

Theory & Practice

Fine and Funky Filters

by Mark Amundson

These days, when confronted by a digital speaker processor or digital crossover, you have multiple choices in the high pass, low pass and crossover filter selections. For those not up on filter lingo, words like Butterworth, Chebychev, Bessel, Elliptical and Linkwitz-Riley sound more like European law firms than filter types. So for those of not possessing an electrical engineering degree with a minor in control systems, this article is to introduce basic “pass” filters and help you make some choices in setting up a drive processor.

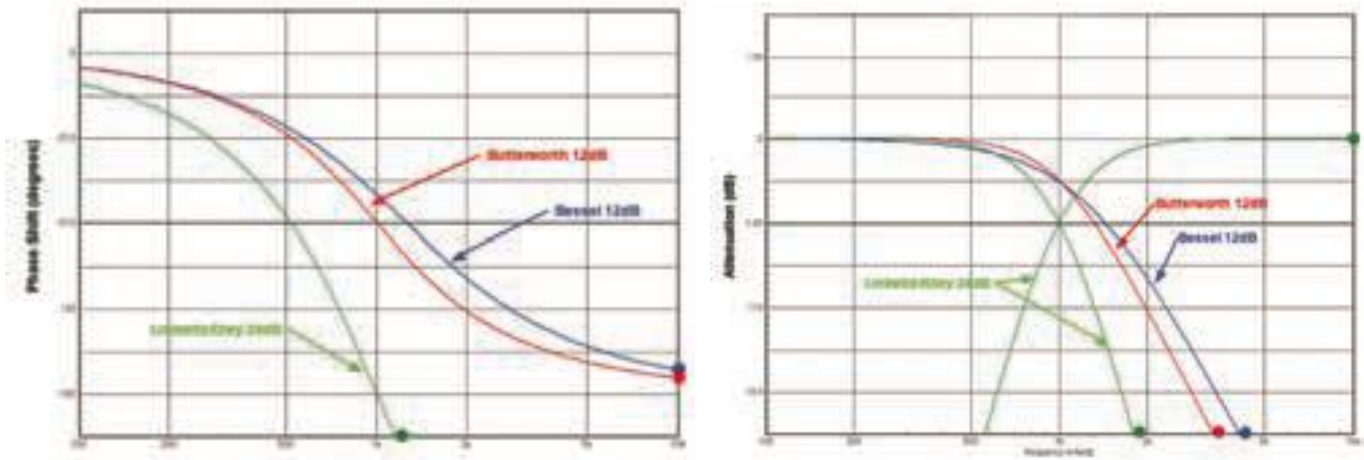
Pass/Cut Filters

The most confusing part in filters for processors is the terminology of highs and lows when combined with pass and cut. In most cases, high pass and low cut mean the same filter. Also, the high cut and low pass terms mean the same thing—the confusion is like the “glass half full/half empty” quandary. From a 20Hz perspective, high pass seems to have more meaning to a filter at 40Hz. But at 100Hz, low cut seems to have better meaning to the same 40Hz filter. Eventually, the terms should merge in your head if used often enough.

Then there is the crossover filter, meaning two filters of opposite types for splitting the audio band for specific types of speaker cabinets. Taking a typical example, a subwoofer cabinet may be set up with a high pass filter (low cut) for about 40Hz that matches up with the sub cabinet’s capabilities. Then a high cut filter (low pass) at 100Hz becomes half of a 100Hz crossover point, so that the top box cabinet starts at 100Hz. Thus the top box filters have a high pass at 100Hz, and, if in passive mode of operation, a high cut at around 16kHz to keep the dogs from barking. So crossover filters generally are set at the same frequency to avoid gaps in coverage of the audio frequency spectrum. But if the efficiencies of the adjoining frequency drivers are radically different, there may be a modest gap in the crossover frequency filters for best matching. This is usually found in two-way bi-amp top boxes with cone and compression driver crossover points and implemented in a passive crossover network.

In filter lingo, the filter name is typically associated with a number for either the number of filter “poles” or the attenuation steepness. In the bad old analog days, filter poles typically corresponded with the number of inductors and capacitors used in a passive filter circuit. Each pole contributes a 6dB drop per octave from the filter frequency. So a two-pole high cut filter at 1kHz would be down 12dB at twice its frequency, or 2kHz. The more filter poles, the steeper the filter drop-off, in general. But adding poles or filter steepness does not always benefit your sound. Beyond two to four poles, some filters start ringing and creating electronically induced overtones that are usually not musical (or desired). Filter ringing is great for keyboard synthesizers, but not for sound systems.

Most typical speaker processors offer choices like: Butterworth 12, 18 and 24; Bessel 12, 18 and 24; and Linkwitz-Riley 24. These labels indicate the filter type name and the db/octave filter steepness in the “stop-band.” I tend to fall into convention and choose Linkwitz-Riley filters at the crossover frequencies, and select 12dB/octave Butterworth filters for the low cut and high cut points (e.g. 45Hz and 16kHz, respectively). One last tidbit to remember is that the frequency associated with the filter is typically at its 3dB point. For our general discussion of stop filters for speaker processing, the -3dB point usually denotes the filter’s corner frequency.



Butterworth

No, not a brand of pancake syrup, but a “maximally flat amplitude” type of filter. Doing my best to avoid confusing engineer terms, Butterworth filters do their best to keep a reasonably sharp but smooth drop at the filter corner frequency, at the expense of letting the phase response wander a bit. Most filters are characterized by their behavior when exposed to impulses of signal (spikes), transient steps in signal (step response), and how much phase delay the filter imposes on signals in various parts of the passband and stopband (group delay). Butterworth filters in the two- to four-pole variety have slightly more phase change than Bessel filters, and are more prone to ring (dampen less) with impulses and transient steps.

Application-wise, Butterworth filters are pretty much the default choice of the far ends of the audio spectrum to ensure out-of-band signals roll quickly off. Because the phase shifts are more dramatic, you are less likely to use Butterworth filters at crossover points. There are exceptions, as you will see in the Linkwitz-Riley description. But if you want a 90-degree shift at the corner frequency, a two-pole Butterworth filter is your ticket.

Bessel

Bessel filters are the natural opposite of Butterworth filters, in that Bessel filters are described as “maximally flat delay.” This means the amount of phase shift in the passband to stopband is minimized compared to other filters. Also, Bessel filters have more dampening (less ring) when exposed to impulses and step transients. Compared to a Butterworth, a Bessel filter has a noticeably less sharp corner in transition from passband to stopband.

Bessel filters are best used in applications where crossover points require minimal phase change from one driver to another. However, Bessel filters are the least used filter since the Linkwitz-Riley filter came about in 1976. Bessel filters are still available, but used much less these days. If you suspect you have too much ringing in your Butterworth filters, switch to Bessel filters to stomp out that problem.

Linkwitz-Riley

Mr. Linkwitz and Mr. Riley are two electrical engineers who worked for Hewlett-Packard back when it was a dominant test equipment manufacturer, not the printer and computer maker it is today. The Linkwitz-Riley filter concept came from cascading pairs of identical two-pole Butterworth filters in low pass and high pass configurations. The result is a crossover filter that uses the same corner frequency in high pass and low pass, has no peaks or dips and is phase continuous at the crossover frequency. In other words, at the crossover frequency, both drivers are in phase, each contributing half the energy they would in their passbands.

Before Linkwitz-Riley filters, the Butterworth and Bessel filters would result in the drivers’ differing phase motions and peaking in amplitude response as both drivers contributed more fully at the crossover frequency. Since the birth of Linkwitz-Riley filters and modern analog “active” filters, almost all analog crossover units use the 24dB/octave Linkwitz-Riley filters at each crossover point. With the new DSP

speaker processors, many analog circuits and quad-matched potentiometers are now replaced by a digital math equation that creates all these filter types with perfection never dreamed of 30 years ago.

Final Notes

Whether it is line arrays or conventional groupings of speakers, the Butterworth, Bessel and Linkwitz-Riley filters should be able to fulfill the needs. Sharper filters can be used, but at the risk of ringing or running out of DSP capability. However, sharper filtering may prevent driver overheating and over-excursion if pushed to the system limits.