Subject: Re: im a crossover dunce
Posted by Wayne Parham on Wed, 16 Jun 2004 05:23:23 GMT
View Forum Message <> Reply to Message

Without doing analysis using dimensions and component specs, I can't give you any specific answers. But I can take a moment to illustrate the issues, and that might be more interesting for you anyway.

Issues to consider:
When you have a speaker with two or more drivers and a crossover, then there is a range of frequencies where two sound sources are active simultaneously. This is the crossover region, and it is where there is interaction between sound sources. This region is more narrow with high-order slopes than with low-order.

There are many things to consider. But the essential problem is the potential of cancellation at specific frequencies. Nulls occur when pressure waves from two sound sources arrive at the listener out of phase. They are caused by one sound source trying to create a positive pressure while the other is trying to create a negative pressure. The depth of the null is determined by the phase relationship between sources and their amount of coupling.

One thing to consider is distance to the sound sources in relation to the speed of sound. The speed of sound in air is roughly 13548 inches per second. It varies a little bit depending on barometric pressure, temperature and humidity. But it's around 13.5K inches per second, so you can easily see that a sound of 13.5kHz has a wavelength of one inch.

Sound waves are alternating cycles of positive and negative pressure. They emanate from a radiating point as a (3D) sphere, or partial sphere if the sound source is directional. The wavefront moves away from the sound source like ripples in a pond. The difference between positive and negative half cycles is 1/2 wavelength. So a half cycle of 13.5kHz is 1/2 inch, and it represents the distance between maximum positive pressure and maximum negative pressure. Likewise, 6750Hz has half wavelength of one inch.

If the radiator is at maximum excursion in the positive direction, then it is at maximum instantaneous pressure on the surface. This pressure zone travels at the speed of sound through the air and by the time the radiator diaphragm has moved back to its minimum pressure position, the initial signal has traveled a finite distance.

So this brings us to the point of the issue with distance and speed of sound. If you have two sound sources, and each of them is driven in phase, then their outputs will be added together when the listener is the same distance from each sound source. But if you move the listener a half wavelength further from one sound source than the other, then the two signals will cancel each other. Using the 6750Hz example above, the distance differential for cancellation is one inch.

As mentioned earlier, the depth of a null is determined not only by phase but also by coupling. If the charged area is small - if all energies were perfectly contained so that the energies fully combined - then the cancellation effect is nearly complete. For example, if two sound sources were pointed into a small box and the listener positioned so that the signals were out of phase, there would be nearly perfect cancellation. But if the two sound sources were actually facing away from one another and highly directional, the acoustic energies may not combine very fully and the cancellation effect may be small. So there is an amount of coupling between sound sources that is determined by their directionality and orientation and by the shape and conditions of the environment.

That pretty well covers the matter of how position and propagation speed set a phase relationship between sound sources. Now we can examine how crossover electronics introduce phase shift too. The amount of phase shift is determined by the filter slope (first-order, second-order, etc.) and the frequency. In the frequency range where the driver is being used, phase shift is small, nearly zero. This is true no matter what crossover slope is chosen. As the frequency gets nearer to the crossover frequency, the phase shift increases and is maximum deep into the cutoff or stop band. So this is the part where crossover phase comes into play. Crossover slope is only relevant near the crossover frequency and in the stop band.

Some suggest counteracting crossover phase with baffle offset so that delay introduced by the crossover is counteracted by staggered placement of the drivers. That's is a pretty neat idea, and somewhat effective. But the problem is that position offsets create fixed delays and crossover electronics create delays that change with frequency. So you can't directly correlate the two.

There's another matter to consider, and that's differences in distance across the driver's radiating surface. When you have a radiator that is larger than a half wavelength across, then you start to run into these kinds of issues even in a single driver. An example is a single driver speaker that's 6" diameter. Remember the example of the 6750Hz signal having 1" half-wavelength. If the center of the cone is 1" further from you than its edge, there is potential of cancellation if the radiator is moving as a rigid piston, i.e. the edge is moving at exactly the same time as the center. And off-axis movement placing the listener much closer to one edge of the speaker than the other causes a similar situation. So high frequency energy begins to cancel out and nulls are formed off-axis, and that's why direct radiators are said to beam as frequency is increased.

Another thing that causes phase differences across a driver's radiating surface is the fact that it begins to flex at high frequencies. The cone acts as a rigid piston at low frequencies, and the entire surface moves together. But at high frequencies, the cone begins to flex and ripples appear across its surface like ripples on a pond. So some parts of the cone move independently of others. This creates a whole other level of interaction between sound sources, since independent sound sources are now developed at different places on a single radiator.

That's a summary of the effects you'll find that come into play. Loudspeakers are a simple system, but there are some interesting behaviors that arise in them. It's like watching a pot of boiling water in that it's a very simple thing, but the patterns made by the boiling bubbles are many and varied.

Conclusions:
I've come to a handful of conclusions that I tend to use when building a loudspeaker system:

First-order crossovers have some advantages that make them attractive for some things. They are very simple and forgiving. They almost always sound good, as long as the number of speakers in the system is small. They are the closest thing to no crossover you can have. Their main disadvantage is that they pass a lot of stop-band energy. For that reason, I don't usually like using them for tweeters. They also have a wide overlap region, and so nulls usually result off-axis. Response anomalies are often noticeable when moving in the listening area, as phase changes between sound sources as the listener moves.

High-order crossovers have a narrow overlap region, so they are generally better for limiting response anomalies off-axis. The overlap region is smaller, and when combined with directional sound sources, the anomalous areas can be limited to regions of low acoustic energy, hence lessening any interaction between sound sources. They reduce stop-band energies more than lower-order filters, so they tend to be better suited for tweeters that can't handle low frequency power. Their main disadvantage is that they are more complex, and generally less forgiving. But when properly implemented, they provide seamless response on-axis, and much better performance off-axis.

Whether using high-order or low-order filters for a crossover, employ impedance compensation with passive crossovers unless you've thoroughly analyzed the circuit and are designing it with a particular response curve in mind. This requires specific information about the driver itself.

A passive crossover is really such a simple device that it doesn't take into account driver characteristics. It's just a handful of passive components and it is often designed for use with an idealized resistive load. The problem is that loudspeakers aren't resistive except in a fraction of their operating range. They should be considered as complex filters in and of themselves, and they do a lot to react with the rest of the components in the circuit. A Zobel or at least a shunt resistance usually helps a great deal in obtaining the expected results.

For that reason, I don't consider a single series capacitor or inductor to be a first-order filter at all. I call it a pseudo-first-order filter because the driver interaction is so high, it has a lot of influence on the circuit's response curve. Series coils generally form shelved response voltage dividers and series capacitors generally form a peaked response, where the capacitor literally resonates with the driver. High-order filters need impedance compensation even more than first-order filters do. I've seen peaking as high as 12dB from second and third-order passive filters run without a Zobel. Sometimes, one of these effects is exactly what is wanted, so there is merit in the circuit in some cases. But generally when someone wants a pure slope crossover, they should use a damper resistor across the driver, or maybe a Zobel.

Here are some other things you might find useful:Spice
Crossover Document
Crossover Electronics 101 Lab Handhout
Baffle spacing, phase angles and time alignment
Baffle spacing, phase angles and time alignment, revisited
Horn Phase
