Sound Reinforcement and Recording of Jazz and Large Ensembles –

Practical and Easy to Use Basic and Advanced Ideas

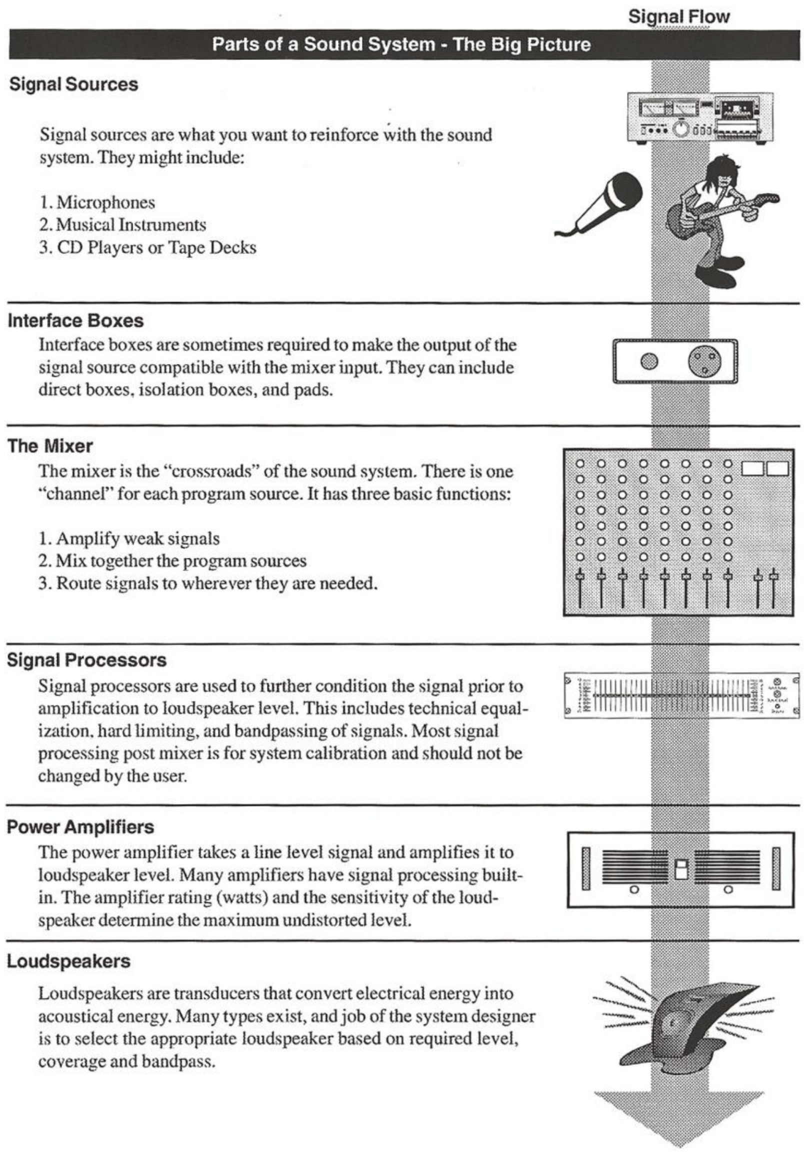
Midwest Band Clinic
December 18, 2008
Grand Ballroom
2:30 – 3:30 PM

Clinicians

Jim Warrick
Mark Morette
Scott Bauer
Scott Steiner

Sound System Components

-speakers- main & monitor, amplifiers, mixers- FOH & monitors, microphones, processing gear- EQ & effects



Signal Flow

Parts of a Sound System - The Big Picture

Signal Sources

Signal sources are what you want to reinforce with the sound system. They might include:

1. Microphones
2. Musical Instruments
3. CD Players or Tape Decks

Interface Boxes

Interface boxes are sometimes required to make the output of the signal source compatible with the mixer input. They can include direct boxes, isolation boxes, and pads.

The Mixer

The mixer is the "crossroads" of the sound system. There is one "channel" for each program source. It has three basic functions:

1. Amplify weak signals
2. Mix together the program sources
3. Route signals to wherever they are needed.

Signal Processors

Signal processors are used to further condition the signal prior to amplification to loudspeaker level. This includes technical equalization, hard limiting, and bandpassing of signals. Most signal processing post mixer is for system calibration and should not be changed by the user.

Power Amplifiers

The power amplifier takes a line level signal and amplifies it to loudspeaker level. Many amplifiers have signal processing built-in. The amplifier rating (watts) and the sensitivity of the loudspeaker determine the maximum undistorted level.

Loudspeakers

Loudspeakers are transducers that convert electrical energy into acoustical energy. Many types exist, and job of the system designer is to select the appropriate loudspeaker based on required level, coverage and bandpass.

Concepts of Live Mixing

-Sound stage (electronic event) vs. Sound Reinforcement (electronic reinforcement of an acoustical event)

-Conductor responsible for the *performance* on stage – Sound Engineer responsible for the *presentation* to the audience *(both parties need to gain an understanding of what is required by each)*

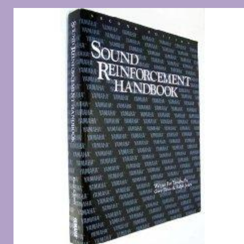
-Mixer should be considered an extension of the band and approached as another musical instrument

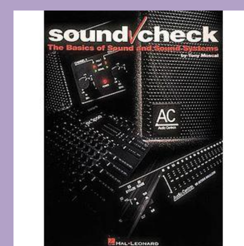
-Mixing is mostly subjective- except for missing a cue or feedback

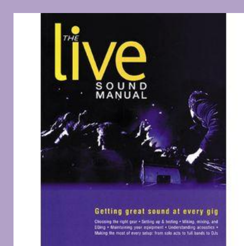
-Mixing considerations include the type of venue, audience size, type of music

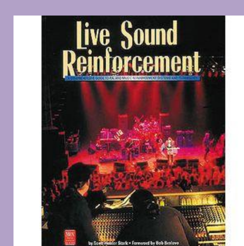
RECOMMENDED REFERENCES

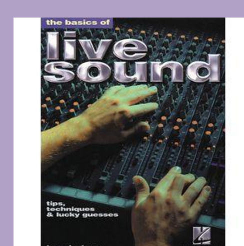
from Hal Leonard Publishing











Microphones as a Mixing Tool

-choose microphones that will achieve a desired effect based on their characteristics *(don't use a mic with a presence peak on a trumpet if you don't want it to cut through; don't use a large diaphragm mic on cymbals if you don't want the tubs)*

Types of microphones

USB Microphone

A USB mic contains all the elements of a traditional microphone: capsule, diaphragm, etc. Where it differs from other microphones is its inclusion of two additional circuits: an onboard preamp and an analog-to-digital (A/D) converter. The preamp makes it unnecessary for the USB mic to be connected to a mixer or external mic preamp. The A/D converter changes the mic's output from analog (voltage) to digital (data), so it can be plugged directly into a computer and read by recording software. That makes mobile digital recording as easy as plugging in the mic, launching your DAW software, and hitting "record!"

Condenser Microphone

The condenser microphone is a very simple mechanical system, with almost no moving parts compared with other microphone designs. It is also one of the oldest microphone types, dating back to the early 1900's. It is simply a thin stretched conductive diaphragm held close to a metal disk called a backplate. This arrangement basically produces a capacitor, and is given its electric charge by an external voltage source. This source is often phantom powered, but in many cases condenser mics have dedicated power supply units. When sound pressure acts on the diaphragm it vibrates slightly in response to the waveform. This causes the capacitance to vary in a like manner, which causes a variance in its output voltage. This voltage variation is the signal output of the microphone. There are many different types of condenser microphones, but they are all based on these basic principles. One example of a highly popular condenser microphone is Neumann's U87, shown here.

Dynamic Microphone

A dynamic mic is one in which audio signal is generated by the motion of a conductor within a magnetic field. In most dynamic mics, a very thin, light, diaphragm moves in response to sound pressure. The diaphragm's motion causes a voice coil that is suspended in a magnetic field to move, generating a small electric current. Generally less expensive than condenser mics (although very high quality dynamics can be quite expensive), dynamics feature quite robust construction, can often handle very high SPLs (Sound Pressure Levels), and do not require an external power source to operate. Because of the mechanical nature of their operation, dynamic mics are commonly less sensitive to transients, and may not reproduce quite the high frequency "detail" other types of mics can produce. Dynamic mics are very common in live applications. In the studio, dynamics are often used to record electric guitars, drums and more. One example of a highly popular dynamic microphone is Shure's hand held SM58, shown here.

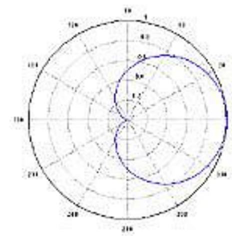
Ribbon Microphone

A type of velocity microphone. A velocity microphone responds to the velocity of air molecules passing it rather than the Sound Pressure Level, which is what most other microphones respond to. In many cases this functional difference isn't important, but it can certainly be an issue on a windy day. Very old ribbon mics could be destroyed from the air velocity created just by carrying them across a room; today's ribbon mics can handle the rigors of daily studio use. A ribbon mic works by loosely suspending a small element (usually a corrugated strip of metal) in a strong magnetic field. This "ribbon" is moved by the action of air molecules and when it moves it cuts across the magnetic lines of flux causing a signal to be generated. Naturally ribbon mics have a figure 8 pickup pattern. You can think of it like a window blind; it is easily moved by wind blowing at it, but usually doesn't move when wind blows across it from left to right. Ribbon mics were the first commercially successful directional microphones. One example of a highly popular ribbon microphone is Royer's F121, shown here.

Understanding microphone polar patterns

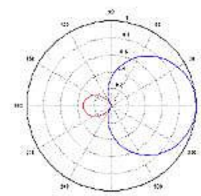
-cardioid patterns are utilized in 95% of live sound applications
-the more directional a microphone the more gain
- as frequency goes higher, patterns become more directional

Cardioid



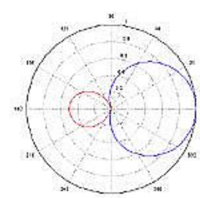
Characterized by strong sensitivity to audio from the front of the mic, good sensitivity on the sides (at 90 degrees, 6 dB less than the front), and good rejection of sound from the rear, the Cardioid pattern can almost be visualized as a "heart-shaped" pattern (hence its name). The ability to reject sound from the rear makes Cardioid patterns very useful in multi-miking situations, and where it is not desirable to capture a large amount of room ambience. Popular in both studio and live use (where rear rejection cuts down on feedback and ambient noise). Keep in mind that like all non-omnidirectional mics, Cardioid mics will exhibit pronounced proximity effect.

Supercardioid



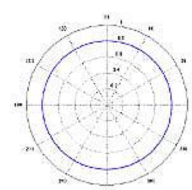
The Supercardioid pattern is very similar to, and often confused with, the Hypercardioid pattern. The Supercardioid pattern is slightly less directional than the Hypercardioid pattern, but the rear lobe of sensitivity is also much smaller in the Supercardioid.

Hypercardioid



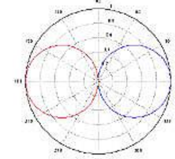
Hypercardioid patterns are similar to Cardioid and Supercardioid patterns in that the primary sensitivity is in the front of the microphone. They differ, however, in that the point of least sensitivity is at the 150 - 160 and 200 - 210 degree positions (as opposed to directly behind the microphone in a Cardioid pattern). Hypercardioid microphones are thus considered even more directional than Cardioid and Supercardioid microphones. Hypercardioid microphones are frequently used in situations where maximum isolation is desired between sound sources.

Omnidirectional



Microphones are said to be omnidirectional if they can detect sound equally from all directions. An Omnidirectional microphone will not exhibit a pronounced proximity effect. Omnidirectional microphones are not normally utilized in live sound applications.

Figure-8



A microphone polar pattern in which the mic is (nearly) equally sensitive to sounds picked up from front and back, but not sensitive to sounds on the sides. This produces a pattern that looks like a figure 8 on paper, where the microphone is at the point of crossover on the 8. The pattern is also known as bi-directional.

Understanding microphone diaphragm sizes

Large Diaphragm

Any microphone with a diaphragm larger than (and potentially including) 3/4" is considered to be a Large Diaphragm microphone. In general, Large Diaphragm microphones tend to have a "big" sound that engineers find especially pleasing where a little more character might be advantageous, such as is the case with most vocals. Large diaphragms are generally more sensitive than small diaphragm or medium diaphragm mics because of the increased surface area. A common myth is that large diaphragm mics capture more low frequencies than small diaphragm mics. Sometimes their coloration may make it sound like this is the case, but a properly designed small diaphragm mic is more likely to be accurate throughout a wide range of frequencies, whereas the coloration of a large diaphragm mic can tend to enhance certain desirable characteristics in a sound, which sometimes amounts to more apparent bass or low end. While there are many great Large Diaphragm microphones available, AKG's C414 microphone (shown here) is one example.

Medium Diaphragm

The definition of Medium Diaphragm is a potentially controversial subject. Historically there have been large diaphragm and small diaphragm mics, but more recently the medium size has begun carving out its own category, though not everyone agrees on the precise upper and lower limits. Most professionals and manufacturers agree that any microphone with a diaphragm near 5/8" to 3/4" can be characterized as a Medium Diaphragm microphone. Generally speaking, Medium Diaphragm microphones tend to do a good job of accurately catching transients and high frequency content (as a small diaphragm would) while delivering a slightly fuller, round and potentially warmer sound (as a large diaphragm might). While there are many great Medium Diaphragm microphones available, RODE's NT3 microphone (shown here) is one example.

Small Diaphragm

While there are no final standards regarding a diaphragm size that defines Small Diaphragm, most professionals and manufacturers agree that any diaphragm smaller than 5/8" would be considered a Small Diaphragm. Generally speaking, Small Diaphragm microphones tend to do a good job of capturing high frequency content and transients. They will tend to have a bit more "air" to their sound and often have less coloration than medium or large diaphragm microphones. Most of this is due to the reduced mass of the smaller diaphragm, which allows it to more closely follow any air disturbances it is subjected to. While there are many great Small Diaphragm microphones available, Neumann's KM 184 microphone (shown here) is one example.

-recording and live sound equipment may appear to be the same, but can have very different specifications and applications

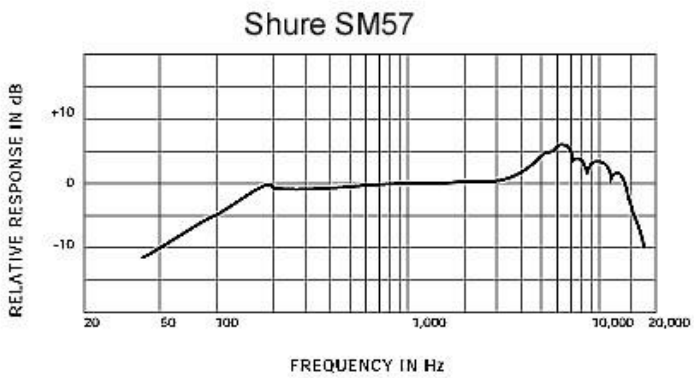
-can a mic be TOO good for live sound? recording mics may be too sensitive for sound reinforcement applications since the mics are in the same environment as the speakers

-utilizing the fewest number of mics will typically produce better sound quality

-evaluate the application and what sound objectives need to be obtained (minimal or maximum sound reinforcement, control poor acoustic situation, similar or diverse instrumentation)

How to read a microphone frequency chart

A microphone's Frequency Chart can tell you a lot about which situations are appropriate for a given microphone and which situations are not. In theory, Frequency Charts are generated at the factory by testing the microphones in an anechoic chamber. An anechoic chamber is a specially constructed room just for audio testing. The idea here is to create a controlled atmosphere where each microphone can be tested equally, so the room is completely dead, without any form of sound reflection. Generally, a speaker is set up in front of the microphone that is being tested and pink noise is played (pink noise is all frequencies with equal energy in every octave). The microphone is routed into a spectrum analyzer that measures the output and a Frequency Response Chart is produced. The chart is usually over the 20 Hz to 20 kHz range, which is the range of human hearing.



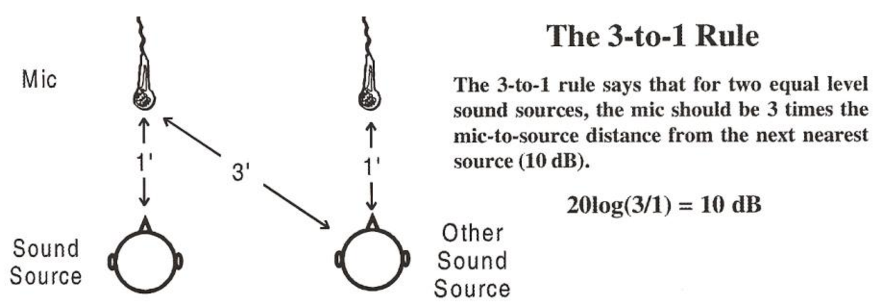
So, how do you read it? The horizontal numbers in a Microphone Frequency Chart represent frequencies (again, usually over the 20 Hz to 20 kHz range) and the vertical numbers represents relative responses in dB (Decibels). As you look at a Frequency Chart, you can tell how a given microphone performs at certain frequencies. How is this information helpful? Well, let's look at the famous Shure SM57's frequency chart:

The frequency response of the SM57 makes it especially good for certain instruments such as a snare drum because the fundamental frequency of the snare resides in the 150Hz to 250Hz range – right where the SM57 Microphone Frequency Chart shows that the SM57 response is flat, or neutral. In other words, at this frequency, what you hear going into the microphone is what you will tend to hear coming out – nothing more, nothing less. The presence bump to the right of the chart is just where the frequency of the "snap" of the snare resides. In addition, it's rolled off low end makes it great for de-accentuating the kick drum which is often very close in proximity. This combination is what most engineers are looking for in a great snare drum mic – the ability to capture the true sound of the snare, accentuate its snap and reject other instruments in close proximity.

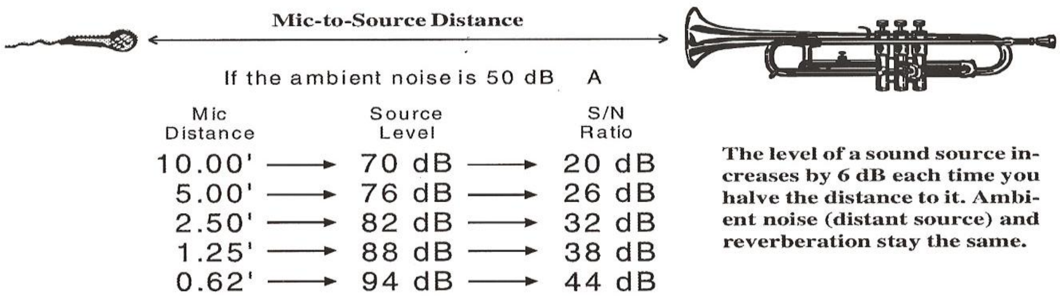
Rules and Considerations

-proximity effect is a factor to consider when using cardioids microphones

-following the **3 to 1 rule** prevents phasing issues when using multiple mics in close proximity to each other, such as a saxophone section

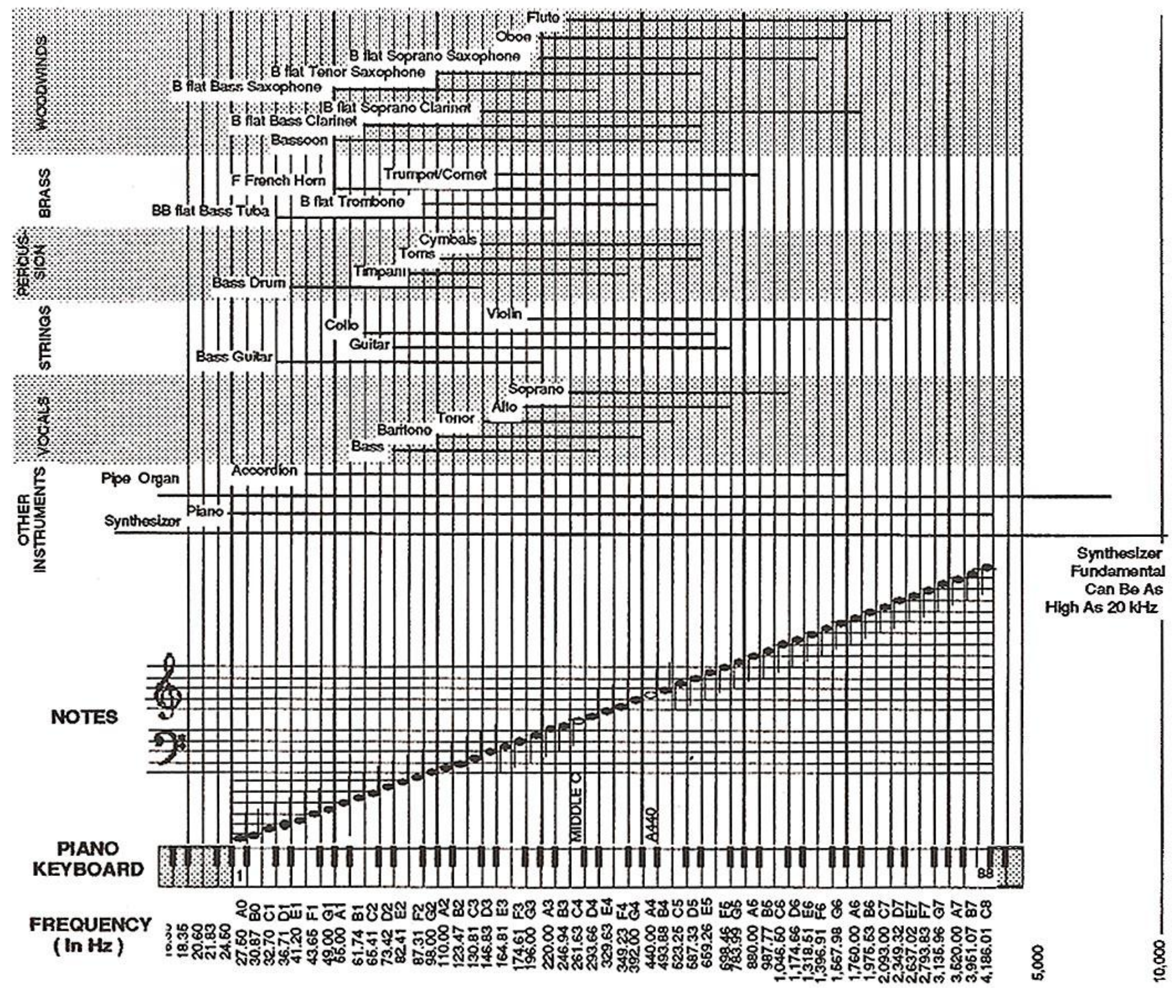


-doubling the mic to source distance (Inverse square law) halves the volume



-proximity to speakers will reduce gain available

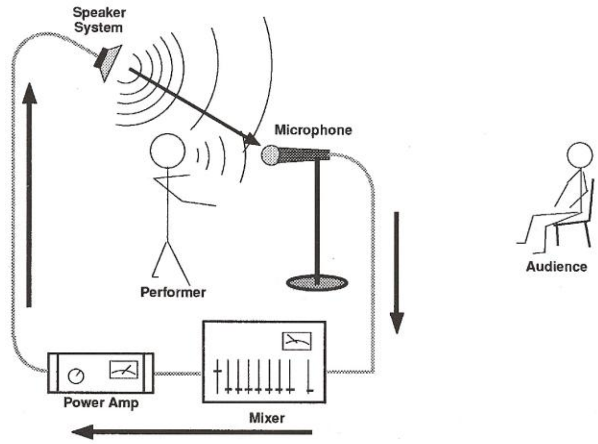
-consider mic placement, such as an acoustic bass mic next to drums



Frequency range of typical instruments and singing voices

Feedback Control

-feedback is caused when the intended source being amplified is regenerated through the sound system and the mic can *hear* the speakers louder than the intended source



Typical acoustical and electrical feedback path

-questions to ask to diagnose feedback
Which mic is feeding back?
Can I turn it down?
Are there too many mics turned on at one time?
Are the tone controls boosted?
Is it too far from the source?
Is the mic too close to the speakers?
Is the mic pointed towards the speakers?
Can it be repositioned?
Is the feedback from a reflection or direct sound?
Can the feedback be equalized out?
Are the speakers or monitors too loud?

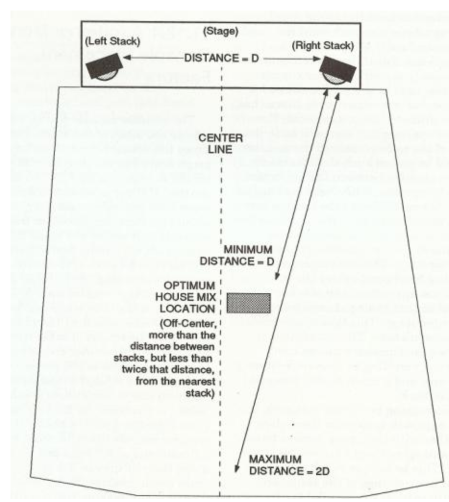
-feedback can be fixed by: changing the distance or orientation of the mic to the speakers; using more directional mics or speakers; reducing the mic to source distance; cutting offending frequencies on an equalizer; changing time of arrival; follow Rules and Considerations in the section abov*e (reducing volume alone may not make it go away)*

Monitors

-place logically and use sparingly (it is OK to set them up and not use them)
-have band play first before adding material into to monitors
-be aware of monitor levels and their wash off stage into the front of house

Mixer Familiarization & Setup

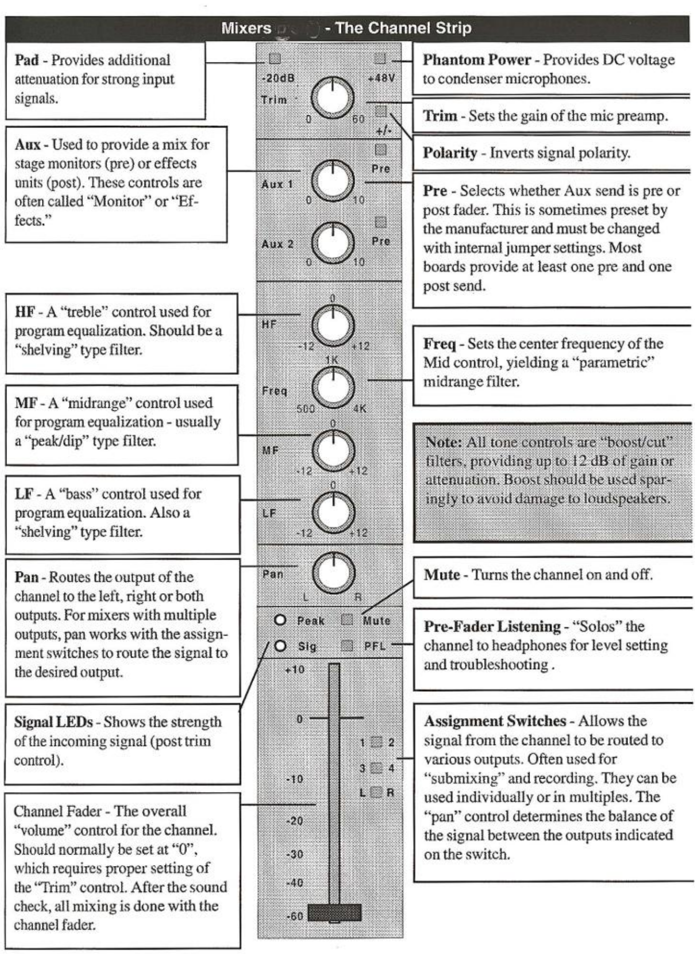
-place mixer in hall just off center while avoiding balconies and booths



Instrument	Signal Processing
Snare top microphone	Noise gate - Equalizer - Limiter
Snare bottom microphone	Noise gate - Equalizer - Limiter
Hi-hat microphone	*
Rack tom left microphone	Noise gate
Rack tom right microphone	Noise gate
Floor tom microphone	Noise gate
Left overhead	Compression
Right overhead	Compression
Electronic drum snare DI	*
Electronic drum kick DI	Equalizer
Electronic drums left mix out	Equalizer
Electronic drums right mix out	Equalizer
Electric guitar left microphone	Noise gate
Electric guitar right microphone	Noise gate
Bass synth DI	Equalizer - Limiter
Synthesizer DIs	Limiter
Lead vocal microphone	Equalizer - Limiter Exciter
Background vocal microphones	Limiter

* No external processing

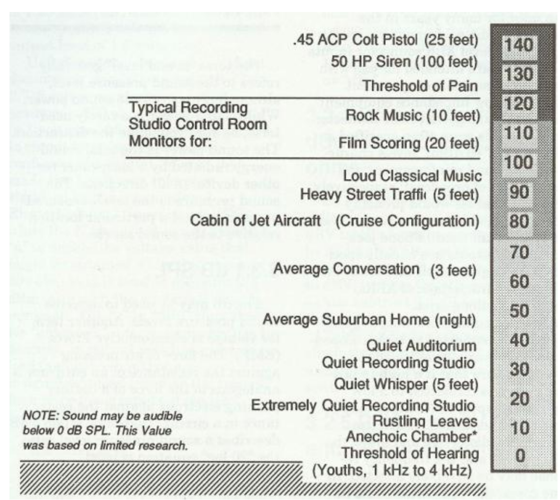
-familiarize yourself with the various sections of the mixer and processing gear



-use a minimal amount of EQ and cut whenever possible instead of boosting (*If you need tons of EQ, adjust positioning or choose a better microphone)*
-grouping inputs for sections and like instruments makes the mix easier *(drums, keys, horn sections)*

Sound Check Sequence & Mix

-use a condenser mic during line checks to ensure phantom power passes
-when setting balances and eq, listen to each instrument individually, then a section, then entire ensemble
- visually check mic placement when it is played into
-input gain should be set with fader at nominal, then bring up trim- remembering that gains will combine at the Master Output
-listen to a chart with the faders down first, then add missing instruments into the mix for presence *(Mix to the loudest acoustical instrument)*
-try to fix problems and balances acoustical first (*elevate bass speaker to de-couple from stage, short stick piano, change microphones, ensure no one is over blowing*)
-when using stage monitors only add what is needed for the band to play together (This is definitely a less is more scenario when it comes to the presentation)



Engineer Objectives

-control overall volume
-use finesse and approach the mixer as an instrument- don't be a throttle jockey (try to be seamless)
-the mix will be greatly a product of the environment, but should be in line with the desires of the director/producer *(or the performer's wife)* as much as possible
-don't over mix a performance
-the ultimate engineering goal is to produce a transparent mix and thus go unnoticed by the audience!

Sources

Davis & Jones. Yamaha Sound Reinforcement Handbook. Milwaukee: Hal Leonard Publishing Corp, 1990.

Sound System Setup. Lab Manual. Greenville, IN: Syn-Aud-Con, 1999