

Live Sound Basics



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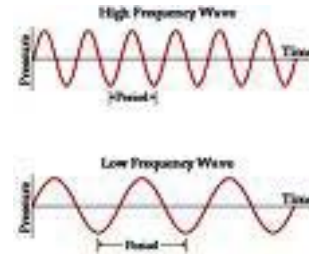
Preface

This book is designed with the novice sound system operator in mind. It contains some basic information to help you get started learning what some components are, a basic understanding of how they work, and how to use them appropriately. This book presents the components of a sound system in the order of signal flow, beginning with a sound source into a microphone and ending with sound waves coming out of loudspeakers.



Characteristics of Sound

Sound is an atmospheric pressure wave. It is similar to the ripples resulting from dropping a rock into a pond, only in 3-D – there is a source, and a subsequent diminishing ripple or *wave form*. *High frequency waves* are close together; *low frequency waves* are far apart. High frequency waves are very *directional*, low frequency waves much less directional, meaning that it is much easier to pinpoint the source of a high pitched sound than a low pitched sound.



Sound can be reflected and absorbed. Hard flat surfaces reflect sound, soft round cushy things absorb sound. Sound waves will also radiate around a boundary and through holes, such as a free standing partition or an open doorway. Remember that sound is 3-dimensional, radiating in the shape of a sphere more or less, from its source.

Sound can be described in terms of:

- * **Amplitude** (volume or loudness)
- * **Frequency** (pitch – high/low)
- * **Timbre** (quality – difference between a violin and a guitar)
- * **Dynamics** (a sound going from soft to loud)

Sound is mainly controlled by its source. For example, pianists control amplitude by how hard they may play the keys. They also control frequency by which keys are played, timbre simply by virtue of the fact that they are playing a piano, and dynamics by playing from soft to loud. This is very important to remember. The sound persons' job is to “reinforce the source.” If there is no source, it cannot be reinforced. A weak source will not sound any better by making it louder.



Microphones

There are two basic types of widely used microphones:

Dynamic and ***Condenser***.

Each has its own general characteristics and best use application.

The **dynamic** mic is constructed very similar to a loudspeaker – utilizing electro-magnetic induction. A voice coil is suspended in a magnetic field; a sound wave hits the diaphragm, connected to the voice coil, moving it, creating an electric signal. Dynamic mics are generally pretty sturdy. They can handle high sound pressure levels, sharp attacks, and survive being dropped. They are well suited to micing things like drums, amplifiers and vocalists.

The **condenser** mic is made by two very thin plates – one moveable and one fixed. The two plates form a capacitor (used to be called a condenser, hence the name). Sound hitting the moveable plate changes the capacitance, resulting in a changing electrical signal. The plates are very thin and very fragile. These mics require power to operate, usually “phantom” power from the board, an external power supply or a battery. Condenser mics are very sensitive and produce a fairly flat frequency response. They are well suited for micing acoustic instruments like guitars and flutes as well as choirs. They cannot handle extremely high sound pressure levels (remember the thin plates) or being dropped.

Things to remember about microphones:

- * **sensitivity**
- * **proximity**
- * **leakage**
- * **polar response**
- * **frequency response**
- * **transient response**

Sensitivity

This is a relative measurement of microphones measured in output voltage. Condenser mics are generally more sensitive than Dynamic mics. A more sensitive mic will pick up weaker signals like acoustic guitars and soft voices.

Proximity

This is the distance between the source and the mic. *The closer the mic, the boomier or bassier the sound*, since low frequencies require more energy to respond to than do high frequencies. Low frequencies are also less directional. Another factor is the inverse square law – the level of sound decreases 50% for every doubling of the distance from the sound source.

Leakage

Leakage is the term referring to a mic picking up signal other than the desired source. The further a mic is from its intended source the more extraneous noise it will pick up. For best control, place mics as close to the source as possible.

Polar response pattern

This is the pattern in which a mic is most sensitive. A **cardioid** pattern is kind of heart shaped; sensitive in the front and sides, but not the back. The SHURE SM57 and SM58 are cardioid pattern mics. **Omni-directional** means just that, 360 degrees of pick up. **Uni-directional** means the opposite; a narrow angle of pick up. It should be remembered that polar response changes at different frequencies. Most diagrams show this.

Frequency response

Frequency response refers to the total range and sensitivity at given frequencies. Generally, condenser mics have a flatter response than dynamic mics. This is to say they are equally sensitive to all frequencies in their range. The SM-58 (dynamic) has a peak in its response curve in the mid-range (somewhere around 8KHz). This is the “presence” sound for vocals; which is why they are often used for that application.

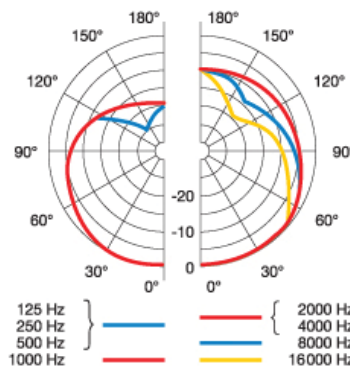
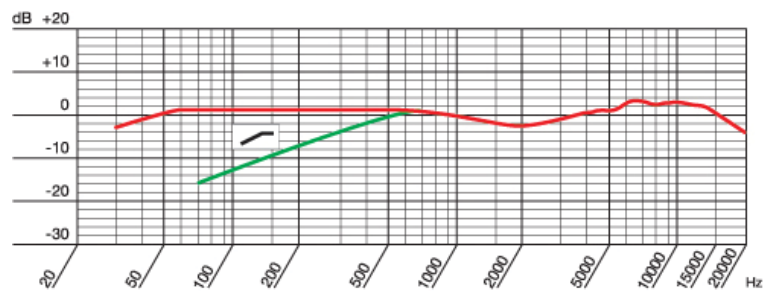
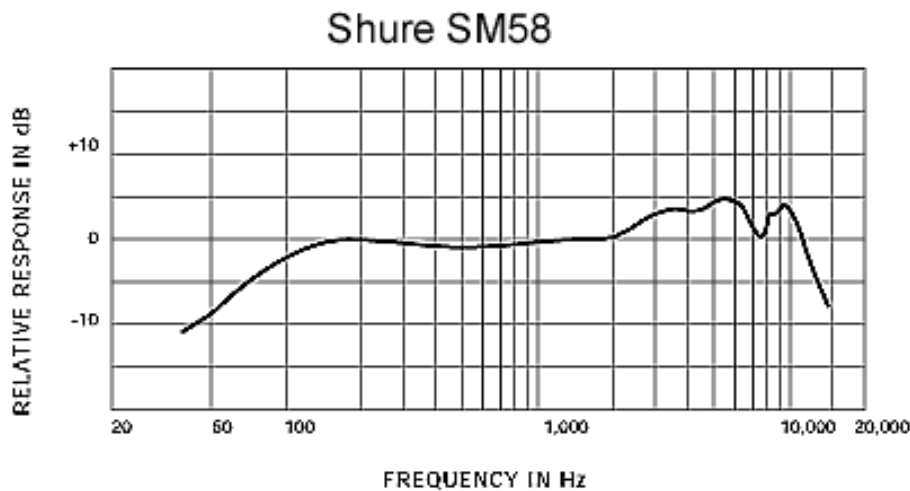
Transient response

This is the measure of how quickly the diaphragm of a mic reacts to a wave form. Dynamic mics have a slower response than do condensers, resulting in a “guttier” sound rather than a “crystal clear” sound.

Below are examples of frequency response charts for Shure SM58 (dynamic) and AKG C3000B (condenser) microphones.

The SM58 has a notch in its response curve at about 8kHz; note also the low end response compared to the C3000B.

The C3000B chart shows a frequency graph as well as a polar response diagram. Notice the different polar pattern for different frequencies, and the resulting response curve obtained by using the low cut filter switch on the mic.



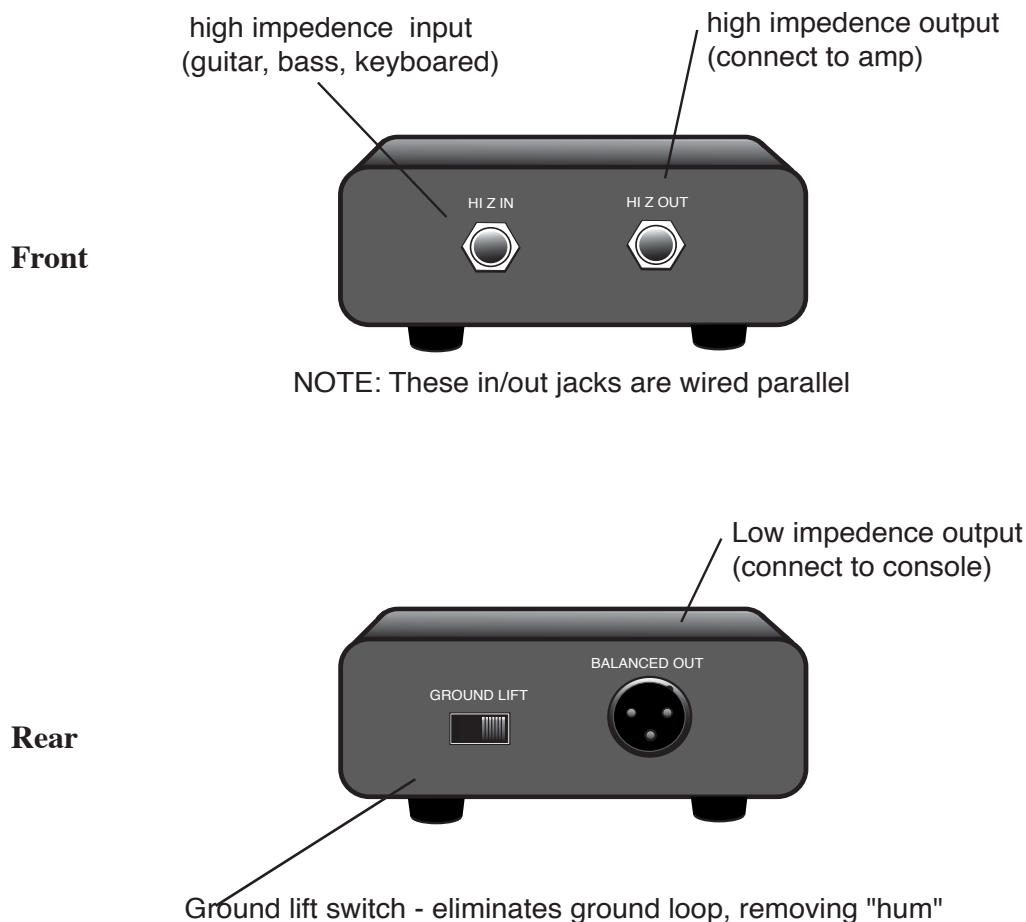
C 3000B



Direct Boxes

Direct boxes are essentially transformers that change a high impedance (unbalanced/Hi Z) signal to a low impedance (balanced/Low Z) signal. Cables coming from instruments like guitars, basses, keyboards, etc. are all Hi Z (Somewhere in the range of 5k-Ohms to 20k-Ohms). To send that signal over a great distance without any signal loss, it must be changed to a Low Z signal. A direct box takes an unbalanced signal and changes it to a balanced signal (up to about 600 Ohms) via a transformer in the box.

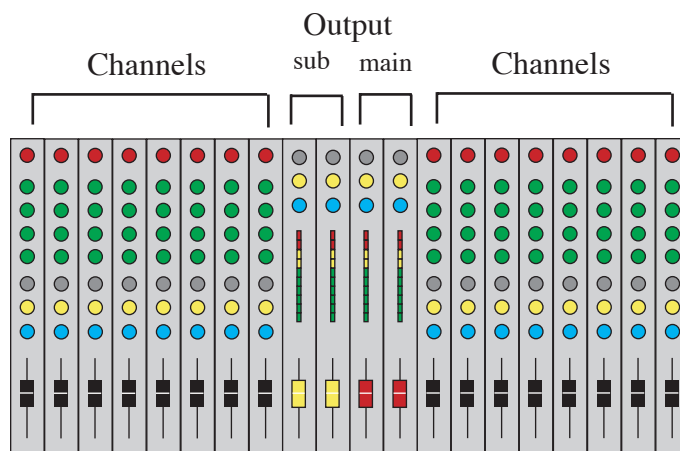
Sample use: from the bass or keyboard to the Hi Z input. From the Low Z output to the board; from the Hi Z output to the amp. This set-up allows for the amp to be used as a monitor for the player, keeping the stage volume to a minimum and the actual bass or synth sound to be heard comfortably by the congregation.





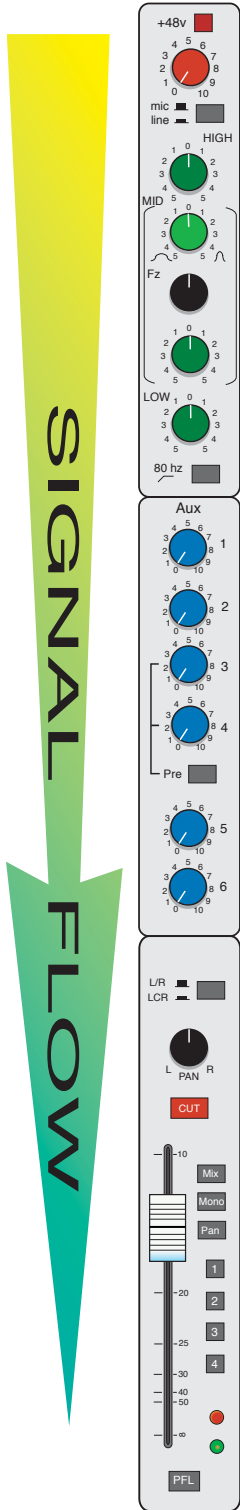
Mixer Basics

A mixer combines several inputs into a few outputs. Mixers are divided into input, output, and subgroup sections. The input section consists of channels; each channel is one input. Think of it as a bunch of streams flowing into one river. Each channel will have certain key elements, varying somewhat depending on the particular make and model. The first is the actual input jack. This is where you physically plug in the microphone cable. The second is the gain knob, which adjusts the input level. Next are the Equalization or E.Q. knobs, usually bass, treble and mid-range controls. After that you will probably find auxiliary channel knobs enabling you to create a separate mix for stage monitors, effect unit or a recording device. Depending again on the particular make and model, there may be sub-group assign buttons sending the channel output to a sub-group before going to the main output. Finally comes the channel volume; this may be a knob or a slider control. All of the input channels will look the same, replicated several times over.

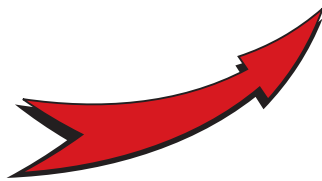
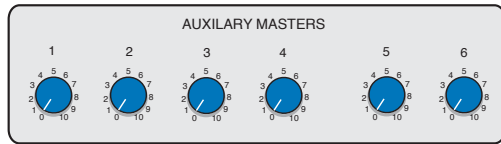


Signal Flow Overview

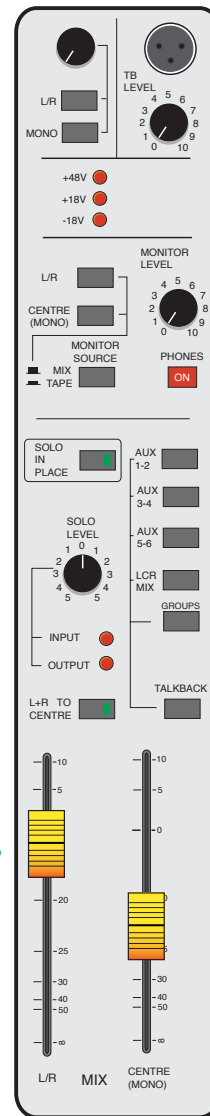
Channel Strip



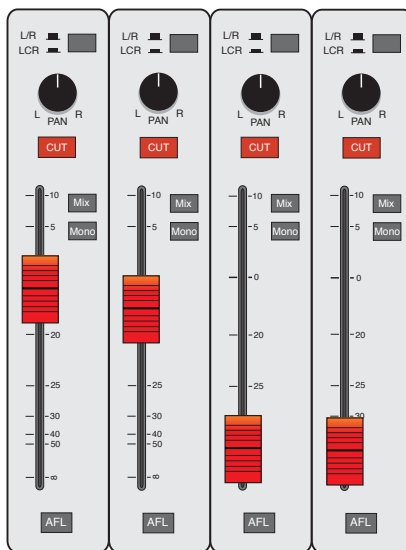
The basic flow of sound to the house speakers is shown by the green arrows; **channel to sub-group to main mix**. A separate mix sent to the monitors is shown by the red arrow; **auxiliaries to auxiliary masters**.



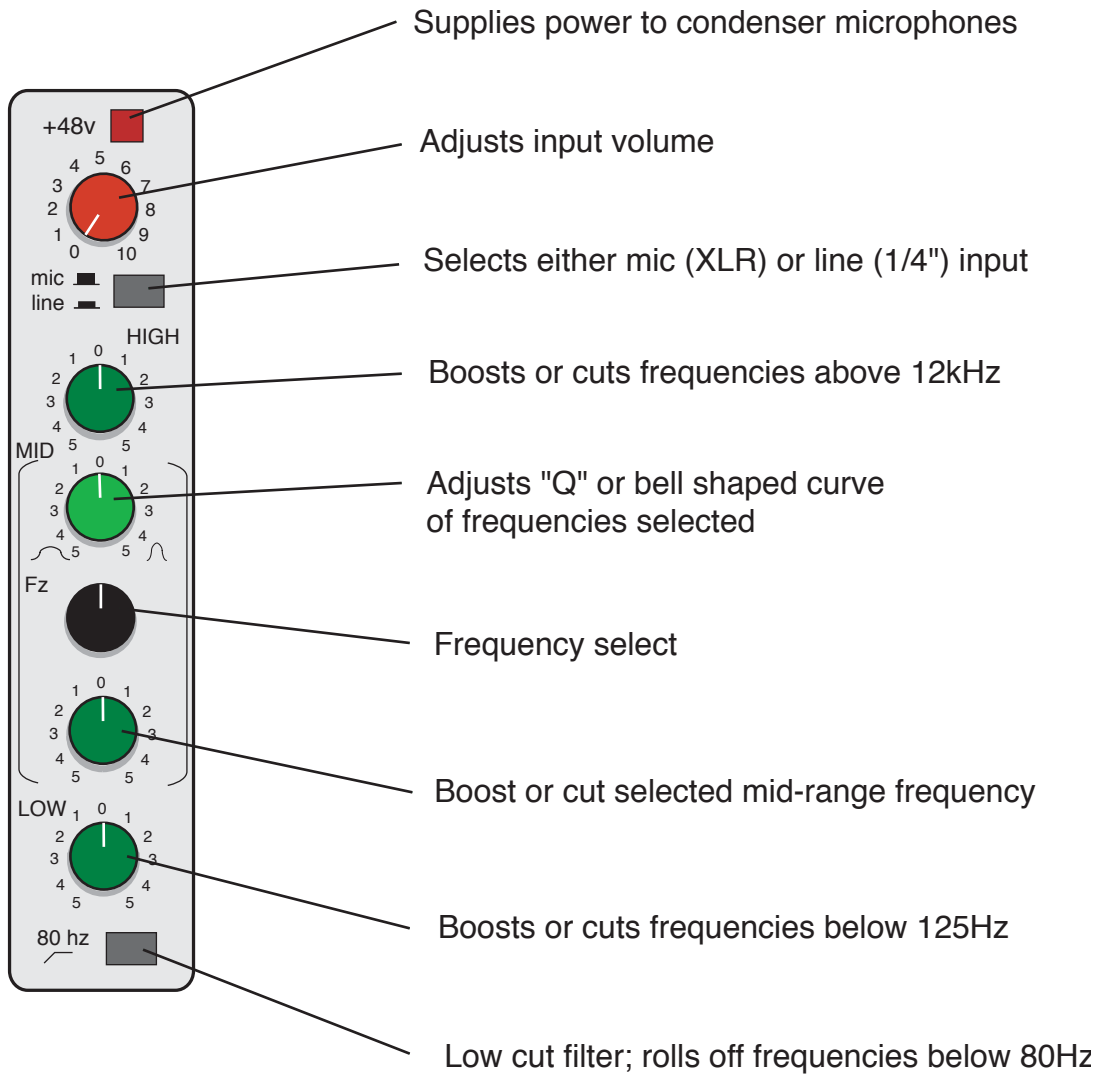
Main Master



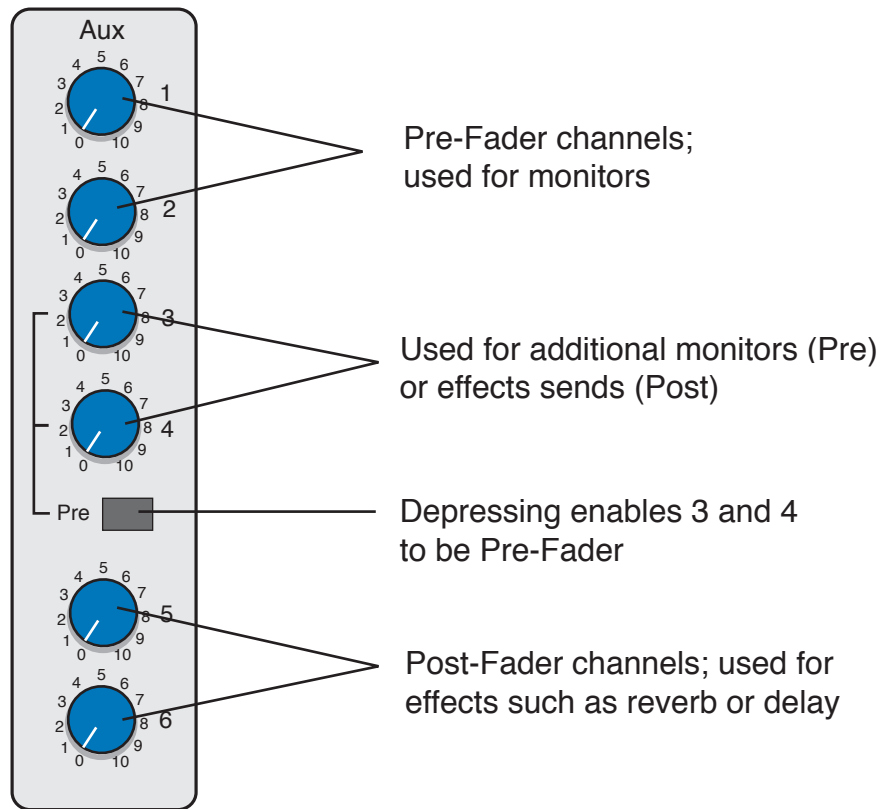
Subgroups



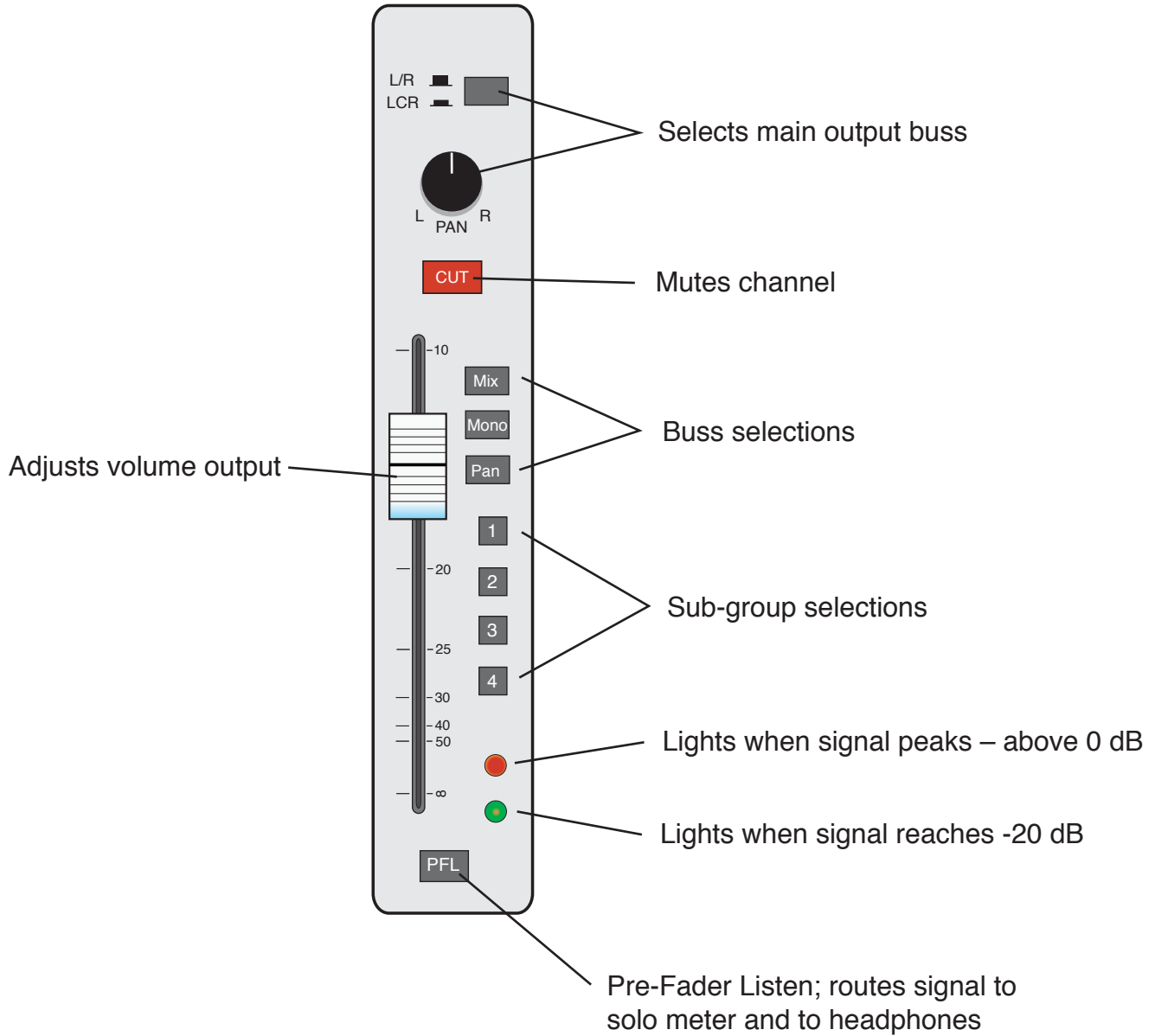
Channel Input and EQ section



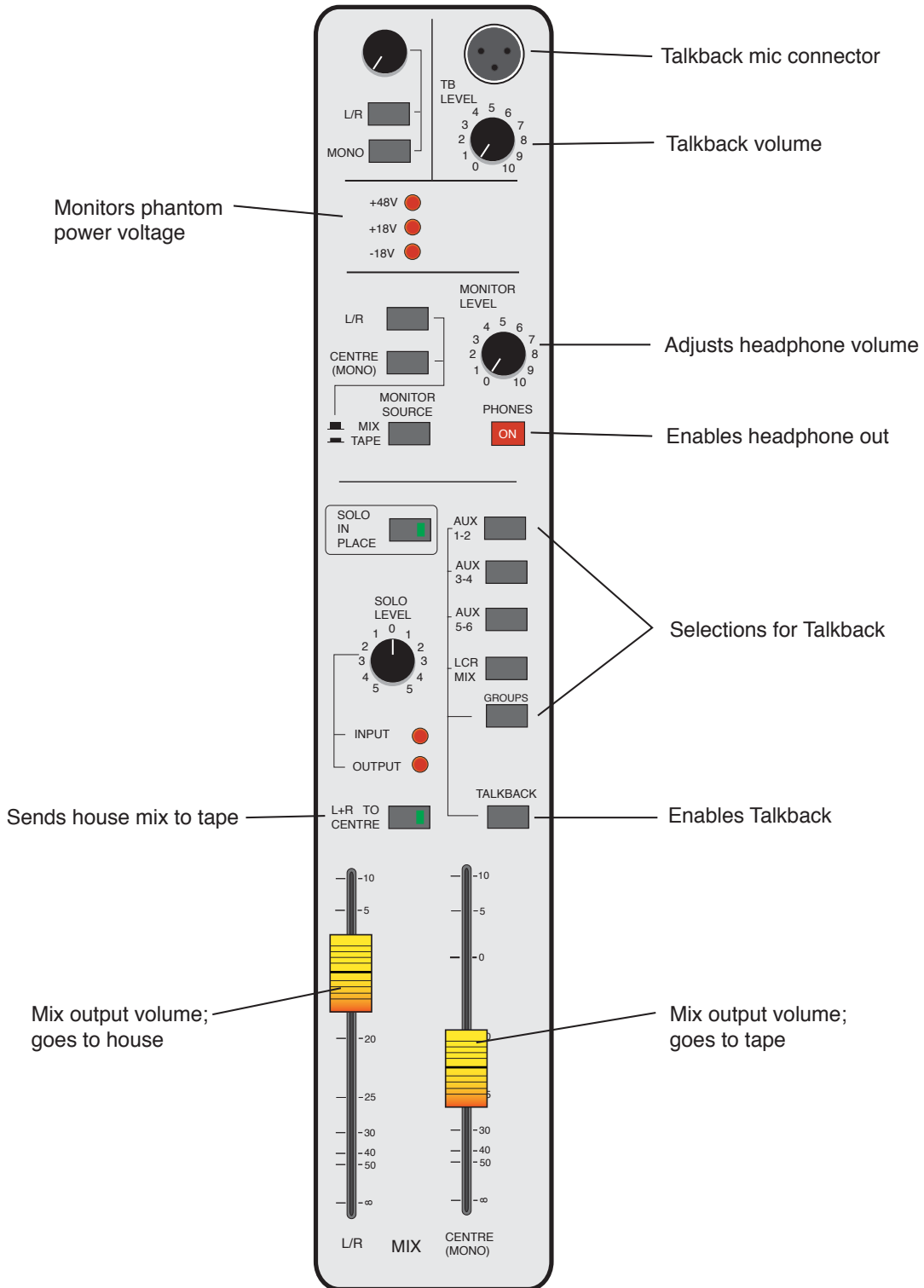
Channel Auxiliary section



Channel Fader section



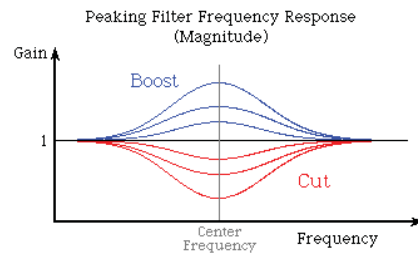
Master section





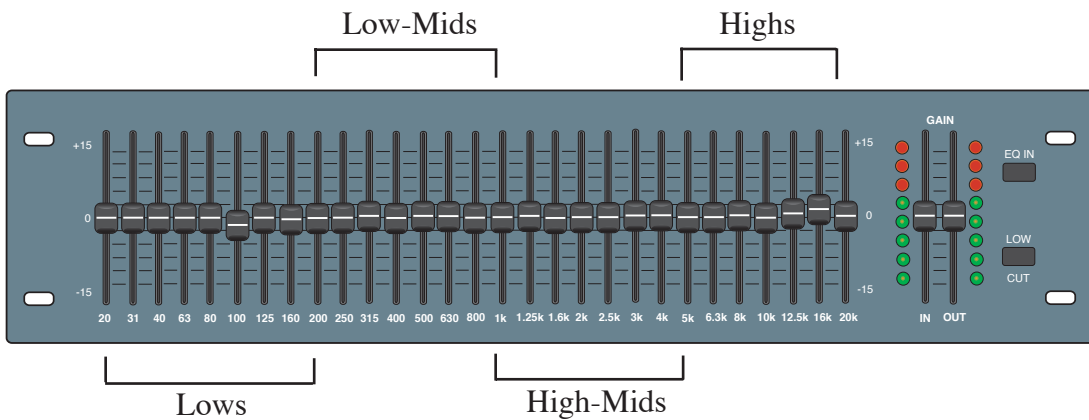
Equalization

Equalization (EQ) is the boosting or cutting of a signal at a given frequency. There are three types of equalizers. **Graphic EQ's** have a bunch of sliders, (usually 15 or 31) one per frequency. When used, these show “graphically” the overall bias; lows on the left and highs on the right. **Sweepable EQ's**, such as the mid-range on many boards, have a boost/cut knob and a frequency select knob. One knob can be “swept” to the desired frequency, which can then be boost or cut. It should be noted (see your mixer manual for more detail) that the frequency knob will select a *range* of frequencies in a bell-shaped curve, centering on the one selected. For example, 800Hz is selected; 700Hz and 900Hz will also be somewhat affected, as will 600Hz and 1kHz but to a lesser degree. Think of it as a curve with the highest point being the one you want. A **Parametric EQ** has an additional knob to eliminate this effect, by changing the shape of the “curve” or **parameter** to pinpoint exactly which frequency is to be affected.

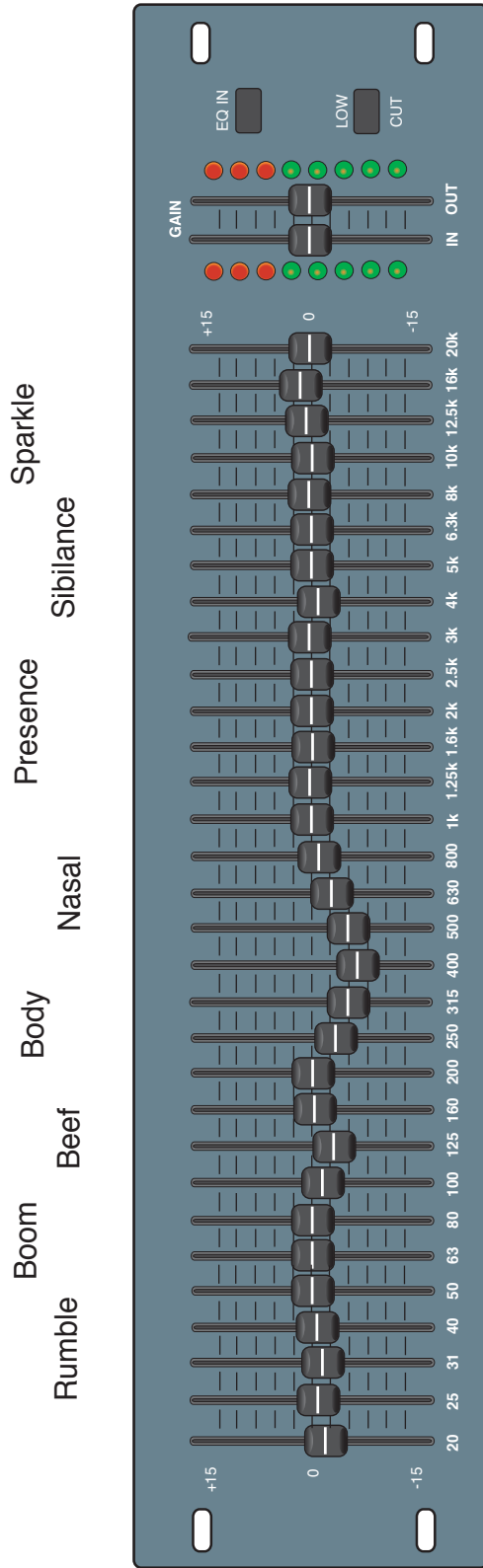


Setting EQ is a tedious and somewhat subjective task. The goal is “natural” sound. Whenever a natural sound is sent through a myriad of components, it is changed. Equalization helps to get back as close as possible to the original signal.

Graphic EQ:



Frequency Map



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Fundamental
 Harmonic overtones
 NOTE: □ These representations are approximate



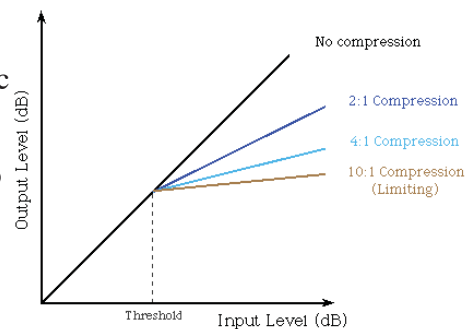
Compression

Compressors are generally **inserted** into a signal path. This puts it post gain, but pre-fader. This allows the unit to do its duty without being affected by fader volume levels. Compressors can be inserted into individual channels, sub-groups, main, and auxiliary channels. For example:

A band consisting of drums, bass, keys, guitars and vocalists could use one compressor inserted into sub-group 1 (drums), another compressor inserted into the lead vocalists' channel, and one used for the overall main mix. This set-up allows the most dynamic instruments (drums & lead vocal) to be “tamed” a bit before being added to the overall mix. Compressing the overall mix “balances” the dynamic of the band as a whole.

Compressing the vocal and instrument subgroups helps to bring the vocals out front and balances acoustic guitars with the other instruments. The lapel mic works well to have a compressor *and* a graphic EQ inserted to it, giving quite a bit of control for that channel.

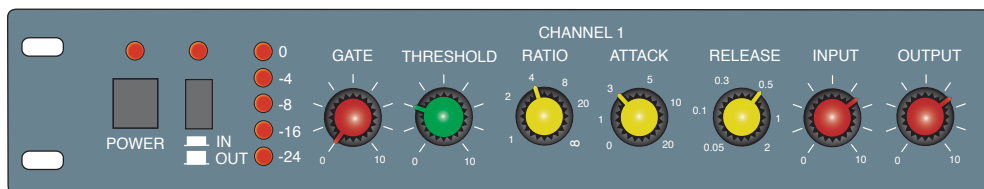
Compressor Input/Output Characteristic



So what does a compressor do?

Essentially, a compressor squishes the dynamic range and averages a signals' overall level. This is accomplished by setting the **threshold** to reduce any signal past that point by a certain amount, or **ratio**. Since the loudest signals are now compressed, the average output signal is perceived to be louder.

This often brightens up the mid-range frequencies, bringing vocals or acoustic guitars more out front. Compression on a bass or kick drum can make them sound a little “beefier”. Like any effect, though, a little goes a long way.





Crossover Networks

Speaker drivers are designed to operate within a certain optimum range; many home stereo tweeters operate from 3500Hz -20kHz, many compression-driver horns for PA use are effective from around 1200Hz-15kHz. Sending frequencies lower or higher than what a driver is designed for won't be effectively reproduced and will cause damage to the unit. This is where crossovers come in.

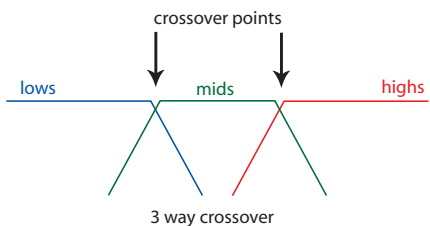
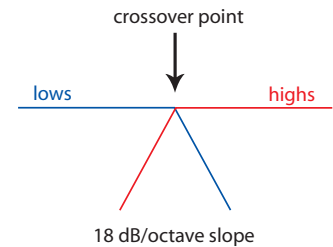
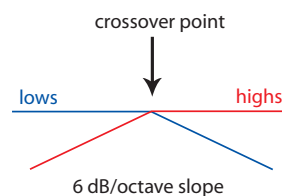
A crossover or frequency dividing network separates frequencies at a given point. Many two-way (horn+woofer) and three-way (horn+mid range+woofer) speakers have a crossover built-in, sending the frequencies into their respective components. High frequencies go to horns and tweeters, lows go to woofers. The advantage of this system is that one amplifier can power a speaker with two or more drivers. Built-in crossovers are designed specifically for the speaker cabinets they occupy. However, many two and three-way systems have separate inputs for lows and highs, requiring a separate crossover unit and separate amplifiers for each driver; the advantage being that different size amps can be used for lows and highs. Low frequencies require more power to reproduce than highs, so a larger amp can be used. These units give much more control over what frequencies get sent where.

What you need to know about crossovers and speakers are **frequency** and **slope**.

You need to know what the specific ranges are for the drivers in the speaker cabinet. The manufacturer will provide this information. Slope is the rate at which a frequency is cut. This is measured in dB/octave.

A 6 dB slope at 1200Hz means that for every octave (2400, 4800 etc.) the signal decreases by 6 dB. A two-way crossover divides into two parts – lows and highs, and so has one crossover point.

A three-way system has two points that separate lows, mids, and highs. Some crossover units will have adjustable frequency points with pre-set slopes. Others have adjustable slopes that can configure differently for lows, mids, and highs. The crossover unit is configured between the equalizer and the amps.





Amps & Speakers

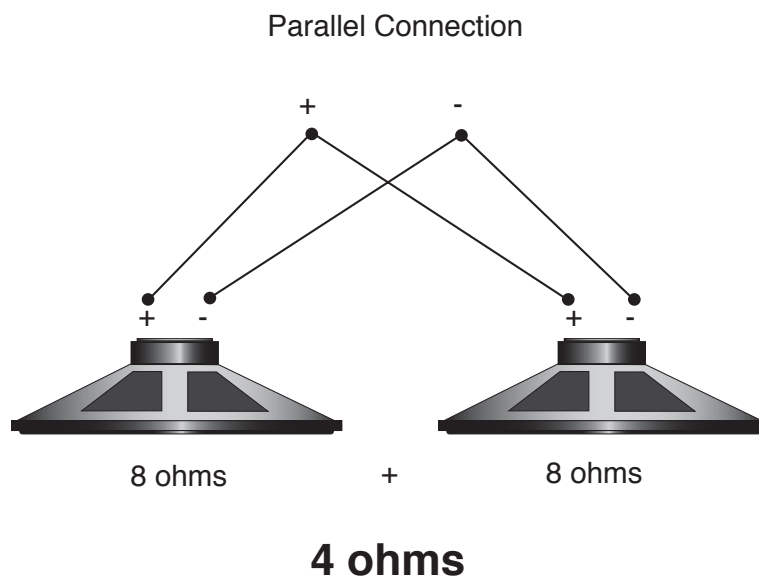
Amplifiers are designed to operate at a certain optimum impedance or **load**, measured in ohms. To prolong the amps life and produce maximum distortion-free power, the total speaker load should fall within range of the amplifier's designed operating load.

Most professional P.A. amps operate within a 4-16 ohm load.

Speakers are the last component in the signal chain, converting electronic signals back into sound pressure waves. Speakers exert resistance (impedance) on the flow of electrons from the amplifier. If the speaker's impedance is too high, output will be reduced. If it is too low, output will be louder but overheating will likely occur.

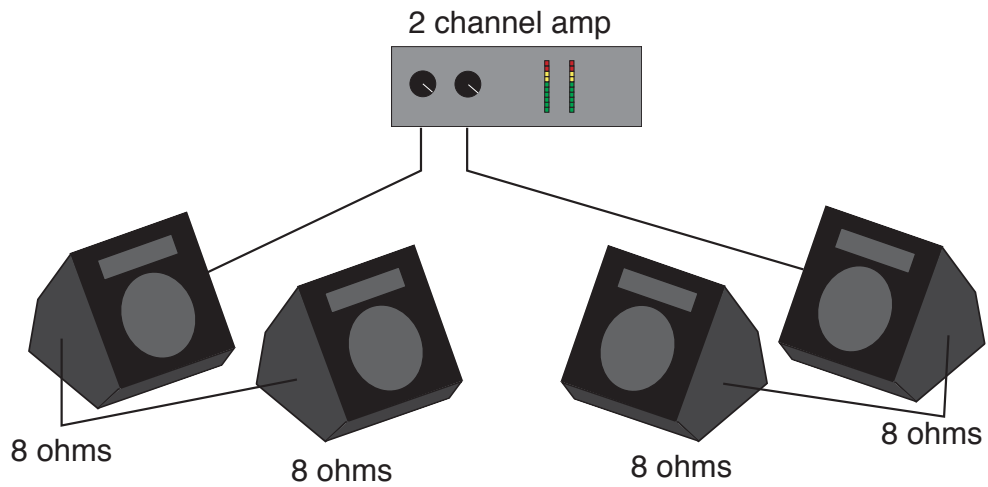
Two Kinds of Loads

Most speakers are connected in parallel – often referred to as daisy chained or in-line with each other. Think of a parallel connection as putting a Y attachment on a faucet and connecting two garden hoses. The important thing to remember when connecting speakers in parallel is that the total load equals the speaker impedance divided by the number of speakers. Two 8 ohm speakers connected in parallel equals a total load of 4 ohms (8 ohms/2 speakers).

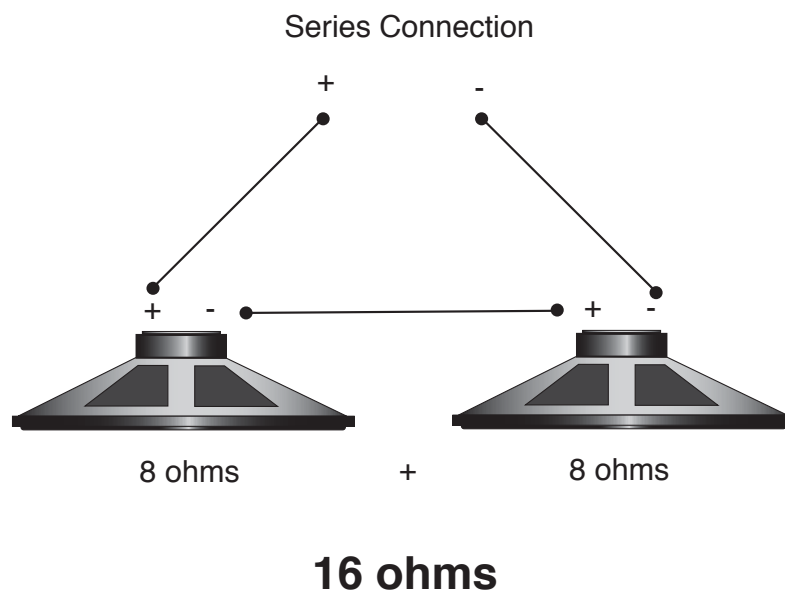


A stereo (2 channel) amp can drive a total of four 8 ohm speakers – 2 on each channel.

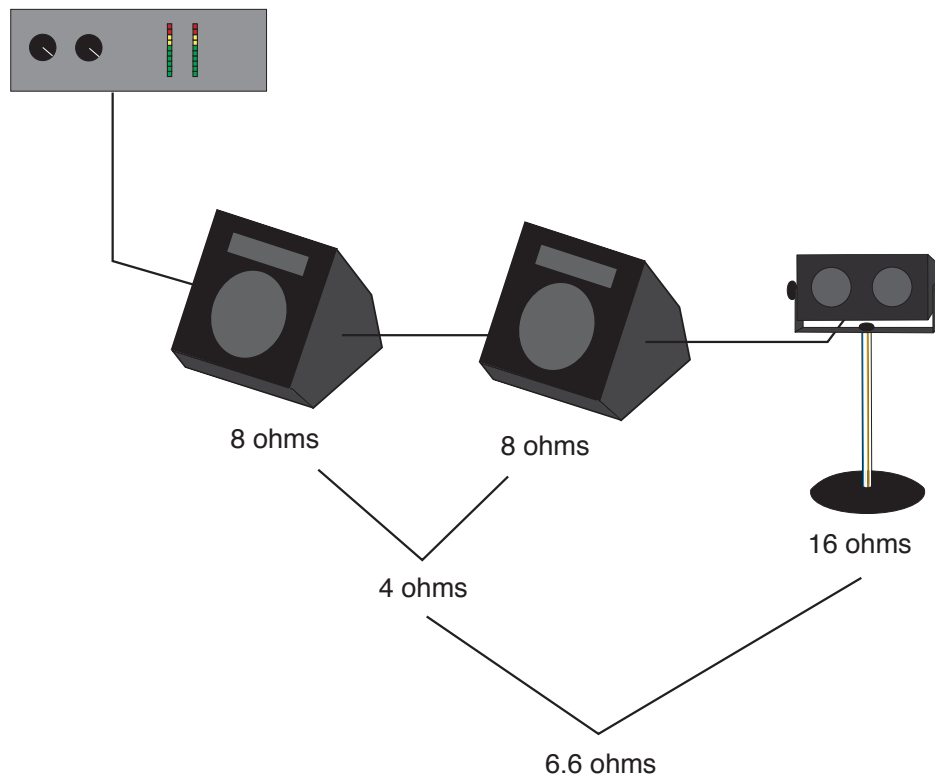
4 ohm load per side of amp



Series connections are like connecting garden hoses end-to-end, resulting in a longer hose. The total load in a series connection is the sum of the speakers impedances. Two 8 ohm speakers connected in series equals 16 ohms.



Some small “hot spot” monitors contain two 4” speakers wired in series, making their load 16 ohms. When these are connected with the bigger 8 ohm floor monitors, a series/parallel connection is obtained. Two 8 ohm monitors (4 ohm load) plus one “hot spot” (16 ohms), combine at 20 ohms. Divided by 3 (number of monitors), results in a total load of 6.6 ohms. The series/parallel routine allows for many speakers to be powered by one amp.



$$\frac{20 \text{ ohms (4+16)}}{3 \text{ speakers}} = 6.6$$



Setting Up

Start by **neutralizing** the board. This means that all of the **gain** knobs are fully counter clockwise; as should be all of the **aux**'s. All of the **EQ** and **pan** knobs should be at 12 o'clock. All of the little **buttons** should be in the up position. All of the **faders** should be off (down).

Next, **assess the situation**: How many mics are needed? What kinds? Should the guitar be mic'd or run direct? Where do mics and monitors need to be placed?

Keep in mind that to some extent, less is more when it comes to setting up mics and monitors. Think of the most efficient way to place them.

Turning On the System

The procedure for powering up any sound system is as follows:

I. Make all your connections

Connect all outboard gear (EQ's, Effects etc.)

Connect all mics and D.I.'s

Connect main and monitor speakers

II. Turn on console and outboard gear

III. Turn on main and monitor amps

After plugging everything in and turning on the power, check each input for signal. Do this by pressing the **PFL** button on the desired channel (this will enable the incoming signal to be monitored by the solo meter). Next, **assign the channel** to a sub-group or to "MIX". **Set the mic level** by adjusting the **gain knob** slowly clockwise (*someone should be talking into the microphone at this point - "test test" or reciting Hamlet or something*), The little green "-20" light should flicker and the solo meter, located in the upper middle section of the board, should peak at "0 dB". Adjust the gain knob until this happens.

Line check

You may notice that at this point there is no sound coming out of the loudspeakers; don't be alarmed. Now slowly raise the **main output fader to "0"** and then slowly raise the **channel fader** to the desired volume level. Volume for the person speaking is controlled by this fader. Use the same procedure for each input. The idea here is to make sure you have a signal coming into the board and coming out of the mains, monitors and to tape; EQ and fine tuning can be done during a sound check, where the band runs through a song.

To power down, do the turning on steps in reverse order:

III. Amps off

II. Console/Gear off

I. Disconnect



Creating a Mix

Adapted from The Art of Mixing by David Gibson

There are basically four ways that sounds can be set apart in a mix:

***Volume level**

***Frequency (EQ)**

***Special effects** (such as reverb)

***Placement** (panning R or L)

In many church and club sound system configurations, **volume levels** and **EQ** are the only adjustable tools available – mostly EQ since there is so much natural sound from the stage in the first place (piano & drums for example). As a starting point for a mix, the following is suggested, using the term **Apparent Volume Level** (a relative loudness scale from 1-5, 1 being loudest).

AVL 1 Emcee or Worship Leader should be the loudest vocal in the mains and monitors. Separate by boosting highs around 5-8kHz or so, depending on the particular individual.

AVL 2 Singers are the next loudest in the mains and in the monitors; EQ for the most natural tones. A bit of compression will help separate vocals from the instruments.

AVL 2 Piano: since the piano has much of its volume naturally, care is needed to separate it by boosting the highs (1kHz or so) for clarity. Too much volume adds to muddiness if other instruments are involved.

AVL 3 Bass should be run direct if possible. Boost around 40Hz for bottom end and 1kHz for presence.

AVL 4 Synthesizer is best run direct and usually needs no EQ. Listen closely and ride the fader for AVL; it should be just recognizable or else it can become overpowering.

AVL 3 & 4 Guitars

Acoustic: Make sure there is a good signal coming from the player. Either mic it with a condenser or run direct. Direct is preferable. Boost 12kHz for clarity, but watch mids around 600Hz for muddiness and obscurity.

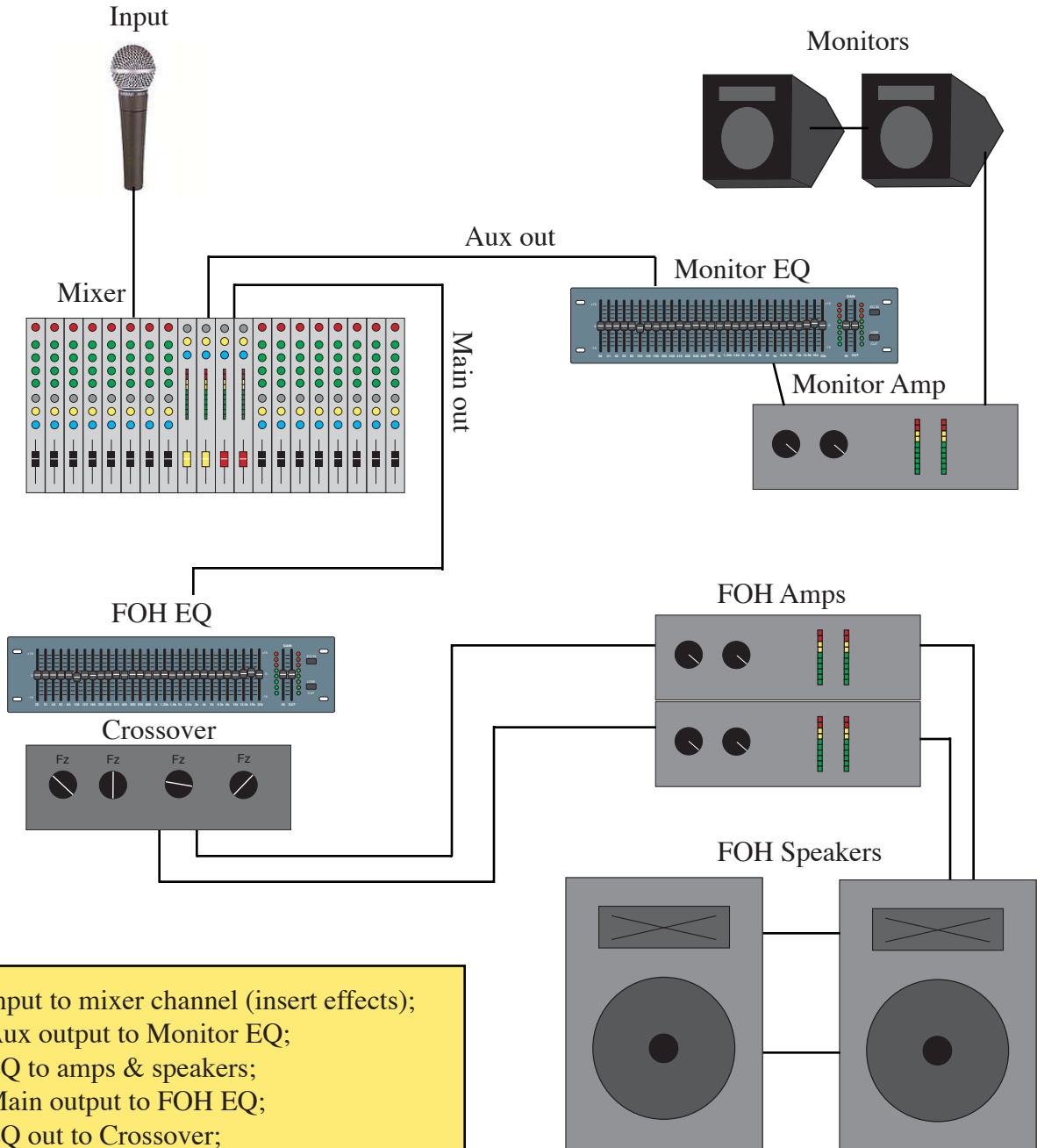
Electric: Either mic the amp off axis or run direct from a line output from the amp. Little or no EQ and ride the fader for AVL; especially if the player is switching between rhythm and lead.

Drums: Unless it is a very large church building, drums generally don't need amplification. These can be used as a reference for volume levels of everything else. As such, drums, bass and the main rhythm instrument (guitar or piano) are best kept at the same AVL. When micing drums, what needs most be heard is the snare/hi-hat while the kick drum felt. *Felt* can translate to frequencies below about 80Hz; this is achieved with a kick drum mic with the hi's EQ'ed out.

NOTE: It is effective to run the vocals through one sub-group and instruments through another sub-group. Drums are dealt with easily in their own subgroup (Kick, snare/hi-hat, toms, overhead all together). Keep in mind that creating a mix is a subjective exercise. Depending on the type of music and venue, your mix will be different.



Basic System Diagram



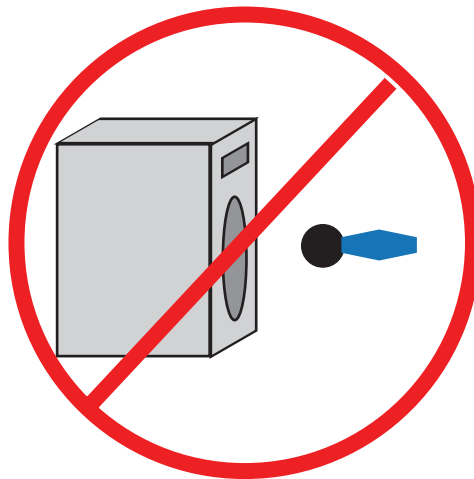
- Input to mixer channel (insert effects);
- Aux output to Monitor EQ;
- EQ to amps & speakers;
- Main output to FOH EQ;
- EQ out to Crossover;
- Crossover outputs to Low & High amps & Speakers



Avoiding Feedback

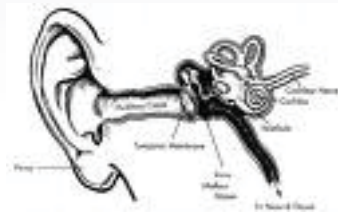
Feedback occurs when the amplified signal enters the signal path at a greater volume level than the original signal. If you speak into a mic with the loudspeaker behind you, facing the mic as you are, a loud howl will occur if the volume reaching the mic from the speaker is louder than your voice. To complicate matters, sound bounces around everywhere, so even if a mic is not directly facing a speaker, sound can bounce around and sneak up from behind as well. It is best to keep mics behind front-of-house speakers and pointing away from stage monitors. Use directional mics when stage monitors are used. Cut the frequencies that “ring” using EQ. Singling out a frequency is done easiest by first boosting it. This lets you know exactly where it is, and subsequently dealt with.

Generally, feedback results from high volume levels in the wrong places. A good way to prevent feedback is to keep monitors away from mics and keep monitor levels low enough to prevent leakage, and mics close enough to their respective sources to keep gain levels to a minimum. If stage monitors are too loud, a singer who inadvertently gets their mic too close to it will cause feedback.



A few words about volume levels or Sound Pressure Levels

The threshold of hearing (where most people begin to hear sounds) is referenced as “0 dB”. The threshold of feeling, the SPL that will cause discomfort in a listener 50% of the time, occurs at about 118 dB between the frequencies of 200 Hz and 10 kHz. The threshold of pain occurs at 140 dB. Keep in mind that dB are logarithmic; so 118 dB is way loud and 140 dB is extremely way loud.



BE CAREFUL WITH YOUR EARS!



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Notes