Book Excerpt

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Live Sound For Musicians

by Rudy Trubitt, published by Hal Leonard.

Chapter 1: Sound System Basics

What is a sound system for? Most simply, it makes soft sounds louder and projects them to listeners who would otherwise be too far away to hear them comfortably. This purpose is at the root of the name "public address system," or PA, as sound systems are often called—a person can stand up and "address the public" with a sound system that amplifies their voice.
The same holds true for music. A PA allows you to play for a much larger audience than you could reach without amplification.

However, the needs of each performance vary. For example, dance bands use amplification to motivate their audiences to get up and move. In this case, the sound system becomes an integral part of the act—No PA? No show. On the other hand, a string quartet may need a little volume boost in a large hall, but they'd be fine without a PA in a smaller room.

This brings us to an important point: The use of a sound system should always complement the performance. Every performance is different, and it's very important to adopt a flexible attitude when it comes to dealing with each event. There is no "right way" to do sound, just the best choices (or compromises) for the circumstances at hand.

Think of live sound as a three legged stool. One leg is the needs of the audience, one is the needs of the musicians, and the third leg is the demands of the room. For best results, all three different needs must be met. Ignoring any one of these three areas can result in a performance that falls flat!

The Tasks

So, projecting a louder sound to an audience is our goal. But how can we break this up into individual steps? Easy. All sound systems must perform the same three basic tasks:

- Pick up the original individual sounds (usually with microphones)
- Combine, or "mix," these sound sources and amplify them
- Play the mix through speakers for the audience, or "house."

Before we look at these tasks in detail, one point must be made. We hear by sensing vibrations, or "waves," in the atmosphere. But sound systems work by manipulating little electrical signals. So, a fundamental task of sound systems is that the musical noise we make (with our voices or instruments) must be converted into electrical signals, processed by the sound system and then converted back into sound (using loudspeakers) for the audience.

In the next section of this book, we'll look at the three tasks above, but in more detail. As you'll learn, most of the difficulties in live sound are caused by the conversion between sound waves and electrical signals.

Picking Up the Original Sounds

This is the first and most fundamental task of any sound system. After all, if the purpose is to make an existing sound louder, we must begin by capturing the original sound. The most common tool used to capture sound is the *microphone*. The mic hears sounds in the air (sort of like your ear does), and converts the sound into an electrical signal.

Not all instruments require microphones—some produce an electrical signal themselves. For instance, you can hear the plinking strings of an unplugged electric guitar, but the main sound from that instrument comes from pickups, which generate their own electrical signal from the strings' sound. Other instruments, like electronic synthesizers directly, create their "sound" as an electronic signal.

So, depending on what kind of instruments you are involved with, you may be using more than just microphones. However, almost every amplified performance uses at least a few mics, so we'll focus on them in the first part of this book.

Mixing and Amplifying Sounds

A sound system must usually accommodate more than one sound source. Even if you're a solo guitarist/singer, you must be able to blend, or "mix," two sounds: your voice and instrument. For larger groups, the number of sounds to be mixed will be many more.

Combining the electrical signals of the individual sounds is done by a *mixing board*, or *mixer*. A mixer's knobs are used to balance the volume of all the different sounds connected to it, so that the desired musical blend is heard by the audience.

The mixer doesn't produce sound waves—just another electrical signal that represents the blend of all the connected microphones or other instruments. But before this signal can be connected to speakers, its power must be substantially boosted.

Boosting the signal is the task of the *amplifier*. Also called a *power amp*, this device takes the low-power signal that comes from the mixer and boosts it enough to operate the speakers.

While the microphones, mixer and speakers are always easy to spot in any sound system, the power amplifiers may not be so apparent. While some power amps are housed in their own cases, amps are often built into to the mixer's case (creating a "powered-mixer"). Sometimes, manufacturers will build the amp right into speaker cabinets, creating "powered speakers."

Turning the Mix Back Into Sound

The final task of the sound system is to turn the electrical signal back into sound that the audience hears. This is done by loudspeakers. Within the speaker enclosure, the speakers themselves move back and forth in response to the amplifier's output signal. This physical motion creates ripples of air pressure, and we perceive the result as sound. And, thanks to the amplification provided by the sound system, the results can be much louder than the original sound was.

What's the Catch?

The problem, as was hinted before, is in the conversion to and from air. Sound waves have so much subtle detail that it's very difficult to design a mic that captures all their nuances. Also, microphones don't really "hear" sounds the way our ears do—even good mics can "color" the sound of an instrument, especially if the mic is not properly positioned.

The flip side of the coin is the speakers. Again, the task of translating the complex electrical signal into an accurate motion in the air is demanding. The loss of accuracy in the translation can result in bad sound. And, just as microphones don't pick up all sounds the way they originally occur, loudspeakers don't necessarily project sound the same way that the original instruments do. The result can be "uneven coverage," meaning that the final sound may be good for those sitting in front of the speakers but not-so-good for the audience off to the sides.

Review

OK, time for a quick review: The purpose of a sound system is to make a quiet sound louder. To do this, you must first capture the original sound (your voice or instrument). This is usually done with a microphone, although some instruments have pickups instead.

The mics or pickups translate the original sound into a faint electrical signal. This is connected to a mixer, which boosts that signal and then lets you balance the levels of all your individual mics. The electrical signal created by the mixer is sent to a power amplifier, which boosts the level of the signal again. This high-power electrical signal is then connected to loudspeakers which turn the signal back into sound for the audience.

Converting sound to and from an electrical signal is difficult; therefore, most of the problems associated with live sound are related to microphones and speakers, and their interaction with the room and the musicians.

Before we go any further, a quick word of caution. There are three basic safety issues related to live sound: hearing protection and conservation, stable positioning of equipment to prevent injury or damage from falling gear, and avoiding electrical hazards. We'll touch upon each of these from time to time, but it's beyond the scope of this book to fully address these safety concerns. Education is a continuous process. Seek out people and resources that can increase your knowledge of these crucial topics.

fundamental concepts and terms

As you've probably guessed by this book's casual tone, we're not going to plunge into the icy waters of mathematical formulas to explain how sound systems work. However, you are going to have to come to grips with a few basic concepts. But fear not! This book is certified Musician-Friendly, 100% math-free.

Sound is pretty simple: An object vibrates. This movement pushes air back and forth, which sends out little ripples of changing air pressure. As these little waves wash up against your ears, your eardrums wiggle back and forth in sympathy. The resulting motion is converted into nerve impulses, which in turn are perceived by your brain as sound.

What makes one sound different from another? Two important things: How fast the vibration is and how vigorous it is. The faster the motion, the higher the pitch we perceive. And the greater the back-and-forth motion, the louder the resulting sound.

The technical terms for these two fundamental properties of a sound are *amplitude* or *level*, which describes how loud something is, and *frequency*, which you may consider analogous to pitch—higher frequencies make higher notes.

Level

Anyone who's played a gig with a rock band has heard the phrase "You guys sound great, but you're going to have to come WAY down!" Hence the concept of volume, or level: How loud is it? (Several terms are used to describe this, including volume, level and loudness. For our purposes, they are interchangeable.)

A sound's level is directly related to how vigorously it is vibrating. If something's barely moving back and forth, it will make very little sound. But if you really whack it, it will vibrate a lot harder. This will move more air and the result will be a louder sound.

Most discussions about volume are relative. For example, "We sound great, but *I* should be louder!" But how loud is loud? Sound engineers measure the absolute level of a sound, or the difference in level between two sounds, using the "decibel." You may have seen charts showing the range of human hearing for sounds of different loudness, starting with a quiet room and progressing to lawnmowers, rock concerts, gunshots and jet-plane take-offs. Next to each of these example sounds is a number that shows approximately how loud that particular noise is in decibels.

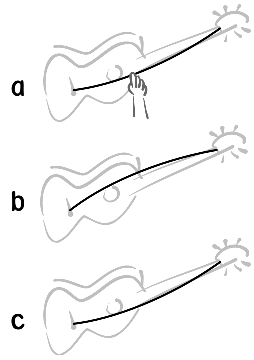
While you probably won't ever be asked to adjust your band's volume to "96 dB sound pressure level," professional sound system engineers often work under these restrictions, especially when concerts are held near residential neighborhoods. Using a device called a sound pressure level (SPL) meter, the actual level of sound can be measured. Radio Shack sells a relatively inexpensive (under $75) SPL meter. If you're curious, this device will give you a much better idea of how loud "loud" is!

Frequency

Besides the loudness of a sound, we are also sensitive to a sound's frequency. In musical terms, frequency is closely related to pitch. The higher a note's pitch, the higher its frequency. For instance, hitting a bass drum has a completely different result than hitting a triangle. Since the bass drum vibrates more slowly than the triangle, we hear its sound as being lower in pitch.

Note that every vibrating object wiggles at more than one single frequency. This explains why you can hear two people singing the same note, yet still be able to tell the performers apart. Both singers produce the same root, or *fundamental,* frequency when they sing the same pitch. However, every person (or instrument) creates its own blend of other, related frequencies called *harmonics*. These harmonics are what let us differentiate one singer from another. There are many instrumental examples of harmonics; an exploration of these is left to you as an extra-credit assignment!

Frequency is measured according to how many vibrations (or cycles) are completed in one second (a *cycle* is one complete vibration, for example, bending outward, bending inward and then bending outward again).



Sounds are created by vibration, in this case, by the motion of a plucked guitar string. Here, a, b and c show one complete cycle of this vibration, starting with the string being pulled and released (a), bending upwards (b) and then returning to the point where it was released (c). Count the number of times this cycle is repeated in one second and you know the frequency of the vibration. If the string was a guitar's open "A," the frequency would be 110 Hz (Hz is shorthand for "cycles-per-second").

Sound frequency is measured in *cycles-per-second* or *Hertz*, abbreviated *Hz*. The open A string of a guitar (in standard tuning) will vibrate back and forth exactly 110 times in one second.

While you can translate between musical pitch and specific frequencies, you'll normally use one or the other. When talking about music, you'll just say "open A," not "110 Hz." Similarly, only frequency is used when discussing sound equipment. You will find controls that apply to specific sound frequencies, but these will never reference musical note names. However, you can use your trained musical ear to identify different frequencies by ear. This will be especially important when trying to control feedback (more on this later).

Our Range of Hearing

You know from experience that we can hear very low and very high-pitched tones. The typical range of human hearing (in younger people) is from 20 cycles per second to 20,000 cycles per second (which is often abbreviated 20Hz-20kHz).

It is convenient to break this "frequency spectrum" into smaller ranges or "bands" to help us talk more specifically about what we are hearing. The simplest division is to split our range of hearing into three parts: bass or "lows," midrange or "mids," and treble or "highs." Your sound system must be able to reproduce each of these three frequency bands with reasonable accuracy.

Midrange Sounds

Let's start with a quick overview of the all-important midrange. Humans rely heavily on midrange for understanding speech, because this is where the consonant sounds reside. If you can't hear consonants, it's very hard to tell the difference between, for instance, "cake" and "ache."

If you get a hearing test (and you should, every year), you'll find most audiologists are only interested in testing your low- and midrange-frequency hearing. Think of the telephone. You have no trouble using it to communicate, but listen to how lousy the "on-hold" music sounds. The telephone is all mids, with zero lows and highs.

So, you would be correct to assume that good midrange response is important for a sound system used for the spoken word, or if you want people to understand the words you are singing.

Midrange is important for more than just speech. Many instruments produce a lot of sound in the midrange. Playing these instruments through a sound system lacking mids will not seem very energetic. However, this is rarely a problem: most smaller sound systems tend to have too much midrange response and produce a sound that is abrasive and tiring to listen to for extended periods.

Just remember that our ears are most sensitive to midrange sounds, so it doesn't take much midrange to wear us out, especially as we get older. Perhaps this is why parents can't understand why their kids like such loud, blaring music.

Bass or Low-Frequency Sounds

Bass or low-frequency sounds add impact or "oomph" to music, especially modern pop or dance music. I'm sure you are familiar with the types of instruments that generate lots of low-frequency sound, including bass guitars, upright basses, electrical keyboards, larger drums and so on.

A sound system that has good low-frequency response will help motivate an audience to dance. However, this will require a sound system with larger speaker cabinets and more powerful amplifiers.

Conversely, a sound system with poor low-frequency response isn't a good choice for an amplified dance band trying to beef up their low end. However, for sound systems that are primarily for vocals, lack of extended low frequency "thump" won't be noticed.

Treble or High-Frequency Sounds

Finally, there are the treble or high frequencies. Remember when I said that most instruments generate sound at more than one frequency at a time? While the fundamental pitch of a guitar's open A string is a relatively low 110 Hz, that sound is rich in harmonics, additional frequencies in the octaves above the fundamental. The high-frequency range of your PA system will mostly be busy reproducing these harmonics.

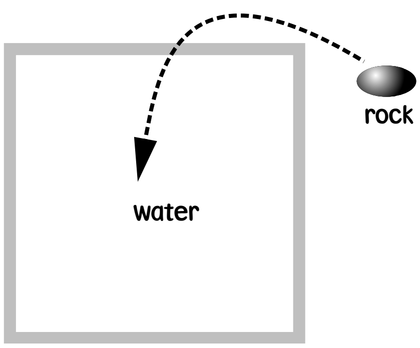
A sound system with good, smooth high-frequency response will have a pleasing, open or "airy" quality. One might also call it "hi-fi." A lack of highs will leave your system sounding somewhat dark, closed or veiled. Most PA speakers don't have the extreme high-frequency response found in good home-stereo systems, but most will do an adequate job. As always, too much of a good thing is too much of a good thing. Excessive highs can make your system more prone to feedback (more about that later) and also make any background hiss generated by your equipment more obvious to listeners.

The Room

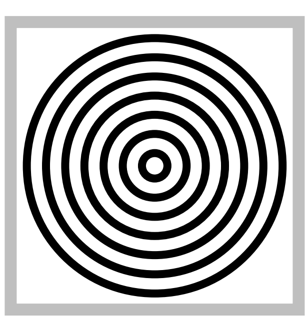
Many live-sound difficulties are imposed by the venue you are performing in. Are you playing outdoors in a field? In a basketball gymnasium? In a tiny club or ballroom? Every space has its own unique "acoustical" properties.

What is important is how a given room responds to sound waves. At first, this may seem like a rather abstract concept, so let's try a little analogy:

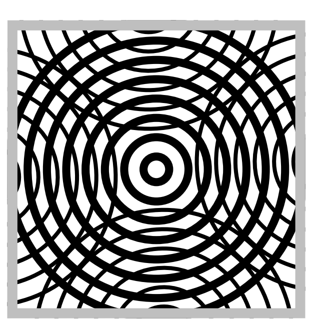
Sound waves in a room are similar to wave action in water. If you toss a stone into water, circular ripples will spread evenly in all directions. But as the individual waves run into obstacles, they are reflected back. What started as a clean set of concentric rings quickly becomes a jumbled mass of overlapping waves, bouncing in all directions.



The behavior of sound in a room can be illustrated by tossing a rock into a box filled with water.



The initial impact of the rock generates a clean set of concentric waves that move outward from their source.



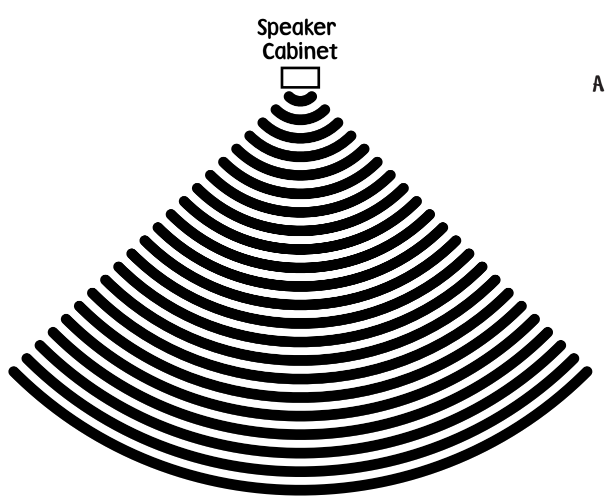
However, as each wave hits an obstruction (a wall, for example), a slightly weakened wave is reflected back. What was simple becomes a complex pattern of overlapping waves. When this example is translated into the world of sound, the reflected sound waves add muddiness and clutter to your overall sound.

The same problem occurs with sound in any partially enclosed space. Your original sound waves bounce off walls, floors, ceilings, etc. and then return to the ears of you and your audience. However, since the speed of sound isn't all that fast (only about 1,100 feet per second), the reflected sound waves arrive at your ears noticeably later than the original, non-reflected waves. This is why you can clap your hands in a large room and then hear an echo back a moment later.

While fine concert halls are known for the smooth and pleasing reflections generated by a performer on stage, most acoustical spaces, ahem, *suck* when it comes to amplified music. The myriad of individual little reflections clutters up your sound, makes it hard for the audience to understand the words, promotes feedback and disturbs the relative balance of different parts of the frequency spectrum. In short, it can be a real disaster. Later, we'll talk about a few ways to minimize the problem, but it can never be overcome completely.

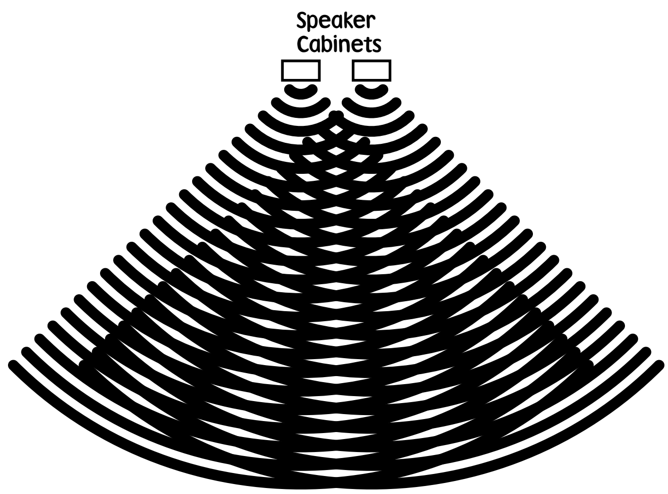
Yet Another Problem

When sound waves are coming from two or more places at once (such as two speakers on either side of a stage, or one speaker and sound reflecting from an opposing wall), the frequency response of your sound is altered at different places in the room. The cuprit is wave interference patterns.



single speaker projects sound without interference,
until it is reflected off walls or other obstructions.

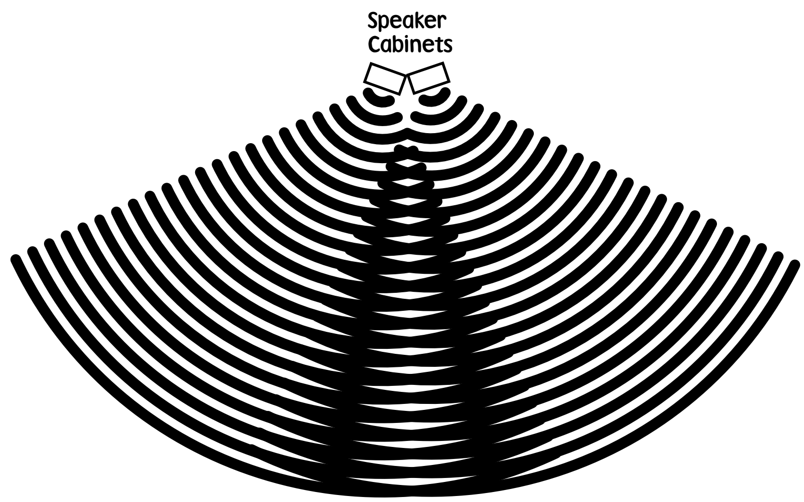
Imaging throwing *two* rocks into a still pond. Two sets of ripples spread from the impact points. These wave patterns overlap. In some places, the crests of each pattern line up, making the wave twice as high at that spot. In other places in the pond, a wave crest caused by one rock lines up with a wave trough created by the other rock. The result? The two cancel each other, and the pond is flat at that particular spot.



Add a second speaker to the equation and all hell breaks lose. The dark "spokes," show areas where sound at a particular frequency will be canceled. Looking at different frequencies or changing the distance between speakers moves the spacing and number of these spokes. Nothing can make them go away!

The same thing happens with sound. At any point in the room, some frequencies will be strengthened by the overlap pattern between your speakers. But in the same spot, other frequencies could be canceled! If you walk around in a room during a performance, you'll hear the tonal balance of the sound change. And the number and spacing of these interference bands changes with frequency. Note that you will still hear sound when standing in these interference zones. This is because the cancellation is at one specific frequency, not at all frequencies. You won't hear *no* sound, just *not-as-good* sound.

The only practical advice I can offer about this problem is this: Avoid setting up two speakers side-by-side that are pointing in exactly the same direction. Either move them at least as far apart as the distance to the closest member of the audience, or if they must be next to each other, angle them away from each other by 30 to 45 degrees. This will reduce the areas subject to the cancellatio caused by the interference patterns.



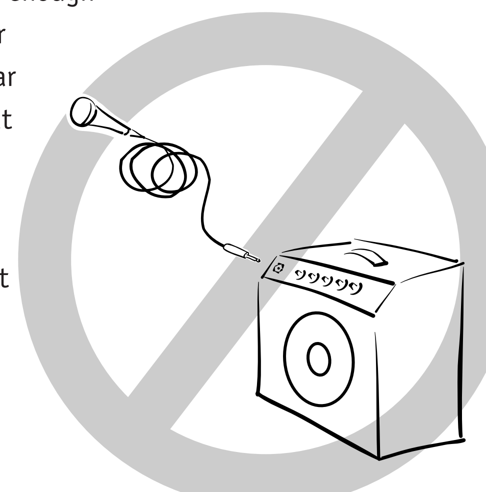
If two speakers must be placed side-by-side, angle them away from each other to minimize the interference cancellation between speakers.

I know, I know. You just wanted to run your PA without feedback, and here I am telling you that it will never, under any circumstances, sound perfect. Sorry. I just thought you'd want to know. But cheer up—we're going to make some noise in the next section!

the basic sound system

Having now examined the most important fundamentals of live sound, we're ready to turn our attention to the main event: using a sound system.

Why Can't I Use a Guitar Amp for Vocals?



Let's look at the requirements for a system that will deliver "good enough" results. If you play music that uses electric guitars, you're familiar with the guitar amplifier. You may have a microphone with a guitar cord-like output and figure "Hey! I can plug my mic right into that guitar amplifier and use that to sing through!"

If you try this, here's what you'll find: First, most guitar amplifiers have only one or two inputs, which won't be enough for most groups. This means you won't be able to connect enough microphones, and those that are connected may not be individually adjustable.

Aside from the lack of individually controllable inputs, there is a second problem: sound quality. A guitar amplifier works fine for an electric guitar. But putting a singer through a guitar amp results in a somewhat harsh, "honking" tone—in short, it just doesn't sound good. This is because a guitar amplifier is designed only to re-create the sounds prominent in an electric guitar. The wider range of frequencies present in the voice are poorly reproduced by a guitar amp.

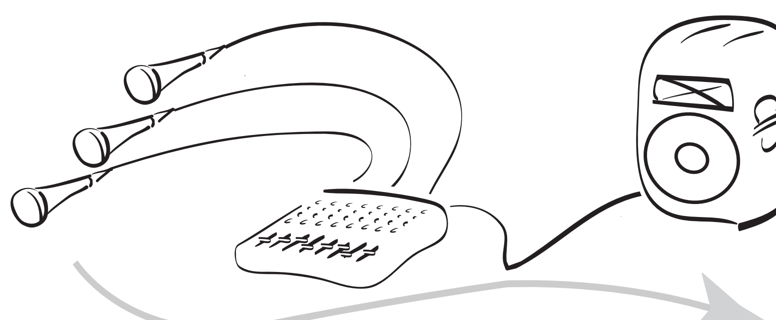
How about using your stereo? Stereos aren't designed to accept microphones, so you'll still need a mixer to run your mics. While the output of a mixer could be connected to your stereo's line inputs, there's a chance you'll damage your stereo amp or speakers if you run them too loud. And if you need to take everything to a gig, you're likely to scratch, dent or break your hi-fi, because it's not built to take the pounding demanded of a portable sound system.

the pieces

So, if a guitar amp or home stereo doesn't provide us with "good enough" sound, what does? There are four basic components required: one or more microphones, a mixer, an amplifier and loudspeaker cabinets (usually two).

To run a sound system, you must know where to point the mics and speaker cabinets and how to set the knobs on the mixer/amplifier. Keeping track of this is easier when you have a mental picture of how sound travels through your system.

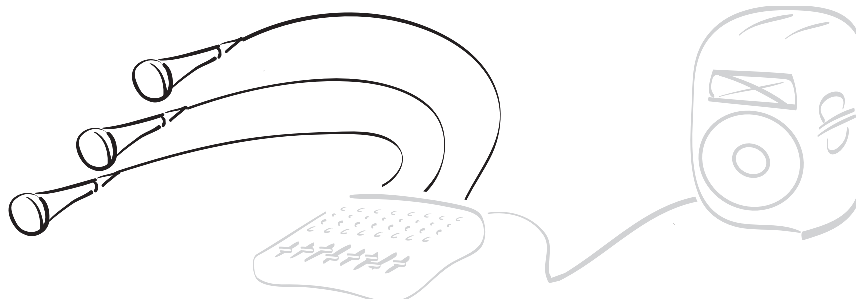
Here is the most important concept in this book, so pay attention! Sound travels through your sound system in one direction. It starts at the microphones and goes into a mixer, an amplifier and then out to the speakers. You may find it helpful to think of this signal flow as a series of streams that all flow to a river and into the sea. Each little tributary is like a single microphone. The small streams join together into a river (the mixer) and evntually reach the sea (the speakers).



Signals in a sound system always flow from the microphones, through the mixer and out through the speakers. A clear understanding of your sound system's signal flow is an invaluable asset for operation and troubleshooting.

Microphones

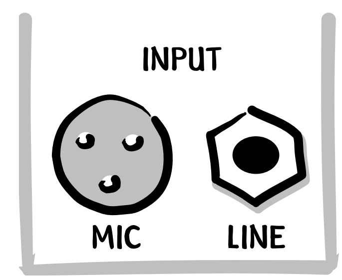
The first step on the journey are the microphones that capture the sound of the performance. The quality and placement of the microphones has a very large effect on the overall sound. Generally, you'll want to place the microphones quite close to te sound source you are trying to capture.



Sound enters the system through microphones.

As an alternative to microphones, some instruments (such as acoustic guitars, violins, etc.) may be fitted with instrument pickups. Pickups convert the sound of the instrument into an electrical signal that is connected to your PA's mixer. We'll talk about the differences between mics and pickups shortly, but from an operational standpoint, they are used in much the same way.

Remember that quality isn't cheap. A good microphone can cost hundreds of dollars. While this may seem like an extravagance, quality microphones will have the single biggest effect on your sound system's performance.

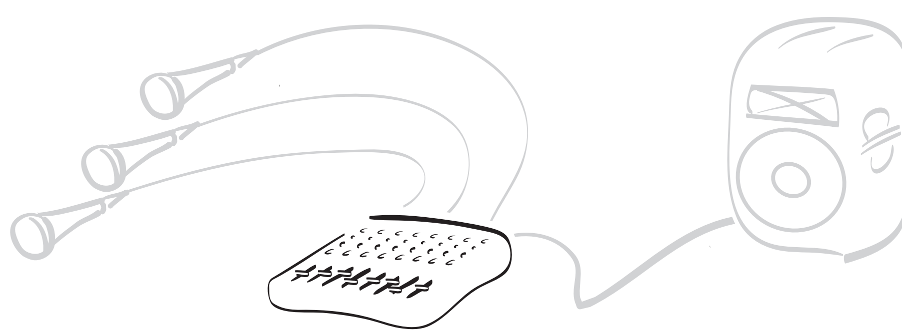


Many mixers will have connections to accept both mic- and line-level signals. If you have mics that use 1/4" phone connectors (that mate with the input marked "line" on the right), an input channel like this may prevent these mics from operating at a loud enough level, since you will be plugging them into a line-, rather than mic-level input. This is one reason why mics with XLR outputs are preferred.

The Mixer

The mixer controls the balance between microphones and, therefore, the musical balance between players. A musical ear is a big asset when adjusting a mixer's controls. However, some technical skills are also required to avoid problems that can result from inappropriate knob-twisting!

The mixer takes the sound from each of the microphones and other inputs and combines them. This mixed electrical signal is then connected to the input of a power amplifier which in turn drives the loudspeakers.



The mixer combines the input signals.

The Amplifier

After the sounds are mixed, they are sent to a power amplifier. The purpose of the power amplifier, or "amp" for short, is to take the electrical signal created by the mixer and boost its power enough so that it can be used to drive your loudspeakers.

The amp may be a separate piece of equipment, or built into the speaker cabinets themselves (as shown here). In other cases, the amplifier may be built into the mixer. The one thing you can be sure of is that there *will* be an amplifier in your PA system, either as a separate piece of equipment or built into another part of the system. If there's sound coming out of the speakers, there's a power amp working somewhere!

Loudspeakers

Finally, the output from the amplifier is connected to one or more loudspeaker cabinets. These convert the powerful electrical signal created by the amplifier into sound that we can hear. Like microphones, the quality and placement of loudspeakers has a big impact on the sound your audience hears.

The internal amplifier often built into speaker cabinets boosts the signal from the mixer and feeds the speakers, converting electrical signal into mechanical motion, creating sound.

Signal Levels: Mic, Line and Speaker

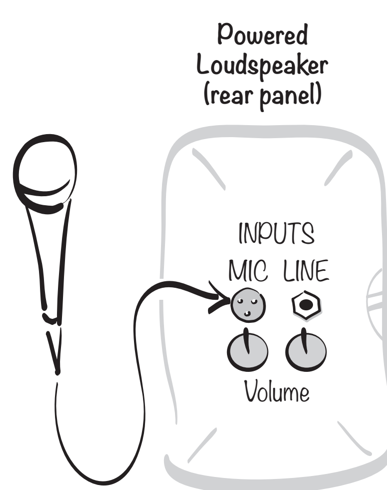
As you can see, a PA works by capturing a sound wave with a mic and converting it into a teensy electrical signal. Then, that signal is manipulated and boosted in power by the mixer, and then boosted once more by the amplifier so it can drive the speakers. This means that the electrical signals representing our sound exist at different strengths—lowest-power signals from the mic, medium-power signals from the mixer, and high-power signals from the power amplifier.

These three different “signal levels” have easy names: they are “mic-level,” “line-level” and “speaker-level.” As you might guess, mic-level signals are the lowest power, generated by the microphones. Line-level signals are stronger than mic-level ones. An example of a line-level signal would be the output of a CD player or tape deck, or a mixer’s output. Speaker-level signals are much more powerful and are created by power amplifiers.

It’s important to understand these differences, because a particular input on any piece of sound equipment will be expecting one of these three signal strengths. Plugging a low-power signal into a device that expects more power won’t work. For instance, plugging a microphone into a speaker cabinet won’t make a peep. The reverse can damage equipment—plugging a power amplifier’s output into a mic- or line-level input could damage that input channel.

Powered Speaker/Tiny Mixer

Some powered speakers include a teeny-tiny mixer on their back panel. In this case, you can plug a mic and/or a keyboard, guitar, etc, right into the back of your loudpspeaker. (Remember our solo singer guitarist from the start of this chapter?) Talk about traveling light! Nevertheless, this sound system still includes all the basic components: mic, mixer, amplifier and speaker. It’s just that three out of four are built into the same piece of equipment.

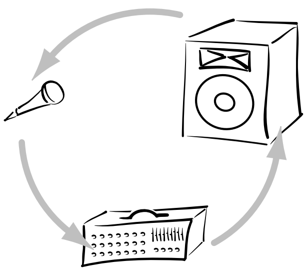


A powered loudspeaker may include a basic mixer (a couple inputs with volume knobs) built right into the speaker cabinet’s back panel.

the problems

Having the necessary equipment (mics, mixer, amp and speakers), properly connected, is the first step towards "good enough" sound. But even with decent gear, there are still major problems to avoid. Some are momentary glitches (like feedback), some are fundamental limits of the equipment (distortion and frequency response) and others affect the sound quality in different parts of the audience area (coverage).

Feedback

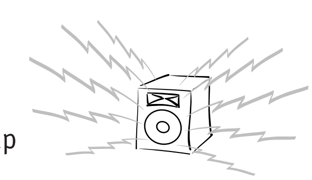


Feedback is the most common problem with live sound. It's that painful squealing or howling sound that makes everyone cover their ears and glare at the sound-person. It's caused when a microphone is turned up too loud or is too close to the PA speaker. The mic "hears" its own sound coming from the speaker and this sets off a vicious cycle that quickly builds to a howling racket. We'll talk more about the causes and solutions for feedback later.

Distortion

Feedback isn't the only problem created by turning your PA up too loud. Even if the system doesn't start squealing, it will begin to *distort*. If you're an electric guitarist, you're familiar with the sound of a guitar amp turned up—distortion adds a bright, crunchy tone to the sound. However, distortion in a sound system reproducing several singers and other instruments results in harsh blurring of the sound (and, in extreme cases, can eventually burn out your speaker's tweeters).

Unclear on the concept? Distortion can be illustrated by a water faucet: As you begin to open the faucet, the flow of water smoothly and steadily increases. But eventually you will reach a point where as much water as possible is splashing into the sink—opening the faucet further just doesn't make the water come out any faster.



The same situation exists with sound systems. Pretend the faucet is your system's volume knob and the water is the sound. As you begin to turn up the knob, the sound gets louder. But at some point (even though the volume knob may be only "halfway" up), turning up doesn't make the sound louder—it just makes it more distorted. Distortion can appear in all parts of your system, from microphones, mixers, amplifiers or speakers. Shortly, we'll discuss strategies to try and avoid distortion occurring in each of these components.

The bottom line for solving distortion problems is, find out which part(s) of your PA is distorting and turn the level of that part of the system *down*.

Tonal Quality

Distortion in your system can and should be avoided. But just because a system isn't distorting doesn't mean it will sound good. We want something that sounds "full and smooth," not "thin, harsh and annoying." *Frequency Response* is one term used to describe how accurately a sound system is reproducing (or "responding to") the original sound at every individual frequency within our range of hearing.

Huh? OK, how about an analogy using less technical terms. Let's use a visual analogy of sound, the rainbow. A full rainbow (with colors ranging from red to violet) contains all the colors of the visible spectrum. This is analogous to the sound of a full orchestra which includes very low-frequency sounds (like a bass drum) up to very high ones (like a triangle or piccolo), and everything in between.

Imagine you've taken a photograph of a full-color rainbow. But when the pictures are developed, the red and violet colors that seemed so clear in real life are drab and muted, or perhaps missing completely in the photo reproduction. Why? Some part of the reproduction process did a poor job of preserving the extreme low- and high-frequency parts of the color spectrum (which are red and violet, respectively).

The same thing happens with sound systems. While the original sound of your music might be full of deep low tones, or bright, shimmering highs, the PA system may not be able to reproduce these accurately, losing them in the process. A good-sounding system will preserve as much of the original sound as possible. A not-so-good system with *limited frequency response* might be adequate for voice (the telephone, for example, has very limited frequency response). But for good sounding music, a wide, relatively even frequency response is required.

Covering the Entire Audience

Finally, the sound system must be able to project good sound quality over the entire audience area. As it turns out, this is a very tricky problem. In fact, *no* sound system on the planet is able to guarantee that each seat in the house will get the same sound quality. It's a fact of life—the character of the PA's sound will change as you walk around the room, and as we already discussed, the acoustics of each room cause their own additional problems. There are no simple solutions to this problem, but I'll give you some tips when we return to the issue of "coverage" in Chapter 3.

Quick hint: if your speakers aren't pointing in the direction of the majority of your listening audience, expect to have coverage problems. So point those cabinets in the direction you want the sound to go!

review

Congratulations! If you've made it this far, you have been exposed to the most important things about sound systems and their operation.
Let's review them.

About Sound

The two most important components of a sound are:

- **Frequency**, which is a measure of how rapidly an object wiggles back and forth while generating sound waves.
- **Amplitude**, which measures how vigorously an object is vibrating. Greater physical vibration (amplitude) results in louder sounds.

Parts of a System

There are four basic categories of equipment required for an adequate sound system:

- **Microphones**, which capture the sound of your voice or instrument and convert it into an electrical signal. Instrument pickups are an alternative to mics for certain types of instruments.
- A **Mixer/Amplifier**. The mixer is used to combine and balance the sound of all connected microphones and pickups. The amplifier boosts the power of this "mixed" signal to a level where it can run a loudspeaker. The mixer/amplifier can be combined into a single piece of equipment (the "powered mixer") or they can be separate.
- **Loudspeaker cabinets**, which take the electrical signal from the amplifier and convert it back into sound waves in air.
- **Cables** and **stands** to hook everything up.

Signal Levels

There are three different "levels" of electrical audio signals:

- **Mic-level** signals are generated by microphones and most types of instrument pickups.
- **Line-level** signals are generated by devices such as CD players, synthesizers and other electronic keyboards and mixers.
- **Speaker-level** signals are generated by power amplifiers.

Common Problems

Finally, there are five major problems to watch out for when setting up and operating your sound system. They are:

- **Feedback**, the squealing sound caused by mics being too close to speakers.
- **Distortion**, a harsh, crunchy sound caused by operating at too-high volume settings.
- Issues of **Tonal Quality** or frequency response, which determines how accurately (or pleasantly) your original sound is being reproduced.
- **Coverage** of the audience, so that everyone gets decent sound, regardless of where they are sitting in the room.
- **Room Acoustics** can sabotage the best-laid plans. Some rooms are easy to perform in; others will be nothing but trouble, especially those that are large and filled with hard, flat surfaces like glass, tile or concrete.

In the next chapter, we're going to cover the equipment and technique required to address each of these critical points.