How Live Sound Systems Work

Jesse Barnes

Imagine you're sitting in a beautiful Broadway theatre watching a thoroughly entertaining musical with vibrant lighting and pristine sound. What would happen if you took away the speakers and microphones and could only hear the sound coming from the stage more than 50 yards away? The show would lose much of its excitement. Most of the music would not be heard at all. You would have a lot of trouble understanding what was happening in the scene.

So how does the sound get from the actors and musicians on the stage to every corner of the theater in a way that is both intelligible and exciting? This may seem like a simple process, but it is very complex. Hundreds of sound waves are being manipulated in countless ways before they reach your ears.

Theater sound systems are designed to go unnoticed by the audience, but if something went wrong with the sound the entire performance would be ruined for most viewers. Because of this challenge, and the inherent physical limitations of sound waves, sound systems demand a high degree of skill from designers. Using tools and techniques like speaker positioning, digital signal processing, microphone placement, and sound measuring devices, a designer ensures that every seat in the house is receiving the sound from the stage as naturally as possible.

Before describing all of these technical concepts and how they are used I will begin by explaining the path that sound takes after it leaves the stage and before it reaches your ears.

An audio signal is a sound wave that has been converted to an electrical voltage by what is called a transducer. A typical sound system has two types of transducers: the microphone and the speaker. Microphones receive the physical pressure fluctuations of a sound wave and, using a sensitive electrical diaphragm, convert these vibrations into positive and negative voltages. This electrical current flows through copper wire to some processing equipment (typically a mixing console) where it is amplified and manipulated in various ways. The signal then travels along another wire to a power amplifier, where the voltage is increased enough so that it can drive magnetic coils inside a speaker. These magnets then cause the cone of the speaker to move back and forth, to convert the electrical signal back to physical pressure—another form of transduction. The result is an amplified version of the sound wave that was picked up by the microphone.

You can trace this signal path through your home stereo. Substitute the microphone for a turntable needle and the processing equipment and amplifier for the receiver. The signal passes through copper wire to a pair of speakers. The only differences between this simple home setup and a Broadway show or touring rock-and-roll concert are the number of microphones, the number of speakers, and the amount of processing equipment.

The sound industry has more competing suppliers and manufacturers than any other entertainment technology field. With hundreds of microphone, speaker, processor, and amplifier types available, each suited to a particular use, sound system designers have to decide what equipment will be suitable for the show that they are designing. After the initial assessment of the event and the venue, they may begin by determining what microphones will be needed.

Since the microphone is the first piece of sound equipment in the signal path, it is often considered the most important. Technicians in the sound field have a popular adage: “Garbage in, garbage out." A poor representation of the sound source from the microphone will normally result in poor sound delivered by the speakers.

There are three important specifications to consider when choosing a microphone: its pick-up pattern, its frequency response curve, and its sensitivity.

The pick-up pattern is the direction in which the mic will “hear” sound. There are three types of pick-up patterns: cardioid, or heart-shaped; omni-directional; and figure-8 or bi-directional.

A cardioid microphone will have a zone of cancellation behind the microphone, so it will pick up sound best directly in front. This is the most common type of microphone used in live music, because it attenuates most of the sound that is not coming directly from the sound source. Almost every vocal microphone on the market has a cardioid pick-up pattern.

Just as it sounds, the omni-directional microphone will pick up sound in a 360º field. With many different designs, these mics can be used for a variety of applications. An area or zone microphone can be placed on any flat surface to pick up sounds in the immediate vicinity. Often used in theatre when there are many actors on stage, it can eliminate the need to “mic-up” each individual performer. The lavalier or wearable microphone uses an omni-directional pattern to amplify individual voices on stage. Because of its size and shape, the lavalier can be hidden in an actor’s costume, hair or make-up.

The figure 8 or bi-directional microphone will pick up sound directly in front or behind, but attenuate sound coming from either side. Live sound engineers and designers rarely have applications for this pattern. In recording studios it can be used when two singers want to sing into one microphone at the same time. They can face each other and sing into opposite sides of the same mic.

Microphones are usually designed with a specific frequency range in mind. Some are best for picking up the lower range of the audible spectrum and are used to amplify the sound of a kick-drum, bass guitar or some other low- range instrument. Other mics are tailored for vocals, and may have a slight increase in amplitude in the frequency range of the human voice. Most microphone manufacturers will illustrate this information with a graph known as a frequency response curve, which can be downloaded as part of a specification sheet from the company’s website. Sound designers and engineers usually have an arsenal of microphones that they know will work well for specific instruments or vocal ranges.

A microphone's sensitivity is defined as the way it responds to fluctuations in volume level. An extremely sensitive microphone is not well suited for amplifying a very loud sound such as a trumpet; however, for some soft-spoken actors it can be very helpful.

You may now begin to understand how complicated the process of sound system design and implementation can be, but I have only scratched the surface. I have covered the first link in the signal flow chain, the microphone. Now I will jump to the end. As with microphones, speakers have a wide range of styles, which the designer must choose based on what the physical space requires.

When a designer is hired to install a sound system in a venue the first thing he or she must consider is the architectural and acoustical features of the space. How large is the room? How deep? How tall? Are there balconies? The goal of the design must be to deliver equally clear sound to every seat in the house. This is rarely easy. A large theatre can require hundreds of speakers. There may be obstacles like pillars, balconies, or walls. The sound designer may have to work around scenic elements, since the sound system is usually installed after all of the scenery has been built.

The physics of sound waves must also be taken into consideration. Sound is cumbersome and slow, traveling at 1160 feet per second. As it hits a surface it will bounce off in every direction. A system designer spends days taking measurements and making calculations before he or she can even begin choosing speakers for installation.

Speakers, like microphones, are designed for different functions. Sub-woofers, which deliver low-end frequencies, are the largest and heaviest speakers. Mid-range and high frequency speakers are often built into one box called a 2-way, or 3-way speaker enclosure. Speakers can be self-powered or amplifier driven. Self-powered speakers have amplifiers built in while non- powered speakers require a separate amplifier to drive them. Amplifiers, as explained in the signal flow section of this article, boost an audio signal to a high enough voltage to "drive" the speaker. Driving the speaker means giving the magnet inside enough power to make the cone of the speaker move back and forth, which reproduces the audio signal so that it can be heard.

Two types of speakers are most often used for live entertainment: the point-source and the line-array. A point-source speaker has a wide dispersion angle and is meant to stand alone or as part of a cluster of two or three speaker enclosures. A line-array speaker has a very narrow dispersion angle and can be combined with many other speakers, to form an array and cover a large vertical area. Line-arrays are needed in venues with tall ceilings and multiple balcony levels.

In large theaters speakers may be required to fill in areas that the main speakers are unable to cover. These are usually patterned in concentric rings with the front-center edge of the stage as the center point. Since each of these rings of speakers needs to deliver the sound from stage at a slightly later time, they are called delay rings. If all of the delay rings played back sound at the same time the people in the back of the house would hear the speakers closest to them first. Sound from speakers farther away would arrive later and would be heard as an echo. For this reason a designer has to calculate the length of time necessary to delay each ring of speakers so that the sound arrives at the listeners' ears at the same time as the speakers over the stage.

I'm sure your mind must be swimming now. Delay is one type of processing that can be done to the audio signal before it reaches the ears of the listener. The next section will cover more processing and how it can be used in a live sound system.

To a layman the processing step in a sound system may seem like the most complicated. The equipment is often large and has many knobs, faders, touch-screens and buttons. All of these processors are made to control only a few parameters of an audio signal: gain or volume level, time, routing, and polarity. For the purposes of this article I will not explain polarity. It is a complex concept, which involves manipulating the positive and negative voltages of the electrical audio signal.

The most important piece of equipment for processing a signal in a sound system is the mixing console, or mixing desk. In many modern sound systems all of the processing is done inside this one unit. The digital sound console can replace a large amount of equipment that was necessary to include in a sound system when analog technology was the only kind available. The basic difference between analog and digital processing equipment is that in digital gear the electrical signal coming from the microphone is converted to digital information that can be manipulated in countless ways by computers before it is converted back to an electrical voltage to drive the speakers. The computers inside digital equipment do the majority of the processing work in a modern system.

Inside the mixing console the first sound parameter that can be controlled is gain. The term "gain" is usually used instead of "volume," because the signal is already at a certain level when it reaches the mixing console from the microphone. Mixing engineers add or subtract gain to set a level that can easily mix with the other audio signals coming into the console. This balancing of gain between all of the audio signals is the mix engineer's primary responsibility.

Equalization of the signal is achieved using gain levels. By filtering out or increasing the gain of certain frequencies in the signal the engineer can dramatically alter the sound. Gain is also used to separate sounds across the stereo field. By increasing the level to the left or right speakers, the engineer can make it seem like a sound is coming from that side of the stage. This process is called panning.

Another way to control gain is by using a compressor, which sets a volume limit, called a threshold, above which sound will be attenuated. Compressors are used when a singer or instrumentalist has tremendous dynamic range. If they suddenly get very loud the compressor can balance out the sound with the rest of the performance.

Delay, as explained earlier, can be used to align a system by making sure all of the sound arrives in a location at the same time. Sound engineers also use time manipulation to create different effects. The term delay can be used to describe an echo effect, often added to the voice for a more dramatic sound. Reverberation is the natural effect created by a sound bouncing off of many surfaces in a space. When simulated digitally by modern processing equipment, reverb can make a voice sound more natural and is often necessary in small rooms when the engineer wants to create the feeling of a larger space.

When a sound system has many speakers in different locations, an engineer or system designer may not want every sound source to go to every speaker location. For example, if the engineer has a loud band with blaring horns he or she may not need to put the horns in the speakers directly in front of the stage. This is where routing comes in. It is a way of steering the sound signals where you want them to go, not just to the various speakers, but also to effects, equalizers, compressors, or other processors either inside the mixing console or through another cable to a different unit.

So now you can begin to understand what the mixing engineer is doing with all of those knobs and faders. As a mixing engineer myself I often feel like some audience members think there is just a large volume knob that can be turned up or down to control the sound. If only it were that simple. Hopefully this article will give its readers a better understanding of how complicated this business can be. Working in the field of live sound is a life-long learning experience.

Nominating faculty: Professor Robert Leston, English 1133, Department of English, School of Arts & Sciences, New York City College of Technology, CUNY.

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