FASST

Field Associate Sound System Training

Pre-amplified Signal Processing

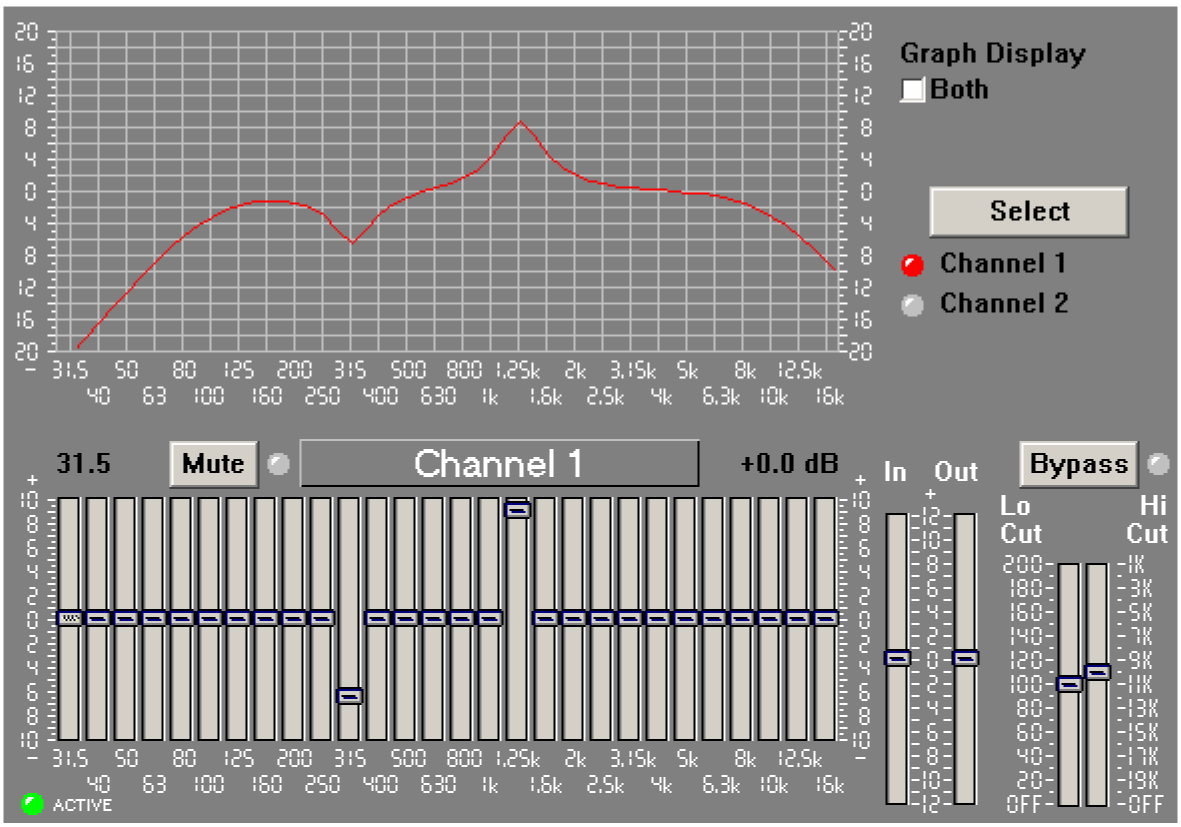
This section will touch on some of the common processing equipment that is used to manipulate audio signals prior to the amplification stage.

Equalizers

Far and away, the equalizer is the most commonly used and abused of signal processing equipment. Equalizers come in two basic designs: graphic and parametric. The graphic is easier to use, but less accurate. An artist might suggest that the graphic equalizer is like drawing a two dimensional object where the parametric allows the artist to create in three dimensions. The basic function of either type is to allow the user to boost or cut a range of frequencies by a certain amplitude. This range of frequencies is called the *bandwidth* and represented by the symbol Q.

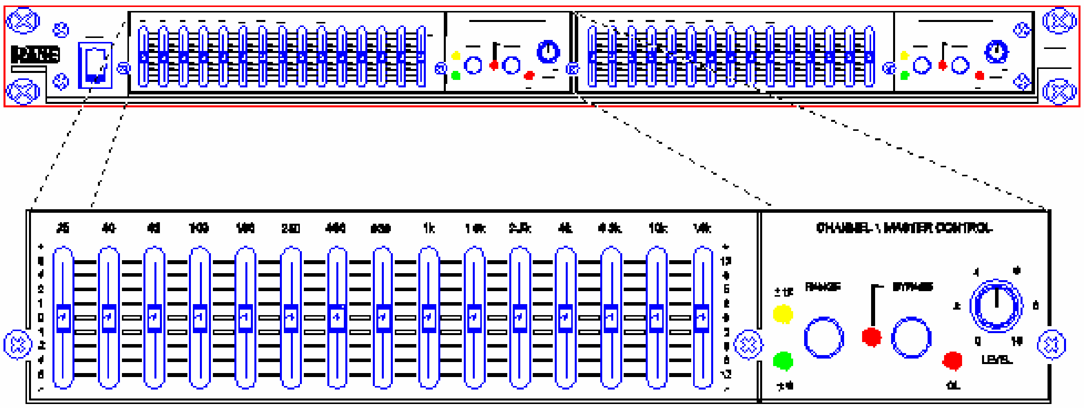
In a graphic equalizer, this bandwidth is fixed dependent upon the number of controls or faders that are available to the user. The more faders there are, the narrower the bandwidth that each controls. Graphic eq's range from the simplistic 2 band bass and treble controls up to a more intimidating 60 bands. Additional bands allow the user to make adjustments with an increased level of accuracy.

The bottom half of the image below shows the faders of a 28 band graphic equalizer. The upper half of the image demonstrates the net effect those faders have on the audio signal.

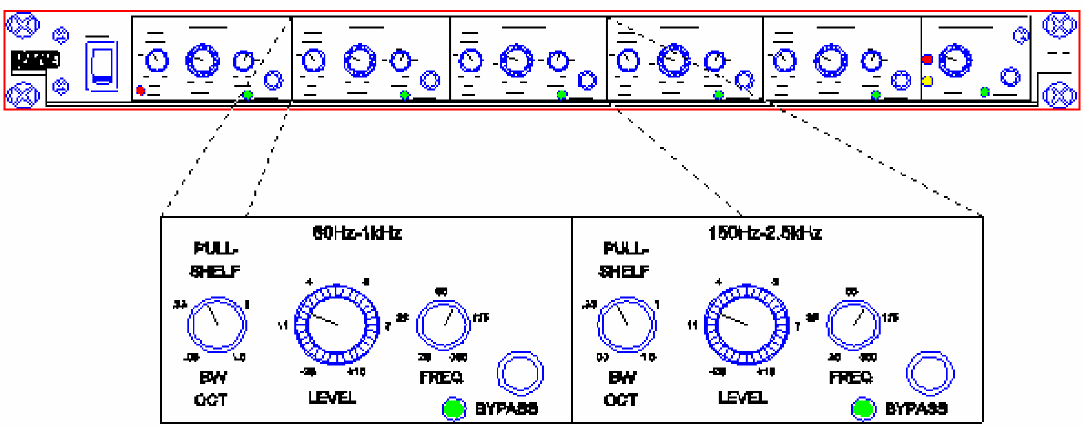


You may have heard graphic equalizers referred to as 1/3 or 2/3 octave. Those terms merely describe the bandwidth or Q of the faders. Each fader in a 1/3 octave equalizer controls 1/3 of the range of frequencies in that octave. Hence, the 1/3 octave equalizer will have more fader controls than the 2/3, because they are controlling a smaller range. It will probably be more expensive too!

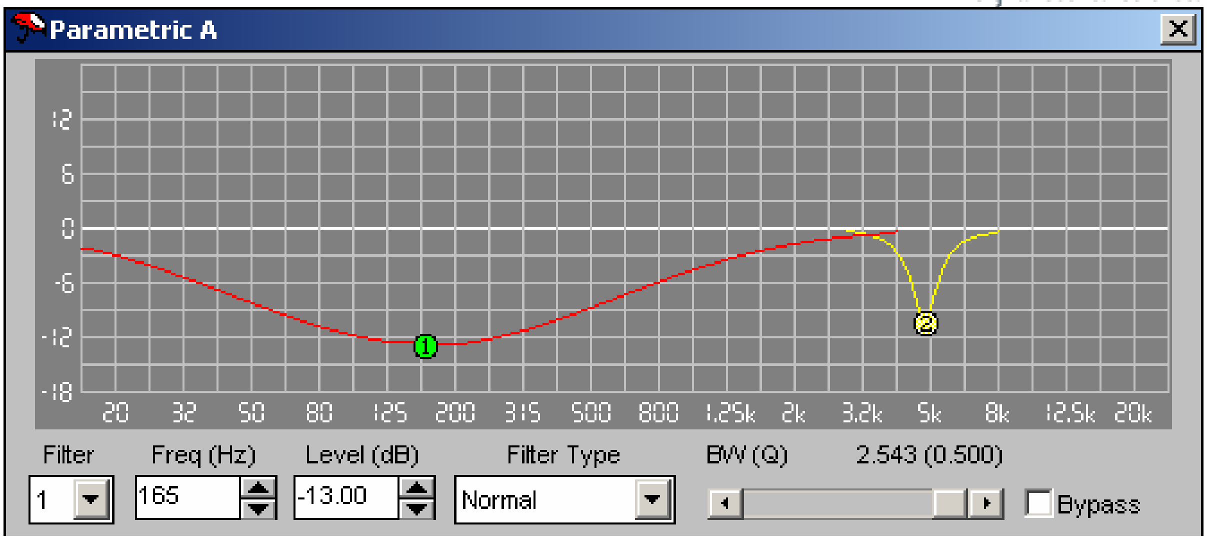
Graphic Equalizer



Parametric Equalizer



A parametric differs from a graphic in the following manner. In addition to selecting the frequency and amplitude, we can also adjust the bandwidth of the filter we are boosting or cutting. The following image shows two filters with very different bandwidth values. The first filter has a very wide Q which allows the cut at 165Hz to impact a large portion of the signal. By contrast, the second filter at 5kHz has a very narrow filter and only minimally affects the neighboring frequencies.



It's clear that this level of adjustment is nearly impossible to do without a Real Time Analyzer. Therefore, the graphics are still very popular, particularly with those who want to look cool. A parametric just isn't nearly as sexy as a graphic!

When setting equalizers there a few rules to live by, regardless of the type you are using.

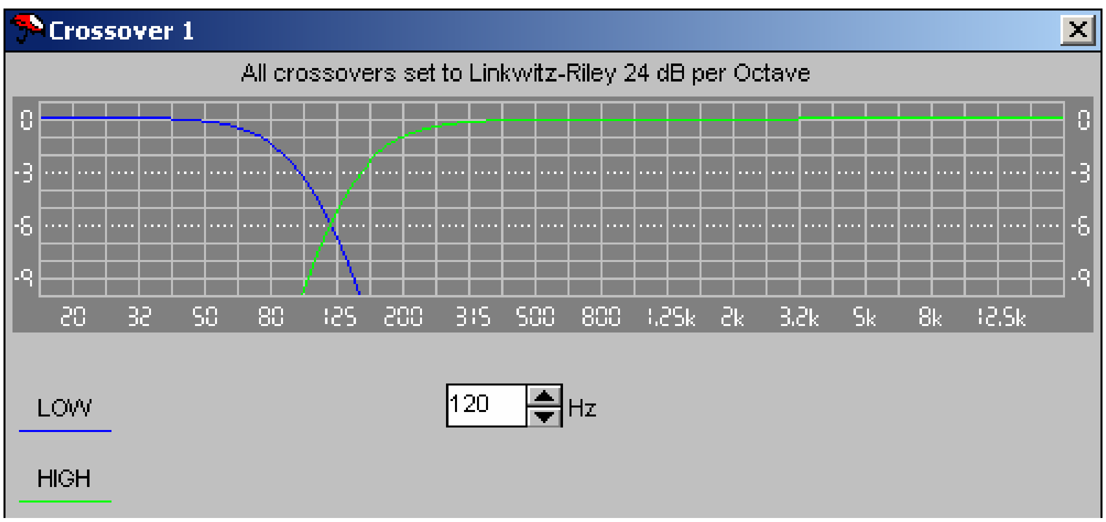
1. Boosting can kill! When a fader is raised, it is amplifying the signal within that bandwidth. This introduces the possibility of clipping the signal out at relatively low volumes. Clipping creates square waves that kill loudspeakers. Make it a policy to cut the frequencies that are too prominent. If you need to boost, limit it to 3dB.
2. From point number 1, we can see that a smiling EQ is not necessary a happy EQ. Most loudspeakers are unable to reproduce the entire audio spectrum anyway. A smiling EQ will only make the signal clip initially on the extreme high and low frequencies that are not going to be heard anyway. If you don't know how to set it, see point one or leave it flat.
3. Don't over EQ. An equalizer can only do so much. There may be characteristics of the room or speaker interactions that are creating problems that cannot be resolved with an EQ. If you don't like the way a system sounds, you may need different speakers…. or a bigger hammer.
4. After setting equalizers, it is good to put a security cover over the controls to keep wandering fingers out. It is amazing how many restaurant and retail employees are 'in a band'.

Crossovers.

Demand for higher fidelity has led to an increased application of subwoofers in both restaurant and retail environments. These higher fidelity systems often utilize an active crossover to improve the clarity and efficiency of a sound system.

Note: Passive crossovers offer no user interaction and are therefore discussed in the loudspeaker module.

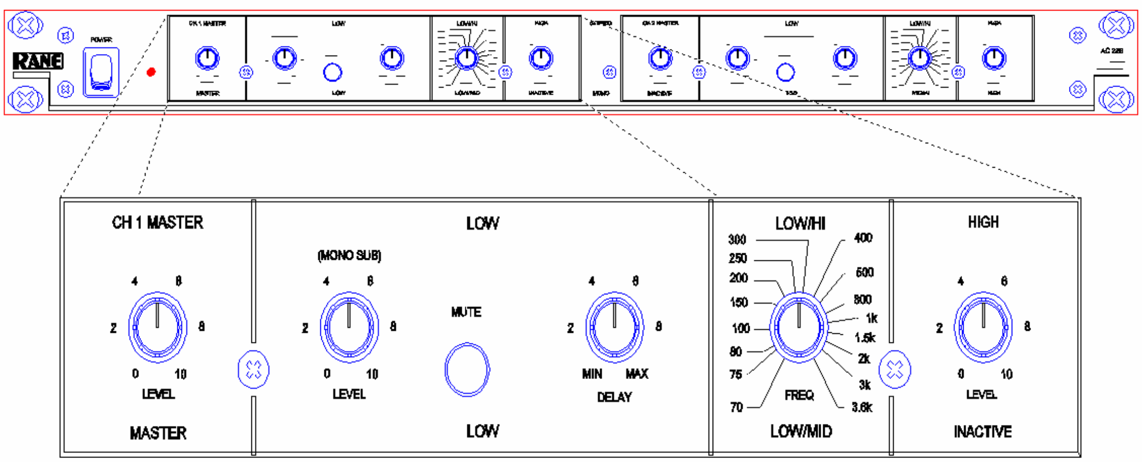
Crossovers are used to route frequencies to specific drivers. They take a full range audio signal and divide it usually into two or three separate signals. The diagram below is a typical 2-way crossover that separates the low frequencies from the high ones. A 2-way design is the most commonly used in commercial sound systems to overcome the inherent low frequency signal degradation experienced in 70 volt transformers.



The frequency at which the signal is split is called the *crossover point*. In an active crossover, this frequency can be raised and lowered to accommodate the characteristics of the loudspeakers that are being used. Most commercial sound systems with a subwoofer application will have a crossover point between 80Hz and 150 Hz.

Active crossovers, like the one shown below often include a level adjustment to balance the low and high frequencies signals.   In a system with proper gain structure, Rane equipment will typically have these levels set around 7.

Normally, commercial systems are small enough that they do not require any delay so this setting should be left on zero.



Active Crossover

Like equalizers, crossovers should have a security cover installed over once they are set. An incorrectly adjusted crossover can quickly turn an expensive sound system into a $3 system.

Other Analog Gear

There are other types of equipment that are occasionally used in sound systems, but are used with such infrequency that we will merely address their existence and application in this document

Delays – Are used to delay audio signals typically between 1/2 millisecond up to 5 seconds. They are normally used in large venues such as stadiums or theatres with multiple loudspeaker clusters. The delay helps to time align the clusters so that the acoustic waves arrive at the listener simultaneously. Venues with high ceilings and loudspeakers at different elevations can also benefit from delays.

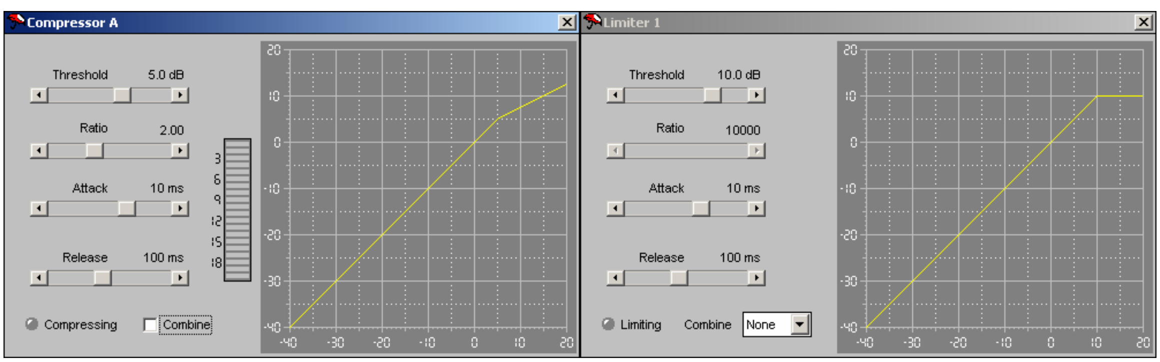
Feedback Eliminators – Sound systems that use microphones risk experiencing feedback, especially if the microphone is located near a loudspeaker. Feedback occurs when a frequency is picked up by the microphone, amplified through the loudspeaker and again picked up by the microphone. This loop repetitiously gains the signal to a point where damage can occur in a sound system (not to mention ones ear drums). Feedback eliminators are very much like a parametric equalizer in that they can lower the amplitude of very narrow bandwidths, eliminating problem frequencies. Older models must be set to locate these nodes in a room, but newer models are designed to automatically sense these frequency gains and make the necessary adjustments.

Ambient Noise Levelers – In theory, these units are designed to act as an automatic volume control based upon the ambient noise level in the room. For example, a restaurant or bar is typically much noisier during their busy times of lunch and dinner. However around 3:00 in the afternoon, it may very quiet. The lever uses either a sensing microphone or the loudspeakers themselves to ‘listen’ to the noise level of the space. Based upon this reading, it will gain or attenuate the program source. These devices have been used with mixed results. Manufacturers suggest that they are simply not set correctly. If this is indeed the case, it appears that most installers do not possess the ability to correctly set them up as they are often in the bypass mode. If you encounter a system that is experiencing random volume fluctuations, you may be able to isolate the problem by putting the auto leveler into the bypass mode. If this corrects the problem, you should probably contact someone who is familiar with the product and its proper setup.

Compressor/Limiters – These are two separate functions that can be in separate devices, combined into one, or even incorporated into other devices. The image below shows a visual representation of their respective impact on an audio signal.

A *compressor* uses a specified ratio to reduce the level of a signal. The user must specify the level or *threshold* of when this compression begins. Compressors are used to reduce or squash the dynamic range of a signal into an acceptable range. Classical music uses a very large dynamic range that my not be acceptable to a specific environment. The compressor will reduce the very loud passages to a point where the listener can turn the volume up high enough to hear the quiet passages without getting 'blown away' during the loud ones. A compressor is also very useful when mixing signals that were recorded at different levels. During playback, the listener would like all songs at roughly the same level so they are not constantly making volume adjustments on their sound system. Audiophiles typically dislike compressors because they change the artist's intent of the program. However, they do have an application particularly in a commercial environment, where there is limited dynamic range to begin with.

A *limiter* is much like a compressor with a hard ceiling. The user must set the level at which the limiter activates. The limiter then does not allow the signal to go above that point, effectively putting a limit on how loud the system can get. A limiter can be a very useful protection tool to keep the user from damaging their system. Limiters are incorporated as a feature in many different processors and amplifiers. Note that a compressor can be turned into a limiter with a very high or infinite compression ratio

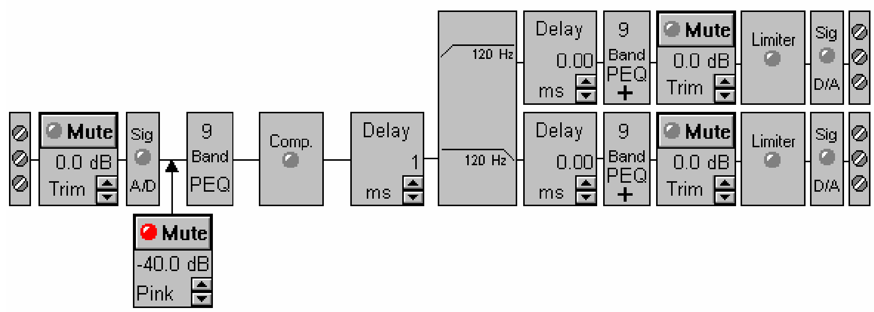


Compressor/limiters can also create havoc on a sound system when set incorrectly. Until you are familiar with setting them up, be modest with settings. Use a higher threshold and lower ratio to minimize their impact.

Digital Signal Processing (DSP)

It is the wave of the future in audio systems. In fact, it is already here and is being used with greater frequency.

DSP’s can perform all functions of the devices that were described above… in a single box. They are able to do this by digitizing the analog signal, manipulating it, and then converting it back to an analog signal that can the be amplified out to the loudspeakers. Once the signal has been digitized, all processing is completed using the manufacturers software on a computer. A sample DSP program is shown below.



Some of the benefits to using DSP are obvious: fewer interconnections between equipment, less rack space needed, fewer power outlets and less power required. DSP systems also offer more security from unauthorized fingers, can be adjusted remotely, and allow a customers design and setup to be easily duplicated at several locations.

The cost of DSP units currently range between $250 and several thousand dollars depending upon the configuration needed. However, with the competition and technology advancing, these costs are dropping rapidly.

Current configurations of the units range from a simple 2 X 2 up to an 8 X 8. However, several of these units are expandable well beyond that. Like any new technology some units are better than other, but the main difference seems to lie in the convenience and flexibility of the software. Most manufacturers allow you to download their software from a web site and experiment with it off line.

There a couple downsides to going digital. Any changes require the use of a computer, so service techs and installers must have a laptop and be relatively computer literate to make changes or adjustments. If the DSP goes down it typically takes the rest of the system with it. Very rarely can a quick patch around or bypass be completed.

But, even with these few risks, the cost of DSP units will eventually make it price prohibitive to use analog equipment, particularly when one considers the labor savings experienced with large systems.

The following block diagram is representative of a typical sound system with DSP.