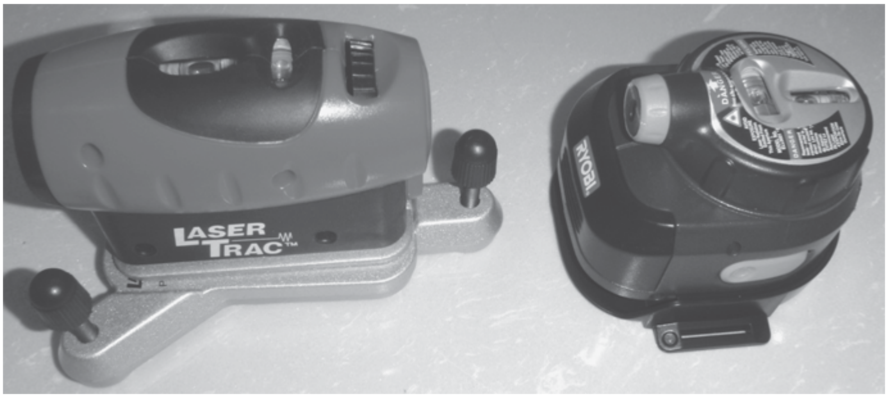


Figure 8.28
Using a laser can help immensely when determining where to aim speaker enclosures for optimal sound dispersion.

The typical horn in a speaker enclosure has a horizontal coverage of 90º and a vertical coverage of 40º. If you know how to use a protractor (or can get a volunteer in the congregation to help out), use it with your laser to figure out how much seating coverage your enclosures will have. Understanding the sound projection of your speakers will give you an idea of what areas of your worship space are getting the best audio coverage. This may mean checking the specs that came with your enclosures or going online and doing some research on how your enclosures disperse sound. As mentioned earlier, there will always be a sweet spot (or two) in your worship space, but properly aimed speakers can make your entire room sound good. Once you have aimed your cabinets correctly, don't forget the subwoofer (if your house system has one).

SUBWOOFERS

Low-frequency enclosures are less directional than two-way and three-way enclosures, but the congregation can still hear their area of orientation nonetheless. So be sure to place them where their sound will evenly travel around the worship space. Whenever possible, subwoofers need to be placed on the floor in front of or under the stage (see Figure 8.29) to sound their best and to prevent noise issues; they can cause the mic stands to vibrate if placed on a resonant stage. Due to their excessive size and weight, subs must reside on a flat surface at all times and should not be flown (except in rare cases not discussed in this book). Also, just like FOH speakers, subs should not be placed behind your microphones so as to avoid any feedback issues (although placing your sub[s] in close proximity to the FOH speakers will help make them sound more natural).

Figure 8.29
Subwoofer placement can vary depending on space constraints. Sometimes, subs may be placed underneath the stage, as shown here.

Having read this chapter, hopefully you are starting to in fact put it all together. From miking the worship band, singers, and preacher to operating all the parameters of your mixing console to tweaking the outboard gear—you are the hero in the background that makes your congregation's worship service a success. You should be good to go…

…until something breaks. Now what? That's what Chapter 9, "Healing the Sick and the Tired…Gear, That Is," is for.

CHAPTER 9

Healing the Sick and the Tired…Gear, That Is

By reading the first eight chapters of this book, you've become a functional sound technician for your house of worship. From setting up to working the board during the service to breaking things down afterward, you have it all covered—until something goes wrong.

With this complex matrix of equipment, it should come as no surprise to you that there's a good chance that at some point, something will go wrong. Apart from this, your system likely has certain weak spots that need to be regularly maintained and at times repaired. By taking care of your system and performing some simple repairs yourself after troubleshooting as needed, you will be doing even more good for your house of worship, as service and repairs for audio equipment can be costly—not to mention extremely inconvenient when you consider the time the gear is out for repair and not in your house, working for the congregation.

This chapter not only serves as a guide to performing simple repairs on various components of your sound system, but also as a preliminary road map for developing a maintenance schedule to prevent problems *before* they happen. In addition, you'll learn about common electrical properties such as grounding. So dive in and get ready to learn some practical ways to keep your sound system healthy as well as explore some advanced techniques and tools for diagnosing and repairing problems.

Tools of the Trade

More often than not, you can be successful at troubleshooting if you have the right diagnostic tools. The tools examined in this section are specific to the health of your audio gear. The tools described here are the must-haves, and are specific to troubleshooting sound systems. That's not to say more generic tools such as a pocketknife or flashlight are not useful. In fact, you'd be doing yourself a disservice by not having those in your audio toolbox as well.

OUTLET TESTER

Let's begin with electricity. Although it's not a tool, it is the fuel that runs your audio equipment. Therefore, it's of paramount importance that you check the health of your electrical outlets, especially if you're mixing in a new room and bringing in your own gear to do so. One of the simplest tools is the electrical outlet tester (see Figure 9.1), which is a device that indicates whether an outlet is energized (that is, if electricity is present), whether it is grounded (the importance of all your electrical outlets being grounded cannot be overstated; this is covered in more depth later in this chapter), and whether the hot and common wires are reversed. This tool will be by far the least expensive in your arsenal, costing you or your worship house only a few dollars.

Figure 9.1
Saying "Big things come in small packages" is an understatement when referring to an outlet tester. With this compact plug-in, you can test an outlet to make sure there's electricity present, that it's grounded properly, and more.

As for using one, it couldn't be simpler. Just plug the outlet tester into a wall socket and see which lights are glowing. The combination that illuminates indicates the condition of the outlet, as per the label.

CABLE TESTER

Your system's audio cabling and speaker wire should be the first place you look when troubleshooting. Just think of the sheer enormity of cabling within your system and, within that, the number of cable types and cable ends—something is bound to go wrong! This is where a cable tester (see Figure 9.2) comes in handy. This indispensable tool can range from \$30–\$150 and is worth every penny. Why? Because cable testers can check the condition of all the common cable types used in a sound system, including but not limited to the following:

- XLR
- Speakon (speakers)
- $^1/_4$-inch TRS and TS
- RCA
- CAT5 (Ethernet)
- MIDI
- Mini plug (CD, iPod)
- Banana plug (speaker)

Figure 9.2
In addition to checking myriad cable types, cable testers can check batteries and provide test tones at varying frequencies, making this tool indispensable.

In addition to using the cable tester to check cables, you can use it to check batteries. Some testers will even produce 220Hz, 440Hz, 1kHz, 4kHz, and other audio tones. Having a tool that can generate an audio tone will enable you to test all the equipment from your mixer to your speaker enclosures when a test signal is necessary. This test signal flows through your mixing console, creating a continuous path of audio to your FOH speaker enclosures and your stage monitors.

SOLDERING IRON

You can fix just about any non-operating cable, provided you have a soldering iron and solder wire (the last section of this chapter explores soldering in depth). A pencil-type soldering iron (see Figure 9.3) is preferable to a soldering gun, as it lends itself to the precision required to repair audio cable and related equipment. When purchasing your soldering iron, pay close attention to the wattage and temperature of your iron. Look for a 25- to 40-watt soldering iron that will produce 500 to 800° of heat. At those temperatures, you'll be able to successfully perform all your audio cable and electronic repair. Soldering irons can be as inexpensive as $10 but can go as high as $50. That said, when it comes to cost, don't be frugal—whatever you solder must be done right, and it takes a quality iron to ensure that happens.

Figure 9.3
A pencil-type soldering gun (top) and/or iron (bottom) is the way to go for the type of precise repairs you will regularly perform as a sound technician.

DUST-REMOVAL TOOLS

The next three tools are designed to remove a sound technician's worst enemy: dust. In the audio world, dust really is our nemesis—it gets into faders, knobs, circuits and other sensitive electronic areas and can cause myriad problems. Dust buildup in gear can become electrostatic (as in static electricity), which can wreak havoc on the operation of your mixing console and outboard equipment. This buildup could make knobs and faders stick and even lose connectivity, ultimately degrading or interfering with the audio signal flow. The following tools of the trade can help alleviate and hopefully eliminate any dust infiltration.

- **Vacuum.** There are many small vacuum cleaners specifically designed for cleaning electronic gear (see Figure 9.4). Although these compact vacuums can remove a great deal of dust from your gear, so too can many consumer-grade units—as long as they have enough power to remove the dust particles from the many hard-to-get-to nooks and crannies of your mixing console and outboard gear. By the same token, make sure your vacuum cleaner is not too powerful, as you don't want to suck the knobs off your board! These types of vacuums should run between $20 to $40; if it costs more than the high range noted here, consider it an indicator that it's too powerful.

- **Paintbrush.** A small or medium-sized paintbrush (see Figure 9.5) can sweep dust and other offending particles from your mixing console and keep them from getting into the more sensitive areas of your board. Paintbrushes are a great solution if you to take swipe during a service without making any noise. A small paintbrush suitable for the job should typically cost less than $5.

- **Compressed air.** The pressure produced by a can of compressed air (see Figure 9.6) can loosen dust or debris from your equipment, reaching spaces a vacuum or brush can't. This type of air can also come in an antistatic form, enabling you to blow away static electricity along with the dust. Canned compressed air is a handy, quick, and easy tool that normally cost from $5 to $10. (Take note: Do not use compressed air on microphones for any reason. In fact, when it comes to mics in general, it's best to have the manufacturer of a certified repairperson do whatever is needed.)

Figure 9.4
Be sure to obtain a vacuum cleaner designed for use with electronic gear, as they are strong enough to remove dust particles from tight places without sucking the knobs off your mixing console.

Figure 9.5
Short-handled, medium-sized paintbrushes are great for brushing dust particles away from sensitive areas on your mixing console, especially during a service when you need to quiet.

Figure 9.6
For those places that no rag, vacuum, or brush can reach, a can of compressed air can be used to eliminate not only dust particles, but static electricity too!

SPRAYS

A worthy alternative to the aforementioned dust-removal tools are liquid-based cleaning products. Spray-type cleaners (see Figure 9.7) are preferable to bottled or canned solvents because they can be easily administered right on the spot, even during a worship service. Many problems, such as a sluggish knob, crackling, static, and audio loss in a fader or a non-operating button can be remedied by a simple shot of spray cleaner. There are a variety of spray cleaners on the market; make sure you purchase ones that are safe for plastics, incorporate high-purity solvents, and contain a lubricant that restores electrical continuity such as NU-TROL from MG Chemicals, which can be purchased at most electronic-supply stores and some hardware stores for about $8. Another popular brand to keep an eye out for is Deoxit. A good cleaner will cost between $5 and $10 per can and can help overcome many potential audio problems.

Figure 9.7
When it comes to liquid-based cleaners, a spray-can type is preferred because you can more easily apply the cleaning agent in hard–to-reach areas.

MULTI-METER

A multi-meter or multi-tester (see Figure 9.8) is a tool used to diagnose electrical problems. It can be used to test electrical properties such as resistance and continuity and to measure voltage. A few of your other tools may perform some of the same or similar jobs as a multi-tester, but it's an important tool to put in your kit nonetheless, as it also performs many unique functions. For instance, while your cable tester can test continuity (the level at which a circuit is closed or continuous), it cannot test a circuit or switch like a multi-tester can.

Figure 9.8
Although some tools in your kit may perform a few of the same functions as a multi-tester, it's important to have a the real thing, as they perform many unique functions that you will surely need in diagnosing your system.

A multi-tester can take yet another tool's function a step farther: an electrical outlet tester. As you know, an electrical outlet tester can determine whether electricity is present at an electrical outlet and if it's grounded; a multi-meter can also determine this, as well as measure how many volts are available at that outlet. To add to the case for the acquisition of a multi-tester, you can use one to determine whether a piece of gear has a faulty ground, also known as a ground fault. Faulty grounds can cause an electrical shock—something you need to avoid at all costs.

The only negative aspect of multi-meters is they tend to be used by people with a greater understanding of electricity than the average person. While this may make tools like outlet testers and cable testers more attractive to a budding sound technician (as many of you reading this might be), don't let that sway you from adding a multi-meter to your toolbox. The day you need one, you'll be glad you did.

Just in case you forgot to make a shopping list as you read through these descriptions, you can do it here:

- Outlet tester
- Cable tester
- Soldering iron
- Vacuum cleaner

- Paint brush
- Can of compressed air
- Spray cleaner
- Multi-meter

While you're at it, add the following tools to your list. Many will be mentioned again in the sections that describe how to solder.

- Micro-screwdriver set
- Utility knife
- Wire stripper
- Needle-nose pliers
- Shears
- Rubber mallet
- Battery-powered hand drill
- Drill-bit set

Don't forget the most important tool of all: a toolbox! Now that you have the tools, let's learn a little about the trade.

Troubleshooting

Take this scenario: You're at your house of worship before a day of services and you're going through your normal, everyday setup procedures. You turn on the sound system and everything lights up as usual. Next, you walk on stage to test the lead vocalist's microphone by speaking some arbitrary words into the mic ("Test, test, one, two," etc.), only to discover you're hearing the un-amplified you, not the sound system!

Perhaps the biggest nightmare for any sound technician is turning on a piece of equipment in your system before a day filled with events in your house only to find it doesn't work. Even worse is when it's a piece, like a power amp, that freezes your entire system. If it hasn't happened to you yet, consider yourself lucky—but at the same time up on deck. It *will* happen.

If and when you find yourself in this situation, keep a level head. It's critical that you not panic. Then get ready to engage in an age-old practice: troubleshooting. The art of troubleshooting is simple in concept: Go through your system testing various components until you have isolated the source of the problem through the process of elimination. Once you've successfully identified the problem, you can then take the steps necessary to fix it.

The process of troubleshooting in the audio arena is all about identifying where in the signal path the audio signal has deteriorated, stopped, or failed completely. Specifically, you'll take the following general steps when troubleshooting:

1. Perform a microphone-to-mixer signal check.
2. Perform a mixer check.
3. Perform a post-mixer-to-speaker-signal check

MICROPHONE-TO-MIXER SIGNAL CHECK

If faced with a scenario like the one outlined here, your first step is to check the signal flowing from the microphone to the mixer. Here's how:

1. Go back to the mixing console and check the input channel for the lead vocalist's mic. Be certain the channel is on, the fader is up, phantom power is engaged (this applies to condenser microphones only), and auxiliary sends are set to the proper level.
2. Go back to the stage and check that the microphone is securely plugged into the XLR cable.
3. Disconnect the XLR cable and test it with a cable tester or multi-meter.
4. If there is no signal, replace the XLR cable.
5. Change the microphone. (While it's rare for a microphone to fail, it does happen.)
6. Make sure the stage snake box and mic input channel on the mixing board match.
7. Plug the lead vocalist's mic cable into another stage snake channel and make sure it's plugged into the corresponding channel on the console you're now working with.

After going through these steps, you might have to consider that the snake is faulty. To diagnose the snake, you'll need a cable tester or multi-meter. It's rare for a snake to have a catastrophic failure, however. You should focus on finding what channel has failed. If you have unused channels in your snake, switch to one that you know is functional to buy some time until you can repair or replace the snake. If there are no unused channels, run a single XLR cable from the platform to the mixer, thereby bypassing the bad snake channel. If the snake passes the test, go to the next section and look to the mixer. Before you do that, though, check to see if you turned on the power amps. If you forgot, you're not the first!

MIXER CHECK

If your troubleshooting quest has led you to consider the mixer, again don't panic. Just follow these steps:

1. Plug a cable tester with the ability to produce an audio signal, a known-good microphone (that is, a mic you know works), or a CD player/iPod directly into an input channel in your mixer.

2. Turn the channel on (if applicable to your board; some boards do not have buttons for this purpose).

3. Turn up the gain knob.

4. Bring up the fader.

5. Make some noise! Activate the signal from the cable tester, speak into the mic, or turn on whatever playback device you've patched in.

6. Look to the PFL/Solo meters to see if there's any signal present.

Are the input lights or meters lit? If so, you know the signal is coming into the mixing board. (If your board is not equipped with input signal indicator lights, look at the master section of the mixer; no board worth its salt will lack some sort of illuminated indicator for signal strength at the master section.) If there are no lights glowing in the locations they should be, you know it's the board. Don't fret; it's not uncommon for a channel strip to fail in a mixing console. Check for a good channel. Some common causes of failure are dust or some mistakenly spilt liquid (e.g., coffee, soda, bottled water, or what have you).

In extreme instances, you may turn on your mixing console to find it does not pass any audio at all. If so, let your preacher know immediately and suggest that the service be performed without the help of a sound system. (This would be an extreme case, but remember: Up until just a few decades ago, all services were performed without an audio system.) At your first opportunity, contact the console's manufacturer if it's still under warranty to start the process of having the board serviced. If it's out of warranty, look to the manufacturer to recommend a qualified repair facility or, better yet, a repair technician who can come on site.

POST-MIXER-TO-SPEAKER-SIGNAL CHECK

If, after performing the preceding steps to check the mixer, you determine it is not the culprit, it's time to continue past the mixer into the outboard gear. Try the following:

1. Test the main output cables with your cable tester or multi-meter, or just replace them.

2. Test the continuity of each piece of outboard gear or, for more expediency, plug your main output cables into the snake returns. If the audio signal is restored, you will know that the problem lies in one of the pieces of outboard gear you have just bypassed.

3. If the previous step led you here, test each piece of gear for continuity input to output (with a tester) or reconnect one outboard device at a time to determine which one has failed.

4. If, after bypassing the outboard gear, you still are without an audio signal and you haven't checked them already, the snake returns would be the next suspect. Like the snake sends to the board, the snake returns from the board must be checked with a cable tester or multi-meter.

5. If the signal flow is solid up to the stage snake box returns, test the cables going to the amplifiers. If you have a self-powered system, you will test or exchange the cables from the snake to your powered enclosures.

6. Check the speaker wires from the amp to the speaker enclosures.

With all cables and wires working, it's safe to assume that the problem is with the power amp(s) or the speaker enclosures themselves. Take note: Testing the actual wattage output of the amplifier is beyond the scope of this book, but there's still more you can do. Try disconnecting any amplifier you are certain is functional and connecting it into the FOH signal path. If you *still* get no signal to the FOH enclosures, it has to be one or more of your FOH speakers. If you have self-powered FOH cabinets, you will be able to bypass the external amplifier testing and go straight to the powered enclosures. Most likely, you will not go all the way to the speaker enclosures; the problem will probably lie in a faulty cable or alien particle buildup somewhere in the console.

So what do you do if the sound system is in fact working, but is just noisy?

System Grounding

To best run and maintain a sound system of any magnitude, some basic understanding of electricity is essential and will prove invaluable. This section explores earth ground, chassis ground, and ground loops. These three subjects will bring to light important aspects of the electrical circuits found in your worship house. Be sure to read carefully and understand the information contained here. A word of caution: If at any time you feel uncomfortable working on the electrical side of things in sound reinforcement, call a professional.

EARTH/GROUND

To best understand the concept of grounding, consider this one simple fact: The incoming electrical current drawn from your wall outlets needs a common return path. Earth and ground are the terms used for this path and, as the former name implies, refer to the physical connection to mother earth (see Figure 9.9), which serves as the destination. If there's no return path for voltages to return to, guess what can make for a great substitute? You, that's what. Like lightning striking a tree, electricity in your house of worship electrical system (as well as any other system based on similar concepts) naturally makes a bee-line for this path and will be more than happy to use your body if there's no other place to go.

Figure 9.9
A ground wire comes from the electrical panel located outside your house of worship and connects to a copper rod dug into the ground, thus grounding your entire electrical system.

Any and all proper electric circuits must have this destination to prevent contact with dangerous voltages. Somewhere, usually around the outside of your worship building, is an electrical panel that contains the circuits for your house of worship (you can find a similar, albeit smaller, panel around your home). As shown in Figure 9.9, a wire from the panel is attached to either a water pipe or a copper rod, grounding the electricity in your worship house. The central, bottom-most terminal on three-pronged electrical wall outlets in your house are physically connected to whatever ground pipe or ground rod you may have outside the building through the ground wire of each circuit running through the walls.

SIGNAL/CHASSIS GROUND

In addition to serving as a return path for current, the ground also reduces or eliminates noise from power cables, audio equipment, and even radio interference. One type of ground is called signal ground and can found in XLR cables and 1/4-inch TRS/TS cables. Along with most other audio cables, XLR and 1/4-inch cables are made up of one or two conductors surrounded by a wire mesh shield. This shield serves as the ground for these cables and eliminates, or at least greatly reduces, the possibility of unwanted noise.

The metal case around your audio equipment can serve as another type of ground that reduces noise. This type of grounding is called chassis grounding. In audio equipment fitted with a three-pronged AC plug (see Figure 9.10), the chassis of the gear is connected to the earth/ground of the AC plug. Specifically, the chassis of your audio equipment is attached to the green ground wire in the third pin of the AC plug, which connects to the ground in the wall outlet and on through the wires of your worship building to the ground or earth rod next to the electrical panel. This path creates safety but can also lead to ground loops (see the next section). Some audio gear, however, will have only a two-pronged AC connector (see Figure 9.11). In this case, the signal ground and earth ground will be combined and/or the internal circuitry of the audio equipment will have a provision for signal ground.

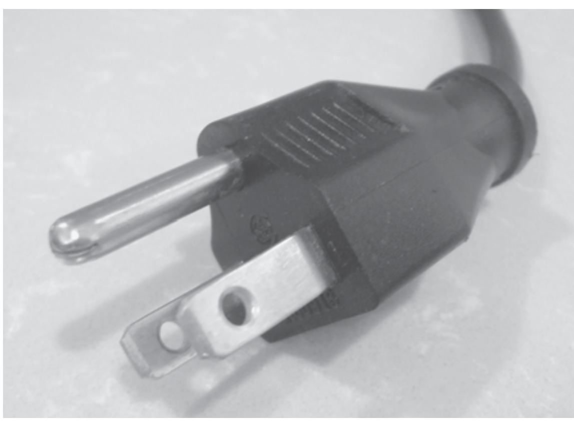


Figure 9.10
When you see this three-pronged plug at the end of the power cord on your outboard gear, you can rest a little easier knowing it's properly grounded.

Figure 9.11
In your work in houses of worship, you will surely run some pieces of outboard gear that still have a two-pronged AC plug like the one seen here. While the signal and earth ground may be combined or the internal circuitry has a provision for signal ground, it's a good idea to modify these pieces with a new power cable that has a three-pronged plug.

GROUND LOOPS

A ground loop occurs when there is more than one ground connection path between two pieces of gear—for example, if you have two pieces of outboard gear and each is plugged into a separate wall outlet, and a balanced XLR cable connects these pieces of equipment. The shield inside the cable carries the ground path from one piece of gear to the other. The ground path continues into the wall outlet, through the electrical ground wire in the wall, and back to the other piece of gear through the wall plug it is plugged into (see Figure 9.12). The loop through the grounding wires now creates an antenna, which picks up unwanted noise and hum, which is in turn amplified by the audio gear.

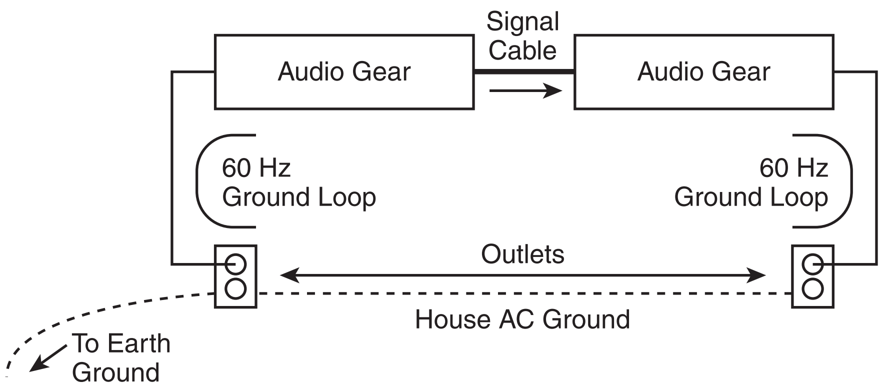


Figure 9.12
To better understand the ground path that creates 60Hz ground loop hum in your sound system, follow the path shown here.

Although ground loops can be difficult to isolate, a simple way to avoid them is to rack-mount your outboard gear and plug all the units into a single-space rack-mounted power conditioner (refer to Chapter 6, "Outboard Gear"), which then plugs into a single wall outlet. This leaves the possibility of a ground loop between your outboard gear and your power amps, though. If a loop occurs here, connecting both outboard gear and power amps to the same phase leg of your AC service panel will do the trick. Of course, most of you reading this book will not know what the phase legs of your AC panel are—*nor should you try to find out on your own by going inside and poking around.* That subject is a little deep for this book. If you have implemented the aforementioned suggestions and you still have a persistent ground loop, call on a more experienced audio tech or an electrician to step in.

YOU GOTTA STAY GROUNDED

If you're frustrated by a hum producing ground lift that you can't easily remedy, you may be tempted to use a three-prong–to-two-prong electrical adapter or even chop off the ground prong of a three-pronged plug to lift the ground. As mentioned earlier, this is more than just unwise; it's extremely dangerous. The earth ground plug is provided for safety—and God wants you to be safe.

Dealing with Unwanted Noises

At some point, after finding the cause of a ground loop and remedying it, you may one day think it has mysteriously returned out of nowhere when, in actuality, you're just encountering other noises in your system. Additional noises in your sound system (other than pure sweet audio) can be at best annoying and at worst distract from the worship service and message. The causes for noise can range from minor interference to potential system failure. Let's begin with some of the more simple solutions and proceed from there.

Hum/Buzz

Luckily, simple problems have simple solutions. In the following list, you'll find some common scenarios that may cause unwanted hum in your system along with some easy methods to eliminate them.

- Power amps, especially old ones that have been in the system for long periods of time, can generate a significant hum field. If your other noise-sensitive components are sitting on top of or are located in the same rack as the offending power amp, it can pick up and amplify the hum, thus causing even more noise. The easiest fix is to rack your power amps separately and mount your other outboard equipment (crossovers, EQs, etc.) in another rack, away from the amplifiers.

- Hum can occur when microphone cables run alongside AC power cords. Separate these cables and cords (if possible) to eliminate the noise. If they must cross each other, have them cross at right angles.

- A common silicon controlled rectifier (SRC) lighting dimmer (see Figure 9.13) switch can introduce buzz into your microphone cables and ultimately your sound system. If you suspect a dimmer switch, simply turn it off to confirm it's the source. If the buzz leaves, you've found the cause of your hum—but unfortunately haven't fixed the problem. Turn the switch back on and adjust the dimmer to see if one position generates less noise than another. Also experiment with positioning or relocating your microphone cables. Lastly, consider replacing the dimmer switch with a regular on/off switch.

Figure 9.13
When trying to pinpoint a hum or buzz in your system, sometimes you need to look no further than your walls to see if there are any SRC lighting dimmer switches like the one seen here.

HISS

The most common cause of hiss-type annoyances are components early in the signal path that have been amplified to unacceptable levels, thus creating a series of poor gain structuring. (Refer to Chapter 3, "Mixers and the Art of Mixing," for a review on gain and signal-to-noise ratio if need be.) If you find yourself having to push the main faders on your mixing console past 0 or Unity (see Figure 9.14), then you have a gain-structure problem within your system, and it needs to be addressed in order to eliminate the resultant hiss.

Figure 9.14
Having to go past the U marker between the two 5dB increments in the master fader's path is usually an indication that there's a gain-structure problem in your system. This causes a hiss issue that must be addressed. (Image courtesy of Mackie.)

At the same time, excessive gain in your power amps can cause hiss. Usually, remedying this problem is a matter of finding a balance between components. For example, if your console is not working hard enough, meaning it's not working properly when set near unity gain, your amplifiers will be amplifying excess noise. In this case, reduce the gain of the power amps and increase the master fader level at the mixing board until you find a balance of the desired signal level that has the least amount of noise. If you have singled out a microphone or component on the other end of the signal path as the noise-maker, adjust the input gain of that particular unit.

STATIC AND CRACKLE

Static- and crackle-type noises can be caused by a variety of sources. Some examples include the following:

- An intermittent short or dirt somewhere present in a microphone cable or the microphone itself can produce static like sounds.

- Dust buildup in a control knob will produce a crackly output when the knob is turned.

- A faulty audio cable can cause scratchy noises when the cable is moved.

The quick solution is to replace the offending cable or to simply spray the knob in question so your system can continue to work.

MORE COMPLEX NOISES

Unfortunately, the sad truth is that even if you solve the simple problems, you may still have other more complex noises in your system waiting for you. Unlike a loss of signal, where you start your investigation at the beginning of the signal path and work toward the FOH enclosures, an unwanted noise confirms that a signal is getting to the speakers, so that method doesn't apply here. To pinpoint an unwanted sound, you need to start from the speakers and go in reverse. Here's how:

1. Turn your power amps on, but not your mixer or outboard gear. This isolates the amps from all the other gear and could potentially isolate the source of the noise.

2. Turn down the gain on your FOH power amp or your monitor power amp (depending on which speaker enclosure the noise is coming from). If the buzz stays the same, you probably have an amp issue. If you or your house has a spare power amp, swap them out.

3. If your power amps check out fine, turn them off.

4. Turn on the mixing console and outboard gear.

5. Turn the amps back on.

6. Using a cable tester, test all the cables going from the power amplifiers to the mixing board for intermediate shorts. No cable tester? Then you will have to replace any questionable cables.

7. Test the audio snake with a cable tester. If you don't have a cable tester, try unplugging different returns to your power amps and see if that silences the noise. Remember, unplugging an offending cable will silence the noise.

8. From the other end of the snake, you will be connected to outboard gear or directly to the mixing console. All cables from the outboard gear back to the mixing board must be tested or exchanged.

9. If you still have noise after testing the outboard cables, mute or turn off each channel individually. If you don't have a channel mute or on/off button, bring down the gain on the channel. By eliminating one channel at a time, you will ultimately find the buzzing channel. A channel that is noisy can be the result of an internal problem, in which case you will need an audio pro to repair your mixer.

10. A noisy channel can also indicate an issue farther up the signal line. Unplug the input cables to the offending channel. If the buzz disappears, you know it is not the board and that it is something plugged into the board.

11. Test the snake line from the channel to the stage. If the line is good, your problem is coming from some piece of gear on the stage.

12. At the snake stage box, you have only one cable between the box and the microphone, direct box, or instrument amplifier. Either the cable or the device at the end of the cable is the culprit.

There you have it—just like Sherlock Holmes, you have solved the mystery. At this point, you have two choices: Fix it yourself or hire somebody to do it. While this book is not designed to turn you into an audio repair technician, the next section will guide you through the simple process of learning how to solder correctly. With this skill you can repair cables that have gone bad and even repair the occasional microphone that fails.

Proper Soldering

This section is a basic how-to on soldering with a pencil-type soldering iron. This type of soldering tool, as opposed to a soldering gun, is the preferred choice because it enables you to execute more precise soldering tasks. From choosing your iron to choosing your solder, you will gain the knowledge needed to become a proficient soldering soldier. Not only is mastering the skill of soldering absolutely invaluable in this line of work, doing so makes you more self-contained, thus making you more attractive to prospective clients if you choose to enter into sound reinforcement as a source of income. The fewer people involved in running and maintaining the system, the more cost-effective it is for the house.

A WORD OF CAUTION

When working with a soldering tool and its related components, be very careful to avoid touching the soldering apparatus. As you will discover, you will be regularly working with temperatures that can severely burn your skin, not to mention burn a hole in that brand-new shirt or jeans. Take it slow and be cautious at all times.

CHOOSE YOUR IRON

There are several types of soldering irons for you to choose from that will work great for maintaining your house system. After reading the following descriptions, carefully consider what type of soldering iron will work best for you and at the same time match your innate tool-using abilities. That is, those of you who don't regularly handle tools may want to ease into this by opting for the simpler plug-in soldering pencil as opposed to the blow-torch-style butane-based pencil iron.

Soldering Station

The best choice for a soldering gun is actually called a soldering station. This device incorporates a soldering pencil with a thermostat that controls the temperature of the pencil tip (see Figure 9.15). A typical soldering station will offer temperature ranges of 500°–800°, which is enough heat to handle all your soldering challenges. A standard soldering iron that you plug into a wall outlet will have a preset temperature of either 500° or 800° and weigh in at 25 to 40 watts. While these units are much less expensive than a soldering station, they're limited to one temperature. It's important to have a soldering tool that offers multiple temperatures because not all solder repairs need the same amount of heat. For instance, an XLR cable can be successfully soldered at 500° to 700°, whereas a heavier speaker wire attached to a Speakon connector or jack can need as much as 800°. Incorrect solder temperatures can cause problems; for example, applying too much heat can damage plastic parts and wire insulation, while applying inadequate heat will cause prolonged waiting periods for solder to melt (reflow), resulting in a poor solder connection.

Figure 9.15
Perhaps a smarter choice for a soldering tool is the soldering station like the one seen here. These table-top units feature a pencil-type soldering tool and a dock that offers wattage and temperature controls.

Cold-Heat Soldering Iron

Another type of soldering tool is a cold-heat soldering iron (see Figure 9.16). The cold-heat iron is battery operated, which makes it portable and great for when you have to make a repair on something that can't be easily removed to a workbench. Another positive is this device is designed to heat up instantly and then cool down in a few seconds, lessening the chance of mishaps when not using the tool. On the downside, cold-heat irons can be a bit temperamental and require a greater level of soldering skill than the more common pencil iron or solder station. Also, they tend to run at a cooler temperature, which is not good for some soldering jobs.

Figure 9.16
While portability and safety are attractive attributes of cold-heat soldering irons, they require some prior soldering experience and may not suit your every need due to inherent temperature constraints.

Butane Gas Pencil Irons

Butane gas or torch pencil irons (see Figure 9.17) are yet another option for your soldering needs. This tool uses a flame generated from butane fuel to heat the soldering tip. The fuel comes in a small tank, making this torch-type setup portable, and can run from 30 minutes to an hour on a butane charge. Quality butane irons never have an exposed flame, and their temperatures can be adjusted as high as 1,000º! Some models can even work as a mini blowtorch or heat gun, which gives them additional versatility. That being said, this type of tool is more difficult to operate and, due to the excessive high temperatures, should be used only by those with prior soldering experience.

Figure 9.17
A butane soldering iron can serve a dual purpose—portable soldering tool and mini-blowtorch!

SOLDER AND TOOLS

No matter what soldering iron you choose, it's useless without the main ingredient: solder! To choose the best solder for the job, let's examine some of the specs you'll need to be aware of:

- The center core of solder wire is known as rosin, which is a non-acid solvent made of various materials. Only use solder with a rosin core, as this enables the deoxidization (cleaning) of metals so they may be joined. The rosin performs this process quickly and ensures that the solder wicks (adheres) to both adjoining metals. Be careful never to use acid-core solders, which are used in the plumbing industry; they will damage your system's electronics.

- The materials used to make the rosin core are available in the following designations: R, RMA, and RA. R refers to non-activated rosin, RMA is mildly activated rosin, and RA is fully activated rosin. This is important to be aware of because the more activated the rosin, the greater its ability to deoxidize the metals to be joined.

- The size of the wire is usually displayed with a number followed by the letters AWG, which stands for American wire gauge. Also known as the Brown and Sharpe wire gauge, this is the standard set by the United States Government for wire sizes in 1857 and used ever since.

60/40 rosin core soldering wire is considered best for audio solder repairs. The 60/40 refers to 60 percent tin and 40 percent lead. As for the best size of solder wire, 18 or 20 AWG is the way to go. This size of solder will melt quickly and flow properly under the right amount of heat.

As mentioned in the toolbox shopping list, you will need a few ancillary tools specific to soldering (see Figure 9.18). Here are few to keep in mind:

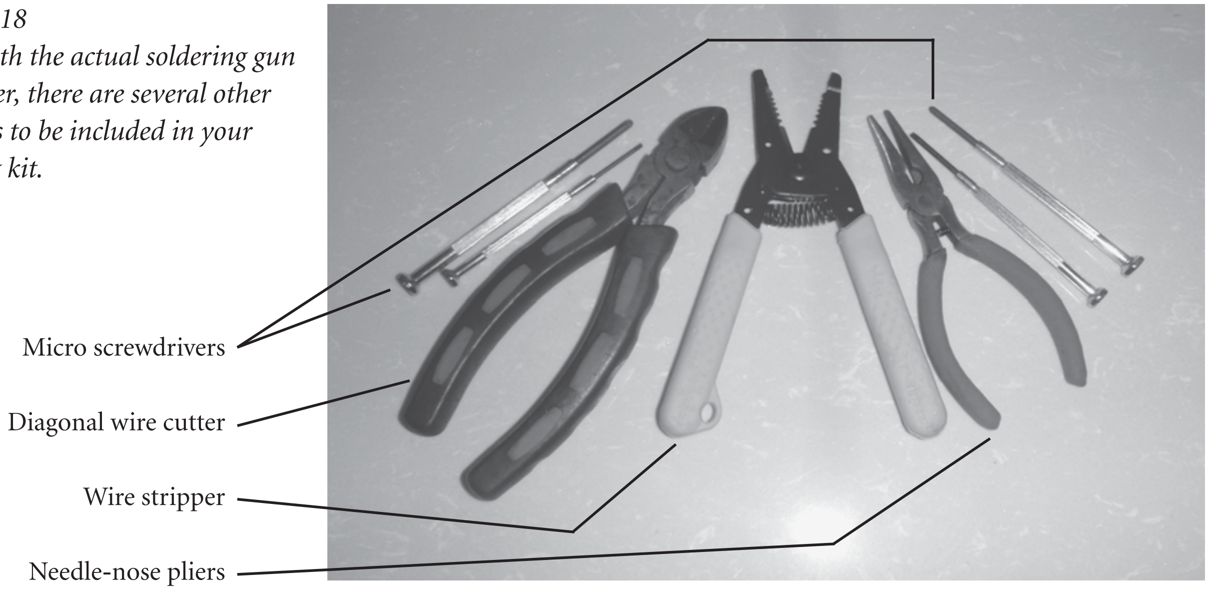
- **Wire stripper.** A self-adjusting stripper will take the guesswork out of wire stripping and can save you from damaging the wire.

- **Diagonal wire cutter.** A good pair of sharp diagonals will make wire cutting simple and precise.

- **Micro screwdriver set.** In addition to a collection of standard-sized screwdrivers, you'll need these smaller screwdrivers for the many miniscule screws you'll encounter throughout your system.

- **Needle-nose pliers.** These pliers can get into small places and hold wires or electronic parts with greater precision than regular pliers.

Figure 9.18
Along with the actual soldering gun and solder, there are several other essentials to be included in your soldering kit.



Another useful tool is helping hands (see Figure 9.19), also known as third hands. These tools consist of two or more mechanical arms tipped with alligator clips or clips and magnets. Some have flexible arms that can be bent to any number of positions to hold whatever you will be soldering, others have a magnifying glass built in to help you see what you are doing, while still others incorporate all these features.

Figure 9.19
With helping hands, you can more easily work on smaller items that are difficult to hold and require you to use both hands while working on them.

REFLOW TECHNIQUES

Now that you have the tools, it's time to turn the heat up and get to work. The term reflow or reflowing describes heating solder until it is in a liquid form and using it to bind two separate pieces of metal or wire together. The techniques described here will take you through the process of making good, proper solder connections. Be sure to follow these instructions carefully. Also, be prepared to have to work at this; soldering takes patience and a fair amount of trial and error—so much so that it's a good idea to practice on some cables that you don't necessarily need before working on a piece of gear that's integral to your system.

Making Connections

Good soldering joints start with good mechanical connections prior to reflowing solder over the connection. If you can crimp or wrap a wire around a solder post, tab, or another wire, then do so with as much care and precision as possible. There is one important exception to mention when talking about this approach: XLR cable ends have solder cups for their wire connections (see Figure 9.20). Wires are placed in these cups and the solder is reflowed over it. Given the fact XLR cable repair can represent most of your soldering duties (because they tend to take more abuse than other cables and wires), it is imperative you pay attention to this fact.

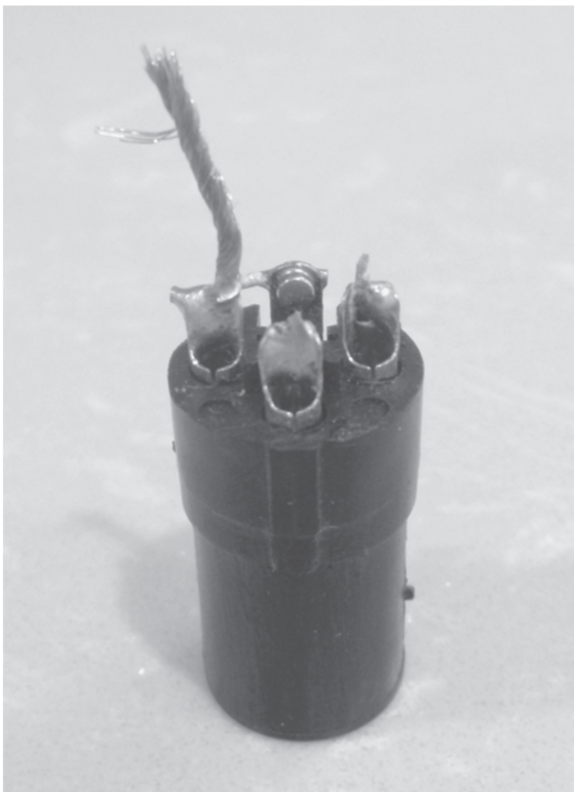


Figure 9.20
When working on XLR cables, you will place the wire inside solder cups, and then reflow the solder over the cups.

Tinning

Tinning is a necessary pre-process in which you melt a small amount of solder on the surfaces to be joined (see Figure 9.21). When tinning wire, remove excess insulation as well as any fraying wire strands. Bend or shape your wire to the close-to-final shape of the solder connection. Once the wire has been tinned, it will be more stiff and harder to manipulate. Tinning does two important things:

- It allows the rosin flux to clean the metal surfaces to be connected.
- It's done just seconds before the final solder reflow, meaning the heat applied during tinning preheats the surfaces for a quick, last reflow connection.

Figure 9.21
Before making the final reflow connection, tin the wires with a small amount of solder for the best overall results.

Applying Some Heat

The soldering iron is used to heat the intersection of the surfaces to be joined. The idea is that the joined surfaces will melt the solder, not the tip of the soldering iron. Ideally, the solder wire is fed into the intersection of the connection after the iron has heated it for a few seconds and the melting solder joins the connections. To ensure this, the soldering iron tip must be freshly tinned (see the previous section) so that a small amount of molten solder on the iron's tip is the heat transfer flow point. Refrain from melting an abundance of solder on the iron tip and then sticking it onto the joined surfaces; this will cause what's known as a cold joint, and will not hold. Solder connections need to be heated for just a few seconds for electronic wires and components. A mere tenth of a second too long with the iron touching both surfaces can transfer heat beyond the connection point and start melting things you don't want melted. With that in mind, once you reflow the solder, remove the iron quickly.

Not Too Little, Not Too Much

The right amount of solder to make a solid connection is measured by your eye and experience. Excessive solder, where you feed too much solder onto the connection surfaces, will create blobs or balls. Too little solder is also problematic, as the surfaces do not have enough solder to hold the connection mechanically or electrically. Too much heat or too little heat can also affect the solder joint adversely. Cracked solder joints are a sign of too little heat or movement of the surfaces to be joined while soldering (steady hands are a must when soldering). Too much heat will cause the solder to spill away from the connection.

All the information in this chapter is designed to help prepare you for a failure in your house sound system. In reality, confidence and preparation are your most powerful tools with regard to healing your gear. Now for the final frontier: A good sound technician can take his or her skills and possibly his or her sound system outside the comfort of one particular house of worship and be able set up and mix live sound elsewhere. Let's see what it takes.

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CHAPTER 10

Outreach

Every house of worship wants to demonstrate the teachings of God in its community. One way to do this is by community outreach. These programs often include taking meals to and spending time with the elderly or helping people in the community with the most basic aspects of living that they cannot handle on their own.

This chapter introduces a more modern trend in outreach programs, where worship bands, singers, preachers, and the like perform in front of tens to thousands of believers as well as non-believers. A music and spoken-word production such as this will require a sound system and technician at a location outside the worship house that involves its own setup and production procedures. From procuring a portable stage, lighting, and generators to scheduling the delivery and pickup of portable toilets, this chapter covers the various aspects of producing an outreach event one step at a time.

If it all sounds a bit overwhelming, don't worry. With proper planning as set forth in these next few pages, your outreach event will be a successful one. Being prepared with a detailed plan including all contingencies will give you the best chance at a smoothly produced outreach program.

One important thing to keep in mind is wherever you set up and run a sound system, the basics you have learned throughout this book are the same. The only difference will be the location.

Leaving Your Comfort Zone

Before you consider leaving your comfort zone to set up and operate a sound system, you will have to have all the equipment necessary for such an adventure. While some worship houses own systems that are totally portable, meaning the gear can be used for an in-house service and also an outreach event, many do not. In that case, you will need to make some important decisions—and in order to do so, you have to be informed. What gear will be necessary for the event? What do you do if your house doesn't have the gear you need and/or its system is not portable? What should you know before you consider renting a system? These questions and many more are addressed in this section, so read carefully.

TO RENT OR NOT TO RENT

If you have a permanently installed sound system in your house, you will either have to purchase a separate outreach system or rent gear for an external event. If your organization plans to produce a number of outreach shows every year, purchasing a separate sound system will be more economical in the long run. If, however, you plan on conducting just a couple of outreach music events a year, renting equipment will be a better and cheaper choice. Following are the pros and cons of renting sound equipment.

Some of the advantages of renting sound equipment include the following:

- The equipment will not have to be stored between events.
- You are not responsible for the maintenance and repair of the sound gear or any lighting, staging, or generators you rent.
- You will never have to disturb the sound system that has been installed and tuned to your worship space.
- You will have the option of renting the latest state-of-the-art sound equipment available.
- You will not incur the cost of purchasing a sound system or, in some cases, a second system.
- You can always rent an operator if you decide the event is too large or complex for your abilities.

And, don't forget: If you do plan to rent a system, you won't have to rent microphones, mic stands, mic cables, or outboard gear (if you have a portable rack), etc. All these things can come from your worship space sound system.

The disadvantages in renting gear are as follows:

- Rental houses assume you can operate the equipment, so you will need to be somewhat familiar with the specific sound system you rent.
- You have to know exactly what you will need to rent for your event.
- Rental houses won't know if you are renting too little or too much gear.
- Should you expand your outreach program, you will be spending more and more money on rental equipment.

In the end, it comes down to what works best for you and your house of worship's needs.

THE GEAR

Whether you rent a sound system or not, the environment in which you will be setting up and operating the sound system will be foreign to you as compared to your worship space. As the saying goes, "Knowledge is power"; to that, we add "So is skill, for that matter." Both sentiments couldn't be truer when it comes to venturing out of your comfort zone to work an event on an unfamiliar sound system in an unfamiliar place. Coming prepared with as much knowledge and skill as you can acquire will ease the inevitable anxiety that will arise from such a scenario and make you that much more effective.

Much of that knowledge and skill comes into play when deciding what gear you will need for any show outside your house of worship. Following is a punch list of gear you might need; be prepared to add to it as you see fit when planning your event.

- Microphones (including wireless)
- Mic stands
- Direct boxes
- Mixing console
- Outboard gear in rack
- Audio snake

- XLR cables
- Instrument cables
- Speaker wires
- Electrical cables
- Tool kit (refer to Chapter 9, "Healing the Sick and the Tired…Gear, That Is")
- Table or stand for mixing console
- Batteries (9 volt, AA, AAA)
- Decibel meter
- CD player (and CDs), iPod, and appropriate patch cables
- This book

This starter list can (and will probably) be augmented with any additional gear you will need for your particular outreach event. Whatever your list contains, make like Santa Claus and check it twice before you leave the safety of your worship space. That said, more often than not an outreach event will be in close proximity to the house of worship sponsoring it (i.e., in the parking lot). This makes things easier should you forget something on the list.

GENERATORS

One piece of gear that you may need to include on your list that is surely not in your house of worship's regular arsenal is a mobile power generator (see Figure 10.1). There may be times when what you take for granted power-wise in your house of worship is simply not available at the location of your outreach event (various location possibilities are introduced in the next section of this chapter). Mobile power generators are a perfect solution for providing the electrical power you will need to properly operate your outreach sound system, lighting, and musical equipment. Although it's more common to use one or more mobile generators at an outdoor event, they could be needed for indoor events as well, making them a valuable and necessary tool.

Figure 10.1
When power is nowhere to be found, simply bring your own in the form of a fossil-fuel mobile power generator.

Powerful Advantages

Generators use an engine fueled by gasoline or diesel to produce electricity. Ironically, the electricity produced by a portable generator is for the most part cleaner and more reliable than the electricity available from the plugs in your worship house or home. This is due to the fact that the generator producing the electricity is generating it for no other reason than for the job at hand. Electricity that comes from the outlets in your house of worship or at certain remote locations such as community centers is subject to competition from other users in the community and abroad. When there's too much demand on the electrical system and not enough power to go around, everyone who is taking from the system suffers. In extreme cases, there could be what's called brownouts, where power through the grid is temporarily diminished or even interrupted. Generally, these happen during the summer months, which are, unfortunately, the months during which outreach events generally take place. Using a portable generator will afford you the luxury of never having to deal with any drop in energy. In addition, because the electricity is being produced where it's being used, there is far less chance that it will be contaminated by electromagnetic interference (EMI) or radio frequency interference (RFI).

Power Considerations

There are several factors to consider when making plans to rent a power generator for your event:

- Generators come in a variety of sizes and power ratings. When renting a generator, be sure you know how much amperage you will need to operate your sound system and lighting in addition to the musical gear on stage.

- When renting a generator, especially a large one, make sure the rental supplier includes delivery, an operator, and a distribution system (see Figure 10.2) in the deal.

- With regard to the distro (the box that has the outlets and circuit breakers in it), the outlets will be from 110 volts to 120VAC (volts alternating current), and the breakers should be 20 amps in size. That said, the distro can also sport 120VAC, 30 amp twist lock connectors (see Figure 10.3) for higher amperage demands.

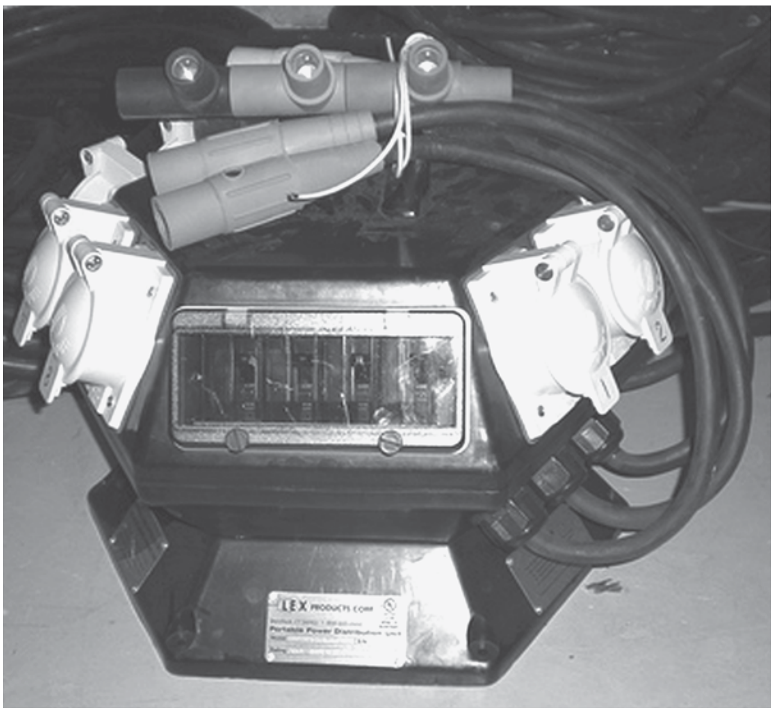

Figure 10.2
A distribution system, or distro box, receives the electricity from your generator and acts as the central station into which various components of your system will be plugged.

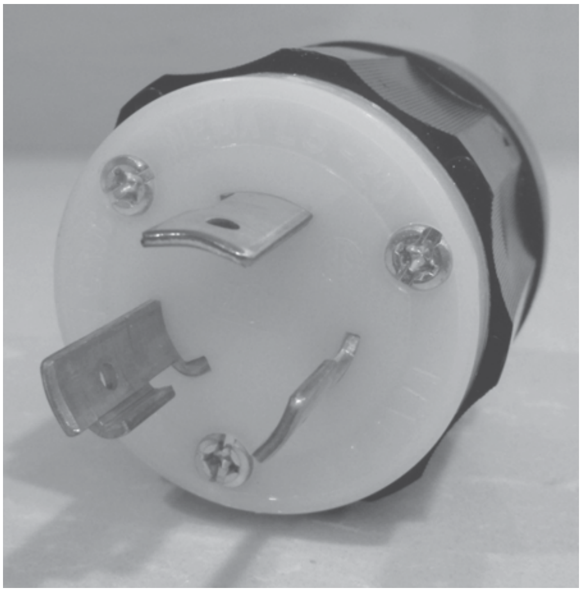

Figure 10.3
When dealing with high amperage levels, you may find twist-lock–type electrical connectors on the end of power cables.

In addition, it's important to be aware of how the cable that connects from the generator to the distribution system (also known as a distro box) comes into play. A heavy electrical cable—usually 10-gauge or larger—brings the electricity from the generator to the distro box. Power cables for sound, lighting, and stage gear will be plugged into this box. When setting this up, do not plug your lighting and sound system into the same circuit (outlet). This can cause noise in your sound system and may draw too much power from a single 20 amp outlet. Also, be aware of your electrical cable gauge and length. For runs from 1 to 50 feet, use 14 AWG (American wire gauge) extension cable. If you go over 50 feet, use a 12 AWG cable; if you are running a distance of 150 feet or more, get yourself a 10 AWG cable. It is unlikely that you will need cables over 150 feet, but plan ahead and be prepared. Whatever scenario you are dealing with, never use a 16-gauge cable. This is just too light for audio and lighting and can be dangerous, as it can get extremely hot.

Outreach Locations

As stated, the purpose of an outreach is to bring your house of worship's particular form of belief in God to your local community. This type of event can be held in numerous locations, with each having its own intrinsic qualities. Although some locations are better than others and lend themselves more readily to musical events, with the right information and some careful planning on your part, you can pull off a successful outreach in any one of the locations described in this section. In reality, the event's location is limited only by your imagination, budget, and local ordinances.

OUTDOORS

Since the basic idea of an outreach is to bring the message to the people of your city, town, or neighborhood regardless of whether they are parishioners, events are often held outside (see Figure 10.4) and for good reason. One obvious reason is it's less intimidating for a non-member of your congregation to walk into an event in your parking lot than to walk into the actual house of worship. Also, just having a live band playing outdoors will attract a crowd, which is what you're after. If you are doing your job correctly, the outdoor sound system and music traveling through it will inevitably draw people in, as your event will be projected throughout your local neighborhood streets and then some!

Figure 10.4
More often than not, an outreach event will be held outdoors. In this case, it's good idea to have a covered stage, not to mention a covered area for you and your gear! Also, make sure you have a sound system with components necessary to accommodate this environment.

POLICE YOURSELF

When planning an outdoor outreach event, it's a good idea to inform your house of worship's neighbors of the event via hand-delivered or mailed flyers. Be sure to disclose the time of the event and what type of entertainment and preaching will be offered. Most importantly, politely ask them to call *you* rather than the local authorities if the music should become too loud. To help ensure no noise ordinances are broken, have a decibel meter (refer to Chapter 1, "Sound and Sound Advice") on hand to monitor the volume of your event. With this device's portability, you can measure decibels at different locations in or near your outreach venue.

Setup Considerations

Setting up a sound system outside is in some ways similar to setting up sound in your worship house. For one thing, you will be using the same equipment outdoors as you would indoors. From the microphones to the FOH speakers, you will be plugging in all the equipment in much the same way as you've learned how to do throughout this book. (You may, however, be using additional equipment, such as a portable stage, a canopy or tent, lighting, and possibly a generator.)

Just as there are similarities, there are differences as well—the biggest one being that your outdoor area will have different acoustical characteristics from your worship space. One advantage to this is that you will probably have fewer acoustical challenges because you will be mixing outdoors. Think about it:

- There are no interior walls or ceilings for sound waves to bounce off and potentially cause mayhem.
- You won't have to deal with any reflective sound unless you set your stage and sound up in front of another building.
- You can apply reverb and other effects to the vocalist and instruments without having to consider the possible clash with the natural reverb of your worship space.
- The FOH speakers will be easier to tune because they will not have to conform to any interior inadequacies.

But alas, these advantages come with a price: Sound waves dissipate more quickly when there are no walls or barriers to contain them. As a result, you will need to push more decibels through your sound system outside as compared to inside—and you may need more power (wattage) to do so. This likely need for additional power and possibly some extra speaker enclosures may influence your decision to rent an outreach sound system. Another dilemma that arises from the absence of walls is that low frequencies tend to suffer from cancellations within the waveform (known as nodes). Unlike with high frequencies, this problem, known as comb filtering, can have areas of cancellation that are several feet wide due to the long wavelengths of low frequencies. While this problem can occur within the confines of your worship space, the reflections usually fill the gaps, making the problem less severe. A common remedy to this problem is to position the subs in front of center stage whenever possible.

Weather Considerations

The weather can play a major role in your outreach program. As you probably know, sound equipment does not like water. At the same time, just like people, it doesn't like excessively hot or cold climates. On those long, hot summer days, you will have to keep your mixing console and your outboard gear (not to mention your performers) under a canopy (by the way, on days like these, you yourself should be wearing a hat and sunscreen). Likewise, your gear will not respond well to cold weather. Digital processors slow up in cold weather, not to mention performers' hands. On days when you have to contend with Mother Nature's fickle nature, don't hesitate to rent a vinyl enclosure and employ portable climate-control units. Of course, rain and high winds will surely hurt your program and may be sufficient cause to reschedule your event. Overall, it's not that you can't protect your equipment in less-than-desirable weather, it's just you want to have people attending your event, and the fact is, your outdoor event will be most successful if the weather is pleasant—not too hot and not too cold. If you are planning an outreach event during the fall or winter months (did someone say Christmas show?), it's a good idea to look for an indoor venue, which brings us to our next location: community centers.

COMMUNITY CENTERS

Just about every community has a community center—a building available to the public that can be used for a multitude of purposes and, in many cases, at no charge to users. Before planning an outreach event at your local community center, it's a good idea to pay a visit to the center. Just walking around inside the building and listening to the environment can give you valuable insight as to how you will make your event sound its best. In fact, it's a good idea to visit twice: once when it's unoccupied and again during a musical performance. When doing so, be sure to take note of the following:

- Where the FOH speakers are set up
- Where the mixing board is located
- How live sound is handled in the room by the engineer present
- Whether there is ample room for a comfortable mix position with *your* gear
- How long a snake you will need

And ask yourself the following questions:

- Can you hear the vocalist?
- Are there a lot of reflective sound waves in the room?
- Are bass, mid-range, or high frequencies exaggerated in any part of the room?
- Can you identify standing waves or other acoustic anomalies?

Take mental notes on the show's attendance level. Just like in your worship space, a full house will handle sound waves differently than a half-full house. You may not be able to talk with the technicians during the event, so make an effort to get a phone number or e-mail address so you can contact that person after the performance. The more information you can gather, the better prepared you will be when it comes time to set up and mix your own outreach show at that location. Also, be sure to find out how much available electricity is at the location; specifically, see how much amperage and how many circuits are available on the stage and in the building. (You may find it will be necessary to rent a portable generator, discussed earlier in this chapter, to ensure that there is enough electrical power for your event.)

SCHOOL AUDITORIUMS

The types of building spaces that fall under the category of school auditoriums can vary in sonic effectiveness. Some schools have auditoriums that are designed for musical performances, while others are multi-use spaces that may host a basketball game one day and a rendition of *Grease* on another. (Gymnasiums are difficult spaces in which to put on a musical production.)

Regardless of how inappropriate the venue may be, you may have no choice but to deal with it. Be prepared, because many houses of worship exist as part of a school complex. If this describes your house, then you will be using the same auditorium for most if not all of your outreach events. The obvious advantage of using the same space repeatedly is that your ears will become accustomed to the acoustics in the auditorium. Whether those acoustics are heavenly or hellish, there will be no surprises as to how the room sounds.

I LOVE IT WHEN A PLAN COMES TOGETHER

The best way to prepare for mixing in a new environment like the ones mentioned in this chapter is to formulate a plan. As mentioned in the "Community Centers" section, if you're involved in organizing an outreach event in a room you've never even been in before, it would be prudent to visit that room to start the planning process. As you walk around, note the shape of the room and the height of the ceiling. Clap your hands, maybe even sing a song—make some noise so you can listen to the room. Check for any room anomalies such as standing waves. Check to see if you will have enough electricity available so you can decide whether you will need to rent a generator. When the day comes, plan to be on location with the sound system early so you have plenty of time to deal with the inevitable problems that always seem to arise. Be sure you have help from your fellow worship technicians, and don't be afraid to reach out to some of the non-audio know-it-alls in your congregation. They can help with the not-so-glorious tasks of loading and unloading. Plan, plan, plan, and plan some more. It will pay off in the long run, significantly upping the chances that your outreach will be a successful one.

Mixing Where You Can

Perhaps the most important rule to keep in mind when working an outreach event is to stay as open and flexible as possible. This not only applies to your mixing duties, but anything else that comes your way. You will be working in a new environment with new people and possibly new rules. Part of this new environment may call for you to mix from a variety of new positions. Remember, whether you are mixing an indoor or outdoor event, the audio will sound slightly different in different locations. This section discusses the various possibilities, starting with the standard and your already established position: front of house.

FRONT OF HOUSE (FOH)

As you know, mixing from the FOH position is the most desirable primarily because you hear what the audience hears (see Figure 10.5). Also, all real-time adjustments are easier from the FOH position. Other advantages include the following:

- You can more accurately make mixing adjustments during the show.
- If a vocal is buried in the singers, you can bring it up and balance the vocalist during a song.

► Should your preacher become louder and more exuberant during the program, you an easily reduce levels and adjust the individual EQ.

Figure 10.5
Mixing from the FOH position affords you not only a clear view of the stage, but more importantly it puts your ears at the same point of reference as your congregation.

Where you will set up your FOH mixing board usually is based on practicality or on the limitations of the location for your outreach program. For instance, in Chapter 8, "Putting It All Together," you learned you couldn't set up your mixing board where it will impede the traffic flow of your audience. In addition, you do not want to inhibit the visual presentation of your event by planting yourself and your gear in the middle of the attendees. Ideally, the mixing board should be set up behind your audience, to the left or right of stage center (if you have a stage) and on a riser so you can view the performers on the stage clearly.

Depending on the venue, you may be required to mix FOH from a balcony position. Remember: The most important thing to consider when mixing from above is that you may hear fewer bass frequencies up there. Be sure to EQ the room from the perspective of the audience and keep that audio perspective in mind when mixing.

MAKING EYE CONTACT

Making eye contact with those on the stage is important. Performers can communicate monitor needs and cues for background music or tracks through body or hand signals. Take this into strong consideration when deciding where to plant yourself and your board and be sure to take some time to work out these physical cues with the worship team before the event.

SIDE OF STAGE

Though mixing from the side of the stage (see Figure 10.6) is not known as SOS, you may feel as if you need to make that call when doing so. The side-of-the-stage location refers usually to behind the FOH speakers on either the left or right side of the stage, much like where you would see the monitor engineer at a major concert event.

Figure 10.6
With a separate monitor feeding you the FOH mix, mixing from the side of the stage affords you some unique advantages such as a closer relationship with the stage participants and the ability to make it onto the stage faster if any problems arise.

While this position may seem awkward, as with all variations and changes you may encounter while mixing outreach events, the most important thing is to not panic when you are faced with a foreign area or location. You may even find that this position for mixing has some unique advantages, such as a much closer relationship with the musicians, singers, and preacher(s).

One way to ensure success when mixing from the side of the stage is to have a separate monitor for your own personal listening. This monitor should have a mono mix of your front-of-house feed, enabling you to hear what the audience is hearing. If you don't have a monitor to spare for this purpose, pack a set of headphones. You may be able to hear some of the stage monitors, but be warned: Don't trust what you hear. It will be clouded by audio coming from the on-stage instruments, and the monitors will be off-axis to your ears. You will do better to listen to your monitor mixes through the aux sends that feed them and a set of headphones. Also, whenever possible, walk out from behind the board into the audience and listen to your mix from their perspective.

EAR PLUGS

When mixing from the side of the stage, it can be loud. You'll be in close proximity to a guitar, bass, or keyboard amp; plus, you have the added decibels from the stage monitors. In addition, your personal monitor may have to be loud if you intend to hear the FOH mix above the stage audio. Given all this, consider investing in a quality set of earplugs—ones that are specifically designed for music listening. These specifically designed plugs allow a full range of sound to enter your ear, albeit at a lower decibel level. Plugs designed for power tools, guns, or yard tools block sound, but at a cost in fidelity not acceptable by audio professionals; make your decision wisely. What may seem cost-effective could cost you your livelihood!

BACK OF STAGE

Mixing from the back of the stage (see Figure 10.7) is rare, but it does happen. As with any foreign mixing location, you need to not be intimidated by your position. To effectively mix from behind, you will absolutely need a personal monitor carrying the front-of-house signal. This monitor will be your only reference to what your audience is hearing. To add to the difficulties, your stage may have a backdrop, preventing you from seeing the performers.

Figure 10.7
When mixing from the back of the stage, it's imperative you have a personal monitor that gives you the FOH signal. Also, be prepared to mix blindly, as there's a good chance a backdrop similar to the one shown here will be in your way.

Considering all this, it's imperative that your stage monitors be set up as perfectly as possible before the event. With a good FOH monitor, you can still hear well enough to adjust tonality and placement of your musicians and singers in the mix. Not being able to see everyone populating your stage will be a definite handicap, but if your outreach performance is being videotaped or Webcasted, you can bring a laptop or computer monitor and get a feed of the show so you can keep an eye on them. As always, a good set of headphones will be an asset when mixing back behind it all. Lastly, as with side-of-stage mixing, try to get yourself out where the audience is. Hearing what they hear, where they hear it will certainly give you your best reference for the overall sound.

Closing Words

The intention of this book is to help you, the budding sound technician, become more knowledgeable, confident, and proficient in your work behind the board in your house of worship and beyond. If you were new to sound reinforcement altogether, hopefully this book has provided you with a solid foundation from which to grow. Regardless of your experience level when you started this journey, you have surely evolved into a more valuable asset to your house of worship.

Throughout your work and further development, be sure to keep this text close at hand for reference, as it was intended to become your audio bible. You have in your hands a useful, common-sense, not overly technical collection of information that will equip you with the tools you need to set up and operate all the aspects of live sound necessary for a successful worship service (and not lose your mind while doing it). Use it well, use it often, and always let you ears be the final judge.

God bless and thanks for reading.

APPENDIX A

Glossary of Terms

The definitions throughout this section were designed to be brief and to-the-point descriptions of terms and items you're likely to encounter in your work as a sound technician in worship houses and beyond. They were written in a non-technical language for quick reference, allowing you to use the terminology to get a head start on understanding the incredible amount of information put before you when you first enter into sound reinforcement. If you find a term within a definition that is foreign to you, chances are it's in this glossary as well.

Take note: Many of these terms have much deeper explanations and will require further study on your part to fully grasp the concepts behind them. With that in mind, it's all the more reason you should keep this book close by, as many of these terms are explained in greater detail throughout.

A

amplifier Usually referred to as a power amp, these heavy-duty hardware units increase the power—measured in watts—of the audio signal to the loudspeaker enclosures.

audio snake Usually referred to as a snake, this component consists of multiple pairs (4–40) of shielded microphone cables bundled in a single casing that terminate either in a group of XLR and/or TRS ends (a pigtail) or in a stage box with chassis-mounted XLR and/or TRS connectors.

auxiliary send Usually referred to as an aux send, this is a function in the form of a knob found on individual input channels that offers you a way to create a separate (auxiliary) mix to be sent to monitors, effects processors, and/or a recording device.

AFL Short for after fader listening. This is a function in the form of a button found on individual input channels on most medium-sized and large-scale mixing consoles that allows the listener to hear an audio signal from a subgroup or aux send through the headphone jack.

attack A parameter found on compressors, limiters, and noise gates. The attack adjusts the amount of time (in milliseconds) it takes for an incoming audio signal to be processed.

AWG Short for American wire gauge. Also known as the Brown and Sharpe wire gauge. Brown and Sharpe is the organization that set the U.S. standard for wire size in 1857.

B

balanced cable A cable that is made up of two conductor wires wrapped in a metal shield, which helps prevent any noise or interference being transmitted through the cable and heard within your sound system.

bus A path within your mixing console that allows you to direct various channel inputs to a specific area on the board, where they can be adjusted as a group known as a subgroup.

C

chorus Both an alternative name given to the choir and the terminology used to name a particular time-based, spatial effect found in many effects processors.

compression driver An electromagnetic speaker, usually attached to a horn, used to amplify mid and/or high frequencies.

compressor An electronic device used to regulate an audio signal's dynamic level.

condenser microphone A type of microphone that requires a power source to operate, either from an internal battery, an external device, or, most commonly, the 48-volt phantom power feature found on most mixing consoles.

crossover A device that separates an audio signal by frequency and distributes those frequencies to the proper transducers in a loudspeaker enclosure or system of enclosures.

D

decibel Usually noted as dB, this is the unit of measurement used to define the amplitude level of an acoustic signal.

decibel meter A device (usually handheld) that measures the loudness of an audio signal in decibels.

delay A common time-based effect found in many digital effects processors that repeats the incoming signal according to user-defined parameters.

direct box Usually referred to as a DI box, this pint-sized signal transformer is designed to accept an unbalanced signal from an instrument and transform it into a balanced output suitable for your mixing console.

dynamic microphone A type of microphone known for its durability that is designed to transform acoustic energy into an electronic signal by way of a diaphragm.

E

earth/ground The common return path required by incoming electrical currents so as not to travel to unwanted destinations such as your body.

equalizer A feature found on either a mixer board's individual channel strips or a standalone outboard processor, this device is meant to boost (raise) or cut (lower) the volume of frequencies in various ways.

F

fader The sliding-type volume control found on mixing consoles' individual channel strips and main outputs as well as most graphic equalizers.

feedback An obnoxious, loud, high- or low-pitched tone that is created when sound waves are picked up by a microphone, projected out of a loudspeaker, and then again picked up by the same microphone, thus creating a continuous loop.

FOH Short for front of house. This refers to the mixing position in front of the stage. This term is also used to describe the loudspeaker enclosures that serve the members of the congregation.

G

gain A measure of a circuit's ability to increase or decrease the amplitude level of a signal.

gain structure The relationship between individual microphone and input gains throughout the sound system.

graphic EQ An equalizer that separates the audible frequency range into specific bands (usually 15 or 31 bands) that can be boosted or attenuated by way of faders.

ground loop The hum that results when two or more ground paths exist between pieces of audio gear.

H

harmonics Also known as overtones or partials, these are the frequencies above a fundamental tone that give an instrument or voice its characteristic sound.

Hertz Named after 19^{th} century physicist Heinrich Rudolf Hertz, it is the universal unit of measurement for frequencies and is more often than not written as Hz.

I–J

impedance Expressed in ohms, it's the opposition in an electrical circuit to the flow of current.

jack A female connector designed to be the recipient of a male connector known as a plug.

L

limiter An electronic device used to regulate an audio signal's loudness potential.

line level A measurement of signal strength considered the standard for optimal sound reproduction.

M

mix A combination of sounds that have been carefully blended together for a superior listening experience.

mixing console A sophisticated piece of equipment that features electronic components dedicated to receiving, blending, and distributing audio signals.

monaural Usually referred to as mono, this describes a mix that features no separation of audio signals directed to the left or right channels. Sound is reproduced equally in all the FOH speaker enclosures.

multi-meter A hardware electronic testing device that can be used for a variety of applications.

mute A function in the form of a button found in various parts of a mixing console that makes the signal passing through that section inaudible.

N–O

noise gate An electronic device that's designed to prevent low-level noise from being heard within your system by blocking the flow of signal that is below a specified decibel level.

ohm The unit of measurement that describes the opposition in an electrical circuit to the flow of the current.

outboard gear External hardware processors, separate from the mixing console, that are used to alter, control, and/or shape sound.

P

pad A function in the form of a button found on individual input channels that attenuates the level of the incoming signal before it reaches the pre-amplifier.

pan Short for panoramic, this is a function in the form of a knob found on individual input channels that directs signal to the left or right channel of the master output, thus creating stereo mixes. On consoles with subgroups, the pan knob also directs the signal to a selected group.

parametric EQ An enhanced type of equalizer that allows you to sweep through the frequency spectrum so you can accurately select specific frequencies to boost or cut.

PFL Short for pre fader listen, also known as solo listening. Found on most medium-sized and large-scale mixing consoles, this is a function in the form of a button that allows a channel of audio to be discretely heard using headphones.

phantom power The power required to operate a condenser microphone, which is generated from a mixing console, but could also be drawn from a 9-volt battery or a separate hardware power source.

plug A male connector designed to be inserted into a female connector known as a jack.

polar patterns The three-dimensional pattern of sensitivity in which a microphone hears an audio signal.

polarity A signal's direction of flow from positive to negative.

R

rack A specifically designed enclosure for safely housing outboard gear.

reflow The term used for connecting two pieces of metal with solder.

reverberation Commonly referred to as reverb, this term is used to describe the lush enhancement to the sounds heard within your house as a result of multiple reflections of sound waves in a particular room. It can also be used to describe the digital emulation of this natural phenomenon found in outboard processors.

RMS Short for root mean squared. The method for calculating the average power of an amplifier.

S

signal path The direction an audio signal travels within your sound system.

signal-to-noise ratio The measurement used to describe the ratio between an audio signal and the inherent operating noise of a particular unit.

sound waves Vibrations that move through the air as a result of a solid object that has been stimulated into movement.

SPL Short for sound pressure level. An alternative name for volume or decibels.

standing waves A combination of sound waves created as a result of a particular frequency leaving a speaker enclosure, striking a dense boundary, being reflected back into the room, and colliding with the original wave, yielding unwanted sonic results.

sweet spot The ideal location(s) in your worship space where the mix sounds superior as compared to other locations.

T

THD Short for total harmonic distortion. The unit of measurement used to describe a piece of audio gear's ability to alter or distort the audio signal being fed in and out of itself.

timbre Usually referred to as tone, this term describes the combination of harmonics that gives a voice or instrument its individual characteristics.

transducer A device that transforms acoustic energy into electrical energy and vice versa, such as a microphone or a loudspeaker.

TRS Short for tip ring sleeve. A type of 1/4-inch jack that splits into three regions for several applications such as a balanced cable, a stereo cable, or a send and return connector in one plug.

true diversity The term used to describe a wireless microphone receiver's ability to automatically change from one signal input to another depending on the strength of the signal.

W–X

watts Most commonly used in describing an amplifier's power potential, this is the unit of measurement used to describe the signal strength of electrical current.

wireless microphone A microphone that operates without a cable, connecting directly to the mixing console by way of an internal transmitter that sends the audio signal to a receiver.

XLR A three-pin connector used by microphones, direct boxes, and other pro audio gear.

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