

DIY BIAMP 6-24 ACTIVE CROSSOVER

by Nelson Pass

INTRODUCTION

The DIY BIAMP 6-24 kit is a stereo analog active biamp crossover circuit designed for high flexibility. Featuring 12 discrete Jfet buffer circuits operated without negative feedback, it offers 16 continuously adjustable filter elements in two low-pass and two high-pass filters with slopes of 6, 12, 18, or 24 dB per octave and various “Q” characteristics. It has flexibility not generally offered by crossover products have “textbook” characteristics, allowing more detailed tailoring of filters around the actual character of different loudspeaker drivers.

The merits of active crossover networks are well understood. They allow:

- flexibility that is not easily achieved with big passive coils, capacitors and resistors.
- the loudspeaker driver to couple directly to the low impedance which is the output of the amplifier, maximizing the damping factor and efficiency.
- two amplifiers to deliver, not twice, but as much as four times the power with lower distortion.
- selection of amplifier types appropriate to the frequency region and sensitivity of the loudspeaker drivers.

If these arguments are convincing, then perhaps the remaining question would be “Why would I want an analog crossover when modern DSP crossovers solve our problems easier and cheaper?”

There are people who prefer analog circuits, particularly if they are running vinyl and don't wish to run the audio through whatever degradation occurs in the analog-to-digital conversion, DSP processing, and then digital-to-analog conversion. Or maybe they just feel it “spoils the cow's milk”. Whatever the reason, there are still discerning audiophiles who want analog.

Analog or Digital, there is still a need for flexible adjustment in the character of the filters. Classic (“textbook”) filter characteristics are a good start, but generally assume some level of perfection in loudspeaker components that is rarely there.

The goal of the crossover network is to seamlessly join two or more loudspeaker drivers so as to produce an accurate (or at least pleasing) summation of the audio signal. Most of the time active crossovers are made up of standard filters which have slope figures of 6, 12, 18 or 24 dB/octave. It is not difficult to make a crossover network whose output sums to a flat frequency response curve, and some of these filter sets can do it while preserving the integrity of the phase character of the signal. For example, simple 6 dB/octave high pass and low pass filters can nicely sum to this level of perfection with no fuss.

Let's note that these filter slopes are often described by the number of *poles* which is the number of 6 dB/octave elements, for example a 4 pole filter is 24 dB/octave.

Of the examples of crossovers with higher slopes, an effort is made to have the frequency response sum flat and the phase shift vary smoothly, an excellent example being the Linkwitz-Riley filters which sum flat, but have a gradual phase shift with the difference in phase being a constant between the drivers. This is a popular crossover design - many of the commercial products use the L-R filters at 24 dB/octave slopes.

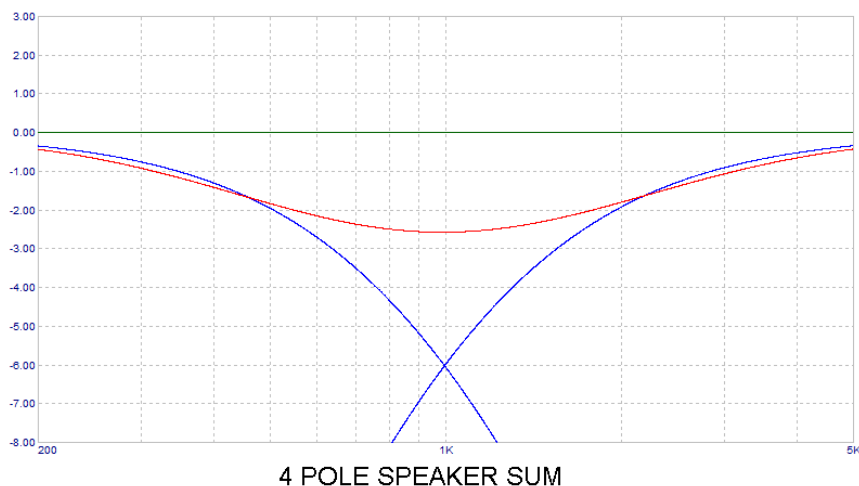
If you have perfect speaker drivers, with flat amplitude and phase response, then you can take one of these standard filters and you are probably done.

If your speakers are not perfect, then this is just the beginning of the process, as the driver frequency response and phase shift are additional unwanted elements in the crossover. My good friend Siegfried Linkwitz dealt with this issue by adding additional filters for the individual drivers to correct for these deficiencies, however, he also noted that adjustment of the crossover itself could often accomplish it or something similar.

Let's take an example, a crossover with 24 dB/octave slopes designed for flat response at the crossover point. The blue curves are the output curves of the low pass filter and the high pass filter. The very flat green curve is the sum of the output of those two filters.

The crossover frequency is 1 KHz, and let's imagine that we have some classic cone type drivers which ideally act as bandpass filters on their own with 12 dB/octave slopes at the bottom and top of each. In this example we imagine that the bandwidth of the drivers extends an octave beyond the crossover point, so that the high frequency driver rolls off at 500 Hz and the woofer rolls off at 2 KHz. Believe me, this is a generous assumption - we don't usually have margins this good.

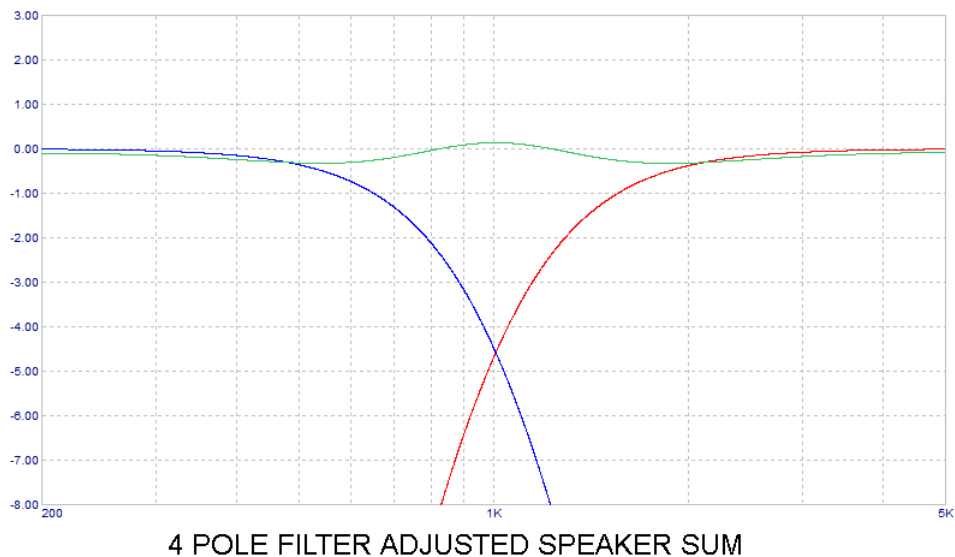
Here we see an LR filter response with low and high pass filters in blue, and their flat sum in green. If we take the acoustic sum of this output through these hypothetical drivers we get the red curve. It is not so flat because the drivers themselves contribute to the crossover.



However we can improve on this by playing around with the filter characteristics, by individually adjusting the filter elements. The phrase "playing around" is apt, because it's the easy way for regular humans to do it - adjust something and see what happens. It helps if you take good notes along the way...

This is the design feature that the DIY Biamp 6-24 brings to the table. Each of the 16 R/C elements that make up the two channel system is independently adjustable with a potentiometer having an adjustable 6:1 resistance ratio. The frequency range is determined by polypropylene film capacitors with the values selected by the builder. A typical example, a 0.01 uF capacitor value gives a range from about 280 Hz to 1.6 kHz.

This extra flexibility comes in handy. Below is the same simulation of acoustic response as above, but with some casual adjustment:



Here the amplitude error is within .5 dB instead of 2.5 dB. Not perfect, but it beats the earlier curve, and probably can be improved upon with a little more work. Keep in mind that we are trying for simple improvement here, and anyone who plays with this stuff knows that +/- 1 dB results for speakers are exceptionally good, +/- 3 dB is still quite good. Even +/- 5 dB is often still acceptable as a trade-off against other effects, and you get to be the judge.

The procedure of making crossovers better generally involves measurement, listening and adjusting (in whatever order you like) over and over again for a lengthy period of time. If you are a perfectionist, that period of time can be months. My longtime business partner Joe spent ten years adjusting his Tannoy crossovers (by ear) but the crossover came out flat.

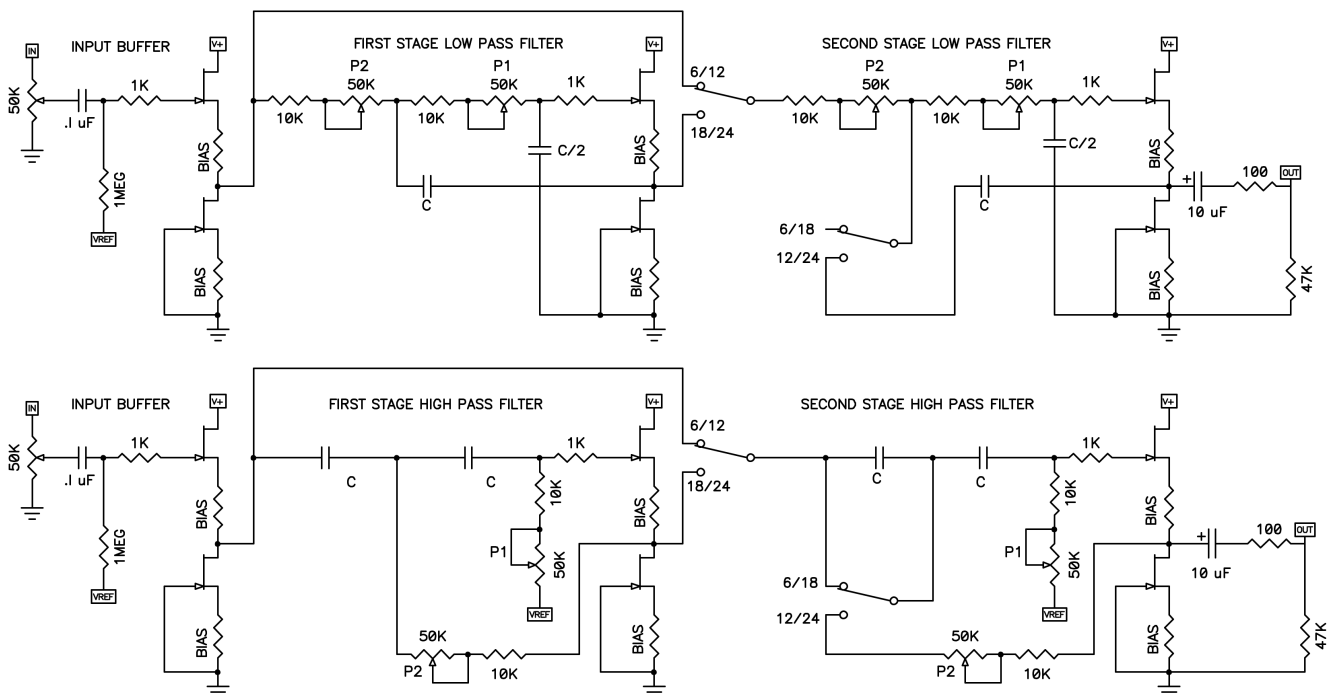
You can do it with just measurements or with just listening, but it's very helpful to have both. It doesn't take amazingly good ears or expensive exotic equipment. A cheap microphone and a laptop and some free software are usually more than adequate, as it is just the crossover points you are looking at.

In my experience, there is great pleasure in making adjustments until you get what you like. I would list it as one of the better excuses to use adjustable active crossovers.

The Standard DIY Version

There is a “standard” version of this circuit, which has predetermined resistor and capacitor values for each range. It is not the only possibility, but has been chosen for greatest ease of use for beginners. Later, we will see some of the potential alternatives, which will include those textbook values mentioned earlier. In this diagram you will see the resistor and potentiometer values and most of the capacitor values except the filter capacitors used to determine the range of adjustment. The variable potentiometers allow a 6:1 frequency range for each pole, although the resulting filters often have less range, depending on variations chosen between the resistance values in a given stage.

Here is a schematic showing the filter circuits for one channel:



ACTIVE CROSSOVER – BIAMP 6–24 DB

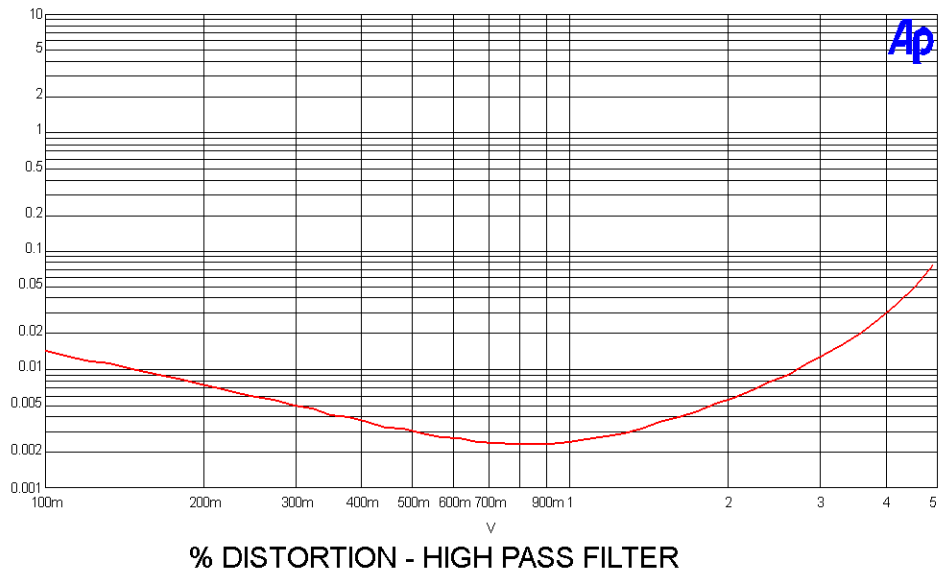
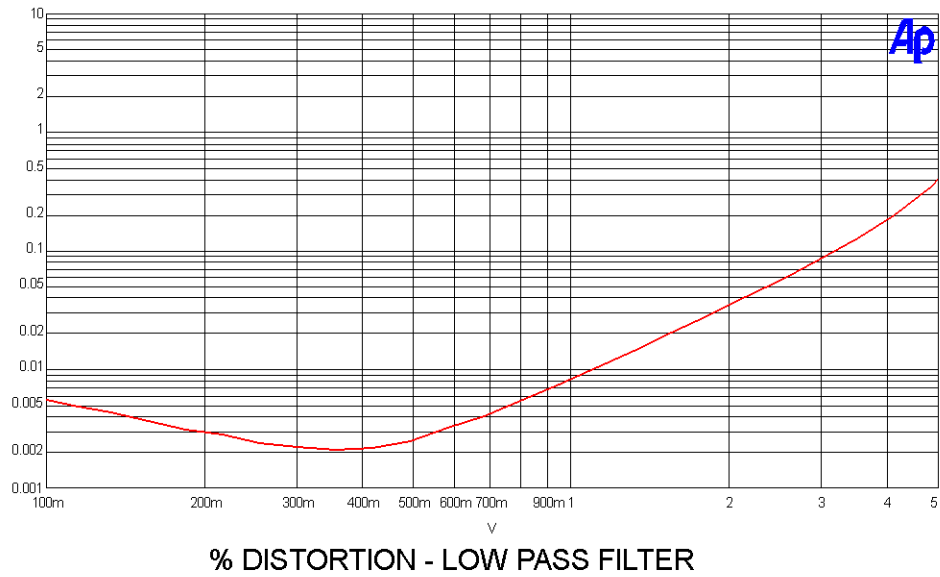
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Here you see the input pots for level adjustment feeding an input buffer which provides a low source impedance for the filters. At the top you see a series pair of low pass filters and below the high pass filters. By use of jumpers, we can select combinations of these filters to get the four different slopes for the filter. Each element has two pots to set the specific frequency of each RC pole of the filter continuously over a 6:1 range. All the values of “C” in this schematic are the same value, determined by the center frequency in the two octave range, and the “C/2” capacitors are generally at 1/2 the value of C.

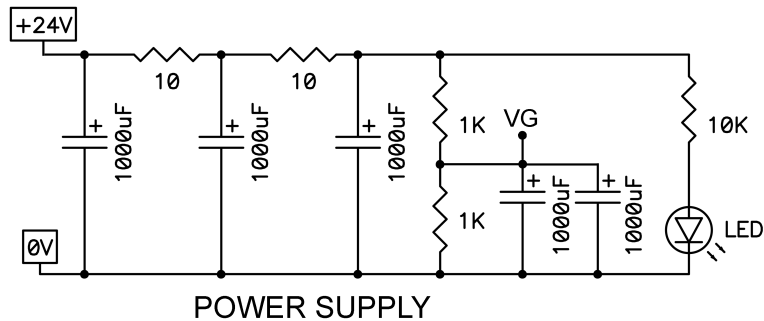
In the kit from diyAudio the Jfets used are selected for I_{dss} and matched for performance. Each is paired with a resistor (labeled BIAS) specific to that Fet for optimal performance.

The bandwidth goes from 1.5 Hz on the low pass to -1 dB at 200 KHz on the top. Nominal input impedance is 12 Kohms and output impedance is about 250 ohms.

Here are the distortion curves with all filters used in the circuits:

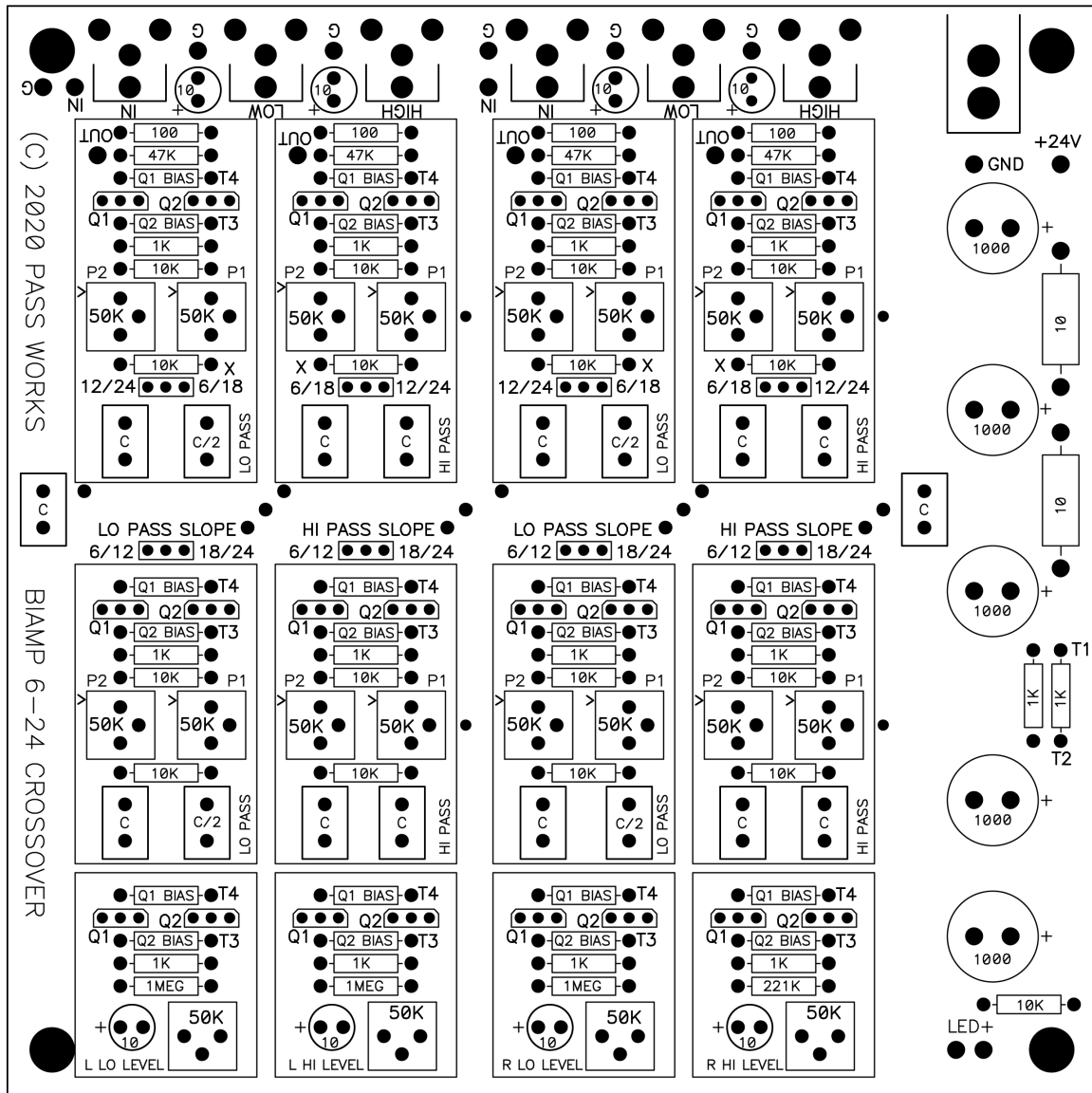


The power supply for this circuit is a filtered wall-wart at 24 volts, and here is the schematic:



The supply is single-ended, with three passive RC filters to remove the switching noise from the wall-wart supply, and a virtual ground reference is provided so that the filters will operate at the halfway point of this supply. You can use lesser values and the crossover will work, but I don't recommend much higher voltage.

Here is a picture of the circuit board, measuring 6" X 6" with mounting holes at 5.5" X 5.5".



You can see all the values of the components, except for the Fets and the resistors used to bias them individually. Also you can't see the value for C in the filters: That is determined by the center frequency that you want. Normally we will use the same value for C everywhere, with C/2 being half that value, as will be discussed shortly.

This board has been provided with pads for REAN PCB-mount RCA connectors and ordinary pads for wires. Same with the connection for 24 volt DC power, and a LED power indicator.

You can see the level and frequency adjustment pots and also the jumpers which select which filters are used for the 6/12/18/24 slope selection. You can set these wherever you like, nothing bad will happen to the circuit - no guarantee that some settings might allow you to damage your tweeter. You must remember to observe the markings for settings on all the jumpers on the board, otherwise you might not get what you want.

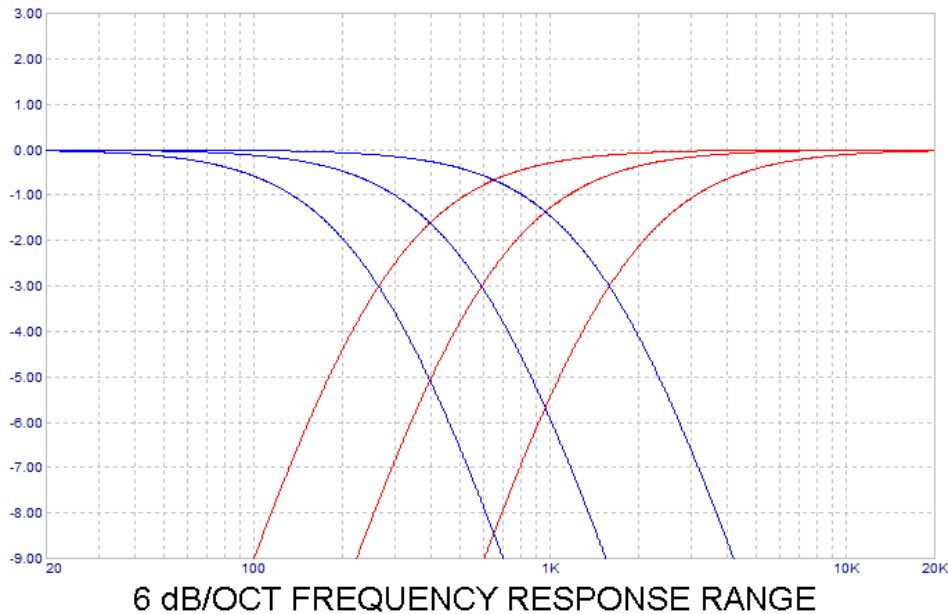
The frequency pots go higher in frequency with clockwise rotation. The center point of the resistance range is close to > mark on the board and using a small flat blade screwdriver you would point the arrow on the rotating top of each pot to this spot.

In the examples here, the value of C is 10 nF. For a higher frequency, the value of C is proportionately less, and for lower it is proportionately larger.

6 dB/octave Filters

For the case of 6 dB/octave slopes, a flat summed response will occur at the point where the two curves meet at -3 dB. To get this, we move each filter 1/2 octave closer to the center, and ideally the low frequency and high frequency drivers are wired in phase.

Ideally is the key word - perfectly summing curves have to assume that the drivers are very flat both in amplitude and phase in the region of the crossover, and with 6 dB/octave slopes there tends to be greater sensitivity to speaker response than with higher slopes.



For 6 dB/octave example, the formula for C is $C = 6000 / \text{Freq}$ where C is in nanoFarads (equals .001 microFarads). In this example C at 10 nF gives a center at 600 Hz, and the filter holds that shape from from 260 Hz to 1.6 KHz. We note that stuffing this filter on the board calls for C/2 to be half, at 5 nF, and the value we can get easily is 4.7 nF, which is plenty close enough, and easily adjusted.

Keep in mind that all these examples aren't necessarily optimal to a speaker system. The goal here is to make the filters very adjustable so as to find "the sweet spot".

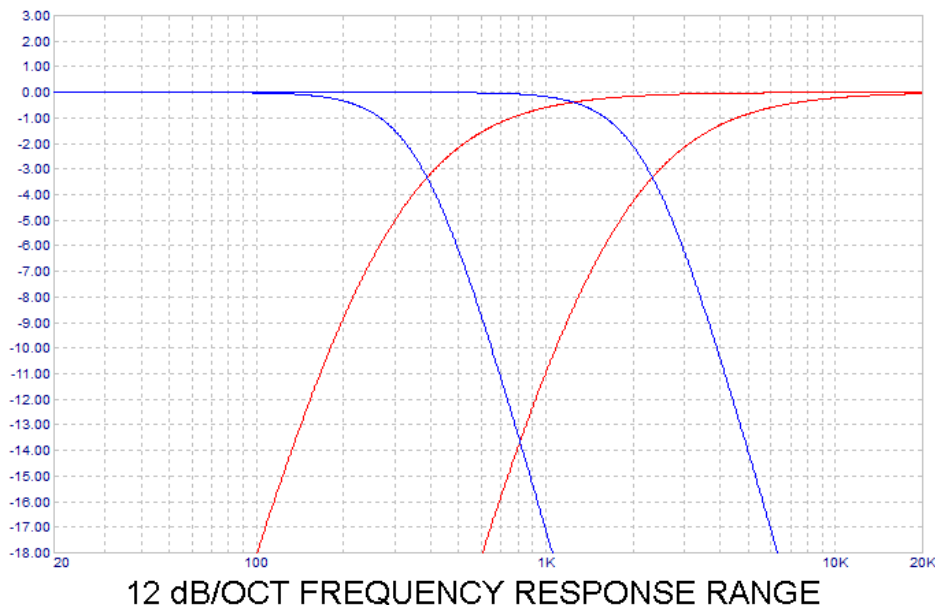
12 dB/octave Filters

Below are the curves of low pass and high pass filters for the case where both are set at 12 dB/oct. Ideally they meet at their -6 dB point, but with this filter it is necessary to adjust the potentiometers to get more symmetry. Once again, we assume perfect loudspeaker drivers, but this time the low frequency and high frequency drivers are usually operated with the electrical phase inverted from each other, for example a woofer would be wired in phase and the tweeter out of phase. To approximate the Linkwitz-Reilly with the stock values you would find yourselves tuning the high pass filter to a higher frequency than the low pass. As with the 6 dB/oct case, the formula for the approximate center frequency (stock values) is also

$$C = 6000 / \text{Freq} \quad \text{with } C \text{ in nanoFarads (nF)}$$

A note: If you are dedicated to a 2 pole L-R filter in your application, the $C/2$ values here could be the same as C , which allows the potentiometers to be at the same value, possibly giving a bit better characteristic.

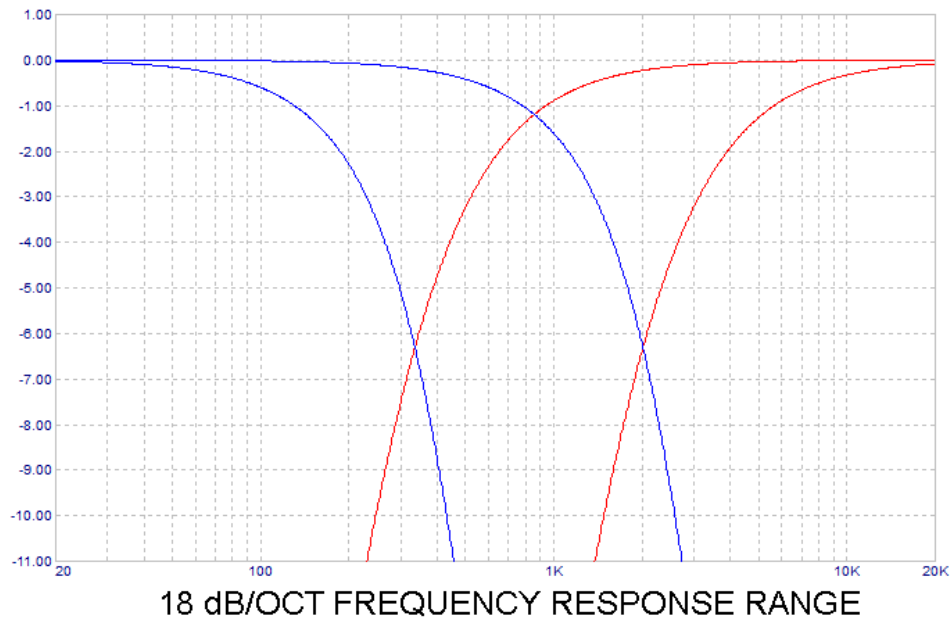
The range of the stock filters, all poles at extremes:



Like the 12 dB/oct and 24 dB/oct crossovers, the 18 dB/oct curves often want to meet around the -6 dB point, and like the 12 dB/oct filters, generally operates one driver out of phase with the other. Keep in mind to explore flipping the phase of the drivers either way when you are experimenting with this - sometimes you will be surprised.

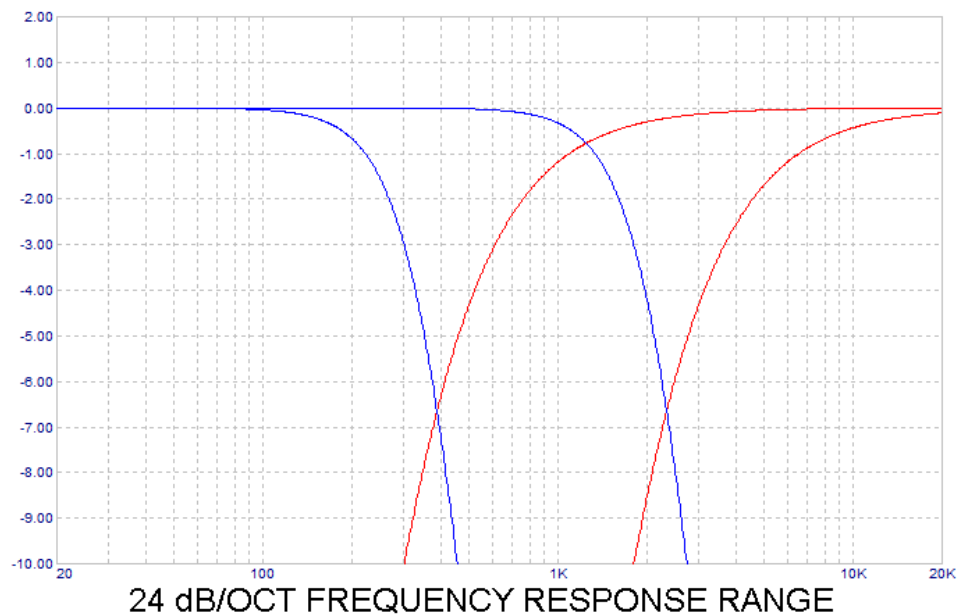
18 dB/octave Filters

Once again, we see that the formula for the stock values of the capacitors is very similar, approximately: $C = 7000 / \text{Freq}$, with C in nanoFarads (nF). Below you can see the range of the stock values at potentiometer extremes for $C = 10$ nF. With a little tweaking of potentiometers, you can get the sum of the filters flat within a 1 dB window, and that will have some small relation to the acoustic result with real loudspeaker drivers.



24 dB/octave Filters

The potentiometer extremes settings give us the following curves:



So far we have been pretty lucky - the formula for the C capacitor value of the 6, 12, and 18 dB/octave filters has been a pretty similar to $C = 7000 / \text{Freq}$, with C in nanoFarads (nF).

Well, here it is again. The 24 dB/octave case has been favored in the compromises in this design, that is to say I expect it to be the most popular launching point for crossover development, and the result is quite good. The only point to note is that the classic L-R alignment will result from the P2 potentiometers to be at around 1/2 the resistance of the

other pots. With a C = 10 nF, we see a functional L-R range from 380Hz to 1.2 KHz, and deviating from that a range from 380 to 2.4 KHz. Like the 6 dB/octave case, it expects the drivers to be wired in phase as a default.

If you are dedicated to the 24 dB/oct L-R alignment, you can extend your frequency adjustment range on this filter if you replace the high pass P2 50K pots + 10 Kohm resistor with 25K in series with 4.7K resistors.

Assembly Instructions

As of this writing, www.diyAudio.com will soon be offering the essential kit, to be followed shortly by more extensive version which contains all the parts on the pc board (the connectors for signal a power included). My prolific friend 6L6 is getting ready to do one of his famous build guides for the project.

Rest assured, that there are no hidden tricks as long as you can solder and read the print on the circuit boards. There will be a thread on the Pass Labs forum at [diyAudio](http://diyAudio.com) where the nice friendly people will help with questions and issues. Or at least sympathy ;)

Part list

1	pcb	(ESSENTIALS KIT)	BIAMP 6-24 CROSSOVER
24	JFET	(ESSENTIALS KIT)	J114 MATCHED N CHANNEL
24	RESISTOR	(ESSENTIALS KIT)	JFET BIAS RESISTORS
4	RESISTOR	100 ohms	DIGIKEY PPC100YCT-ND
14	RESISTOR	1 K ohms	DIGIKEY PPC1.00KYCT-ND
17	RESISTOR	10 KOHMS	DIGIKEY PPC10.0KYCT-ND
4	RESISTOR	47 KOHM	DIGIKEY PPC47.5KYCT-ND
4	RESISTOR	1 MEG OHMS	DIGIKEY PPC1.00MYCT-ND
2	RESISTOR	10 OHMS 3 WATT	DIGIKEY BC4940CT-ND
20	POT	50 KOHM	DIGIKEY 3386P-1-503LF
5	CAPACITOR	1000 uF 25V	DIGIKEY 493-10989-1-ND
8	CAPACITOR	10 uF 25V	DIGIKEY 493-10824-1-ND
1	HEADER	40 PIN	DIGIKEY S1011EC-40-ND
8	JUMPER		DIGIKEY S9341-ND
14	CAPACITOR "C"	(SEE TEXT)	DIGIKEY/MOUSER, WIMA FKP SERIES
4	CAPACITOR "C/2"	(SEE TEXT)	DIGIKEY/MOUSER, WIMA FKP SERIES
1	LED		DIGIKEY 160-1852-ND

Optional connectors:

6	RCA CONNECTOR (PICK COLORS)	DIGIKEY CP-1419-ND (RED) DIGIKEY CP-1420-ND (WHITE) DIGIKEY CP-1421-ND (YELLOW) DIGIKEY CP-1422-ND (BLUE) DIGIKEY CP-1423-ND (GREEN)
1	DC POWER INLET	DIGIKEY CP-037B-ND
1	DC POWER SUPPLY	DIGIKEY 237-2249-ND

Customizing Notes

The board is amenable to customization, and here are a few notes to make it easier:

With regard to supplies, you can use whatever you like, but if you exceed 24 volts you will want to check the voltage ratings of the supply filter capacitors. I don't recommend exceeding 30 volts DC regardless.

The Jfets are J113 and we want them operating at about 8 mA current in the buffers. The ones in the kit are selected along with Source resistors to give this with a matched characteristic. If you want to use something better, such as 2SK170 or LSK170, you will want to remember that they may have a different pinout. Also, there is little need to upgrade the J113's that are used as constant current sources, as they just do constant bias current.

The film capacitors in the part list are all Wima polypropylene, and you are welcome to find something better, but of course you will have to figure out how to fit them into the space.

The standard version of the crossover is very flexible and is likely to give you at least an approximation of what you want, but this involves some practical trade-offs, and so can't be everything to everyone.

Previously I have mentioned the options of altering the alignments of the filters in a manner not supported by the standard version. To possibly make that task easier, here are a couple of example MicroCap simulation circuits for the Linkwitz Riley filters showing custom values and the resulting response and phase characteristics. As of this writing you can download the Microcap 12 simulator and LTSpice for free.

If you are not prepared to spend time learning some LTSpice or MicroCap, DIY patron saint Mike Rothacher has put together a filter calculator for these circuits, available for download. Works like glue, and no rocket science involved

You can download Microcap for free from: <http://www.spectrum-soft.com/index.shtm>

You can download LTSpice from:

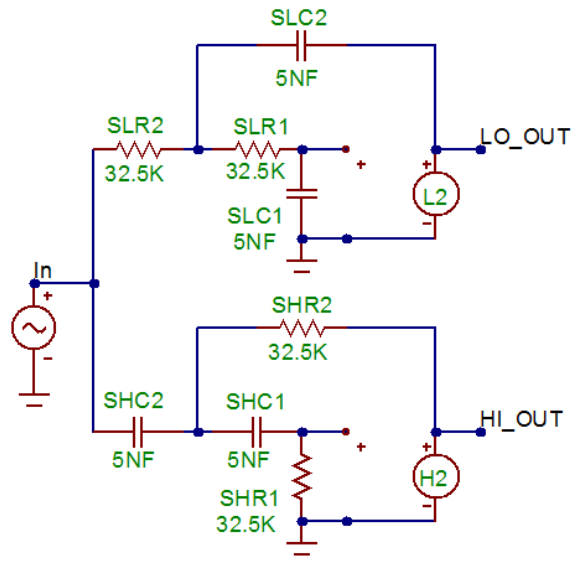
<https://www.analog.com/en/design-center/design-tools-and-calculators/ltspice-simulator.html#>

and Mike Rothacher's filter application from: <http://doublesecretlabs.com/apps/passxo>

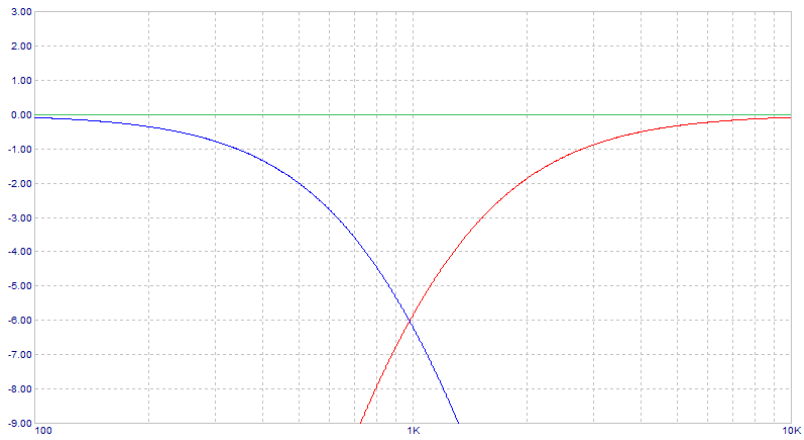
For the purpose of making customization of the board easier, on the following three pages you will see two examples of circuits from Microcap simulations that have parts designations that match the custom pc board layout labels on the third page. Since the Linkwitz-Riley alignments are very popular, I have chosen the 2 pole 12 dB/octave and the 4 pole 24 dB/octave as the examples, but of course you are limited only by your imagination.

Note in these examples that the active stages are MicroCap models for voltage controlled voltage sources (VCVS) with a gain fixed at 1 to simulate the Jfet buffers used in the crossover. You can also simulate this with op amps operated at unity gain.

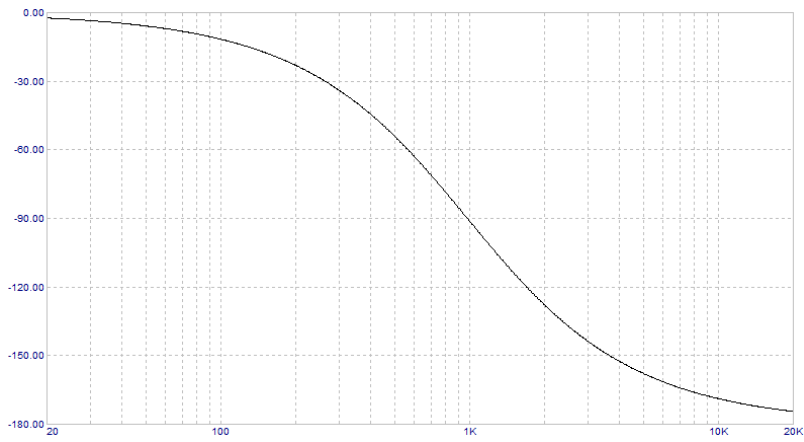
Here is the 2 pole L-R filter:



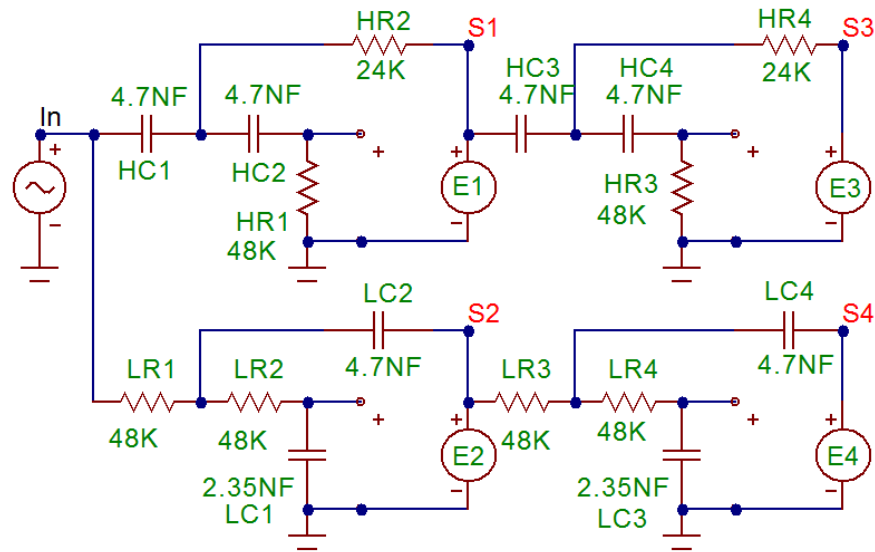
And its response curves and sum:



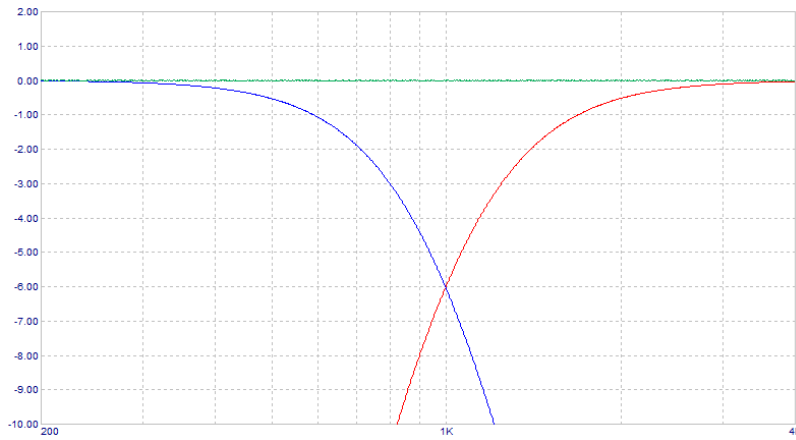
And the phase of the summed filters:



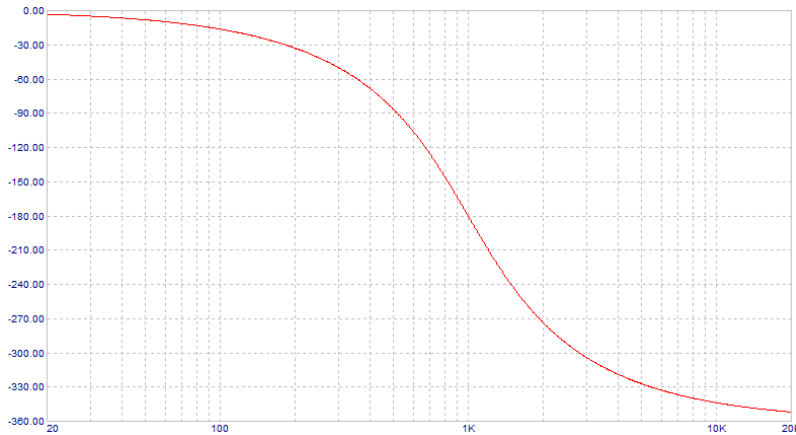
Here is the 4 pole L-R filter:



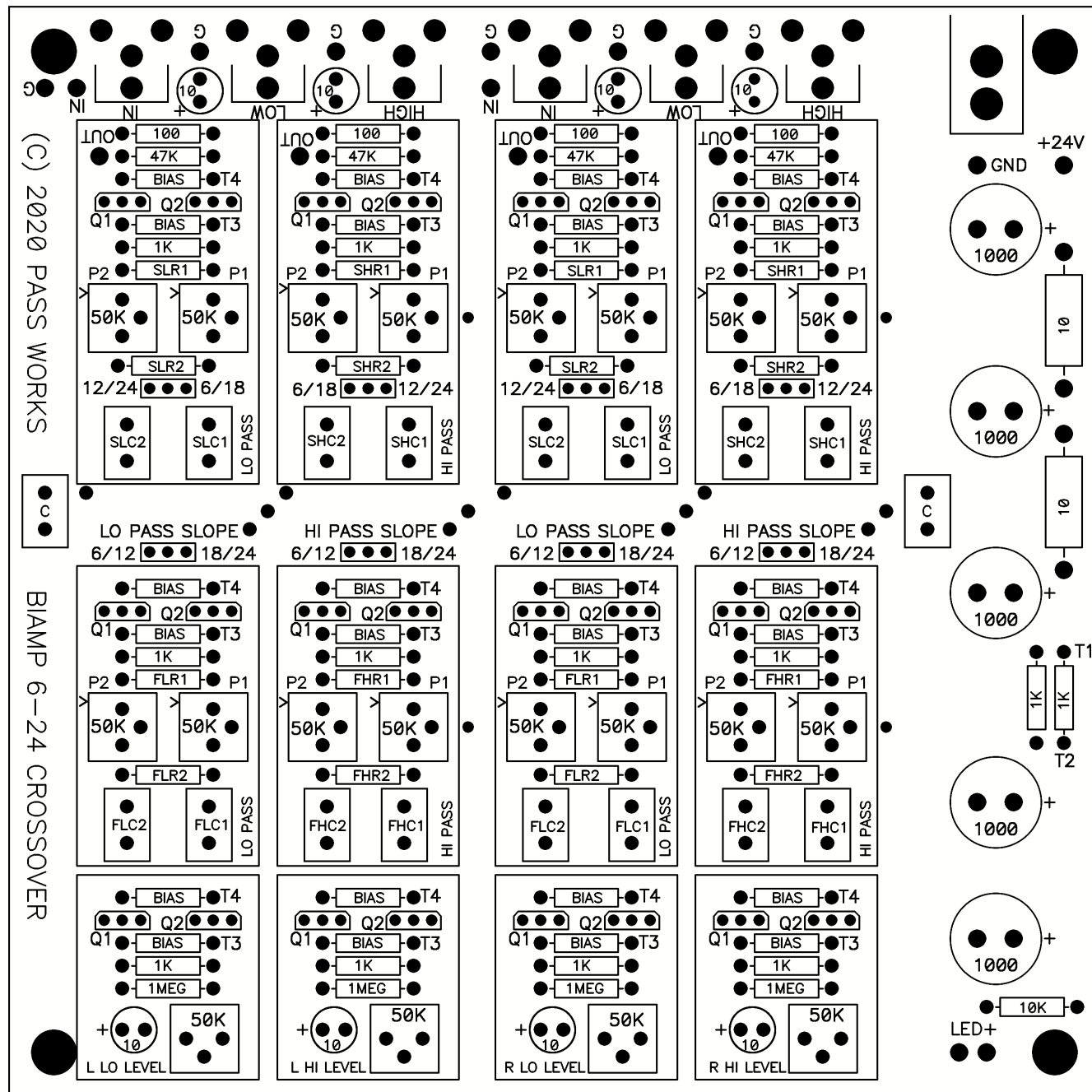
And its response curves and sum:



And the phase of the summed filters:



Here is the board layout with those custom part locations. Note that resistor values SLR1, SLR2, SHR1 and SHR2 are those resistors in series with the P1 and P2 potentiometers on each filter. In the standard version they are 10 Kohm in series with the 50 Kohm potentiometers.



Enjoy.