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Processor Setting Fundamentals

-or- What Is the Crossover Point?

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There are many misconceptions about what a crossover is, and how they should be set for use with a multi-amplified loudspeaker system. Today's user modifiable DSP processors have allowed the general user the ability to adjust signal processing that at one time was reserved for the professional designer. Unfortunately, subtle changes to a loudspeaker manufacturer's recommended settings can have detrimental effects on the system's performance. This paper attempts to explain some of the details of crossovers and point out some commonly made errors that can seriously inhibit sonic quality.

What is a Crossover?

A "crossover" is literally a single pair of filters that separate an electronic signal into two separate signals; each with a bandwidth smaller than that of the original signal. The term "crossover" also refers to the actual electronic device that separates the electronic signal and may be comprised of one or more filter pairs. 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, a topology, 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 0.707 or 0.5 times the maximum or "-3dB" and "-6dB" respectively as shown below. The topology defines the shape of the filter around the cutoff frequency. Over the years, several filter topologies have been developed. The most common filter topologies used today are Butterworth, Linkwitz-Riley, and Bessel. Examples of these are illustrated in **Figure 1**. The slope of a filter defines the rate at which the filter response falls beyond the cutoff frequency. This is usually 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. Very 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 the same as a 4th order Butterworth filter.

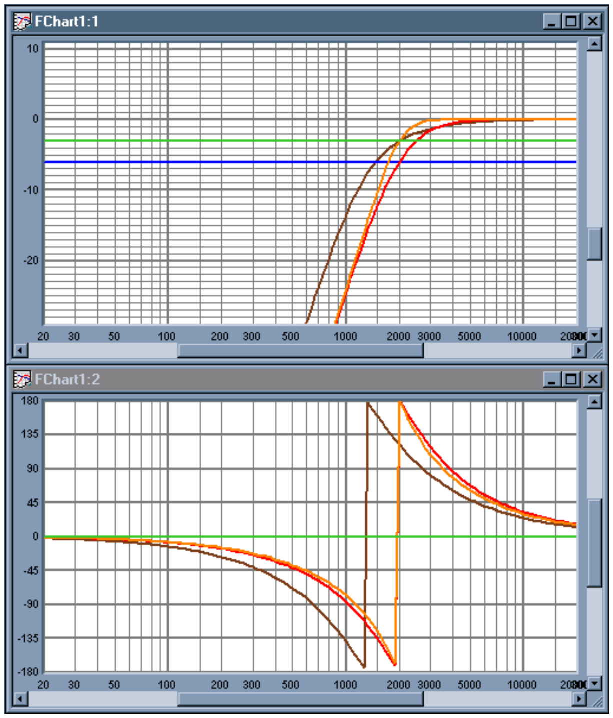

Figure 1: Red- 2kHz 24dB Linkwitz-Riley high-pass filter, Orange – 2kHz 24dB Butterworth high-pass filter, Brown - 2kHz 24dB Bessel high-pass filter, Green – "-3dB", Blue – "-6dB"

Crossovers are necessary in a full range loudspeaker system since acoustic transducers are not 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 transducers.

Ordinarily crossovers are classified as being *passive* or *active*. Generally speaking, a passive crossover is used to separate the audio spectrum post-amplifier (or at speaker level), and can usually be found mounted within a loudspeaker cabinet. An active crossover separates the audio spectrum pre-amplifier (or at line level), and is usually a separate electronic device, located between the signal source and the amplifier. The signals from the crossover eventually feed the appropriate transducers, 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. Many other factors such as power handling and beamwidth are also greatly affected by the crossover. These are all taken into consideration during the design process.

Phase – a Brief Explanation

Summation occurs when the phase responses have similar value and slope at a particular frequency. The phase response allows one to interpret the phase difference and/or the time difference in the arrival of two signals. When the phase responses of two filters are similar, they will add coherently, otherwise, they will tend to cancel one another. Each of the filter topologies and slopes discussed above have their own unique phase responses, as illustrated in **Figure 1**.

The following examples illustrate some common phase variances that might be seen in a loudspeaker system. These measurements can be obtained using an acoustical measurement system such as SIA Smaart. Consider the phase response, shown in the lower graph, of the two filters in **Figure 2**.

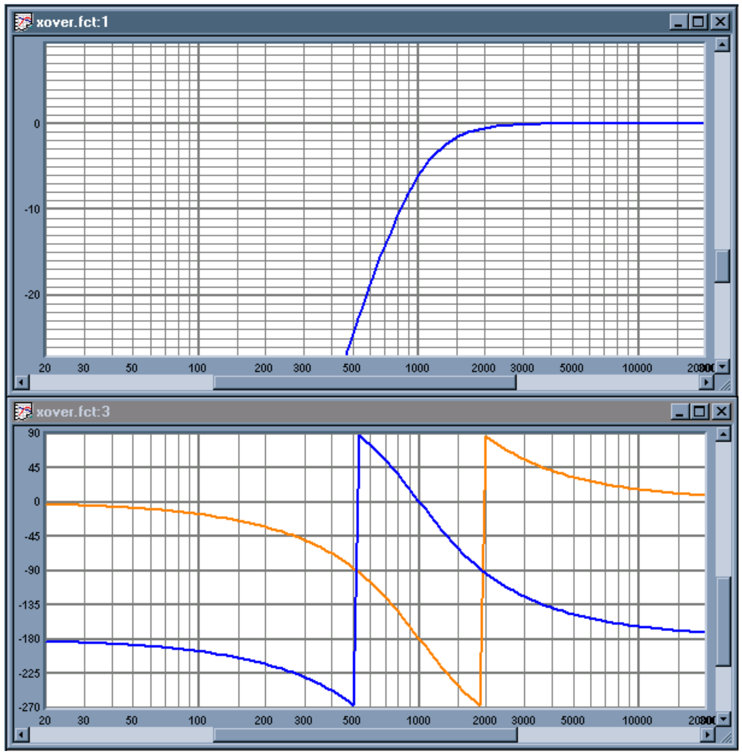

Figure 2: Two identical filters. Orange - normal, Blue - inverted polarity.

Although the magnitude responses of the two filters are identical, there is an obvious difference in their phase responses. Careful inspection of the phase response shows that the slopes are the same, and that there is a consistent 180 degree difference in the two signals. This difference is characteristic of a polarity inversion. This should not be confused with a simple 180 degree phase shift, which can occur at a single frequency as the example in **Figure 3** illustrates.

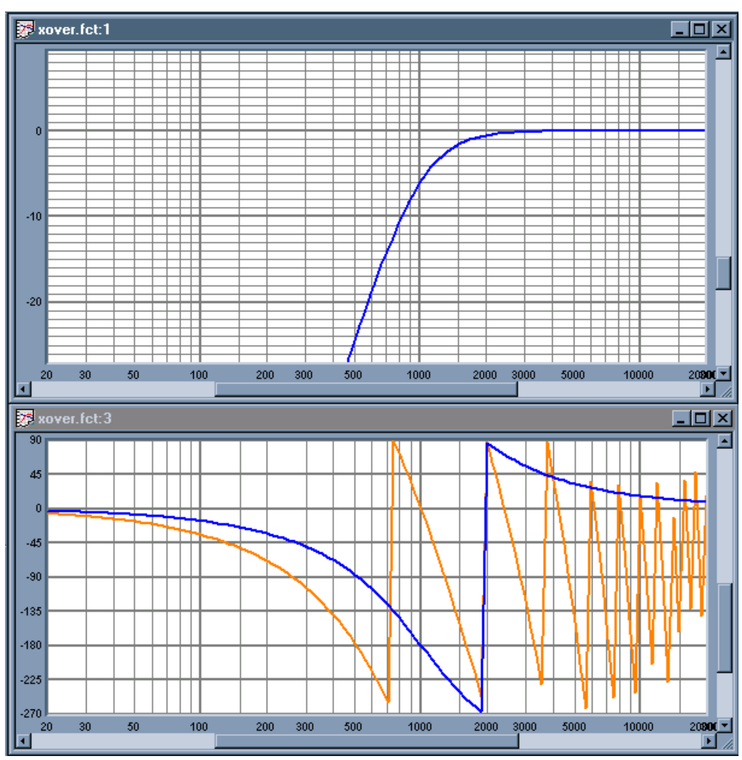

Figure 3: Two identical filters. Orange - delayed, Blue - normal.

Here the difference is not consistent, the slopes and difference in phase change with frequency. This is characteristic of a time offset, or delay, indicative of a difference in spacing of the two devices. This offset can be calculated by the following equation.

$$t = \frac{|\Phi_1(f) - \Phi_2(f)|}{(f \cdot 360)}$$

The equation states that the time offset is equal to the absolute value of the phase difference at a particular frequency, divided by the product of that frequency and 360 degrees. Let's assume that 500Hz is chosen as a frequency. The plot shows that at 500Hz, the phase of the blue curve is –90 degrees whereas the phase of the orange curve is –180 degrees. This is a difference of 90 degrees. Therefore, the time difference of these two signals is 90/(360*500Hz) = 0.5ms. This calculation can be done at any frequency; the result will be the same.

Attention needs to be given to the "wrapping" of the phase plot. The y-axis of this plot ranges from –270 degrees to +90 degrees. Phase plots usually only represent 360 degrees of phase; since 0 degrees and 360 degrees are equal, as are 90 and –270, 180 and –180, etc. It may not be intuitive that the phase responses shown consistently decrease. The phase angle of the orange curve at 2kHz is actually –630 degrees (-270 from the first wrap at 700Hz, plus –360 from 700Hz to the second wrap at 2kHz). These phase values must be used in the above equation to yield proper results.

The Crossover Point

The crossover point can be 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 (passive or active), or the crossover point of two acoustic filters. Any transducer is in fact also a filter. Every transducer has inherent high and low pass filters with their own cutoff frequencies, slopes, and topologies.

Often times the question "What is the crossover point(s) for system X" is asked. The answer to this question is generally not what the user is looking for. Normally, this question is answered by providing the overall acoustic crossover point for system X. The overall acoustic crossover point of a system is interpreted from the mathematical combination of the acoustical responses of the transducers and the responses of the electronic filters. When an electronic filter is applied to an acoustical filter, the responses of the two are combined, resulting in an entirely new response curve as shown in **Figure 4**.

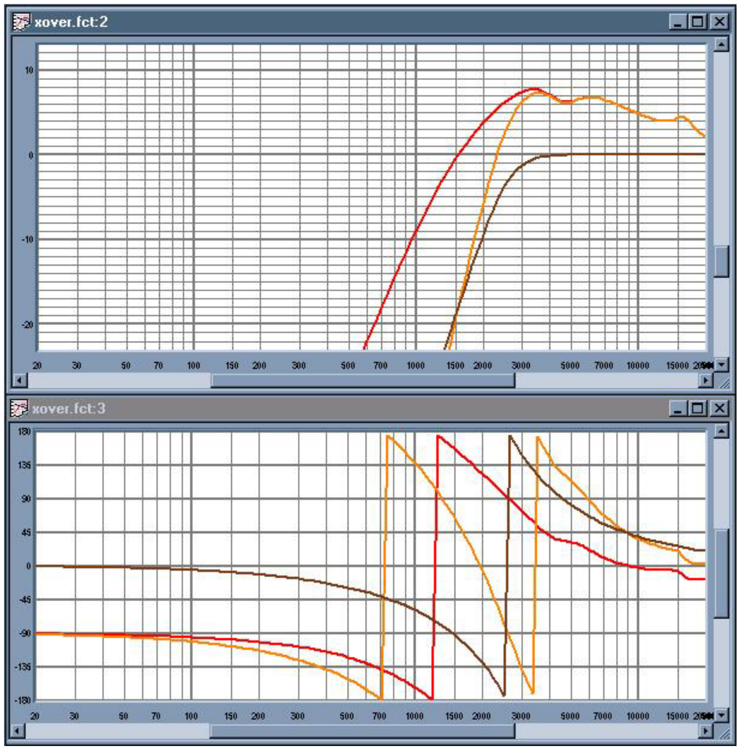

Figure 4: Red – HF driver response, Brown – electronic high-pass filter, Orange - resultant response

The answer the questioner is generally looking for is "what are the crossover settings for system X". System settings cannot be summed up with crossover points. As explained above, a crossover is made up of a high pass filter and a low pass filter; each of which can only completely be described with three parameters.

System Example

Consider the following example. **Figure 5** shows the raw responses of a particular high frequency (HF) device and a particular low frequency (LF) device mounted in an enclosure.

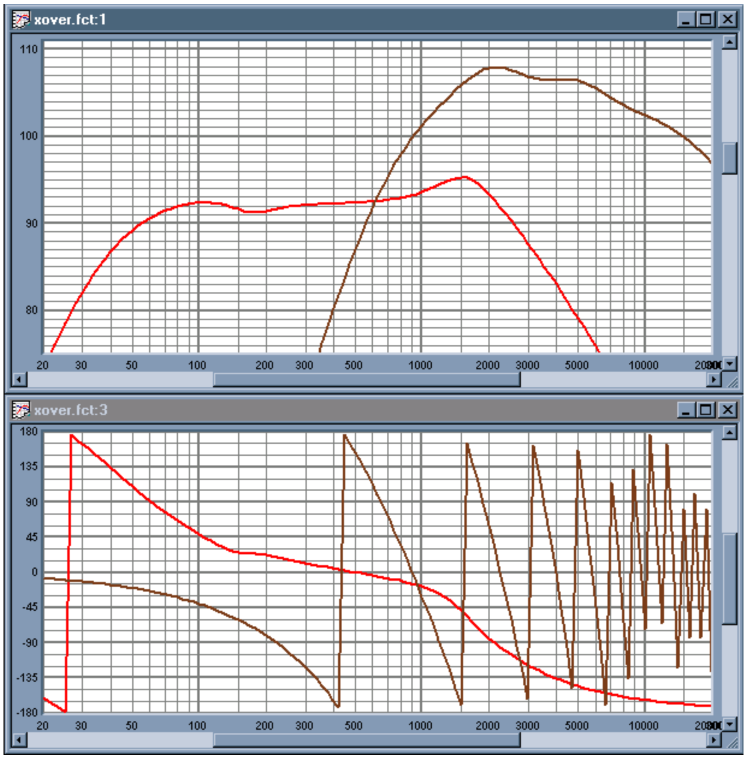

Figure 5: Raw acoustical response of two transducers. Red- LF, Brown - HF. Crossover point at approximately 613Hz.

The level/sensitivity differences between the two transducers as well as the phase lag of the HF device are evident. The HF device is probably mounted on a horn with a long throat and is therefore delayed with respect to the woofer. In order to use this system for reproduction purposes, a crossover needs to be developed to "flatten" the response. Processing shown in **Figure 6** and listed below is used to obtain the results shown in **Figure 7**.

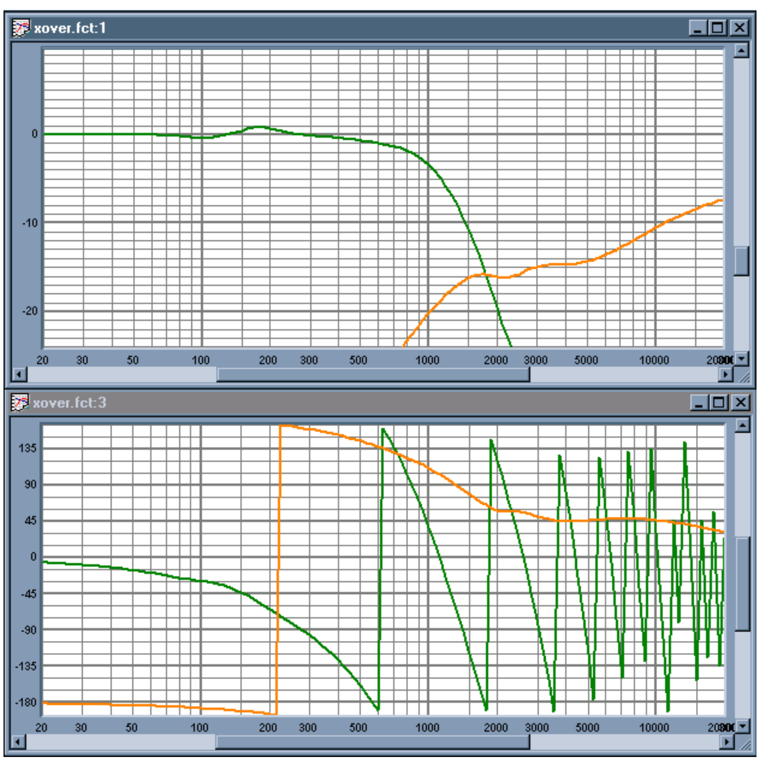

Figure 6: Response of electronic crossover. Green – LF, Orange HF. Crossover point at approximately 1.8kHz

Electronic LPF

Gain	0.0 dB		
Delay	0.5 ms		
Polarity	Positive		
HPF	None		
LPF	944 Hz	24 dB Bessel	
PEQ1	2239 Hz	-5.0 dB	1.19 (Q)
PEQ2	917 Hz	+1.0 dB	1.19 (Q)
PEQ3	103 Hz	-0.5 dB	2.00 (Q)
PEQ4	178 Hz	+1.0 dB	2.00 (Q)

Electronic HPF

Gain	-5.5 dB		
Delay	0.0 ms		
Polarity	Positive		
HPF	2053 Hz	12 dB Butterworth	
LPF	None		
PEQ1	4597 Hz	-8.5 dB	0.67 (Q)
PEQ2	2239 Hz	-3.0 dB	1.68 (Q)

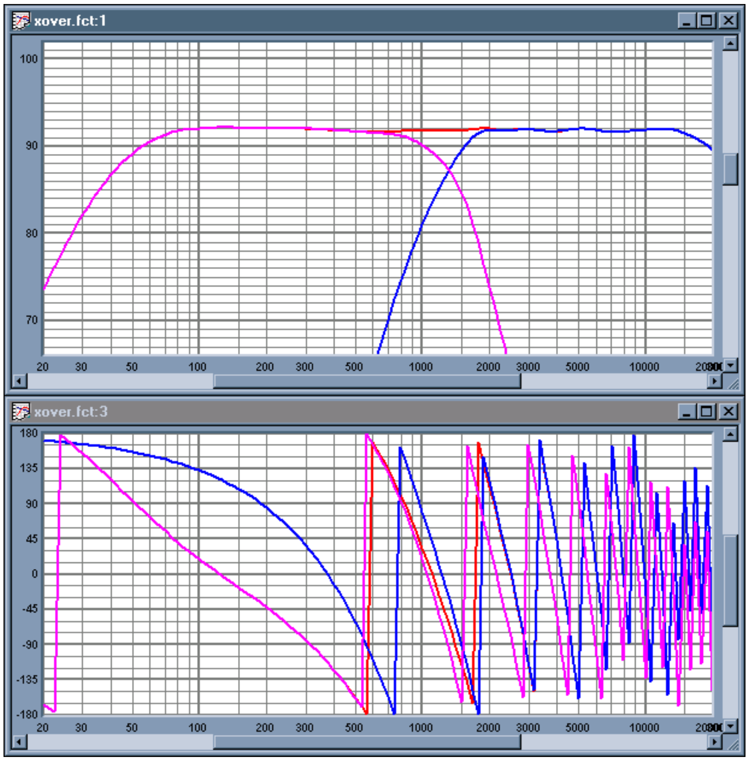

Figure 7: Overall acoustical response of system with crossover. Pink – LF, Blue – HF, Red – Overall. Crossover point at approximately 1.3kHz

One will note that the overall response is optimally flat from 50Hz to 20kHz (-3dB). Note that the phase responses of the HF and LF have similar slopes at crossover and differ by less than 90 degrees. This was accomplished through use of delay on the LF section to align it with the HF. One should realize that this is just one possible crossover solution for this system; many others surely exist. Note that the acoustical crossover of 1.3kHz in **Figure 7** really gives no information relevant to the 944Hz LPF used on the LF, nor the 2053Hz HPF used on the HF. In addition, it does not correspond to the raw driver crossover point (**Figure 5**) nor the electronic filter crossover point (**Figure 6**).

Why Asymmetric Filters?

Note that a 12dB Butterworth filter is used on the HF, and a 24dB Bessel is used on the LF in the above example. This use of asymmetric filter slopes *and* topologies is quite common, since very few transducers share filter slopes and topologies. Referring back to **Figure 5**, it is shown that the inherent slopes and topologies of the HF and LF devices are not equal. It's already been stated that the overall acoustic response of a system depends on the combination of the electronic response of the crossover and the acoustic response of the transducers. In order for an electronic filter set to be symmetric, the transducer set would also have to be symmetric. This is generally not realizable and therefore asymmetric electronic filters need to be utilized to complement the asymmetric characteristics of the transducers.

Unfortunately, less expensive electronic crossovers do not allow for asymmetric slopes and/or topologies. Many inexpensive crossovers simply have a “frequency” knob that allows one to dial in a center frequency. This is likely where the request for “crossover points” comes from, since only one frequency parameter can be set on these units. Usually these devices utilize symmetric 24dB Linkwitz-Riley HPF and LPF because of their high cutoff slope and identical phase response at a given frequency. This is shown in **Figure 8**.

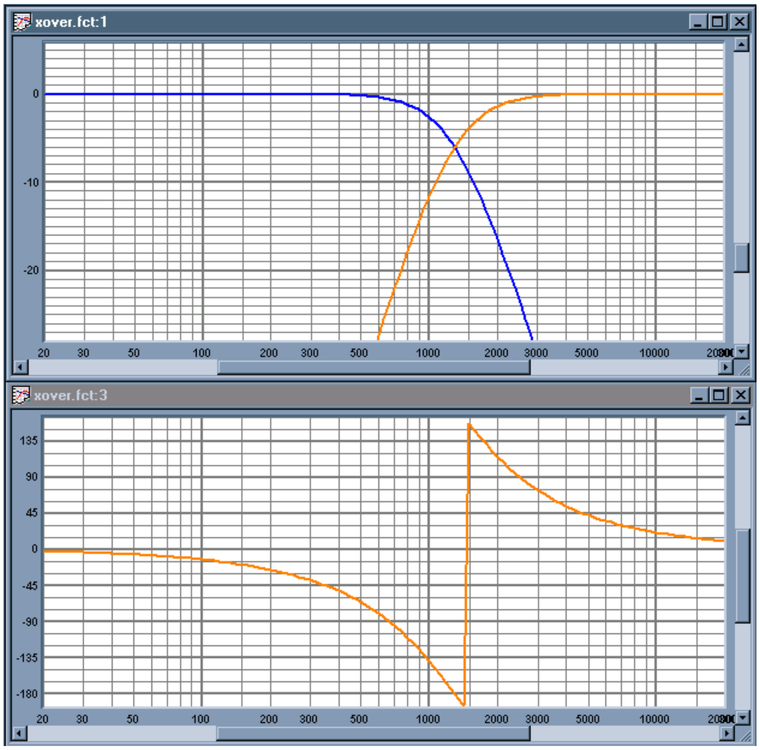

Figure 8: 24dB L.R. HPF and LPF, both at 1.3kHz. Note that the phase responses overlap, so the blue phase plot is not visible.

Let’s consider the effects of using such a crossover device, with a single frequency knob, on the system depicted in **Figure 5**. This is shown in **Figure 9**.

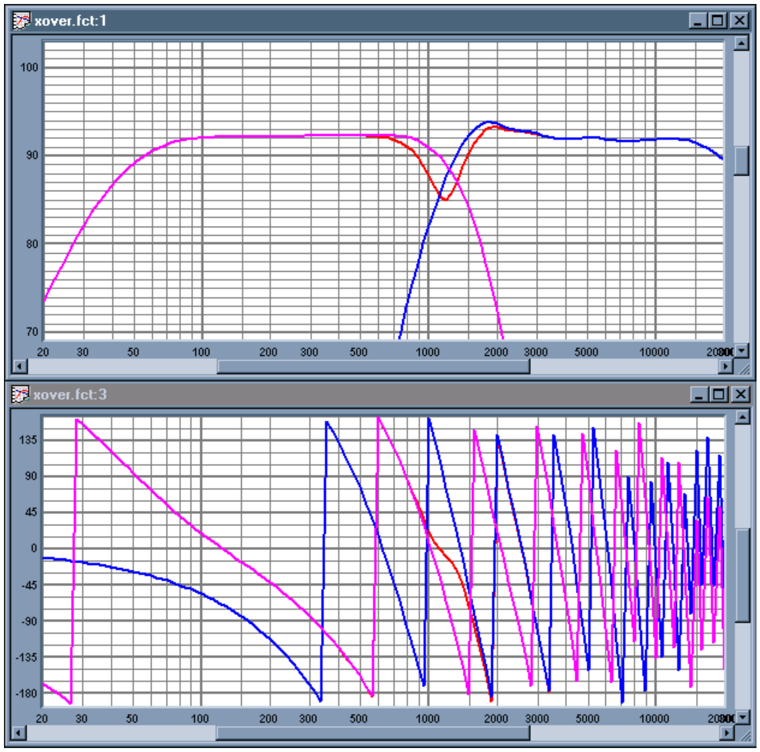

Figure 9: Processing shown in Figure 6 with HPF and LPF replaced with 1.3kHz 24dB L.R. HPF and LPF. Pink – LF, Blue – HF, Red – Overall. Note the difference in phase response.

These curves represent the overall response of the system with identical processing to that shown in **Figure 6**, except the HPF and LPF have been replaced with symmetric 24dB L.R. filters at 1.3kHz. The frequency of 1.3kHz was chosen since it is the overall crossover point of the system depicted in **Figure 7**.

Let's consider two other possibilities. The first is to eliminate the delay in the setting in the previous example. Since an inexpensive crossover would not necessarily have a delay adjustment, or at least not one that would allow a precise setting of 0.5ms, this examines the effect of using no delay at all. This is shown in **Figure 10**.

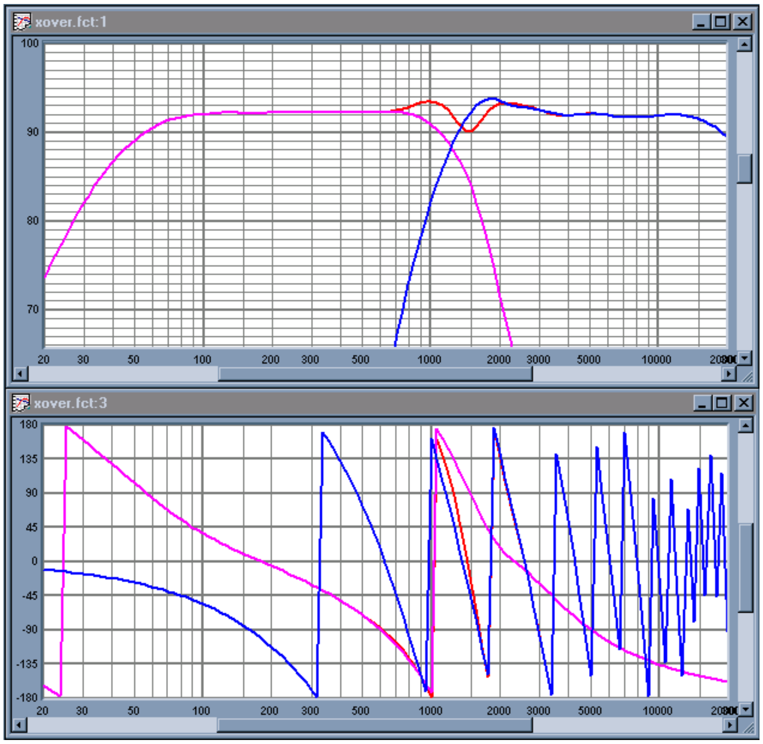

Figure 10: System depicted in Figure 9 without delay.

It may also be worthwhile to consider setting the symmetric 24db L.R. filters at 1.8kHz – the crossover point of the electronic filters shown in **Figure 4**. This is shown in **Figure 11** with and without 0.5ms of delay.

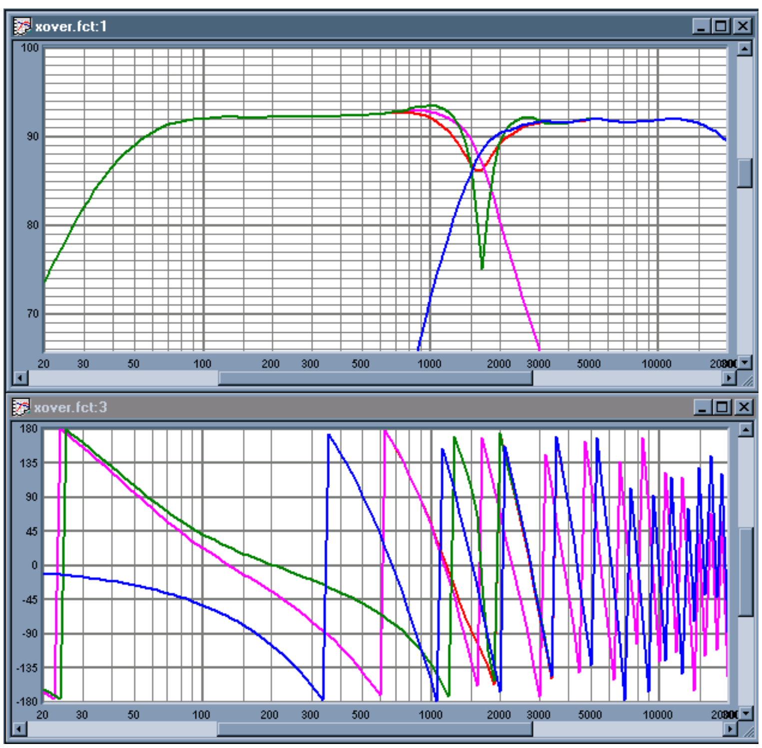

Figure 11: System shown in Figure 10 with crossover point set to 1.8kHz. Red – with 0.5ms delay, Green – without delay.

Finally, let's consider the original example of using the symmetric 24dB L.R. filters depicted in **Figure 9**. Assuming that this system could be "fixed" with an external parametric or graphic EQ, let's consider exactly what would need to be done to make the system flat. The phase response in **Figure 9** shows that the phase difference in the crossover region (800Hz – 2kHz) is a fairly consistent 180 degrees. This, as stated previously, is indicative of a polarity inversion. **Figure 12** illustrates the effect of inverting the polarity of the HF signal, then inserting an additional EQ at 1.49kHz. In addition, the existing EQ was manipulated slightly. The result is a flat response, but these changes are in no way intuitive; a user could not possibly make these changes without the use of quality measurement equipment.

It might also be worth noting that the low Q parametric at 1.49kHz could not easily be implemented on a graphic equalizer utilizing standard ISO frequencies. It is also unfortunate that an EQ needs to be utilized to eliminate excess summation at crossover, when separating the HPF and LPF frequencies would probably suffice (lowering the cutoff frequency of the LPF and increasing the cutoff frequency of the HPF). In addition, one can not know how these alterations will affect other system parameters such as power handling, off axis response, beamwidth, etc.

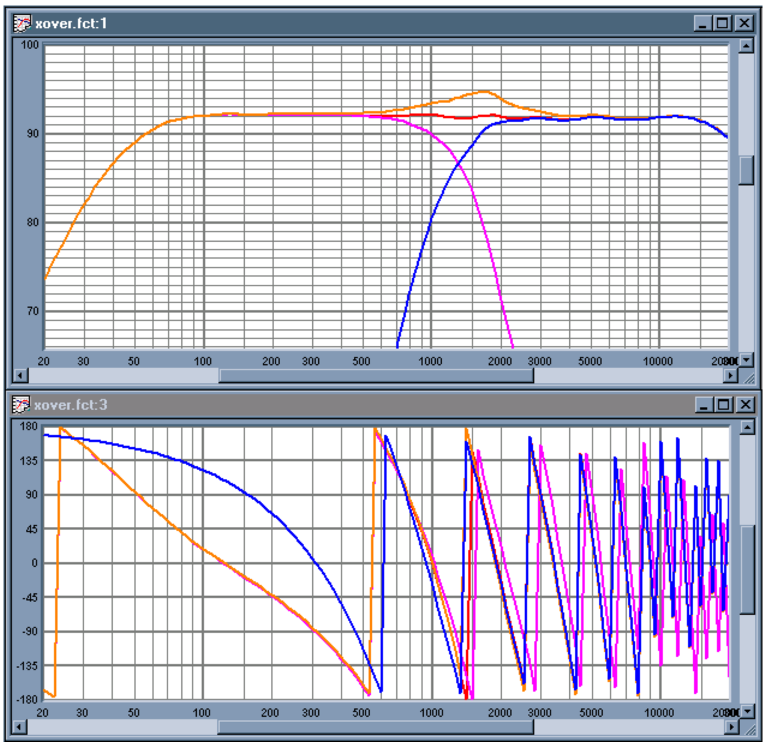

Figure 12: Orange, system using 24dB symmetric L.R. filters with HF inverted. Red, final system with additional parametric EQ.

These settings illustrate the changes made to those shown in **Figure 6**. The parameters in **red** were changed. The **blue** parameters were added.

Electronic LPF

Gain	0.0 dB		
Delay	0.5 ms		
Polarity	Positive		
HPF	None		
LPF	1296 Hz	24 dB Linkwitz-Riley	
PEQ1	2239 Hz	-5.0 dB	1.19 (Q)
PEQ2	917 Hz	0.0 dB	1.19 (Q)
PEQ3	103 Hz	-0.5 dB	2.00 (Q)
PEQ4	178 Hz	+1.0 dB	2.00 (Q)

Electronic HPF

Gain	-5.5 dB		
Delay	0.0 ms		
Polarity	Negative		
HPF	1296 Hz	24 dB Linkwitz-Riley	
LPF	None		
PEQ1	4597 Hz	-8.5 dB	0.67 (Q)
PEQ2	2239 Hz	-3.0 dB	1.68 (Q)
PEQ3	1496 Hz	-3.5 dB	1.26 (Q)

What’s Gain got to do with it?

Many times the gains of particular crossover output channels are changed in order to accommodate for changes in room acoustics, etc. It needs to be realized that changing the gain of a channel, whether at the amp or in the processor, also changes the crossover point.

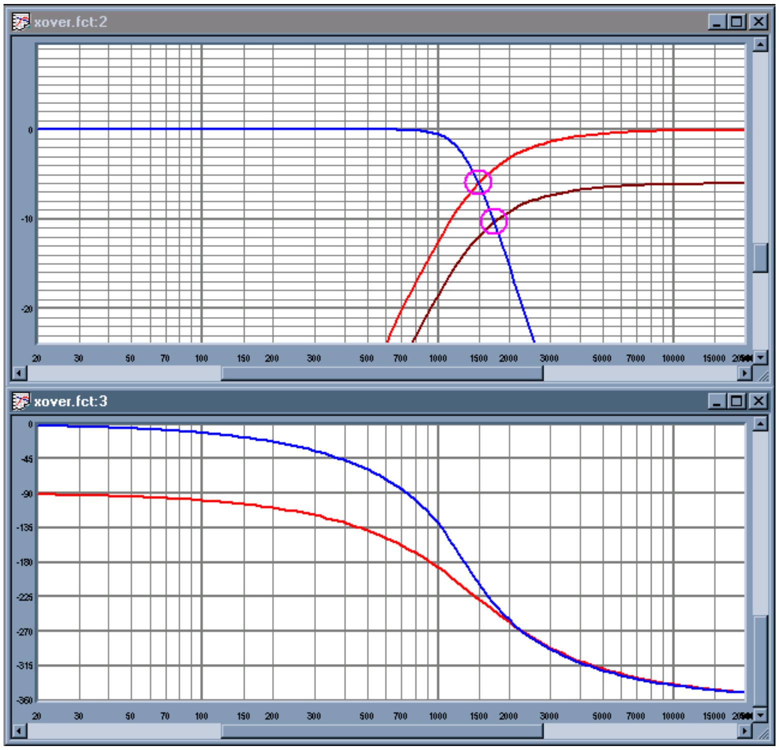

Figure 13: Blue/Brown – LPF at 0dB, HPF at –6dB, crossover point at 1.7kHz.   Blue/Red – LPF and HPF with +6dB gain change on HPF, crossover point now at 1.5kHz. Note that the red and brown phase responses overlap, so the brown phase plot is not visible.

As the above graph shows, a gain change in the magnitude response does not yield a change in the phase response. Therefore, if the crossover point changes and the phase relationship between the two filters remains the same, then it is possible that the phase relationship will differ sufficiently at the new crossover point that proper summation may not occur. For example, the phase responses of the individual sections may be similar between 1.6kHz and 1.9kHz, allowing for proper summation within this region. Outside this region, the phase responses may differ considerably. Moving the crossover point to a frequency above 1.9kHz or below 1.6kHz will not necessarily yield adequate summation. Although this is usually taken into consideration during system design, not all systems allow for the same level of flexibility. Caution needs to be taken when adjusting levels of individual channels. This example again demonstrates that the crossover points of a system are not only incomplete information, but also vary greatly when subtle changes are made to system parameters.

Parametric Equalization

A very important aspect of system settings is parametric equalization or parametric EQ. A parametric EQ is a type of filter that has non-zero gain for some frequency ranges, and zero gain for all frequencies above and below that range. As seen in the previous examples, an EQ is used to eliminate non-linear behavior in the response of transducers. A parametric EQ is defined by three parameters; Q or bandwidth, center frequency, and gain. Q or bandwidth (BW) defines the width of the filter. There are several methods used to calculate bandwidth and Q. Since there is no apparent standard for these calculations, the methods will not be discussed herein. Simply put, a “low Q” and a “high BW” filter covers a broad range of frequencies; whereas a “high Q” or “low BW” filter covers a narrow range of frequencies. The gain of the filter, expressed in dB, defines the amount of boost or cut the filter exhibits at the center frequency. **Figure 14** shows some examples of parametric filters.

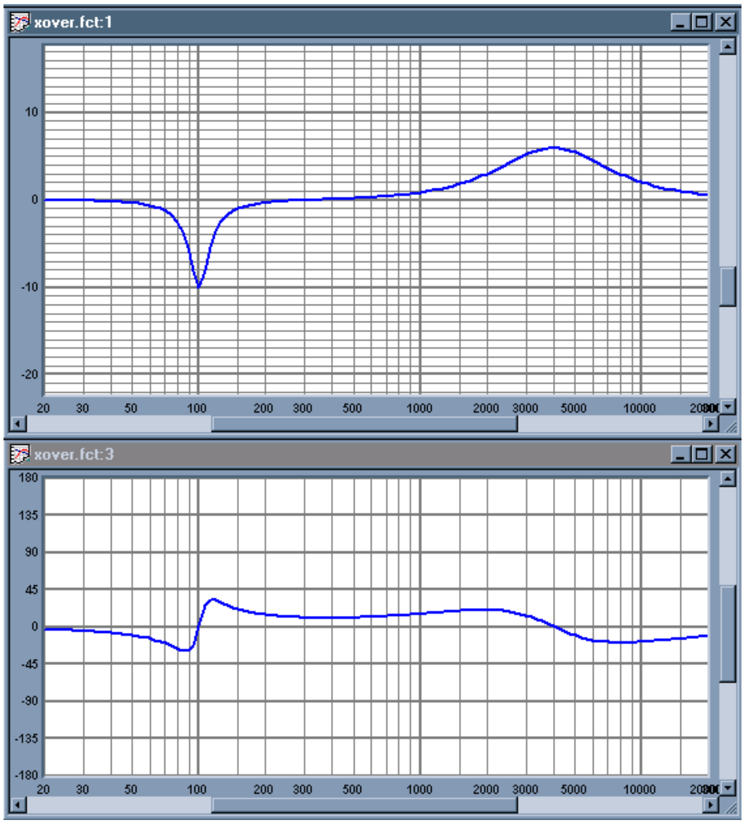

Figure 14: Two parametric equalizers. 100Hz, gain of –10dB, Q of 6.3; 4kHz, gain of +6dB, Q of .67. Note the phase variations that accompany the magnitude changes.

There are two parametric EQs shown. The 100Hz filter could be described as high Q or low BW. Conversely, the 4kHz filter could be described as low Q or high BW. This is illustrated with the actual settings shown here:

	Freq.	**Gain**	**Width**
PEQ1	3981 Hz	+6.0 dB	1.50 (BW)
PEQ2	100 Hz	-10.0 dB	0.20 (BW)

	Freq.	**Gain**	**Width**
PEQ1	3981 Hz	+6.0 dB	0.67 (Q)
PEQ2	100 Hz	-10.0 dB	6.31 (Q)

Note the phase changes that occur with a parametric EQ. This change is critical since any insertion of a parametric into a processor setting may change the phase response at crossover and therefore compromise summation of devices. On the other hand, sometimes this can be used to the designer's advantage. Placing a high Q filter with a *negative* gain at or near crossover may create just enough phase and/or magnitude change to facilitate summation. However, one should *never* utilize a parametric EQ for the purpose of boosting a frequency to alleviate a dip in response at crossover. Many times, as this paper has revealed, dips in response at crossover occur from improper phase alignment between two devices. A parametric EQ can very rarely fix such a dip, and if it could, the resulting system would likely not sound very appealing.

Summary

This paper has attempted to show some of the complexities involved in determining loudspeaker processor settings. It was suggested earlier that many crossover solutions exist for any given loudspeaker system. One could certainly approach a crossover design from differing angles, with greater or lesser results. EAW strives to provide optimal settings for their loudspeakers, and many hours of engineering time, utilizing state of the art measurement facilities, is spent optimizing settings for unparalleled system performance. Without access to like measurement equipment, it is strongly recommended that EAW customers adhere closely to factory-recommended processor settings so that they may enjoy the full performance capabilities of their loudspeakers.