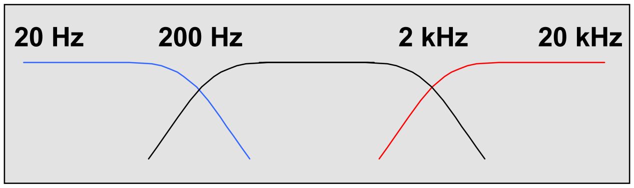
Active Crossovers and Filters



What are active crossovers? In short, they electronic circuits that divide the audio spectrum up into discrete bands of frequencies and they function in place the passive crossovers found in loudspeakers. The underlying motivation for the switch to an active crossover is the improved accuracy and flexibility of the active crossover holds over the passive crossover.

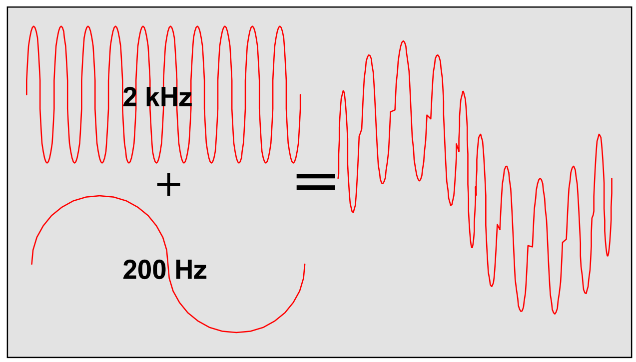
Loudspeakers represent truly complex impedances, which only an equally complex passive crossovers can match. Thus deriving the required crossover frequency and slope is difficult with a passive crossover and changing a preexisting passive crossover frequency is anything but easy. Active crossovers, on the other hand, remove the large passive components that make up the passive crossover. This is for the good, as the hundred feet of magnet wire that makes up the inductors and the two back-to-back electrolytic capacitors that make up most crossover non-polarized capacitors are not missed: these components are far from ideal. And the power amplifier, once freed from having to work through this dreck, exercises a better control of the loudspeaker drivers. For example, a damping factor of 100 means little, if the passive crossover adds 1 ohm of DC resistance to the mix, thereby decreasing the effective damping ratio to 8.

Active crossovers also allow for a frequency tailoring that would be altogether impossible or at least incur efficiency penalties with a passive crossover. For example, with a network designed by Linkwitz, we can effectively shift the resonant frequency and Q of a loudspeaker driver. Furthermore, a high Q crossover can boost a drooping low frequency response of a low Q speaker while filtering away sub-sonic garbage, such as record warp.

Bi-Ampping Psycho-Acoustics

An added advantage is a psycho-acoustic phenomena wherein bi-ampped systems seem more powerful than the sum of the amplifier wattages, e.g. two 36 watt amplifiers sound much more powerful than a 72 watt amplifier.

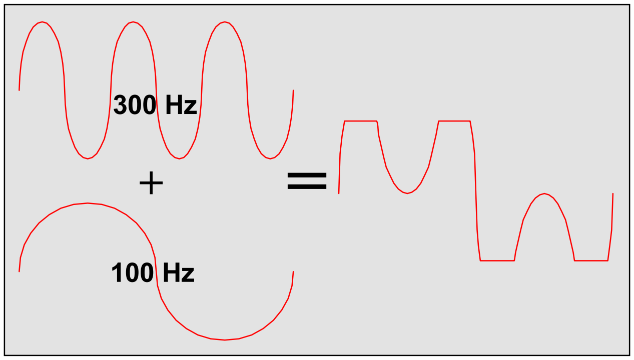
How is that possible? Half of the answer lies in the output voltage adding together rather than the wattages adding together. Wattage is based on the voltage squared. For example, 24 peak volts of output signal equals 36 RMS watts, but 48 peak volts of output signal equals 144 RMS watts. In other words, doubling the voltage quadruples the wattage; tripling, increases the wattage by ninefold.



200 Hz + 2 kHz added together

Let's imagine a crossover point of 500 Hz. If a 144 watt amplifier is presented with a signal of two 24 volt tones, say 100 Hz and 2 kHz, the signal will trace a 2 kHz sine wave superimposed on a 100 Hz wave. The total peak voltage is 48 volts, which equals 144 RMS watts. Remember that a given instant the amplifier's output is at only one specific voltage. (The plate cannot be at +24 and at –12 volts at the same time. We are dealing with cooper wire and voltage, not fiber optics and light.) A passive crossover splits these two frequencies from the output from a 144 watt amplifier into two 24 volt signals. The active crossover also separates the two frequencies so that two 36 watt amplifiers can play these two tones at the same volume as the single 144 watt amplifier can, i.e. 24 volts at 100 Hz and 24 volts at 2 kHz.

Of course, if both tones had been at frequencies below the 500 Hz crossover point, say 100 and 300 Hz, one 36 watt amplifier would clip its output while the second amplifier sat idle. While this seems to limit the power increasing effect, this indirectly leads to second power increasing effect.



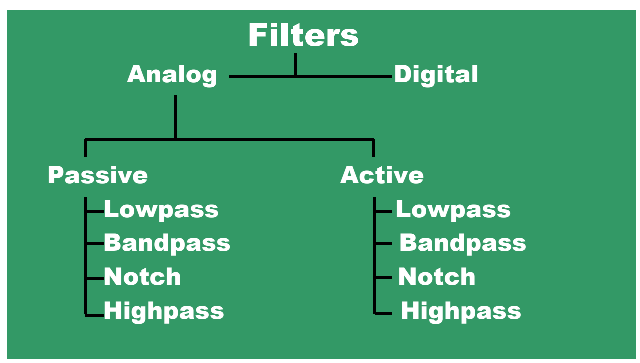
100 Hz + 300 Hz added together and clipped

The ear is most sensitive at about 3 kHz (roughly, the resonate frequency of the ear canal). As the frequency falls from this frequency, so does our sensitivity to it. So if a bi-ampped system sees a signal that only clips the woofer amplifier, the ear will not complain as much as it would if the tweeter amplifier had clipped. A further reason is that woofers are slow heavy devices that cannot reproduce the harsh high harmonic of the clipped low frequency waveform. So while a low frequency amplifier may clip repeatedly, the woofer will not readily betray that fact. And since it is usually the low frequencies the demand the greatest share of power, the two 36 watt amplifiers may even seem more powerful than a 144 watt amplifier as long as the high frequency amplifier has not clipped. (This might help explain why a 3 watt tube SE amplifier can sound as powerful as a 30 watt solid-state amplifier: the ear hears clipping as the limit to power. Transformer coupled, feedback free amplifiers have the softest clipping characteristics of any amplifier, as the transformers limit the transfer of high frequencies and the absence of feedback means the driver stage will not try force a sharper slope on the output tube's flattened waveform.)

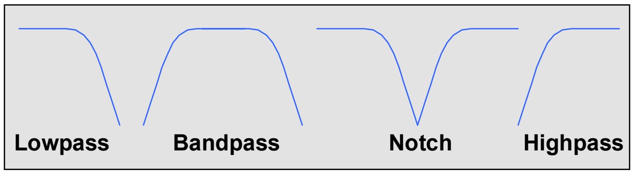
One last argument in favor of active crossovers: headphone listening can be superb, but it suffers from the lack of a visceral component; the floor just doesn't shake with headphones…but it can. For over two decades my preferred way of listening to headphones has been to use them with powered subwoofers. With a high frequency limit of 80 Hz the subwoofers seldom even move with pop music, but with Mahler's 3rd symphony, they work overtime. (The missing octave is filled in and yet your neighbors have a hard time locating you as the source of the low growls, as the ear has a hard time localizing ultra-low frequencies.)

Filter Types

What is the difference between a crossover and filter? A crossover is specifically meant to work with loudspeakers and it is usually made up of several filters. In other words, a filter is the more general term, which suits a circuit that has such a wide application. Almost all electronic devices employ some filters. The radio and TV set are filters of sorts, as they select only a narrow band of frequencies at a time. A filter can be made out of as little a single capacitor or inductor (and a load resistance). A filter can also consist of twenty resistors and twenty capacitors and many amplifiers.



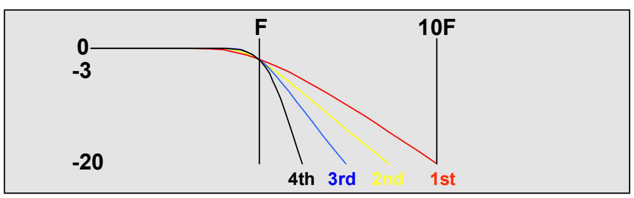
In all filters, however, some frequencies are allowed to pass, while others are attenuated. The portion that is allowed to pass is named the passband and the portion that is excluded is named the stopband and the portion of overlap between these two bands is named the transitional-band.



If only low frequencies are passed, we call this filter a low-pass filter, as that is what it passes. If only high frequencies are passed, we call this filter a high-pass filter. If only a band of frequencies, say 500 to 5000 Hz, are passed, we call this filter a band-pass filter. If all frequency are passed save for a very narrow band of frequencies, we call this filter a notch filter.

Beyond these four basic divisions, a further classification is applied based the steepness of the rejection of undesired frequencies.

Since the slope's steepness is marked by the "order" in a filter's design. (The number of poles is also sometimes used as a short hand description of the steepness.) For example, a single order filter, also called a single pole filter, has a crossover slope of -20 dB per decade, which equals -6 dB per octave. Thus a second order filter attenuates at –12 dB per octave; a third order filter, -18 dB per octave; a fourth order filter, -24 dB per octave; a fifth order filter, -30 dB per octave; and a sixth order filter, -36 dB per octave.



The next taxonomic division is made based on the filters alignment. The alignment is based on the relative position of the poles in the filter, which in turn gives rise to the different shapes each filter exhibits across its frequency range. Some filters offer a sharp transition from passband to stopband, but at the cost of some ripple and even ringing in the frequency response. Other filters offer a soft transition that droops at the transition frequency that is ripple free. In short, an analog filter's design hopes to optimize one or two parameters, but always at the cost of some other parameters.

The most common filter alignments are the Butterworth and the Bessel (AKA Thomson). Less common alignments are the Gaussian, Chebyshev and the elliptic (AKA Cauer). A recent addition to the palette of filters is the Linkwitz-Riley alignment (AKA Butterworth squared), which was specifically designed for use as loudspeakers crossover (More on this type of filter to follow.).

Common Audio Filter Types

The Butterworth filter is the most commonly used filter type in audio applications. It has a fairly flat time response, a fairly crisp transition shape and the flattest passband response. In contrast, the Bessel filter offers a flatter time response, but not as sharp a transition shape or flat a passband response. The Chebyshev and the elliptic filter are seldom used in everyday audio gear, as these filters bring too many undesirable characteristics such as ripples in the passband or wild phase shifts.

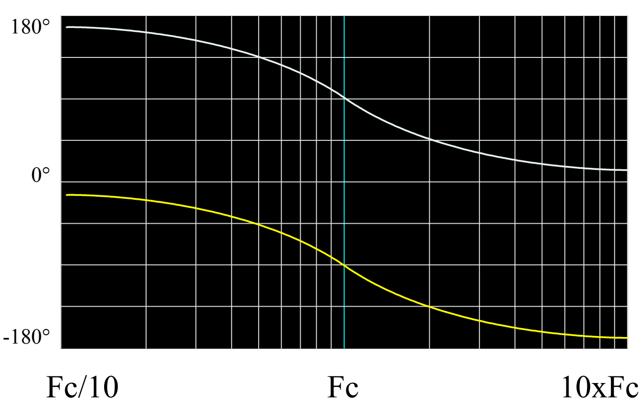
The Linkwitz-Riley filter improves on the Butterworth when feeding loudspeakers. Odd order crossovers (6 dB and 18 dB per octave filters) are marked by of 90 degree multiples in phase differences between outputs. The sum of two 90 degree equal amplitude signals is +3 dB boost in output. Thus the –3 dB point of the filter works to yield a flat transition region.

Even-order crossovers (12 dB and 24 dB per octave filters) are marked by 180 degree multiples of phase differences between outputs. The sum of two 180 degree phase shifted, but equal amplitude, signals is a deep null in the output. The solution is to reverse the phase of one of the loudspeaker drivers to eliminate the phase difference at the crossover frequency. However, when two loudspeakers play the exact same signal (no amplitude or phase differences), the result is a twofold increase (+6 dB boost) in volume. So if we wish to reconstruct the outputs of one lowpass and one highpass filter, we must tailor the attenuation at the crossover frequency to match the intended summing device.

Thus, when using loudspeakers that see the same phase and amplitude signal, we cannot use a –3 dB down point, as it would yield +3 dB; thus, we need a –6 dB attenuation at the crossover frequency in order to ensure a flat frequency response. The same phase is the key point here. If a Butterworth second or fourth or sixth order crossover is used, the frequency response will display a bump at the crossover frequency. On the other hand, using a Linkwitz-Riley 2nd or fourth order crossover will yield a flat frequency response at the crossover frequency.

Well at least that is the theory. Complications arise: the distance between drivers, the frequency response of each driver, the front-to-back spacing of the driver's voicecoils. If the distance between drivers exceeds the wavelength of the crossover frequency, or if one or both drivers droop at the crossover frequency, or if the tweeter acoustic center is substantially in front of the woofers, then the Butterworth aligned crossover might actually prove flatter.

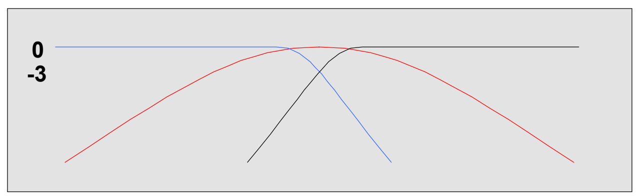
Do not falsely imagine that an even order crossover is phase flat; it isn't. The actual phase relation between 2nd order lowpass and highpass filters is a constant 180 degrees phase difference, which when one driver's connection is inverted, yields a constant phase relation between drivers, but not a flat phase response.



Phase differences between 2nd order Butterworth high-pass filter (white line) and its low-pass complement (yellow line)

A Phase Flat Loudspeaker

One interesting loudspeaker configuration was created by Philips. An ostensibly two-way loudspeaker was designed using a second order Butterworth crossover, but without the tweeter's phase reversal. Yes this resulted in a deep suck-out. The suck-out was then filled in by using a bandpass filter (-6 dB slopes) feeding a full-range driver. This extra speaker saw only the crossover frequency unattenuated, as all the frequencies below or above this frequency were attenuated at –6 dB per octave. The sum of all three driver's output equaled a phase-coherent flat frequency response. In this speaker system, the woofer was prevented from going too high, while the tweeter is protected from low frequencies, while fullrange driver filled in the hole. Unfortunately, this solution, like most speaker innovations, disappeared long ago.

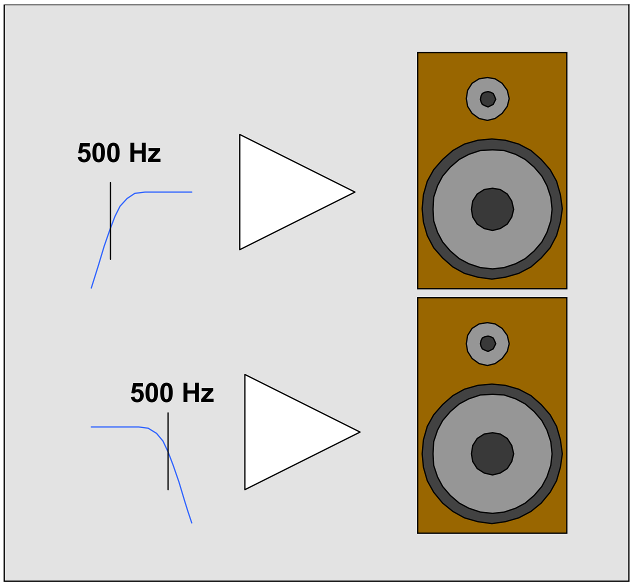


Crossover Testing

Evaluating active crossovers under actual use is difficult. Which component is making the biggest difference? The active crossover itself or the extra amplifiers and speaker cables? A listening test I have used is to take a pair of fullrange conventional loudspeakers and apply the crossover under test using my present amplifier and cables. This means there is no worry about relative efficiencies and frequency response limitations.

The procedure is simple enough; place one speaker on top of the other and set the crossover frequency to some value between 300-700 Hz (too high a frequency will result in lobbing effects due to greater than wavelength spacing between drivers). This test requires a mono signal source and I recommend any of the mono Verve recordings of Ella Fitzgerald.

First listen to just the loudspeaker on top (fullrange, without the active crossover in other words). It takes some time to get used to mono, so play about 30 minutes worth. Next introduce the active crossover and feed the highpass signal to the top loudspeaker. Now listen to the same 30 minutes of music.



General crossover test setup

If the crossover is working well, the sound should be very similar to that from the single loudspeaker. The usual result is that the sound is much better than that from the sole speaker. Why? The power increasing effect is part of the reason. The rest of the answer is found in the reduction in distortion that results from freeing the top loudspeaker from the onerous task of reproducing low frequencies. In other words, the top loudspeaker is less likely to mechanically clip. Furthermore, there is a large reduction in a seldom mentioned distortion: frequency modulation distortion, FMD.

FMD

This distortion is related to Doppler distortion: the phenomenon of frequencies increasing in pitch as a sound producing object nears us and the sound decreasing in pitch as the object retreats. The sound of a race car approaching and passing is the usually given example. RRRRR R R R R R R R R R R.

Loudspeaker diaphragms move in response to signals and the large diaphragm movements caused by low frequencies raise the high frequencies while the driver's cone moves toward us and then lower the high frequencies when the cone moves away from us. Consequently, in absolute terms, it is impossible for anyone driver to reproduce more than one pure tone at a time. This is one reason why electrostatic and planer speakers sound as good as they do; the large surface areas do not need to move very far to produce high sound levels. Three-way, four-way, five-way, and six-way loudspeaker also greatly reduce this distortion, while admittedly adding huge crossover problems. And the worst offenders are small fullrange single driver based loudspeakers.

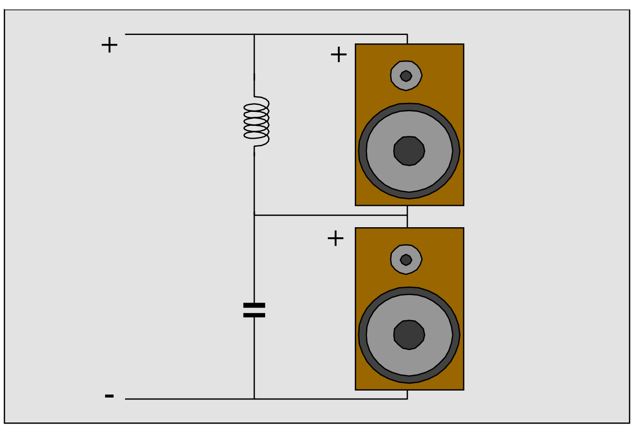
Bi-Ampped Identical Loudspeakers

I once heard something like the test setup with four large Advent loudspeaker from the early 80's. A friend bought four speakers for quadraphonic use and instead used all four in a stereophonic setup with a active crossover set at 100 Hz. The improvement in midrange clarity and bass definition was markedly better. In fact, it sounded much better than it should seem possible. (A similar experience occurred when I heard a speaker setup that consisted of eight high quality mini-monitors configured as two speakers. No crossover was used and four speaker enclosures were stacked on each other per side, with only one facing the listener. The resulting impedance was identical to that of a single speaker, as they were wired in series-parallel. Because each speaker saw half of the available output voltage from the power amplifier, its response was down 6 dB; but as the radiating area had increased fourfold, the output response gained +12 dB of gain, which in the end, yielded +6 dB of gain.)

In same vein, if you are pleased with your present loudspeakers, but wish that they provided a little more headroom or clarity at high volumes, then this doubling of loudspeakers might be the best way to go. Active crossovers are certainly nice, but not strictly necessary, as one choke and one capacitor are all that is needed to make a passive single order crossover. My recommendation is to select a crossover frequency that is the geometric mean of the lowest and highest frequencies produced by your loudspeakers. For example, 600 Hz for speakers that extend down to 20 Hz and up to 20 kHz; 1400 Hz for that only extend down to 100 Hz (mini-monitors).

$$F_{geometric} = \sqrt{(F_{low} \times F_{high})}$$

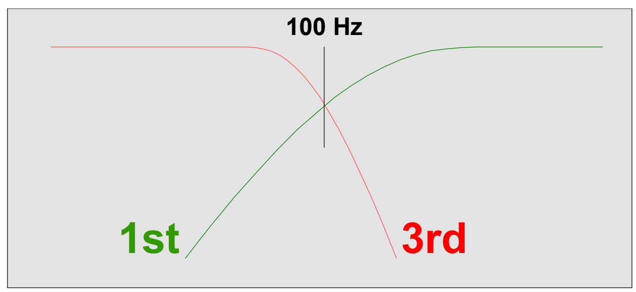
I also recommend using the shunt arrangement rather than the series, as it provides better damping. (The shunt arrangement attenuates the stopband by shunting the driver with an ever decreasing impedance; where the series arrangement attenuates the stopband by presenting the driver with an ever increasing impedance.) Of course, with a passive crossover, none of the power increasing effects are bestowed.



Choosing Crossover Slopes

In theory, nothing beats a simple 1st order crossover. It results in a flat, time coherent frequency response. Actual practice differs. The 1st order filter provides a shallow attenuation slope that is too gradual to protect fragile tweeters from brutal low frequencies and too gradual to keep woofers from working up into their high frequency limits, as a two to three octave overlap is recommended to make this filter work well. However, as the number of crossover points increase, the more likely it is that this gentle filter slope will provide sufficient protection.

On the other hand, if you are adding a powered subwoofer to an existing fullrange loudspeaker, the 1st order highpass filter is a good choice for the fullrange speaker and a 3rd order filter for the subwoofer. This combination seems to work well in terms of amplitude response. It also has a minimalist aspect that is gratifying: the subwoofer contributes only its low frequency heft and then quickly gets out of the way of the lower midrange; the satellite loudspeaker sees a very gentle highpass slope that allows it extend down into the deep bass as it very slowly falls off in volume with decrease frequency.



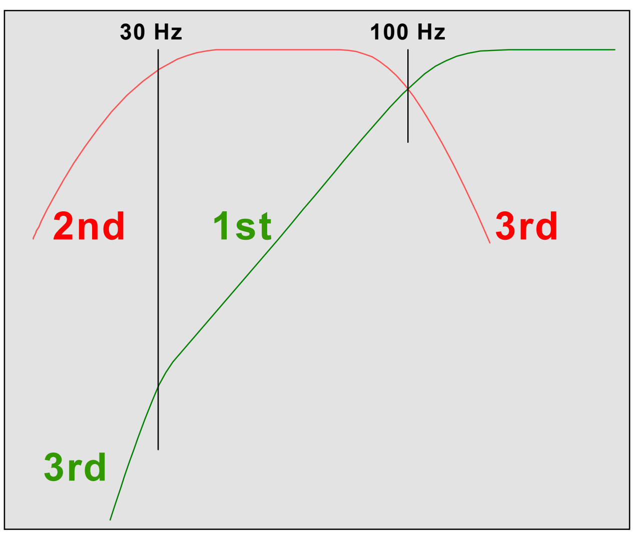
Those using zero feedback tube amplifiers, will find a preexisting 1st order highpass filter in their power amplifiers. The coupling capacitor that connects to the grids of the output tubes defines a highpass filter that is usually set to some very low frequency, such as 1 to 5 Hz. Decreasing the value of this capacitor raises the crossover frequency. By choosing the right value, we can forgo the need for an external crossover. The formula is an easy one:

$$C = 159155/Fc/R,$$

Where Fc equals the desired crossover frequency, C equals the value of the coupling capacitor in µF, and R equals the value of the grid resistor. Truly minimalist. One trick I have wanted to try for some time now is using a satellite loudspeaker (a sealed box with a Q of 1) along with a 1st order filter, which comes with a Q of 0.707 (set to the Fo of the enclosure).

The Qs are multiplied against each other, which will give the loudspeaker an effective Q of 0.707. The advantage of this arrangement lies in the smaller enclosure needed for a higher Q loudspeaker alignment and the natural 2nd order highpass filter function of the sealed box enclosure adding to 1st order filter to make a 3rd order highpass filter with a -3 dip at the crossover frequency.

To work best this 1st order filter must come before the power amplifier, as using a capacitor in series with the loudspeaker will not work due to the impedance spike at the speaker's resonance. The subwoofer's lowpass filter should be actively realized with an active 3rd order Butterworth filter. A further refinement would be to add an active 2nd order or 4th order highpass filter whose cutoff frequency would equal the Fs of the subwoofer. This would both work to protect the satellite speaker from excessive low frequency cone excursions and work to bring both subwoofer and satellite into the same phase relationships, as the subwoofer's own box resonance defines a highpass filter (2nd order for sealed boxes and 4th order for bass-reflex enclosures.)



Using the speaker's own effective filters to complete a crossover . Here the speaker's 2nd order function is added to a 1st order filter to yield an effective 3rd order slope.

For most drivers that suffer from either limited power handling or limited frequency response, sharper cutoffs are needed. The 2nd, 3rd, and 4th order slopes both protect and isolate the drivers. Many audiophiles, however, distrust higher order filters, fearing the signal's phase reversals and complexity of the crossovers. Yet they do not realize that the speakers themselves bring phase aberrations because of resonances driver breakup at higher frequencies. For example, most tweeters have a resonant frequency between 800 to 2 kHz and all cone and dome drivers must flex once the circumference of the diaphragm becomes larger than the twice the wavelength of the frequency being reproduced. (Strictly speaking, for example, we should not let an 8 inch woofer extend beyond 1400 Hz; but we do so regularly.)

Here is where active filters shine. As has been mentioned, the problem with using passive filters with loudspeakers is that they seldom function entirely to plan and this poor performance is due to the passive components falling short of their ideals.

Inductors (chokes) are made up of long lengths of wire, which adds resistance. Capacitor also carry an ESR (effective series resistance) that upsets inter-relations within the filter. And loudspeaker drivers are anything but pure resistances. All of these failings add up to a filter that does not match its textbook model. On the other hand, the active filter can match its textbook template easily. Additionally, active filters can make do with only resistors and capacitors to define their poles. Doing without inductors is a relief in terms of money and space, as the lower the frequency, the bigger the inductor. And because the active filter presents an almost infinite input impedance to the frequency tailoring parts, high valued resistors can be used, which only require small valued capacitors to match. For example, an 8 ohm loudspeaker driver crossing over at 1 kHz requires a 20 µF capacitor, whereas a 10k resistor only needs a .0159 µF capacitor. In other words, active filters present a win, win situation.

Mix and Match

Active filters can be cascaded to yield a degree of frequency tailoring that is all but impossible with passive filters. The passive filter works best with known, fixed input and terminating impedances, which a cascading passive filter cannot readily do.

For example, two active 3rd order filters cascaded equals a 6th order filter. An active highpass and a lowpass filter cascaded can yield bandpass filter; and these same filters fed into a mixer can yield a notch filter.

Conclusion

In this article we have looked into the why bother with active crossovers and filters? We have seen that active crossovers and active filters in general offer some real advantages over their passive brethren, as they are more accurate, flexible, and easy to implement. In the next article, we will look into designing tube crossovers.

//JRB

Suggested Readings

Active Filter Cookbook
By Don Lancaster, 1975, 95, 96
IC based active filters explained in detail. A classic that is readily available new and used.

Audio/Radio Handbook
By National Semiconductor, 1980
This book has a chapter whimsically named "Floobydust," which covers some interesting active crossovers, including asymmetrical filters.

Passive and Active Filters
By Wai-Kai Chen, 1986
The book to own, if you are serious about filters. A classic, but not for those fearful of math.

Understanding Electronic Filters
By Owen Bishop, 1996
This book is the best place for the novice to start. Should be read from start to finish.

