The Basics of Sound Reinforcement

By John Dziel ©2017

Contents:

This book is designed to be a base-line primer on sound reinforcement, not a comprehensive guide.

Concept- Public Address vs Sound Reinforcement

Although often confused, the two terms “PA system” and “Sound Reinforcement” refer to two very different types of systems. To understand this, we need to look at the applications of each:

PA System:

The concept is to take a spoken message and broadcast it to a large audience via sound. The first “PA” was invented in the mid 1800’s and consisted of a cardboard cone or horn you would speak through. Often called a “Megaphone”, these simple horns allowed a person to be heard in a large room. In the early 1900’s, after the invention of the vacuum tube, the first amplified PA systems went into use. These consisted of a microphone, some type of amplifier, and some form of speaker. These allowed a person to speak to many thousands of people and still be heard somewhat clearly.

Modern PA systems have come a long way but the goal remains the same: Allow someone to broadcast their voice to a large area via sound. The best example of this is the announcer at a ball game. Fidelity is not important, only the clear transmission of the message.

Sound Reinforcement:

The earliest sound reinforcement systems appeared in the 1940’s. The goal was very different. Instead of simply making a voice loud, the goal was to faithfully expand the sound of a performer or performance in such a way that the quality of the sound was not lost. Early sound systems borrowed heavily from the equipment used in movie theaters as this was the only source of high output, high fidelity equipment.

The perfect sound reinforcement system would make the audience member unaware that a system was being used, and deliver that feeling to hundreds, if not thousands of audience members.

It is the job of the person running this equipment to produce a balanced mixture of the sound being created on stage. Unlike a recording engineer, the live performance sound technician must balance the sound coming off of the stage with the sound being produced by the system. In the case of smaller venues, what is actually coming through the sound system may be a very incomplete mix consisting of whatever is not coming off of the stage. Picture a vocalist and a piano. In a small room, the piano is able to produce a far more powerful sound then the vocalist. To achieve balance, what would be coming through the sound system would primarily be the vocalist. Recorded by itself, this would sound very strange, but to the live audience member, the sound would be balanced. Now, imagine taking the same show into a 50,000 seat arena. In this case, the piano would have no chance of carrying the room. The result would be the piano and the vocalist would both be carried in the mix, and a recording of that mix would closely match the balance you would expect in a studio.

The "Other" System:

No discussion about performance sound would be complete without talking about the "other" burden a live sound technician has to deal with. Stage monitors. Unfortunately, this is not a matter of folding a little sound back to the stage from the main mix. Stage monitors require a completely different approach that has little to do with the sound the audience hears. The performer is not as concerned with the fidelity of the sound. Their concern is that they are able to hear themselves, as well as every other instrument they need to remain in sync with. Often, a large touring group will have a second sound technician located at the stage running a monitor mix. In many cases, individual monitors may each have their own mix as each performer may need to hear a different sound profile. Often, there may be as many as 16 different mix-downs occurring at the same time. Conversely, a small group may only use one mix that everyone has to "agree" on. Unfortunately, that job may fall to the tech that is operating the main mix and is a significant distance from the stage.

The Diplomacy of Sound Reinforcement

As will be covered in this guide, sound reinforcement involves both science and art. Unfortunately, it also involves diplomacy. A seasoned performer has been down this road many times and has a more realistic concept of what can and cannot be done. Unfortunately, they are few and far between! Here are some of the diplomatic issues facing a sound technician:

The Loud Instrument:

As covered before, the larger the room, the more control the tech has over the mix. Conversely, the smaller the room, the easier it is for things to get out of control. Certain instruments pack a lot of punch. Often it may be percussion, a brass section, or electric guitar. In any case, if one instrument is too loud, it may become impossible to achieve a proper mix. You may have seen Plexiglas walls set up around drummers as it is a common practice. The attempt is to decrease the sound level coming off the stage while allowing the drummer to be bathed in their own sound! A loud instrument can not only be a problem for the sound tech and the audience, it can be a problem for the other performers on the stage. If the singer cannot hear him/herself over the sound being generated on stage, it is very hard for them to stay on pitch or stay in sync. There is only so much stage monitors can do. If this is exceeded, the performance suffers. The first sign is often a singer who puts their finger in their ear in hopes of hearing their own voice through their head. Some singers have gone through this so many times they may do it out of habit.

Although most groups that have worked together for a long time have already worked these types of issues out, there is never any guarantee. It falls on the sound technician to point out that a particular instrument is making a balanced mix impossible.

The Musician's Preference:

The bane of the sound technician is the performer that comes off stage during the sound check and proclaims that they are not loud enough in the house mix! Tragically, it is almost a stereotype. To each musician, their life is their instrument. When they come out and hear an even mix, the first conclusion is that they are under-mixed. On stage, they are usually located near their equipment and have spent many years getting used to what that area of the stage sounds like. To them, the mix sounds “off.” Again, if falls on the sound technician to explain the issue.

The Venue Owner:

This problem varies depending on the type of act, but you can be almost assured that if you have a large enough crowd, someone will not like the mix and complain. Possibly, the venue owner themselves has had bad experiences with complaints from neighboring establishments. This can be especially true in multipurpose buildings that are conducting other types of business while an act is performing. You best ally is to stick to achieving a transparent mix that doesn’t “sound” like a system is being used. Ironically, it is often underpowered systems that cause issues as compared to over powered systems. This is the classic difference between “loud” and “powerful.” An underpowered system that is straining will tend to introduce large amounts of distortion. This is perceived as “loud”, and not in a good way! People know something is painfully wrong, but tend to simply feel the act was “too loud.” Conversely, a system that is producing a far more powerful output may not be noticed as “loud” as long as the sound is clean, free of distortion, well balanced, and been equalized properly.

Another factor has to do with equipment placement. For example, if members of the audience are close to the speakers, those members will be subjected to a very elevated sound level that is most likely out of balance with what they are hearing from the stage.

Equipment

I will cover each component in more depth a bit later, but it's best to know what the general components do and how they fit into the picture. Basically, sound is translated from an acoustic medium to an electronic signal, processed, amplified, and translated back to an acoustic medium.

The Microphone:

This is the first element in the system. There are many types of microphones and it is important to know which on best suits what you are trying to achieve. Prices are all over the place, and for a very good reason. Generally, there are two parameters you are paying for. The first is that the microphone picks up all frequencies in the same manner. Inexpensive microphones may lack bass or high end. Even worse, they may pick up a handful of frequencies disproportionate to the rest. Second is distortion. This can come in many forms. Often, a microphone may add fuzziness to the sound, or be prone to picking up the sound of the air flowing near it or mechanical noise from being handled. In other cases, the microphone may not be able to deal with the sound level it is receiving and will "clip" the sound. (This problem can also occur elsewhere in the system.)

Another characteristic that varies with price is how well it complies with its description. A cardioid or directional microphone is supposed to only pick up sound from certain directions. An omni-directional microphone is supposed to pick up sound from all directions equally. Each is chosen for different purposes. Another factor is "proximity effect." This is where a microphone "sounds" different when the sound source is close as compared to when the sound is at a distance. (Some of this is a natural effect of any directional microphone.)

One other factor is how the microphone is designed to connect with the mixer. Most professional microphones use a balanced 3 conductor cable to reject noise. Many higher cost microphones also need to see power from the mixer to run electronics within the microphone. (Known as "Phantom Power.") Needless to say, that microphone will not work if it doesn't see power. Another style of microphone is the transmitter microphone. These are actually a series of components including a microphone, a powered transmitter, and a receiver. Obviously this type of microphone needs a lot more attention than a wired mic as well as knowledge of the pitfalls of radio frequency transmission.

In summary, the microphone is the critical element that takes care of converting acoustic sound to an electronic signal, and as such is often the weakest link in the system. Even the best microphone will give poor performance when used incorrectly or the wrong style of microphone is chosen. In addition, it is your main interface with the performer and often requires the performer to have knowledge of how to use it.

The Mixer:

Once a signal is converted to electronic format it is delivered to the mixer. Often, the mixer will see many different types of inputs. Some may be powered and unpowered microphones. Other times, a "Direct Box" may be used to bring a signal in from an electronic source such as electric guitar, bass, keyboards, etc *without* the use of a microphone. Other sources may be receivers for wireless equipment, or playback devices such as a tape deck or digital player. The job of the mixer is to straighten out all the differences in levels as well as differences in equalization. The job of the soundperson is to control the mixer.

Electronics have their limitations, so most mixers first have to deal with the raw level coming from the source. Two primary levels are "Microphone" and "Line" inputs. Generally, microphones need to be pre-amplified to line level before being mixed. Conversely, line level signals do not, and would completely overload a "microphone" level preamp. Typically, you would see a "line level" signal from a playback device but you would also see it from a transmitter microphone! Despite the confusing name, "Microphone Level" and "Line Level" really have little to do with the actual device but are simply legacy terms for how strong a signal is.

Most mixers have at least the following controls on each channel: Input Gain, Equalization, and Level Control. A far more detailed look at mixers will come later. One important note has to do with equalization. Channel equalization is for modifying the sound of one given channel so that it sounds appropriate for the source. Channel equalization should never be used to try to change the overall sound of the system. That is the job of-

The Equalizer:

This usually comes in the form of a 31 band graphic equalizer. Its sole job is to match the speaker system to the room. This is where you would take care of global equalization problems, such as minimizing frequencies that are booming, or add highs that are eaten by some sound absorbent rooms. The bulk of EQ problems have to do with deficiencies in the speaker system itself (They ALL have them.) As such, settings on the graphic do not usually need modification as long as the same speakers are being used, unless a problem with the room in general shows up. Often a second Graphic Equalizer may be used for the stage monitors, which will be covered later.

The Power Amplifier:

This is the workhorse of the system. Once the signal has been straightened out, it is time to make that signal strong enough to drive the speakers. Because of the power levels involved, most issues related to power amplifiers have to do with making sure there is sufficient electrical power available to run them, and making sure enough air is moving to keep them cool. Prices on power amplifiers are driven by two factors: How much output power they produce, and how reliable they are.

The Speakers:

By far, this is the weakest link in the system. While mixers, amplifiers, and other equipment measure distortion in hundredths of a percent, it is not uncommon for a speaker to be introducing distortion levels of 20 to 30% THD. Distortion specifications are rarely published for speakers and when they are, it is usually some test level (like 1 watt) which is totally unrealistic for use in the real world. Speakers used in sound reinforcement are VERY different that what might be used in a home stereo system. Most high-end home stereo speakers would not last more than a few seconds, and would not sound too good. The reason is simple. When a recording is made, the sound is carefully engineered so as to fit a profile that will work with most home speakers. Signal levels are divided into demographics, with x% appearing at the high end, x% in the midrange, and x% in the bass region. Although transients may occur in any region, the general "program" of the sound will still fall into a broad band of frequencies. In sound reinforcement, anything goes, and a speaker may be called on to handle the full output of the power amp at one given frequency, such as the sustained note a singer or keyboard player may be holding. Since the speakers have to be designed to handle this, compromises have to be made in other areas.
The graph of the frequency response of a given set of speakers is anything BUT flat. As such, equalization is a necessity. With the right amount of power and

equalization, a very reasonable sound can be achieved. More on speakers, bi-amping and tri-amping later.

Speaker placement and type are critical in sound reinforcement, as well as remembering that the shape and materials used in the room play a big role as well. Often overlooked are crowd size and temperature/humidity which can radically alter the sound, making the performance sound totally different than the sound check.

Generally, speakers handling vocal and higher frequencies need to be elevated above the heads of the audience otherwise, the audience in the front will absorb the sound (usually unpleasant for them) and those in the back will only hear a muffled version of it. Conversely, low frequency speakers are better on the floor as those frequencies pass around objects, and do not produce a location "image."

The Wires:

Connecting all these things together are the wires, or properly known as cabling. These can generally be divided into four sub categories with one exception:

- **Microphone cables:** Almost always a three conductor cable with XLR connectors. The male connector is always the source, so the microphone will have a male connector and the mixer will be female. Various lengths are used and it is always good to have several feet of slack when laying them out so that changes can be made. These are always the LAST cables laid out during a setup, and the first to be taken down.
- **Speaker cables:** These cables are usually quite thick. Unlike microphone cables, they use the same connector on both ends. The most common connector is the NL4 or SpeakOn connector. They are a locking 1/3 twist connector. Extending a cable is achieved by using a barrel connector designed for the NL4. Speaker cables are usually laid out at the same time as power cables as there are not many of them.
- **Power cables:** As the name implies, these are simply cables that bring the house power to the area of the power amps, and in some cases they help to power onstage instruments.
- **Patch cables:** Usually short (under 6 feet) and with many different types of ends. These are the interconnect cables used to patch equipment together at the sound board and sometime between instruments and Direct Boxes.

- **The Exception:** One cable that is sometimes used is called the “Snake” cable. This is a multipurpose cable that allows the sound board (mixer) to be operated from the rear of the room without having to run a massive amount of microphone cables. It is often helpful to have the mixer, equalizers, and any effects in the back of the room so the board operator can hear a better representation of the sound the audience is hearing. (Power amps remain near the stage.) There are alternatives to this arrangement, such as iPad based mixers that operate wireless, but each system has its pros and cons.

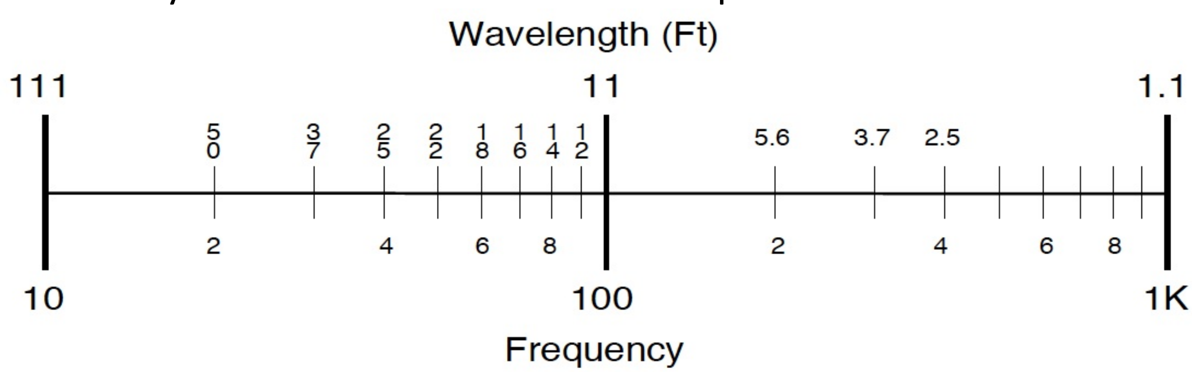
Road cases and travel:

It may seem odd to think of this as part of the sound system, but it can make or break a performance. When you buy a new set of Christmas lights, the new set is neatly wrapped. Often, once the season is over, they are tossed in a box with many others. Next time they are used, you find it quite a mess to untangle them before you can use them. What may be an inconvenience with Christmas lights can turn into a nightmare with sound equipment. In addition to the added time spent untangling the mess, real damage can be done that often does not show up until the middle of a the next show! What looks like a pile of black wires often represents an investment of several thousand dollars. The lifespan of a microphone cable, or most any cable is directly related to how many times the cable has been flexed, and how tight a curve the cable was forced into. Having a damaged shield conductor may show up as intermittent noise on a particular channel. Losing a signal conductor on a balance line may result in a distorted or low output, or background noise. It is important to never knot a cable to tie it off. It is important to loop the cable in a wide loop during teardown and make sure it is free of kinks. If a cable is damaged or compromised (such as something dropping on it), the cable should be marked and set aside until it can be tested or repaired. The #1 cause of sound system failures has to do with bad cables. It is key to develop a routine to setups and teardowns. There is almost a need to discourage the sudden “volunteer” from helping with a tear-down when it comes to cables. Best practice is to make a mental note on what order the cables were laid out during setup, and use the reverse order when tearing down. Simply grabbing a random cable can cause quite a mess and waste the time of those who are trying to get the job done.

Likewise, all equipment should be packed into its respective road case before it is removed from the stage area. Microphones should be removed and put away after the system is powered down but before the stands are taken away and the cables are wrapped. No equipment should leave the stage area prior to all the equipment being packed and stacked on the carts. At the appropriate time, the equipment should leave the stage and go to the truck in a solid stream so nothing is left behind.

Microphones

Now that we have an overview, let's look at some specifics. We'll start with getting the source of the sound into the system. Microphone and microphone placement are a critical issue in setting up. There is math involved in this one. No matter how good the equipment is, the microphone must be able to hear the source of the sound much better than in hears the speaker system. The closer the microphone is to the source, and the farther it is from the speakers, the more available gain you will have. Gain, is basically the term used to describe how much you can amplify the source. The limiting factor is feedback. This occurs when sound from the speakers re-enters the microphone creating a fixed frequency oscillation that runs away. Sound can make its way back to the microphone either directly from the audience speakers, the onstage monitors, or through reflections in the building. Some feedback can be suppressed through equalization, but there is the cost of making the system deaf at that frequency, which may leave the overall mix sounding mushy. If there is a persistent ring at a certain frequency, relocating the microphones may be a better solution. For example:



Lets say you had feedback from two overhead microphones at 400 Hertz. We can see that the acoustic waveform of 400 Hz is 2.5 feet in length. So, try moving one of the microphones 1.25 feet closer of farther away. It may not be exact as the sound may be coming from an angle, but moving it ½ wavelength away will suppress pickup at 400 Hz!

Let's look at which microphones will give you the best ratio. The simple answer is the one closest to the performer's mouth, irrelevant to the type. Technique aside, a handheld or lavalier microphone will give you the best control over feedback. Another thing that will help reduce feedback is minimizing the number of "open" microphones on stage. If you have a solo singer, playing with piano, then the only microphones that should be active on the mixer should be that of the soloist and the one on the piano. All other microphones should be muted.

Unfortunately, it is impractical to have a microphone on every member if you are dealing with a choral group or other large ensemble. Now we move to the world of area pickup and mic selection becomes more critical. Let's look at pickup patterns:

Omnidirectional :
These microphones are great at picking up large areas and tend not to have much proximity effect. Unfortunately, they are also very good at picking up speakers and are therefore more prone to feedback. When picking up a large group of people at a distance, stage monitors are impractical. The elimination of monitors gives you more headroom before feedback, provided your speakers are at a distance. It is worth mentioning that stage monitors only become important in two applications. The first is during a solo, where the singer may not be able to hear themselves due either to other stage noise, or may be thrown off by the timing of their voice being reflected off a distant object like a back wall. The second is when an instrument is too weak to be heard by the other musicians.

Cardioid:
Often thought of as a "close talk mic", the cardioid gets its name from its heart shaped pickup pattern. It is a directional microphone which rejects sound coming from side and back angles. If close monitors are required, it may be the only choice that will give you enough gain to properly represent the sound source. These mics are often used where rejection is key, such as when picking up a guitar amp, drums, acoustic piano, or other onstage instruments that you want to be isolated in the mix. In loud environments, such as a rock concert, they are the microphone of choice. They tend to be more expensive due to the design need to reject sound, but there is another cost. Proximity effect. The sound characteristics change radically as you move away from the microphone. Because of this, in a quiet environment, even a soloist is best mic'ed with an omni microphone.

Transmitter:
Not really a "type" of microphone, but a system of equipment. Transmitter mics come in two styles; Handheld and Body-pack. The handheld style is almost always a Cardioid microphone designed for close use, and has an integrated preamplifier, audio processor, and radio frequency transmitter. The body pack style is a two part system. Usually, an omnidirectional microphone worn near the neck and hard wired to a small pack that contains the preamp and transmitter electronics. Both systems require an additional device called a receiver. Each transmitter requires its own receiver, so if you are using six microphones you will need six receivers. Although very "freeing" for the performer, each is a burden for the board operator / sound person who now has several additional tasks. These include stocking / monitoring microphone batteries, laying out a frequency chart (each mic must operate on its own frequency.) Insuring that the chosen frequencies are clear to use in a given building, and monitoring the RF level being received in order to spot "dropouts." In addition, training performers in their use, insuring they don't get switched off or muted, and insuring the mics don't get buried under clothing.

One additional note: Receivers usually end up next to the sound board (mixer) and put out a "line level" signal. As such, they are usually *not* patched into the microphone inputs on the board.

Other microphones:
There are many other types of microphones such as super cardioids, shotgun microphones, Pressurized Zone Microphones (PZM) but they all fall into one of the two main categories: Omnidirectional, or Directional.

Direct Boxes:
When the source is an electronic system, such as a keyboard, electric bass or guitar, it becomes easier to tap into that signal as compared to using a microphone. There are exceptions to this as many musicians like how their own equipment colors or changes the sound, and therefore would prefer to have it mic'ed over using a DI (Direct Input) box. However, the use of DIs can reduce the amount of equipment on stage and can provide a very clean signal. In some cases, on stage amplification can be eliminated although this puts an extra burden on the board

operator as they now must insure there is sufficient levels of that instrument passed into the stage monitor system to keep the musicians happy.

The DI box usually has a pass-through consisting of ¼ jacks (sometimes known as TS, TRS, or phone jacks.) The musician plugs into the box, and then loops out of it to his on-stage equipment if needed. The box also has a XLR microphone connector which feeds into the sound system the same way a microphone would. There are controls on the box. Usually, there is a “pad” which is used to attenuate the signal. Often, what may be entering the DI is “line level” and what you want to see is a “microphone level” output. The numbers are inverse of the amount of attenuation, so “0 Db” is no attenuation and “40 Db” would be the most attenuation. Often there is also a “ground lift” switch. Minor power differences between the instrument and the sound system may end up superimposed on the audio signal and show up as a hum or buzz. The “ground lift” switch electrically isolates the two and hopefully eliminates this noise.

Pictured- Two brands of direct boxes as well as a SM81 microphone.
The SM81 is a sample of a phantom powered mic that also has a low end roll-off selector and a gain adjustment ring near the element.

Speakers

This may not seem to be the next logical thing to talk about, but they have a lot in common with microphones. Both are used to interface between the acoustic and electronic world. The type of speakers you chose are based on what your source is and who you want to deliver the sound to. A basic knowledge of how different frequencies behave is helpful in choosing the correct equipment.

Below 30 Hz (Sub bass):
Although most stereo systems boast performance down to 20 Hz, few deliver at a level that can be heard. Sub bass is something you feel as compared to hear. Very few instruments produce any sound in the region. The exception may be some of the largest pipe organs. Sub bass is more useful in the special effects world than the music world. At 30 Hz, each acoustic wave is 37 feet in length. Because of this, the room size becomes a factor. It is easy for standing waves to form. This happens when the waveform matches the room dimensions. Because of this, most live-performance sound systems intentionally roll off, or attenuate, frequencies below 50 Hz (well above the 30 Hz sub-bass region.) The other problem is that these frequencies require massive amounts of air to be moved , which uses a lot of power and greatly increases intermodulation distortion. IM occurs for the same reason an ambulance siren sound higher in pitch as the vehicle approaches and lower in pitch as it leaves. Commonly known as the Doppler effect, sound waves are compressed if the sound is approaching and stretched as the sound moves away. With large cone excursions, other notes are raised in pitch as the cone moved forward, and stretched as the cone moves farther away. In the home system, often a separate subwoofer is used to get around this effect.

50 Hz to 150 Hz (Bass):
What we tend to think of as deep bass actually occurs in the region. Drums, bass guitar, lower notes on the keyboard all occur in this region. It is very rare for the human voice to venture into these frequencies, but it does happen. Low frequencies, especially those below 100 Hz, are omnidirectional. At 100 Hz, a waveform is still 11 feet in length so it is very hard for a human to tell what direction this sound is coming from. Because of this, often bass cabinets are left on the floor as there is no "sound image" in that region. Sound systems for arena and

stadium size venues often “crossover” at 150 Hz and frequencies above this are sent to the “flown” speaker system and those below are sent to cabinets at or below stage level. As an added benefit, cabinets on the floor also take advantage of the “mirror” effect, which can double their output by utilizing the sound reflected off the floor itself.

150 to 1000 Hz (Midrange):
By far the most critical area for sound. Most all vocals are in this frequency range. It also provides much of the “sound image” in that humans can detect the location the sound is coming from. This is true with higher frequencies but there is less happening up there. Midrange does not cut through obstacles as well as lower frequencies, so it is important to get these speakers off the ground and above people’s heads. Midrange is also the peak human hearing range, so problems in the midrange will be noticed real fast!

1000 to 10,000 Hz (Highs):
These frequencies provide the “definition” to the sound. Many vocal overtones and harmonics show up here and if underrepresented, the sound will be perceived as muffled. Unfortunately, the ability of speakers to evenly disburse their output rapidly decreases as the frequency goes up. In addition, due to phase cancelation, additional speakers may actually make the matter worse. This is why “single point” or “vertical arrays” come into play. Horizontal dispersion is most critical, so vertical placement leads to less phase cancelation.

Above 10,000 Hz (ultra highs):
Although depressing, few humans hear well above 10,000 Hz. Home systems often brag about reproducing sounds up to 20,000 Hz, but few deliver. In a quiet listening environment, these frequencies add the extra sizzle to sound, and it is good to try to achieve in that type of environment. In commercial sound reinforcement 12KHz to 15KHz are good goals to shoot for in a system. Above that some serious trouble starts. Specifically, something called ultrasonic oscillation. This occurs when some of the output signal leaks into some of the input signal causing electronic feedback. The higher the frequency, the more prone the system is. As this usually occurs at a frequency between 20KHz and 40KHz, the first symptom may be speakers and amplifiers blowing up for no apparent reason. The

attentive sound person may hear a dull hum from the speakers which is the sound of the power amplifier's power supply straining under the load. This can occur even when there is no audio signal being processed through the system. Most modern sound equipment suppresses these frequencies but it can still happen, which is why it is important to keep the rat's nest of stage wiring as clean as possible and to keep speaker lines away from microphone lines as much as practical. Although some crossing and mixing is unavoidable, long runs of microphone and speaker cables together should be avoided as the speaker lines can induce signals into the microphone lines.

Types of speakers:

Professional Name	***Often called***	***Freq Range***	***Notes***
Subwoofer	Subs	< 100 Hz	Sealed or folded horn cabinet
Bass Bin	Woofer	50 Hz to 150 Hz	Sealed or bass reflex cabinet
Mid Bin	Midrange	150 Hz to 1000 Hz	Cabinet or horn loaded cabinet
Horns	Tweeter	> 1000 Hz	Fiberglass or metal horns with drivers
Stand Speaker	Speaker	50 Hz to 12,000 HZ	Speakers mounted on a stand
Line Array	PA system	100 Hz to 15,000 HZ	Large vertical grouping of hung speakers
Stage Monitor	Monitor	50 Hz to 12,000 HZ	Located near performer
Cross-Stage Monitor	Monitor	50 Hz to 12,000 HZ	Positioned at sides of stage
Hot-spot Monitor	Monitor	150 Hz to 10,000 Hz	Very small fill monitor

The choice of cabinets is determined by the type of performance and the size and layout or the room. If the performance is of a vocal only group in a smaller venue, then a simple set of stand speakers will usually do the trick. Once you add instruments that will be run through the system you may need to increase the range of the system. If you are adding electric bass, keyboards, or mic'ing drums, then at the least bass bins should be added. Choices change as the venue size is increased. In moderate sized locations where a great deal of amplification is needed, using a component system (Bass bins, Mid bins, and Horns) often works best. In large venues, such as a stadium, Line arrays supplemented with bass bins is the commercial choice. Power demands of such systems can run into the tens of thousands of watts.

Powering the speakers:

There are two basic ways to power speakers. The first is to use self-powered speakers. These speakers have a built in amplifier. They are heavier than conventional speakers and require that you run both signal lines and power lines to them. Some people prefer these as the general layout is easier to understand and there are fewer items to carry around. The second are known as passive speakers. Here you would have a separate power amp but would only have to run one cable to the speakers. Both types of systems are available from stand speakers all the way up to line arrays.

Biamping and Triamping:

To divide frequency bands up and send them to the right speakers (Bass bins, Horns, etc.) a crossover network is needed. These can be located before or after the power amplifiers. When they are located after the power amplifier, they are known as “passive” crossovers. The disadvantage to passive crossovers is that they eat up a lot of power. Often about 50%, requiring the use of larger amplifiers. The advantage is that they can be built into the speaker cabinets and you don’t have to give them a second thought. No special wiring is required. The disadvantage is that in addition to using more power, all the sound is being driven through one line. If a sudden heavy bass demand is put on the system, midrange frequencies carrying the vocals may be disturbed. This is especially true when an amplifier is driven near or above its rated output.

On very large systems, frequency bands are often divided before the power amplifiers. These type of crossovers are known as active crossovers. A system that uses separate power amplifiers to drive the lows and the highs is known as a “Biamped” system. A system that uses separate amps for Lows, Mids, and Highs is known as a “Triamped” system. Since there is nothing between the power amp and the speakers, 100% of the power can be delivered to the speakers. Since the midrange has its own amplification, vocals remain clear even if the bass amplifier becomes overwhelmed.

In practice, the larger the system the more there is a need to use a multi-channel amplifier approach. The downside is that precautions must be made to make sure the correct outputs reach the correct speakers. Without protection, even a split second of bass drive will destroy a high frequency driver.

Cabinet positions:

As covered before, the lower the frequency the lower the position. Bass bins on the floor, mids and highs elevated. All in one units such as stand speakers are elevated. Now that we have the vertical down, let's talk about the horizontal position.

Generally, the farther the speakers are from the microphones the better. There are some limits however. You want the sound image to be coming from the stage. Too far away will make the performer seem fake. Also there are timing issues. Sound is slow, and you want your performers to be in sync with what the audience is hearing. One technique to get even dispersion is to locate four speakers in the corners of the room. This can be done in rooms that are under 30 feet if the resources are available. Your stage side system would contain your bass bins, and your rear system would usually just be stand speakers. Beyond 30 feet you end up with an audible delay or echo. This can confuse audience members that are not in the center of the room, and confuse the performers as well.

Best practice is to locate the speakers in the front of the room at an equal distance to the audience as the performers themselves.

Stage Monitors:

These speakers exist for a completely different reason then the house speakers, and must be conceptually thought of in a different manner. Generally, the simplest stage monitors are a pair of speakers, one on either side of the stage facing the performers. These are known as cross-stage monitors. It is the only type of system that should be attempted to be run by the sound board operator.

More complicated system are in common use where each monitor gets its own mix, but this really requires a monitor board operator who is located at the side of the stage and can communicate with the performers during the show in real time. Needless to say, stage monitors can be a large source of feedback. There are several "rules of thumb" regarding what should and should not be put through the monitors. In general, any microphone being used to pick up an area of sound should not be mixed into the monitors. Close talk cardioids solo microphones should in the monitors. Even Omindirectional microphones can be added to the monitors if they are being used for solo work, although care needs to be taken. If a musician is using a Direct Box and not using his own stage equipment, then the monitors are his/her only lifeline.

The need for monitors varies greatly. In a very small venue where the house speakers are close enough to be heard on stage, monitors are not required. Conversely, if the on-stage sound level is high, or the reflection of the sound off a far wall generates a timing problem, monitors are required even if the room is relatively small. In larger rooms where the house sound cannot be heard on the stage, monitors are required.

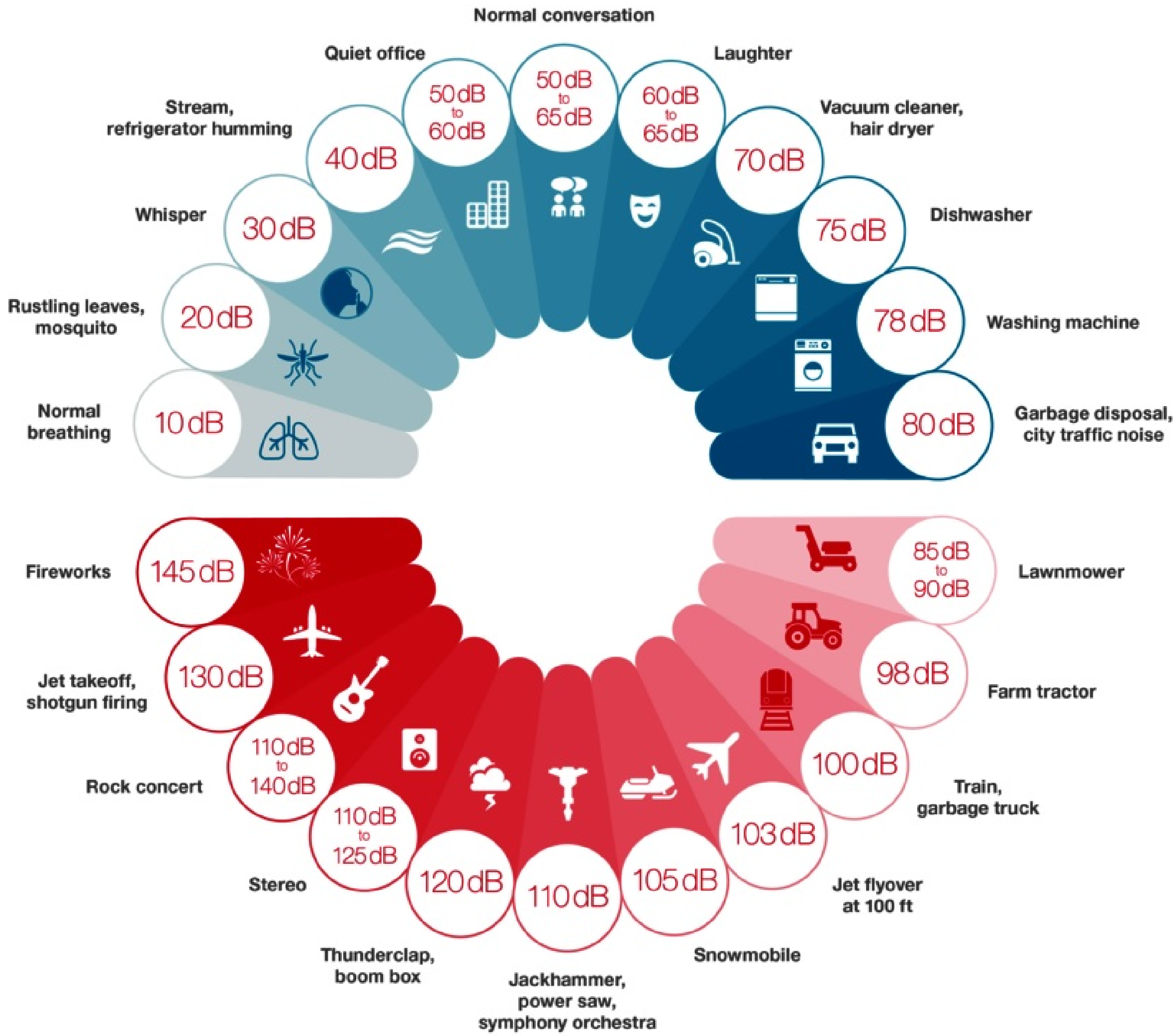
Dynamic Range, Calibration, and Distortion

Nothing can wreck a show faster than a distorted sound system! Often, the problem in not that the speakers or power amps are too small, but that the system is out of calibration with itself. Usually, little to no calibration is needed in the field as a well designed system would have this done before it leaves the shop, but things do go wrong in the field and a good knowledge of system calibration can help troubleshoot problems before they become headaches for everyone. There are several terms and phrases related to sound that are good to know:

Decibels (Db):

The decibel originates from methods used to quantify a signal in telephone circuits. In the Audio world, it ends up with three different uses. The important thing to remember is that it is a logarithmic progression. We most often think of decibels when it comes to actual sound produced, so let's cover that first:

In the acoustic world, 0 Db does NOT mean there is no sound. 0 Db is the established point where you are at the lower limit of average human hearing. Noise can be lower than 0 Db and those levels are represented as <u>negative</u> numbers. The best way to think of 0 Db is the Centigrade temperature at which water freezes. Things can get a lot colder but the freezing point of water is the index point. When it comes to sound, each increase of 6 Db means the sound energy level has doubled. (This is different than its electrical meaning.)



In the electrical world, 0 Db also does not mean there is no signal, in fact a 0 Db electrical signal is actually quite hot! 0 Db represents "Line Level" (Specifically, 0.775 volts into a 600 ohm load.) Signals lower than those are represented in negative numbers. The average microphone being driven with 94 Db of acoustic sound only puts out a signal of -40 Db, or 40 decibels below 0. There is a difference in progression as well. A signal level is considered to have doubled with each increase of 3 Db. Why? Well it is a quirk of how power is delivered to a speaker. If you were to double the voltage level going to a speaker, Ohm's law says that speaker would then draw twice as much current. Wattage is voltage times current (measured in amps.) The result of doubling the signal level is that the speaker is now putting out four times as much power!

In "Signal to Noise", decibels represent how far below the maximum signal level the noise level is. For example, if something has a S/N or 140 Db, then the sound of the background noise is 140 decibels below the unit's rated output. The goal is to have the background noise below the threshold of human hearing.

In "Dynamic Range", decibels indicates how far above the noise level the maximum signal can be. Pretty much the same as signal to noise level.

In signal measurement, the 0 Db mark is considered to be the "Maximum normal level." This may be considered to be strange, but it is actually why the decibel measurement was invented. The inventor? Western Electric, the old Ma Bell system! What the telephone company needed was a way to measure the content of a signal. Volts and amps only tell you how much power is flowing, it doesn't tell you what part of that power flow is actual audio content.

The concept was, if a telephone produced a normal signal (0 Db), how much of that signal content was lost after running through 10 miles of wire? So, if the signal lost 9 Db, then it would have to be amplified by 9 Db to return it to normal.

On most commercial sound boards, the individual level controls have the 0 Db mark near the top of the slider. This is considered the "normal" range. Likewise, level meters are often broken into groups of green, yellow, and red LEDs with the transition from green to yellow occurring at 0 Db.

Generally, the closer the signal is to 0 Db, the better your available dynamic range will be.

Clipping:

This is the point where the signal is so strong, it exceeds the ability of the circuit to handle it. The sound produced by a signal being clipped is agonizing to listen to! Clipping can occur at any stage of the audio process even while other stages are working correctly. This is why a system needs to be calibrated. Let's look at how many times the signal can be modified in strength from Mic to Speaker:

Microphone:

The output of a microphone can vary greatly. First, the general sensitivity varies between models and types of microphones. Second, what is put into a microphone can be anything from a soft acoustic guitar to a blasting trumpet! Microphones themselves can be overdriven, so one must look at what the manufacturer says is the maximum level the microphone can handle.

Input Gain Control:
Sometimes called a "pad" this control is different than the channel level control as it varies the sensitivity of the preamp. Too much gain, and the preamp will clip, too little and background noise will be high. Basically, it is used to match the microphone to the mixer. These controls are usually located near the input jack.

Channel EQ:
These controls alter the emphases of amplification at various frequencies. This can set up a situation where certain frequencies exceed the ability of the electronics and again cause clipping.

Channel Level Control:
Channel Level, sub-master, and master controls should all have moderate settings. Running the master control at a very low level while running channel level controls very high can also cause clipping.

Sub-Master and Master Controls: (see above)

Graphic EQ:
Generally, if too hot a signal hits the graphic EQ and the gain is lowered on it to compensate, input clipping can occur.

Power Amp controls:
If the sensitivity is set too low, the prior equipment may end up driven into clipping trying to get the proper output.

As you can see, even when the output volume of the system is not high distortion can occur within the electronics if the individual stages are not calibrated properly.

Initial System Calibration

This process is usually done at a shop as compared to trying to do it at the job. It involves a lot of noise! Once properly calibrated, a system should be good to go for a long time or until someone adjusts the wrong thing! Oddly enough, calibration is usually done from the output backwards.

Start with the Graphic EQ, Power amp, and speakers.
It is necessary to inject a line level signal (0 Db) into the input of the graphic EQ. If you have a known good mixer, you can play a tape through it and adjust it to get an average meter reading of 0 Db. Next adjust you graphic EQ to be flat and with the graphic engaged, see that the level meter on the graphic is showing 0 Db. At that point, with the controls turned down and the speakers connected, turn the power amp on. Bring the controls on the power amp up to a level that is moderately loud. (It's normal maximum.) You should not see much activity on the clip lights of the power amp. At this point, these stages are calibrated. Mark the levels on the power amp. At that point you can shut it down as long as you have meters on your graphic. It is now time to calibrate the front end of the system.

Microphone and board calibration.
Set all of the EQ controls flat on the board, set the Master, Sub Masters (if used) and Channel level to 0 Db and provide a signal source, such as a tape deck, to the line level input of the channel. As the signal is brought up in level, the meter indicators on the board output and graphic EQ should match. Now plug a microphone into the microphone input and adjust the "input gain" until the microphone is bringing the meter level to 0 Db when you are providing a substantial input to the microphone. At this point, you have a baseline calibration.

EQ calibration

This process is also very noisy! Although individual rooms will need tweaking, general EQ calibration between the speakers and graphic should be done at the shop. Here are the three common methods:

Pink Noise:

White noise has equal energy across the full spectrum. Pink noise has equal energy between each octave and therefore a lot more energy in the bass region and less in the high end. Pink noise works better for setting up a system as it is not quite as painful to listen too! Generally, what is needed is a noise generator with calibration graph, a measurement microphone, and a spectrum analyzer. Find a room that does not color the noise, is free of echo, and does not resonate much. Set the microphone in the middle of it. With all EQ's flat, bring the noise level up on the system and adjust the graphic EQ to give you a pattern on the spectrum analyzer that closely resembles the calibration graph. Although this method is "scientific", it often produces some serious real-world errors. Usually moving the microphone will affect the results, so several tests should be made and averaged.

Sweep Generator:

In this case, your sound source is an audio sweep generator and you monitor the sound using a measurement microphone connected to an audio level meter. What you want to do is start at 100 Hz and sweep down, looking for any peaks or troughs that need to be boosted or suppressed. You should roll off your gain as you cross through 50 Hz. Next you sweep upward again looking for any peaks or troughs that need to be boosted or suppressed. Be careful not to push the high end at too hard a level as most high frequency drivers are only designed to handle a small percentage of the overall power.

By Ear:

This is the preferred method of a seasoned sound technician. It is the least scientific however it usually produces the best results if the person is competent. Basically, a known musical track is played through the system and the system is then set to achieve a natural reproduction.

At this point, you have achieved a baseline EQ. Next, it is time to set up a few microphones. These should be the make and model you intend to use live. Set the microphone to be approximately equal in level and bring up the overall gain. You will quickly notice a prominent feedback frequency. Try moving the microphones a bit. If the frequency remains the same, then you want to lower that frequency on the graphic by a few Db. Increase the gain until feedback occurs again. If it is the same frequency, then nudge it down a bit more. You want to be careful not to develop a hole in the sound at that frequency, but if the system is too energetic then you need to notch it down a bit. Try removing different models of microphones to see if a persistent problem occurs with only one model. If this is true, then that problem should be addressed on the individual channels that use that microphone. A system will never be free of all feedback, but when you reach a point where multiple frequencies want to feed back at the same time, you have likely achieved a good system equalization.

And then there's this:
A nice flat EQ is not always what you want! Still, adding color to the mix should only be done after the system has been calibrated. Also, you want to take a snapshot of your settings so you can find your way back should things get out of control.

Channel EQ

Channel EQ is a whole other topic. Generally, channel EQ has a specific purpose. For vocals, a low end cut is advisable as it reduced microphone handling noise, noise transmitted through mic stands, and wind noise (including the wind from the singers themselves.) For bass vocals, you may need to let the low end remain in place, or use the low end cut on the board then boost the bass level. This will achieve a warm sound, but still roll off the sub-bass that is useless. For most vocals, some high end boost is useful for adding definition to the sound. If a windscreen is in use, some of the highs are lost. Generally, windscreens are not needed indoors if the vocalist is using the microphone at a 12 to 16 inch range. Close talk can present popping problems depending on technique and a windscreen will help with those. Most handheld vocal microphones have a built-in pop filter and employ some high end emphasis.

Instrument microphones can be tricky. Usually, they are kept flat, but you may need boost especially when the mic is used on an acoustic guitar or piano. Generally, electric guitar and bass need very little EQ. Drums can be a different story. An entire chapter could be written on drum mic'ing technique and equalization. It can be as simple as a kick drum mic and an overhead, or as complicated as mic'ing each individual component of the set. The preferred system for electronic keyboards is to use DI boxes on each individual keyboard, usually with very little EQ.

The Mixer

It all comes together at the mixer (sound board.) Before we get into the controls, a few words are needed about the types of cables that get plugged into them:

XLR Microphone Connectors:
These have become the standard of the industry. Sometimes called a "balanced line" as they are used in other locations as well, they contain a positive phase signal, and a negative phase signal. The idea is that any noise the cable picks up would appear on both and self cancel.

TRS Phone Jacks:
Standard TS phone jacks have fallen out of favor. (Think electric guitar cable.) TRS stands for Tip, Ring, and Sleeve. Older phone jacks had the signal on the Tip, and the shield on the Sleeve. Balanced jacks carry a negative phase signal on the Ring. TS and TRS are <u>not</u> fully compatible! When in doubt, always use a TRS cable. There are some exceptions to this. For example, some mixers allow a signal processor to be "inserted" into a channel or master buss. Often, these jacks may carry the outgoing signal on the Tip and the returning signal on the Ring. Special cables are needed for these. In almost all cases, a Phone Jack input is a "Line Level" input.

Other Inputs and Outputs:
Most mixers provide additional lesser-used inputs and outputs. Some of these include Direct outputs for use in multitrack recording, and insert patch points.

Mono vs Stereo:

Although Stereo is useful in recording, it can be a problem in live performance work. Unless you are sitting in the center of the room, the last thing you want to hear is more of the instrument you are right in front of! Some sound techs do a reverse stereo mix. For example, if you are on the left side in front of the piano and there is a guitar player on the right, the piano would be panned to the right and the guitar to the left so that each side of the room is balanced in reverse of what is coming off the stage. Generally, you are better off mixing in mono unless you have a few years of mixing experience.

A Channel of Mix

On the next page is a typical mixer channel. The first two jacks are for Microphone level (XLR) and Line Level (TRS/Phone jack) inputs. The push switch next to the jacks engages the **Low End Cut.** The rotary control under that is the input “Pad” or **“Gain” control** for setting the signal range of the microphone.

Compressor:
Some mixers provide a compressor control. Sometimes called a limiter, this control reduces the gain as the source gets louder. The position of the control determines the level before this limiting effect begins.

EQ:
This is your channel EQ and allows you to add or subtract bass, midrange, and treble to a channel. The “flat” or “no EQ” position is the center range of the control. Often sound boards will give you a little “parametric equalization” using the midrange control. What this means it that you can select the frequency it is operating at and then add or subtract the emphases at that frequency. This is very helpful with stubborn feedback problems. By setting it to subtract about 10 Db, you can then use the frequency control to sweep through the range and identify the problematic frequency. The rule of thumb is to use as little EQ as you can to achieve your goal. Large EQ settings can leave holes in a vocalist or instrument and notes that fall into those holes are underrepresented.

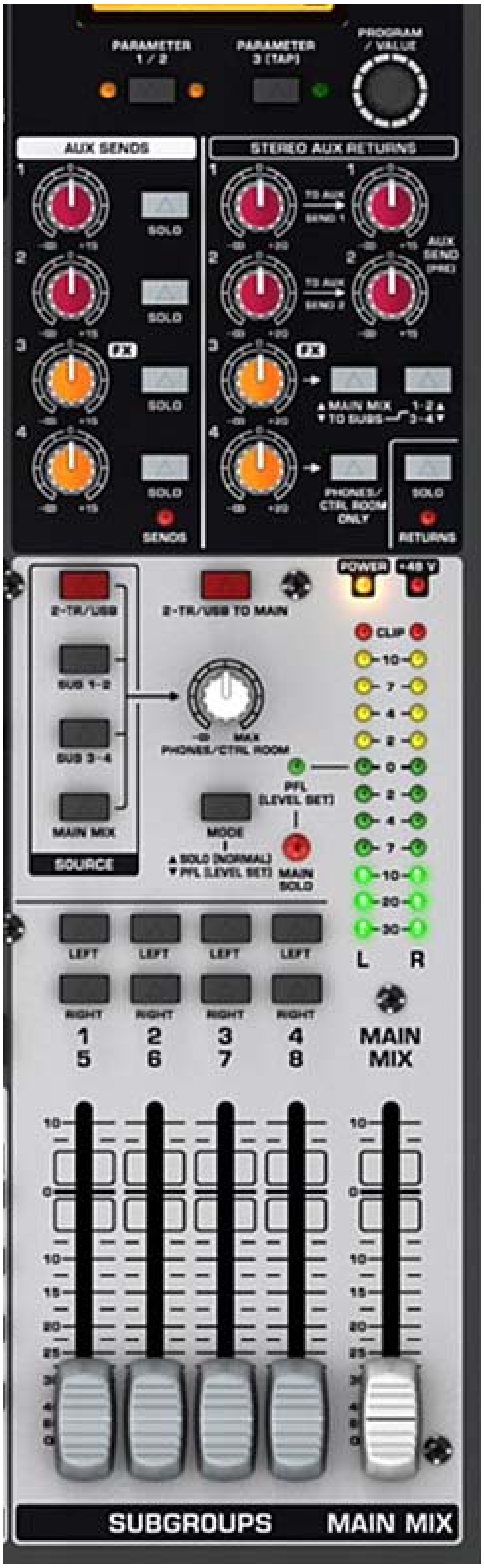
Aux / FX / Monitor sends:
Technically, these work the same and provide a way to produce alternative mix-downs that are separate or partially separate from the main mix. A small Pre/Post push switch allows you to separate them completely. In the "Pre" position, the output is completely independent from the main mix. In the "Post" position, the channel level control will also affect the level this control sends. (FX is short for Effects.) It is normal to use the "Aux 1" as your stage monitor feed. If you are using a separate effects rack for echo, reverb, or any other number of effects, you would send selected channels to that rack using the FX control. For example, you may want to send vocals to the effects, but not instruments.

Pan:
If a board is set up for Stereo, then the pan control will allow you to change the amount of signal seen in your Left and Right channels much like the "balance" control on a home stereo.

Mute:
This button allows you to take a channel out of the main mix without having to move the level slider. It is advisable to always mute microphones that are not in use.

Assignment buttons:
These allow you to assign the output of that channel directly to the main mix or via sub-masters. The "Solo" buss allows you to listen to a lone channel through headphones to help identify problems.

Clip LED – Some mixers have a visual indicator of when a channel is overdriven.



Master Section

As the name implies, everything comes together in the master section!

Why Sub-Masters? Boards can be patched without sub-masters but they come in handy to group things. For example, you may want to put all the vocals on one, the drum set on another, and a complicated keyboard rig on a third. This way, everything in an entire section can be brought up or down with one control.

Aux Sends:
If you are using Aux 1 for the monitors, then “Aux Send 1” would be the master volume for your monitors. In that application, “Aux 1 return” would serve no purpose.

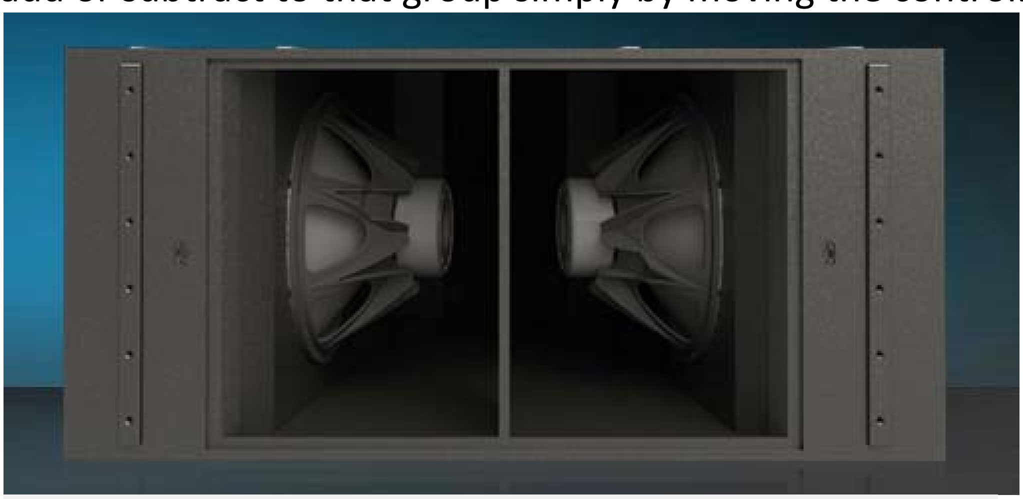
If you were using an outboard effects rack and FX 3 as your effects buss, then send #3 would be your master volume to the rack, and the output of the rack would be patched into #3 return. This control would then set the level of the effect in your main mix.

Solo controls: (center section)
These allow you to select a source and send it to headphones or some other output. It is a good diagnostic tool for when something is going wrong in a live performance as you can send things to the headphones without disturbing the main mix, and quickly identify where a problem is occurring.

Power & 48V:
The power indicator lets you know the board has power. The 48V tells you that a phantom voltage is being sent to the microphones. Some mics require a phantom voltage to work.

WARNING:
Phantom voltage should never be switched on until the microphones are connected and should be turned off prior to disconnecting the microphones. The spike caused by connecting or disconnecting a microphone with the voltage on can damage the mic or the mixer preamp!

Sub-masters:
If you are using the sub-masters, start by setting them all to 0 Db, and then assign whichever channels to whichever sub-master fits the grouping. In the case of a mono-mix, simply push-on both the left and right buttons above them. At that point, the board will operate as if they were not even there, but if need be, you can add or subtract to that group simply by moving the control.



Two views of a *Danley Sound* sub-woofer called the "Cine-monster"

Each cabinet contains two 21 inch drivers and weighs 480 pounds.

They are rated at 3000 watts continuous duty and have a "program" rating of 12,000 watts

The Setup and Sound-check

With a good knowledge of the components and a working calibrated system, it is time to venture into the real world. The first stage of this is to assess the room as well as the group's needs. First, let's look at the room:

Acoustics:

How live is the room? If you clap your hands, is there an echo or reverberation? A room with a lot of flat hard surfaces can be difficult to deal with. The echo and reverberation will tend to muddy up the sound making it difficult to get a clean mix. A dead room will tend to require more power and higher gain, but the end result will give you a cleaner sound. Temperature and humidity also play a roll. A cooler, dryer room will emphasize high end and you may end up with a shallow sounding mix. Large flat surfaces that face the stage may create a "slap back" effect that will throw the performer's timing off. This would require louder on-stage monitors. Also, you need to assess how the room may change by show-time. A sound-check can be thrown off by conditions that change once the room is full of people and may be warmer and more humid.

Layout:

Where would your equipment be best placed? What cable runs might cause a tripping hazard? How far can the speakers be placed from the performers? Are there obstructions that will interfere with the sound? All of these questions should be addressed prior to setting up equipment. A sound tech may prefer to be in the back of the room, but is this practical given the shape of the room and the cables that need to be run? Also, look for a "staging area" (usually in front of the stage) where the equipment will be placed during the load-in and collected during the load-out. This should not be the stage itself as you will be tripping over it!

Power:

A single 20 amp circuit can provide 16 amps continually, and 20 amps for short durations. That works out to about 2000 watts of power. As long as lighting is not involved, 20 amps is a lot of power. Audio equipment is rated at "program draw" which is actually five times the average draw. In other words, a 1000 watt amplifier will usually be drawing 200 watts. As you can see, a 5000 watt sound system would still give you plenty of headroom on a 20 amp circuit. There is one other concern. What else is on that circuit? If there is a major appliance also plugged in somewhere out of view, you could reach your limit faster than expected. In most commercial buildings, each wall is on its own circuit. A quick visual survey is always a good thing. The second note on power is that you should always prefer a "single point source." It may seem convenient to plug the stage equipment in one outlet, the power amps in another, and the mixer in yet a third. This is a recipe for "ground loops." Most often, these will manifest themselves as a background hum or buzz in the audio system. The hum may not appear during sound check, but appear later once the house lights are dimmed. Elimination may require last minute running of extension cables as well as use of "ground lift" features on direct boxes.

Once these observations and decisions have been made, it is time to set up. Locate and set up the speakers, amps, and mixer first. Cable runs should be done in three stages. First, you power runs. Next your speaker runs, followed by a snake run if needed. Volunteers can be tasked with setting up microphone stands as well as putting runners over cable runs that may be a trip hazard. Next comes the microphone runs. It is good practice to develop a sequence to these that can be reversed during tear-down. The final stage is to patch things up and test the system. Make sure all the microphones are plugged in before powering up the mixer if phantom power is being used. The last thing to turn on is also the first to be turned off at the end of the night, and that is the power amps.

The next stage involves injecting a signal like a tape deck or MP3 player. This assures you that the mixer and power amps are working and also gives you a clue on how smoothly the room is being covered by the speakers. Don't forget to send some to the monitors as well.

Pre Sound check testing:

Many things are best done before the group arrives or there is an audience in the room. Once the system is running, it is time to test each input. If you are using transmitters, this is a good place to start. Turn on the receivers first. See what type of RF signal they may be picking up from other sources. Some background RF is normal, but if there is already a strong signal present, then that frequency will not be available to use that night. Consulting the RF charts, look for a frequency that is two channels away and retune the receiver. If that frequency looks good, then retune the microphone or body pack to that frequency. Next, try out the mic. Walk around the room and make sure you don't have a dropout problem. If you do, try reorienting the antenna on the receiver. If this does not help, make a mental note to tell the performer not to venture to that area. Turn off the mic. The receiver should mute. If it does not and there is a blast of noise, then adjust the squelch on the receiver and re-test. Once your transmitters are tested, it is time to check the onstage microphones. The hardest to deal with will be area pickup microphones. See how much gain you can get before feedback. Try to minimize feedback by relocating the microphones slightly. If there is a specific feedback problem with all of the microphones, then you may need to adjust the graphic EQ. Memorize your current setting and make minimal adjustments. There is a price to pay for having large EQ "gaps" that needs to be compromised against the tendency of feedback. No system is feedback free, but the goal is to be able to have a little headroom between where the show will run and where the feedback starts. REMEMBER: The more open channels you have, the less gain you will be able to achieve before feedback. Area mics should be turned off when they are not being used, such as during solos. Solo mics should be off when they are not being used unless they are also being used for area pickup. Announcement mics should be off when announcements are not being made. This not only reduces feedback, it also reduces the amplification of onstage noise such as paper shuffleling, footsteps, and stray conversations.

The last thing to test is the instrument mics and DI boxes. One of the reasons for this is that you really don't want to bring the performers on the stage until you are ready for them. Also, they tend not to be around during the early stages of the setup. If a drum set is being used and is being mic'ed, these will need to be tested. More on that later.

Rule of 3:

Microphones that are close to each other present a unique problem. A singer may be using one microphone, but if a second microphone is close by, it may pick them up as well. This can be a real problem! Whatever the distance differential is, frequencies that are ½ wavelength will be missing from the singer's voice. Although there is a technique called "double micing", it relies on the distance differential to be out of the human hearing range. In general, the nearest second microphone should be three times the distance as the primary microphone is to the singer. If a singer is one foot from the mic, the nearest second microphone should be three feet away. If a line of singers are being mic'ed, and the singers are standing two feet from the microphones, then the microphones should be 6 feet apart.

The Sound Check:
Never expect perfection! The type of sound check varies greatly depending on the group. An instrumental group with vocals, one vocalist per mic, is handled very differently than an ensemble. In the case of the instrumental group, it is best to start with a song and mix the instruments first. Then bring in the vocals and finally address stage monitor feeds. Remember, you are trying to achieve a balance between what is coming off the stage and what is coming through the system. For an ensemble, the technique is different. Your system level is going to be based on the sound level of the singers plus whatever gain you can achieve without feedback. Once that is established, any accompaniment, such as a piano, is blended in. Often in smaller rooms, the piano may stand on its own and the challenge may be in bringing up the level of the singers to match the piano. Once that is established, any solo singers should be checked. If the singer will be leaving the stage, have them do so during the sound check to insure there are not feedback problems or signal dropout points.

The Show

If a sound check has been done correctly, the show itself should simply be a matter of following a cue sheet and hitting the mute switches of whatever mics are not in use. Of course, things are never that easy in the real world! As the room fills with people, the addition of heat, moisture, and ambient noise will change the mix. In addition, individual instruments as well as vocalists will tend to play or sing at different levels during the show than they did during the sound check.

You should secure a set list at minimum. Make notes on which mics will be used at which points during the show, and which can be safely muted. Always be aware that groups like to mix things up so you may have to "roll with the changes." This is especially true with whoever is filling in the gaps between songs. Announcers tend to be very unpredictable, and often more than one person does announcement from more than one location. Be very aware of “after-ring.” This is a tone heard after a loud noise and usually fades in less than a second. It is also a precursor to feedback. Try to figure out which microphone is a little too hot and back off the gain a bit. If there are open microphones not in use, mute them. As a rule of thumb, having something under-represented in the mix is more desirable than runaway feedback.

The Teardown

This may seem unimportant, but it really sets the stage for how your next setup is going to be. Also, a proper teardown extends the life of equipment and reduces the risk of equipment being lost, left behind, or stolen. In general, spontaneous “help” should be avoided. It is always better to use people who have a background in equipment teardown and who are familiar with the show.

The first consideration is background music. If the show does not use an after-the-show audio track, then the teardown should begin by muting all the audio channels, turning off the power amps, then turning off the mixer and other signal processing equipment. If the show does use a track, then mute all the microphone channels and shut off the phantom power for the microphones (usually a switch on the mixer.) It may take a few seconds to bleed off before it is safe to disconnect microphones.

Using the “staging area” you picked out during the load in, disconnect the microphones and monitors and move all of the onstage equipment off the stage and into this area. Leave the cabling where it is. This equipment can be packed away even if a music track is still being used.

The next step is to shut down the system if it was still being used. Again, power amps off, then mixer. Disconnect all cables from the mixer and amp rack. Move them to the staging area as well. At this point the only thing remaining should be the wiring. Using the reverse order of how the cables were laid out, wrap each one being careful to let the cable tell you how it wants to be wrapped. In other words, if a cable is not looping correctly, rotate the loop to avoid setting a kink or twist in the cable. Once all of the equipment is packed and stacked in the staging area, it is time to load out. The biggest mistake is to let people move some of the equipment towards the door before you are ready to move ALL of it there. This is how things get lost or left behind. The load-out should be one swift move. With the truck loaded, a final “recon” is done by walking the path in reverse and looking around the venue for anything left behind. Always good to check the dressing room as well!

About the Author

1982 picture directing the Witness show

John Dziel first entered the sound business in 1971, founding the company DAE. In the mid 1970’s, DAE provided sound reinforcement equipment for many shows in the Allentown / Bethlehem area. These shows included every possible act from Rick Derringer to Thin Lizzy. In addition, DAE provided equipment for many touring club acts as well. In 1976, John and DAE expanded their business to include concert lighting. All in all, John did sound, lighting, show production and pyrotechnical for over 2000 shows and 135 national acts, spanning 25 years.

Notes

Classic "Line Array" sound system with ground level subs

Publishing Notes: