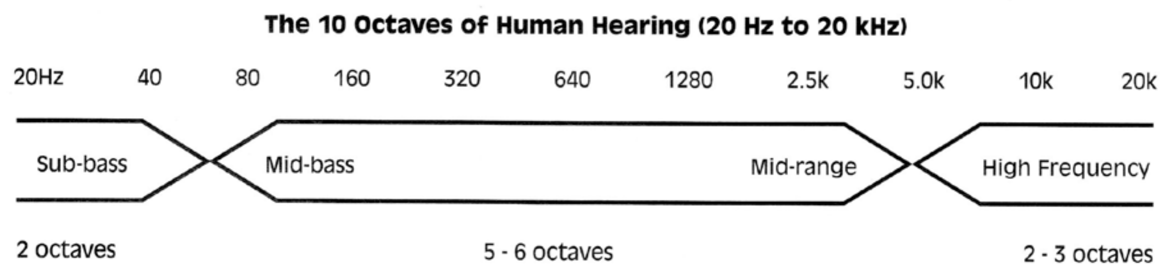
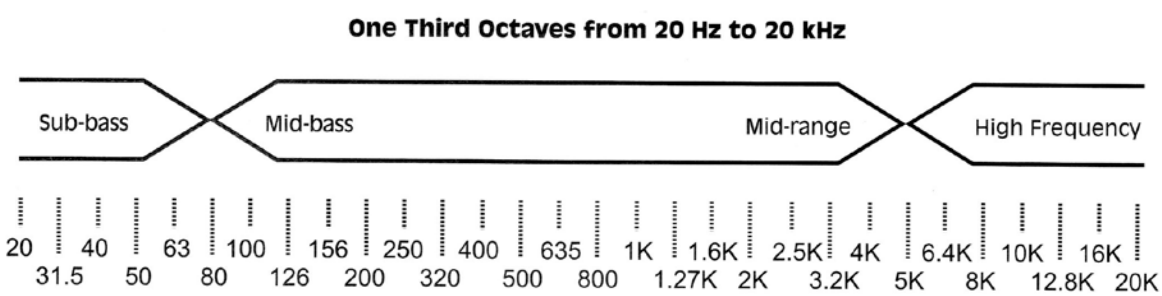
Introduction to Equalization

Tools Needed: Real Time Analyzer, Pink noise audio source

The first thing we need to understand is that everything we hear whether it is musical instruments, a person's voice or the constant howling of the cat next door is comprised of sounds that fall within the 10 octaves of frequencies that we as human beings can hear. An octave is simply a halving or doubling of a frequency from a given starting point, and for most of us that starting point is 20Hz (the lowest frequency we can hear). An octave up from 20Hz would be 40Hz. The next octave up is 80Hz, then 160Hz and so on until we reach the limit of our hearing which tops out at 20kHz (20 thousand Hz, or the highest frequency we can hear). Everything we can hear falls into this 20Hz to 20kHz range.



OK, so now we know what an octave is and what the 10 octaves of human hearing range are. What is 1/3 octave? We… it is simply the same 20Hz – 20kHz range broken into 1/3 octave slices. Calculating 1/3 octave intervals is pretty easy too. You simply start at 20Hz and multiply it by 1.26 to get 25.2Hz. We round this to 25Hz. So 1/3 octave up from 20Hz is 25Hz. The next step up from 25Hz would be 25 x 1.26 or 31.5Hz. The next 1/3 octave step is 31.5 x 1.26 or 39.69Hz. Like above, that is rounded to 40Hz. This goes on and on all the way up to 20kHz.



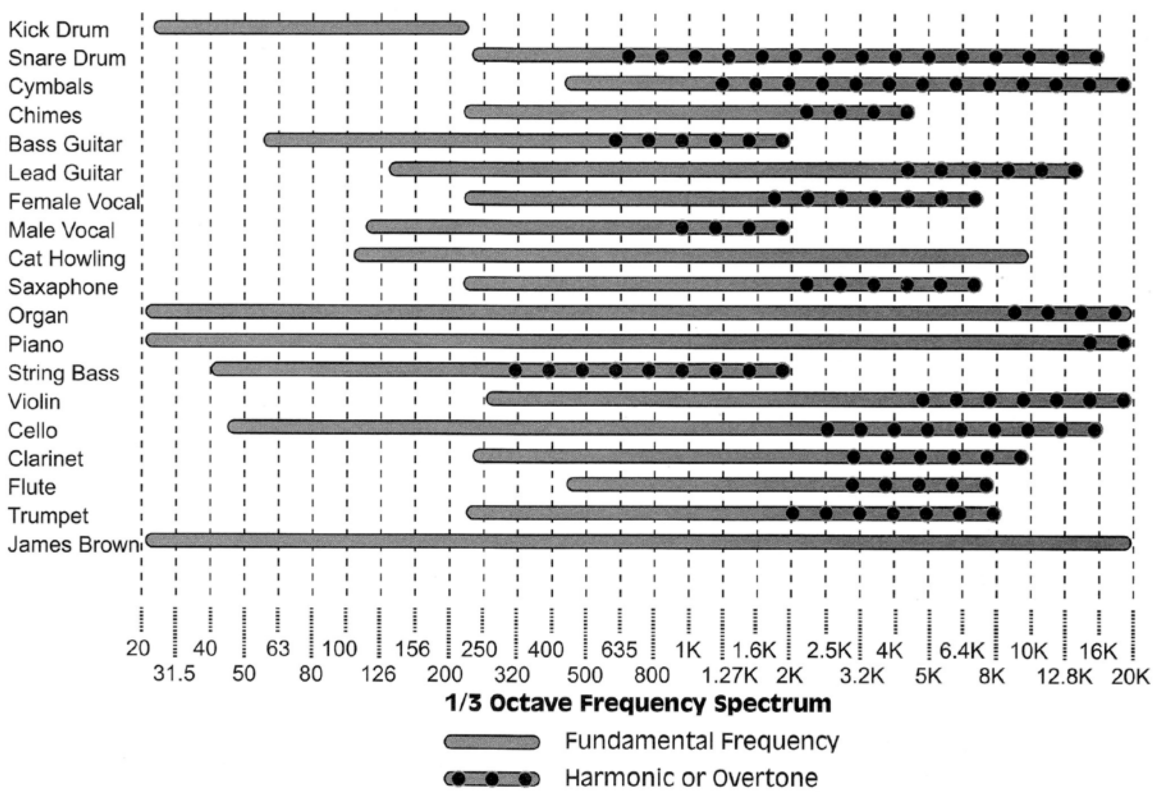
Not fun math notice:

Right now you must be saying to yourself just where do we get 1.26? Isn't .33 equal to 1/3? Just trust us when we say to use 1.26… we are skipping a bunch of difficult math. We hear in a logarithmic scale, not linear, and 1.26 breaks the logarithmic scale up into 1/3 slices. OK? Good. Let's move on…

So if our hearing works within these 10 octaves, where does 1/3 octave come into play? Why 30 bands? It has been determined that 3dB changes between 1/3 octaves are the smallest changes the human ear can easily detect.

A 1/3 octave is designed for tailoring your sound system to better fit your listening environment, or to just make it sound like you want it to sound. Experimentation has shown that the way the human ear sums energy in the Critical Bands to determine the loudness of a sound are about 1/3 octave apart. Because of this, one third octave spacing is all that is required to tune a system within the capabilities of the average human ear. You could use an equalizer with tighter spacing (like a 1/6th octave), but this would only present you with twice as many controls and no audibly better result.

The chart below gives you a good idea where some of the musical instruments and voices operate within the 1/3 octave scale. The fundamental frequencies are the primary sounds the instrument makes when played. The harmonics or overtones are multiples of the fundamental sound. Think of it as striking a hollow drum. When you hit the drum it makes a distinct sound, but there are extra sounds that reverberate from the original strike. These extra sounds are harmonics. These harmonics plus the fundamental give every instrument, voice or sound its own unique sound.



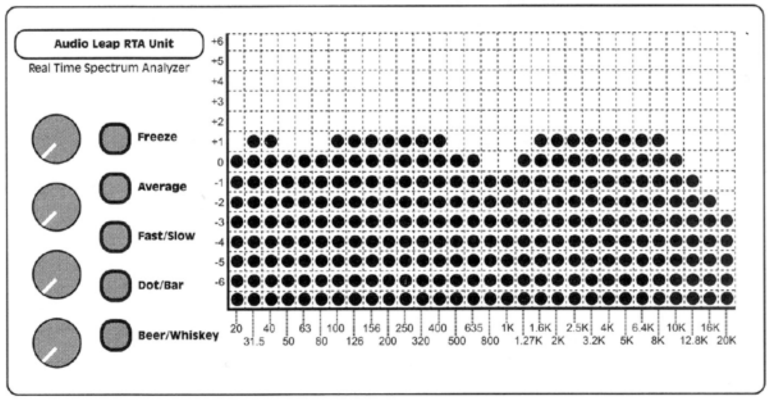
The above chart gives you an idea where popular instruments and voices are located in the audible spectrum. Knowing this can help give you some direction when tuning your vehicle to that certain sound you're after or adding a little definition and character to a particular instrument.

The chart below breaks the frequency ranges down into smaller groups and gives you a general idea of how each range effects the musical spectrum. The last column on the right gives you an idea of what can happen if you apply too much equalization to that area.

Equalization is like adding spice to your food. Not enough and the food tastes bland… too much and you're reaching for the water. The right amount of equalization will really make your system shine.

Frequency Range	Affected Area	Results Of Excessive Boost
16 Hz - 60 Hz	Sense of power, music is felt more than heard.	Makes music sound muddy.
60 Hz - 250 Hz	Fundamentals of the rhythmic section. Equalizing here can change the musical balance making it fat or thin.	Makes music sound boomy.
250 Hz - 2000 Hz	Low order harmonics of most musical instruments that are horn-like. Listening fatigue may result if improperly equalized.	Gives telephone like quality to the music. Can make the music sound tinny.
2 kHz - 4 kHz	Speech recognition.	Listening Fatigue. Will add a lisping quality to voices. "M", "V" & "B" will become vague.
4 kHz - 6 kHz	Affects clarity and definition of voices and instruments. The music will seem closer to the listener with proper setting.	Sibilance on vocals (harshness). Adding boost at 5 kHz will make the music seem louder.
6 kHz - 20 kHz	Brilliance and clarity of sounds. Gives air and pressence to the music.	Sibilance and/or harshness on voices

To properly adjust your 1/3 octave equalizer, it is highly recommended that you, someone you know, or your local dealer have and be familiar with the use of a 1/3 octave RTA (Real Time Analyzer). The RTA shows you where the peaks and dips in your systems response curve are located and dial in the proper amount of boost or cut at the correct frequencies to get a smooth overall system response. The RTA should not be the only source of input to tell you what needs to be tweaked… you must also listen.



There can be times where no matter how good it looks on the RTA display, it simply sounds bad to the ear. By the same token, a curve that may show some uneven response may actually sound pleasing. You always want to start with the RTA and look for major peaks and valleys. Fix these first and then listen. Do this back and forth until you get a response that is smooth as possible and sounds realistic Here are 5 basic pointers to keep in mind as you start your quest to EQ perfection:

1. Your goal is a smooth curve with no more than 3dB difference between bands. Flat lines, happy faces or ski slopes are not necessarily the goal, unless you are into that sort of thing.
2. Always cut first, boost last and keep your boosting to as little as needed to get the desired results. For every 3dB of boost you are asking your amp to double its power output at that frequency. If you notice that most of your frequencies are boosted, simply lower all frequencies equally to reduce the overall level. Remember… cut before boost…
3. Equalization does not change the basics. A properly adjusted equalizer is not a band aid for poor components or installations. Address these issues first… EQ last.
4. If you find certain frequency valleys that do not respond from boosting, you may have some cancellation at that frequency due to speaker placement, speaker phasing, crossover phasing or simple vehicle acoustics. If this happens, address these issues separately.
5. Your ears are always right. No matter how good or bad it looks on the RTA, if it sounds bad… it sounds bad. Period. Let the RTA guide you and your ears tell you if it's right.

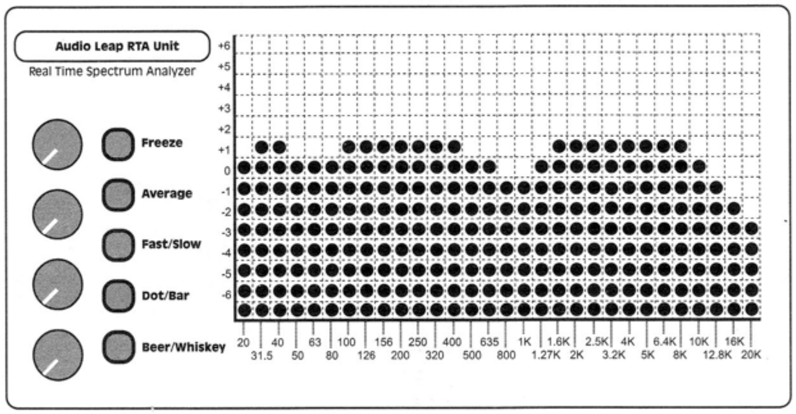
Let's get started…

You'll need a source of pink noise. That could be a CD, MP3, etc. The pink noise should be played through the head unit (usually the first component in the audio system). Turn on the audio system and set the volume of the pink noise to a good working volume. Not too loud and not too soft.

Now for the RTA…

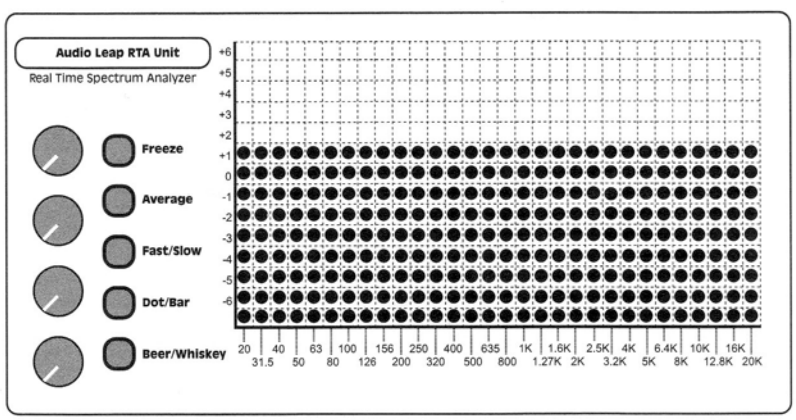
With the SPL set and the pink noise continuously playing through your system, put the RTA into analyzer mode. Set the speed to medium or slow and the resolution to 3dB/step. You will now see a graphical representation of how your system is working at playing back the pink noise test signal. Figure 1 gives you an idea of what you should be seeing, except yours should be moving up and down at the various frequencies.

Fig. 1



The pink noise that you are playing through your system is a recording of equal energy at all octaves being played at the same time. If you were to look at it as a pure signal; before it went through all your amplifiers, crossovers and speakers, it would look like the display in figure 2.

Fig. 2



How is this useful? Knowing that the signal coming from the audio source before it goes through your system is supposed to look like a flat line gives you a known reference. Looking at it with a microphone after it has gone through your head unit, equalizers, crossovers and finally your speakers gives you an idea of how the signal is being affected by these components.

You also see the effect that speaker placement and phasing has on what you hear as well as how the car itself is affecting the signal.

The end result is the ability to look at how everything is affecting the sound you hear in your car. Does this mean we should equalize the car back to a flat line? Definitely not. Car audio competitors will shoot for the flat line so they can get a perfect “score” in the RTA judging section of a contest, but for listening, a flat line just sounds bad.

So if I am playing a source wich is essentially a flat line and you don’t want me to equalize it back to a flat line, just what am I supposed to do? The key to using the RTA and pink noise is to look for major problems with the response curve and smooth them out. Remember from the introduction 3dB changes between 1/3 octave bands was the smallest change the human ear can detect? Well that is what we are trying to do here, simply keep each 1/3 octave band within 3dB of the other as we go from 20Hz to 20kHz.

The overall shape of the curve is not as important as keeping each band within 3dB of those around it. You may like a real bass heavy sound while your friend does not. So you can both have smooth response curves that look different because they are based on how you like it to sound. (Figure 1 and 2)

Fig. 1 your response curve

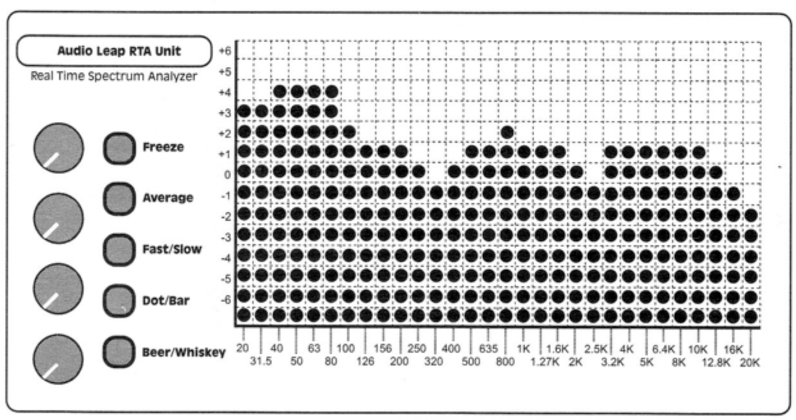
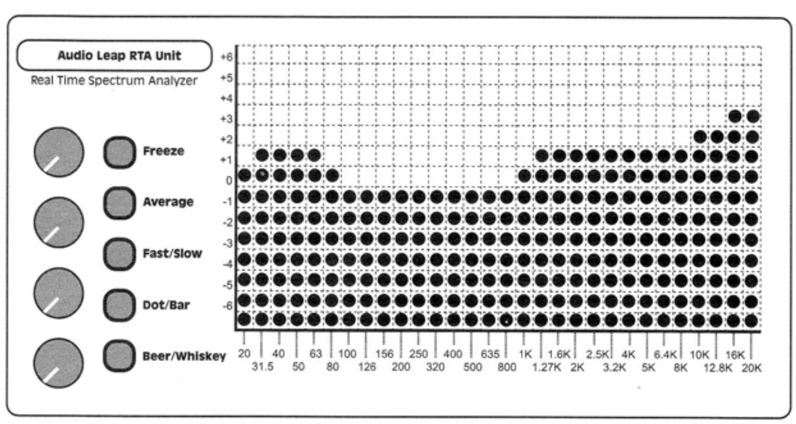
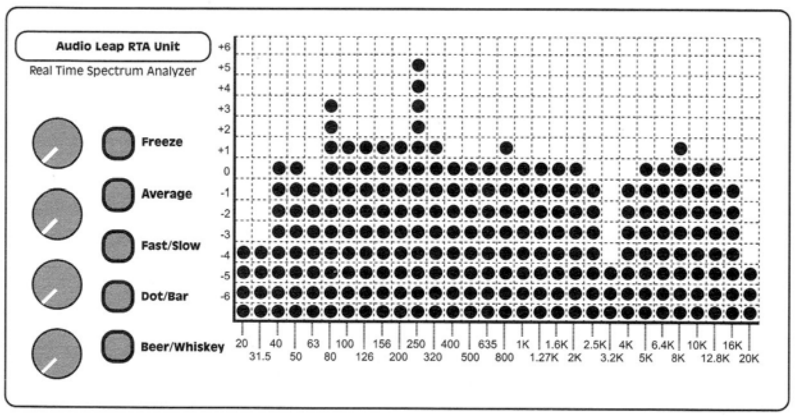


Fig. 2 your friend’s response curve



Neither curve is more correct; it just depends on how you like the sound. The one thing you do notice is that each 1/3 octave band is within 3dB of the next band.

Fig. 3



So what does a bad curve look like? Check out figure 3. Notice how 31.5Hz is more than 4dB down from 40Hz. See where 80Hz is more than 3dB above both 63Hz and 100Hz? How about thet nasty spike at 250Hz? These are problem areas that need to be corrected.

Your first step should always be to equalize out the peaks by cutting (reducing the gain) of the controls at or near those frequencies. Next you would try to bring up the valleys by boosting in those areas (or cutting around them) Your end result is to keep the sound you like but smooth out the overall response so there is no more than a 3dB variation between bands.

So if you start with figure 3 you want to end up more like figure 1 or 2. Not necessarily the same overall curve, but smooth transitions from band to band.

If, after doing all your equalization and level adjustments you feel that your system lacks “bite”, even with your source unit turned up to 90% of its output potential, then the gain settings on your amplifier may be turned up slightly. Keep in mind that the gain settings on any amplifier are for level matching only, they do not increase the power output of your amplifier. The lowest gain setting that will allow your amplifier to make full power is always best for sound quality, lowest system noise and reliability.

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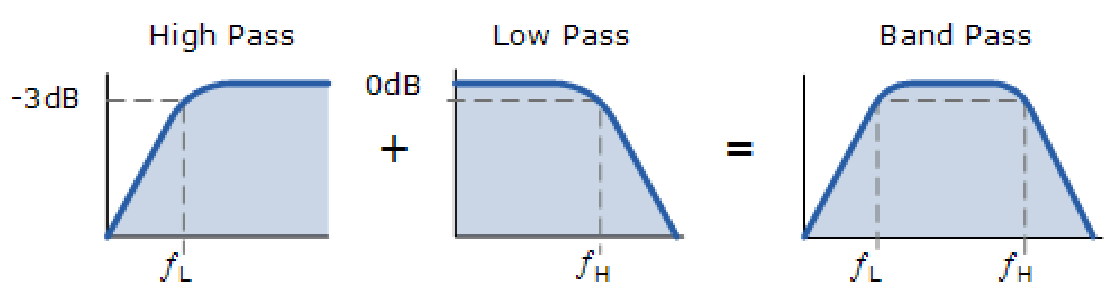
Crossovers… The Basics

There are many misconceptions about what a crossover is, and how they should be set for use in a multi-amplified loudspeaker system. IQ Series amplifiers feature programmable Digital Signal Processing (DSP) allowing the general user the ability to adjust features that at one time was reserved for the professional audio designer. Unfortunately, changes to a loudspeaker manufacturer's recommended settings can have detrimental effects on system performance. This paper attempts to explain some of the details of crossovers and point out some commonly made errors that can affect sonic quality.

What is a Crossover?

A "crossover" is literally a filter that separates an electronic signal into any number of segments each with a bandwidth smaller than that of the original signal. The term "crossover" also refers to the actual electronic device that separates the signal. Crossovers are also called "frequency-dividing networks". The filter pair that makes up a crossover consists of a high-pass (or low-cut) filter and a low-pass (or high-cut) filter. These are sometimes abbreviated HPF and LPF. Filters are frequency selective devices that pass certain frequencies while rejecting others. They are generally defined by three parameters; a cutoff frequency, an implementation, and a slope. The cutoff frequency identifies the frequency at which the response of the filter falls to some point below its maximum level. This is generally "-3dB" and "-6dB" respectively. This defines the shape of the filter around the cutoff frequency. The most common filter implementations used today are Butterworth and Linkwitz-Riley. The slope of a filter defines the rate at which the filter response falls beyond the cutoff frequency. This is defined in dB/octave and common slopes are 6, 12, 18, and 24 dB/octave. It is common for the terms "filter slope" and "filter order" to be interchanged with one another. An increase in filter order refers to a 6dB/octave increase in slope. Simply, a 1st order filter refers to a 6dB/octave slope, 2nd order to a 12dB/octave slope and so on. For example, a 24dB/octave Butterworth filter is a 4th order Butterworth filter.



Crossovers are necessary in a multi-speaker system since acoustic transducers (speakers) are not usually capable of equal level, full bandwidth (20Hz-20kHz) output. Woofers are generally used to reproduce low frequency signals, whereas tweeters are used to reproduce high frequency signals. Crossovers allow the proper frequencies to be delivered to the proper speakers. Ordinarily, crossovers are classified as being passive or active. Active crossovers are inserted into the signal path between the music source and the amplifier. Passive crossovers are wired between the amp and speakers. All Kicker amplifiers feature built-in active crossovers. The crossovers in IQ Series amplifiers are DSP controlled. The signals from the crossover feed the appropriate speakers, which reproduce the appropriate portion of the audio spectrum. When a crossover is properly designed, the signals from each transducer are able to "sum" and accurately reproduce the original signal in its entirety.

The Crossover Point

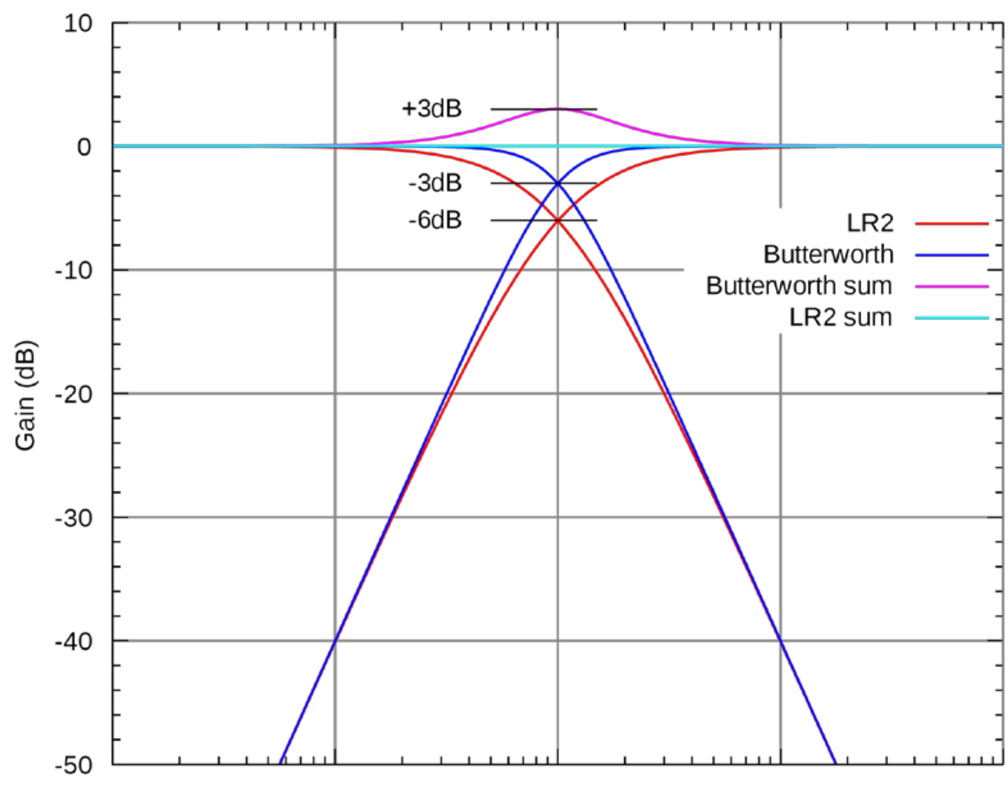
The crossover point is defined as the frequency at which the responses of two filters, usually a HPF and a LPF, cross one another. This can be the crossover point of two filters in an electronic crossover, or the crossover point of two passive filters. Any speaker is, in fact, also a filter. Every speaker has its own cutoff frequency, slope, and implementation.

Since Kicker IQ Series amplifiers utilize full digital control via Digital Signal Processing, the music is manipulated in the digital domain without the signal being degraded in any way. Analog to Digital and Digital to Analog conversions are done at the same resolution, therefore, there is no data loss.

The crossover sections of all IQ Series amplifiers allow the user to select any frequency from 20 – 20kHz. The slope (roll-off) choices are 6, 12, 18 or 24dB/octave or OFF. Selecting OFF will disable that crossover.

Linkwitz-Riley or Butterworth?

We offer two different crossover options within the Tweeq software. The diagram below shows the difference between the two implementations. It basically comes down to how the high-pass and low-pass filters



sum together at the crossover point. Electronically speaking, a Linkwitz-Riley implementation sums together to a flat response at the crossover point and a Butterworth implementation sums with a +3dB rise at the crossover point. However, in a vehicle, drivers could be located almost anywhere, so perfect driver physical alignment and control of acoustic interaction may not be possible, so all bets are off. The best course of action would be to experiment in the vehicle using your favorite music. Select Linkwitz-Riley and try different crossover points, then select Butterworth and try different crossover points. Pick the setting that sounds best.

Here is a little information that may be helpful when choosing your crossover slope:

6dB/octave

First-order crossovers have a 6 dB/octave slope and are considered by many audiophiles to be ideal. This is because this filter type is 'transient perfect', It has minimum phase change, but allows more unwanted signals to pass than do higher order configurations. While woofers can easily take this, smaller high frequency drivers (especially tweeters) are more likely to be damaged.

In practice, speaker systems with true 6dB/octave slopes are difficult to design because they require large overlapping driver bandwidth, and the shallow slopes mean that non-coincident drivers interfere over a wide frequency range and cause large response shifts off-axis.

12dB/octave

These have a 12 dB/octave slope and can have Linkwitz-Riley or Butterworth implementation. This order is commonly used in component set passive crossovers as it offers a reasonable balance between complexity, response, and high frequency driver protection. When designed with time aligned physical placement, (drivers mounted on the same baffle) these crossovers have a symmetrical polar response, as do all even order crossovers. Refer to the section below regarding time alignment.

18dB/octave

Third-order filters have an 18 dB/octave slope. These crossovers usually have Butterworth filter implementation and have a +3db bump at the crossover point.

24dB/octave

Fourth-order filters have a 24 dB/octave slope. A 24dB/octave crossover with −6 dB crossover point and flat summing is also known as a Linkwitz-Riley crossover (named after its inventors). Butterworth is another choice in this crossover implementation.

A 24dB/octave slope has the potential for a lower crossover point and increased power handling for tweeters, together with less overlap between drivers, reducing unwelcome off-axis effects. With less overlap between adjacent drivers, their location relative to each other becomes less critical and allows more practical installation possibilities.

A few additional thoughts…

Ordinarily, speaker placement, orientation, distance from each other, etc. are the first hurdle to overcome when building and tuning a mobile audio system. Unfortunately, most vehicles make this job extremely difficult.

Kicker IQ Series amplifiers are audiophile quality, fully functional amplifiers right out of the box. All controls necessary for setting up a great audio system are on the end panel.

But…If you want to really take your system to the next level, IQ Series amplifiers provide far more control over typical amp parameters like crossovers and basic equalization. One very powerful feature is time alignment. All you will need to access these additional features is our free TWEEQ software, a mini-USB cable and a computer. Simply measure the distance from each speaker to your ear, enter these measurements into the TWEEQ program, and basic time alignment settings are made for you. You can always “TWEEQ” the computers results to further fine tune your system.

Speaker protection

All mobile audio packaged component sets provide passive crossover networks optimized for the speakers in the kit. The provided crossover slops are generally 12dB/octave. These crossovers usually provide some type of tweeter protection. If the decision is made to “go fully active” using the amplifiers internal crossovers instead of the passive crossovers supplied with the component set, the protection provided with the components will not be able to protect your tweeters. This is where crossover point and slope selection in the IQ amp can help.

Generally speaking, lowering the crossover point for a tweeter will allow it to blend better with a midrange. Unfortunately, lowering the tweeter crossover point will reduce its power handling. This can be compensated for by increasing the slope of the tweeter crossover. If you are happy with the manufacturers recommended crossover point, you can increase the tweeters power handling by increasing the tweeter crossovers slope and/or raising the crossover point. This also holds true for a midrange or mid bass speaker.

Overlapping and underlapping

The crossover section of the IQ Series amplifier allows you to overlap or underlap crossover points. What the heck is this? Say you have a four channel IQ amplifier and are running a pair of components fully active. Amp 1 is running a pair of front tweeters and Amp 2 is running a pair of midrange drivers. The left front high-pass crossover is set to 3500Hz. The right front high-pass is also set to 3500Hz. The left and right rear mid-range crossovers of Amp 2 are set for a band-pass of 100Hz to 3500Hz. If the low-pass (upper limit) portion of the mid-range driver’s band-pass is raised to 4000Hz, it will “overlap” the high-pass frequency setting of the tweeter high-pass from 3500 to 4000Hz. This will result in increased output between 3500 and 4000Hz. If the Low-pass portion of the midrange band-pass is lowered to 2500Hz, there will be a gap in the frequency response between 2500 and 3500Hz resulting in a drop in output. This is referred to as “underlap”. These are just a few more tips that can be helpful when fine tuning your system. IQ’s powerful internal DSP allows great flexibility.

In conclusion…

Crossover setting can be one of the most frustrating aspects of audio system tuning. Many things need to be considered when dialing them in. Kicker has made this task much simpler with our IQ Series amplifiers. Our TWEEQ software quick start menu will allow you to answer a few questions and enter a few measurements resulting in the starting point for an unbelievably great sounding audio system.

IQ Series amplifiers allow you to “open up” your tuning choices. DSP provides flexibility. Practicality, ease of use and a very high cost/performance ratio gets you the most out of your audio components.

David E. Gumienny

Kicker

Credits:

I would like to thank Nathan Butler, Design Engineer for EAW (Eastern Acoustic Works) for allowing me to use portions of his “Processor Setting Fundamentals” paper.

Dynamic Range Compression

The word "compression," as it relates to audio, is often misused, overused and even misunderstood. Likewise, compression itself is often misused, overused and misunderstood. The process of compressing an audio signal is used in the recording studio, during live performances and in the playback of recorded music. So then, what is compression? Simply stated, compression, or Dynamic Range Compression, is the process of reducing loud sounds and amplifying quiet sounds to bring them closer together. A simple analogy would be turning down the volume on your TV because the commercials are just too darn loud. The effect can be used to make recordings sound better at low and high listening levels, to make the quiet passages of music audible in a noisy concert venue or to protect the components in an audio system. For our purposes, we will focus on using Dynamic Range Compression as a way to protect the components in an audio playback system.

Let's begin by breaking compression down into five parts: threshold, attack, release, ratio and output level.

- Threshold: The level at which the incoming signal triggers the compressor.
    - The threshold is generally set close to the upper limit of the system's capabilities. This is the point at which we want to reduce the signal to protect speakers from being over driven. If the threshold is set too low, the compressor will kick in before the system reaches full potential.
- Attack Time: Once the threshold has been reached, attack time determines how quickly the compressor begins reducing the signal.
    - If the attack time is too fast, the transients are eliminated and the music sounds lifeless and dull. If the attack time is too slow, a crashing cymbal can toast a tweeter or a kick drum can launch a woofer's cone.
- Release Time: Once the signal has been compressed or reduced in level, release time is how long the compressor maintains this reduction before releasing the signal level to normal or uncompressed.
    - A release time that is too fast may allow the compressed signal to return to its speaker damaging level before the music naturally reduces in level. On the other had, a release time that is too slow can make a fast paced, articulate bass line sound like one, long, muddy note.
- Ratio: The amount the compressor reduces the original signal, like turning down the TV's volume when a commercial comes on.
    - A compression ratio of five to one would be written as 5:1. If an input signal exceeds the compressor's threshold by 10 Volts, the output will be reduced to 2 Volts above the threshold. A ratio of 10:1 or greater is generally referred to as a limiter and prevents the output from ever going above a set level. This can save the life of many a speaker.
- Output Level: A volume control to increase the output of the compressed channel to match the volume of the other channels.
    - A compressed signal may need an increased level to balance its output with the rest of the system.