# Active Filter Four Version 1.0

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# 1. Introduction

This document is a guide on how to use the Active Filter Four printed circuit board. The PCB has room for a complete 2 way active cross over network for one channel, and features the options of using a first order all pass section and up to 5 equalizers of various types. The filter sections can be first through fourth order.

Two PCB's are needed for a stereo two way crossover, 4 PCBs for a stereo three way crossover and 6 PCBs for a stereo four way. You will also need a +/-15V power supply. An enclosure for the crossover unit is recommended.

In this document you will find design examples that demonstrate how and why to use the features of the board. They are not intended as recommended projects.

Before designing a filter and equalizers with this PCB, you must determine the required filter transfer function. This document assumes that you have some knowledge of crossover design and have determined the target cut off frequencies, Qs and gains of each section that are needed for the different filter sections on the PCB. If you have prototyped your system with a series of Active Filter Two boards (A.K.A. MOX), you can simply transfer the values selected there to this board.

A spreadsheet in Microsoft Excel format is available to assist in the calculation of component values.

Happy building.

# 2. Table of Contents

# 3. The Schematics

## 3.1 Input filter, Buffer and Buffered Output

The input filter has three major functions in the active filter.

1. Set the input impedance.
2. Cut of HF noise signals.
3. Boost the signal to improve the signal to noise ratio.

The input filter and buffer are made by the components around IC3A, and the opamp IC3.

IC3B provides a buffered signal output with response tailored by the user's choice of input buffer function.

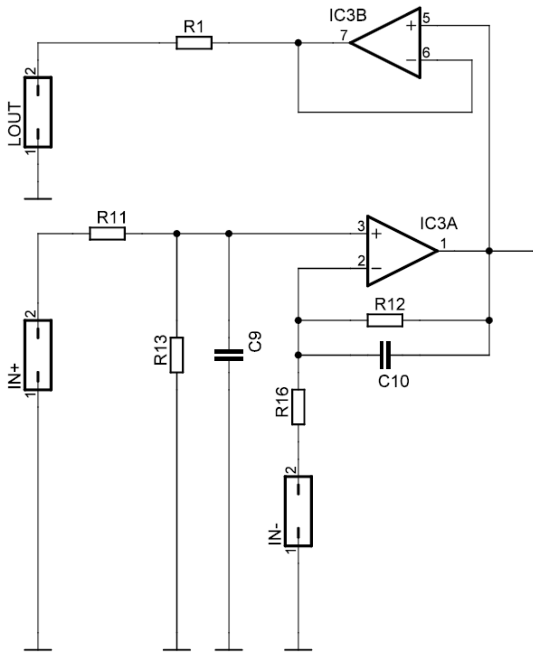



**Figure 1: Input buffer and gain**

### 3.1.1 SINGLE ENDED OPERATION

For single ended inputs, a jumper must be placed between the -In pad and the adjacent ground pad (near R12 and R16).

### 3.1.2 Low Pass Input Filter

R11 and C9 form a low pass filter that can be used to limit the bandwidth of the crossover to limit RFI susceptibility or any other purpose for which limited bandwidth is required. It can even be used to add another pole to the low pass filters if required, but obviously this is not appropriate for most tweeters.

The cutoff frequency (F3) of this filter is determined with the following equation (which assumes that a very high impedance follows) is:

$$F3 = \frac{1}{2\pi R_{11} C_9}$$

R11 in ohms, C9 in FARADS

Keeping values practical, 100 ohms and 10 nF gives an F3 around 150 KHz if R13 (Sets input impedance) is 10K.

### 3.1.3 Buffer with Gain

IC3A forms an input buffer. If you do not need any additional gain, simply jumper R12 and omit R16. Providing some gain in this stage helps the noise performance, so typically 6 db of voltage gain is used.

$$Gain(dB) = 20Log(\frac{R12 + R16}{R12})$$

If R12=R16, then the gain is 6 dB. Virtually any reasonable values will work here, but there are competing considerations. For minimum DC offset and distortion, the impedance to ground seen at both inputs should be the same. This implies that for our 6 dB gain scenario, R 13 should be twice R12 and R16. Lower resistance values will lead to lower noise, but too could be more than your preamp can handle. Perhaps this is why Linkwitz uses so many 2.37K resistors…

In practice you probably won't hear a difference between the ideal case and just using 10K for all three.

C10 should be chosen to give the same corner frequency as R11/C9. This works out to be 100 pf for 10K R12/R16. It forms a shelving high pass filter, reducing the buffer gain to unity at high frequencies.

### 3.1.4 Shelving Low Pass (Baffle Step, Dipole EQ)

As noted above, C10, R12 and R16 form a first order (6 dB/octave) shelving low pass filter. This can be used to compensate for Baffle Step, rising frequency response and dipole equalization.

There are several ways to design shelving low pass filters, one reference for the theory is: http://www.t-linespeakers.org/tech/bafflestep/bstepcompo.html, and the design process is simplified here.

First choose a value for $C_{10}$. In the following equations use the value in Farads. So, if you choose 100nf, use $1x10^{-7}$ in the equation.

$\omega_0$ is the frequency where the gain required is half the total gain required.

$A_0$ is the total gain required. Convert from dB = $10^{(\text{gain in dB/20})}$

$A_1$ is the geometric mean between no gain and $A_0 = 10^{(\text{gain in dB/40})}$

$\omega_0 = \frac{230\pi}{W_b}$ where $W_b$ is the width of the baffle in feet.

$$R16 = \frac{1}{\omega_0 C(A_0 - 1)} \sqrt{\frac{A_0^{\,2} - A_1^{\,2}}{A_1^{\,2} - 1}}$$

$$R12 = \frac{1}{\omega_0 C} \sqrt{\frac{A_0^{\,2} - A_1^{\,2}}{A_1^{\,2} - 1}}$$

The spreadsheet posted in the construction thread will enable you to input the baffle width, a value for $C_{10}$ and the desired boost and determine the values of the resistors for you. Keep greater than 10nF and adjust it to give reasonable values for R12 and R16 (Kohms, not 100K)

The board layout also gives you the option to make a non-inverting shelving high pass by placing a capacitor instead of a jumper between R16 and ground. See http://www.linkwitzlab.com/images/graphics/shlv-hpf.gif if you need this function. It would be useful for compensating a tweeter in a waveguide above the effective frequency of the waveguide.

### 3.1.5 BALANCED OPERATION

#### 3.1.6 Input Buffer with Bandwidth Limiting

To use the input buffer in the balanced mode, the signal connects to the In+ and In- pads appropriately. If R11=R12=R13=R16 you will have a unity gain buffer. Make them all 10K and C9=C10=100pf and and you'll have the same ~150KHz rolloff as the single ended version.

If you need gain (generally not as the balanced signal is 6 dB higher than single ended) you'll have to work out the details on your own.

### 3.1.7 Baffle Step Compensation

The balanced input can also be configured to provide baffle step compensation, although not all components will be on the board. Figure R12 and R16 as in the single ended case and then set R11=R12 and R13=R16. Place a capacitor identical to C10 across R11.

To provide bandwidth limiting, place a 5nF capacitor across In+ and In- and connect one end of a 100R resistor to the In+ and In- pads. The other ends of your resistors are now the input connections.

## 3.2 FILTER SECTIONS

### 3.2.1 The Sallen-Key High Pass Filter

The high pass sections on this board are as follows:

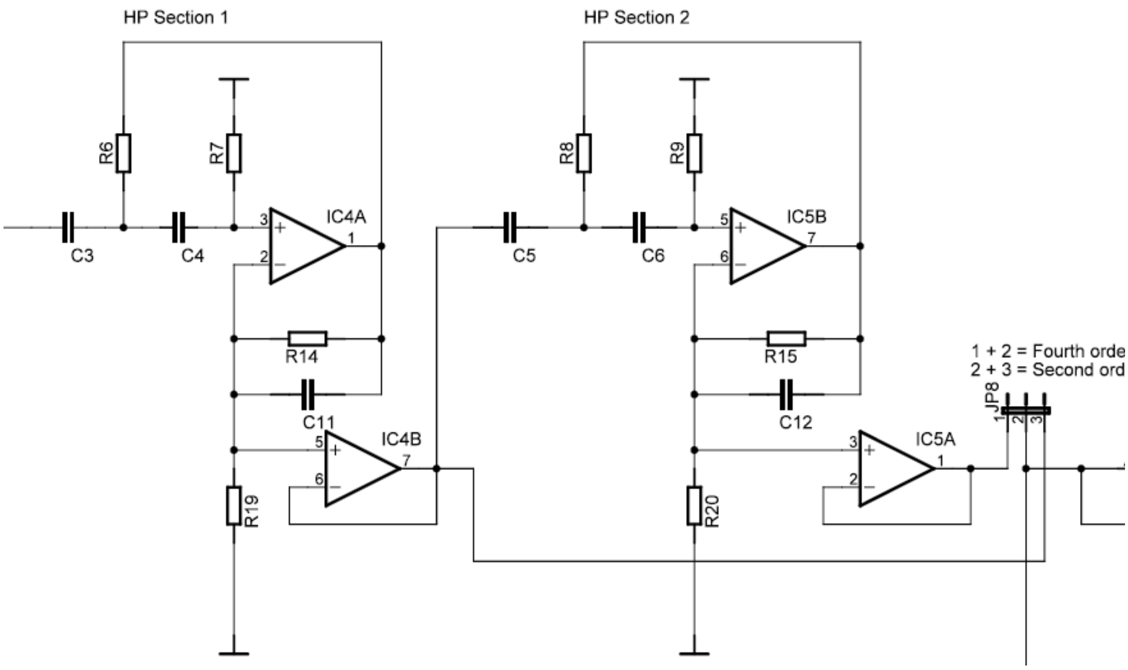



**Figure 2 – High Pass Sections**

IC4A and IC4B and components left of IC4B pin7 form a second order filter, as do IC5A and IC5B and the remaining components. Discussion of determining component values will use the component designations of the first section. Calculations for the second section are similar. There are several sources for derivation of the formulas, including the other Active Filter documents at www.delta-audio.com should you desire a more in depth understanding.

C11 and C12 provide bandwidth limiting, rolling off the filter gain to unity at high frequencies. Values of 100-220 pf are appropriate.

You should have determined the cutoff frequency ( $f_c$ ) and desired Q of each section analytically. Now choose a convenient value for C3 and C4 and a value for R14.

$$R6 = R7 = \frac{1}{2\pi f_c C}$$

$$R19 = \frac{QR_{14}}{2Q - 1}$$

You may need to run through a few iterations to get reasonable component values. Aim to keep Capacitors >10nF and Resistors under 100K.

If you need an odd order filter, determine the value of C4 as above, jumper R14 and C3 while omitting R6 and R19. This will make this section a first order filter. When combined with the 2$^{nd}$ order HP section 2, the result is a third order filter.

Jumper JP8 is provided to select whether HP section 2 is in circuit. When a jumper is placed between positions 1 and 2 both sections are included. When positions 2 and 3 are jumpered HP Section 2 is bypassed.

### 3.2.2 The Sallen-Key Low Pass Filter

The Low Pass Sections are similar to the high pass sections.

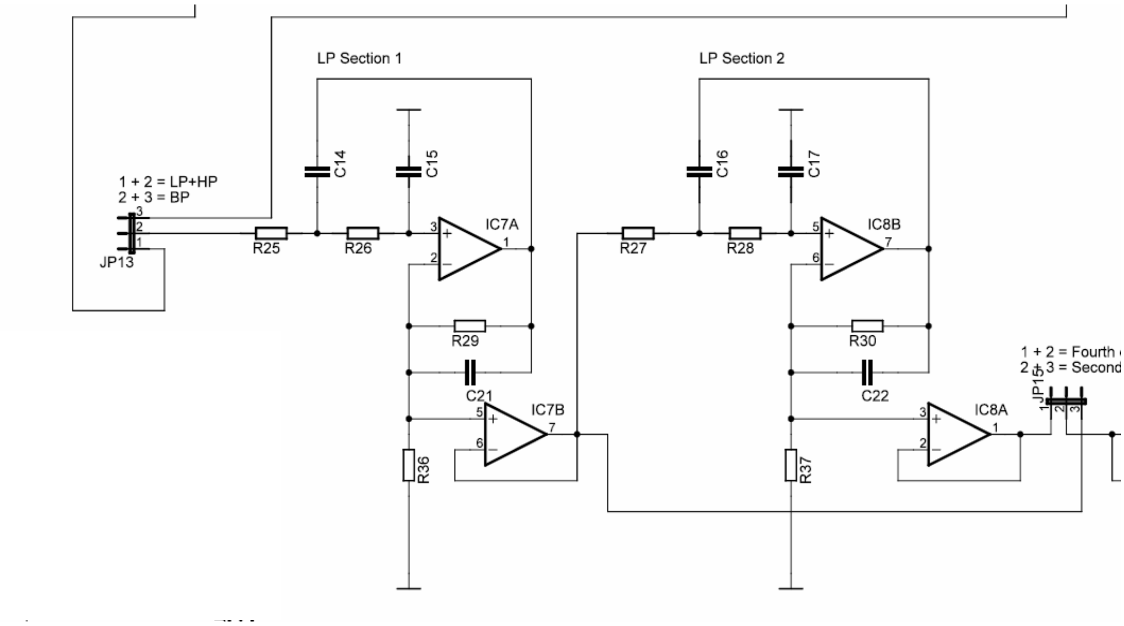



**Figure 3 – The Low Pass Sections**

The input to the low pass section is selectable to allow the board to make either a high pass and low pass filter or a bandpass filter. The jumper settings are shown in the schematic above.

Calculation of the component values is the same as for the high pass case, except component designators are changed.

You should have determined the cutoff frequency ( $f_c$ ) and desired Q of each section analytically. Now choose a convenient value for C14 and C15 and a value for R14.

$$R25 = R26 = \frac{1}{2\pi f_c C}$$

You may need to run through a few iterations to get reasonable component values. Aim to keep Capacitors >10nF and Resistors under 100K.

To set the Q of each section, use the following equation:

$$R36 = \frac{QR_{14}}{2Q - 1}$$

If you need an odd order filter, determine the value of C15 as above, jumper R25 and R29 while omitting R36 and C14. This will make this section a first order filter. When combined with the 2nd order HP section 2, the result is a third order filter.

Jumper JP15 is provided to select whether LP section 2 is in circuit. When a jumper is placed between positions 1 and 2 both sections are included. When positions 2 and 3 are jumpered LP Section 2 is bypassed.

### 3.2.3 The Band Pass Filter

A bandpass filter is formed by cascading a high and low pass filter. Usually the low pass follows the high pass to filter out the first stage's high frequency noise. Bandpass filters are useful for midrange channels in a 3 or more way design and to limit the lowest frequencies and prevent damage to a subwoofer. They can also be used with a fairly high Q high pass section to boost the low frequency response of a driver. However, this approach usually results in more group delay than using a Linkwitz Transform (also provided on this board)

Another possible use is to roll off a tweeter with an ultrasonic breakup mode, such as the Seas 27TBFC/G. At least one commercial speaker does this with a more benign tweeter breakup.

To make a bandpass filter section on this board, choose the appropriate high pass section, Jumper positions 2 and 3 of JP13 and choose the appropriate low pass section.

## 3.3 EQUALIZER SECTIONS

### 3.3.1 The Notch Filter

Again, theoretical derivation of the notch filter component values is provided at www.delta-audio.com in the active filter one documentation, so it will not be repeated here. The spreadsheet posted in the construction thread includes a notch filter calculator.

Notch filters are often used to suppress cone breakup modes in stiff cone woofers. You need to know the center frequency, Q and depth of the desired notch to design it. Determining center frequency ($F_0$) is fairly obvious, look at the frequency response curve and eyeball the center of the peak. To determine the required Q, you look for the frequencies where the response is 3 dB down from the peak. Call these frequencies $F_{lo}$ and $F_{hi}$.

$$Q = \frac{F_0}{F_{hi} - F_{lo}}$$

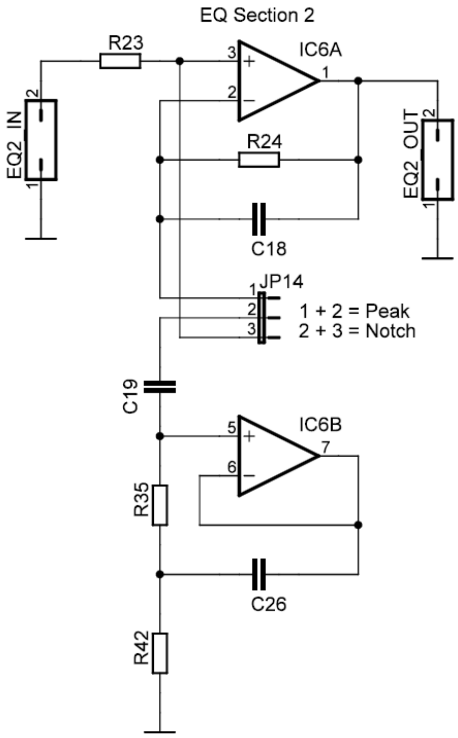



**Figure 4 – The EQ Sections**

For this filter R42=R35 and R24 is jumpered (omit C18) and jumper positions 2 and 3 of JP14. Gain in the R23 equation is V/v or $1x10^{(gain/20)}$

$$R35 = R42 = \frac{1}{4C_{19}Q\pi f_0}$$

$$R23 = \frac{1}{2C_{19}Q\pi f_0} * \frac{1 - \text{Gain}}{\text{Gain}}$$

$$C26 = 4Q^2C_{19}$$

Several iterations may be required to come up with reasonable component values. The spreadsheet in the building thread makes this relatively painless.

### 3.3.2 The Peaking Filter

A peaking filter can be used to fill a dip in frequency response. The spreadsheet can calculate the component values for you. The design equations are shown below. The Q is figured the same way as for the notch, except choose the frequencies where the response is 3 dB up from the bottom of the dip. Again the gain is $1x10^{(dB/20)}$

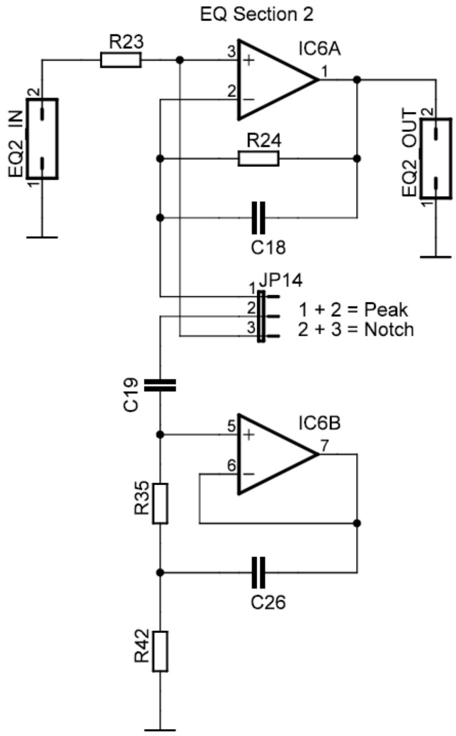



**Figure 5 – The EQ Sections**

In this case, R23 is jumpered, R35=R42 and C18 is 100-220 pf. C18 reduces the stage gain to unity at high frequencies.

The equations to solve for component values are as follows:

$$R35 = R42 = \frac{1}{4C_{19}Q\pi f_0}$$

$$R24 = \frac{\text{Gain} - 1}{2C_{19}Q\pi f_0}$$

$$C26 = 4Q^2C_{19}$$

As with the notch, several iterations may be required to come up with reasonable component values.

## 3.4 The Cauer-Elliptic Filter

In the days before computers mad it easy to design a filter to make virtually any desired transfer function, designers chose from a limited number of “standard” configurations. Names such as Bessel, Butterworth, Chebychev (in various spellings) and Linkwitz-Reilly describe a certain transfer function shape.

Cauer and Elliptic refer to the same transfer function that has a very steep initial rolloff with relatively low group delay. As with all things, TANSTAAFL[1]. The price paid for a sharp corner is some ripple in the pass band and at some frequency in the stop band the transfer function bounces up a bit before rolling off at a rate lower than the initial roll off. A Cauer filter is a standard high or low pass filter with a notch approximately an octave into the stop band.

This type of filter is useful for running tweeters near their low frequency limits and woofers up near their breakup modes. There are several designs on the web that use passive Cauer filters to achieve low crossover points needed to avoid lobing issues with an MTM using 7" woofers. These designs have a transfer function that approximates an 8th order Linkwitz-Reilly crossover initially, and keeps the bounce at least 50dB down. The advantage a Cauer design over an 8th order Linkwitz-Reilly filter is lower group delay. The overall group delay is close to the group delay of the basic filter.

The easiest way to design a Cauer filter is with a circuit simulation program. Start with a standard 4th order crossover topology with HP and LP sections followed by a notch filter.

Set the notch about an octave away from the desired crossover frequency. It needs to be fairly high Q – I ended up at 6.5, with 32.5 dB of cut.

Set one section of your HP and LP filters to give an F3 of the desired crossover frequency with a Q of 1.6. Set the second section at 15% higher (for HP) or lower (for LP) with a Q of 1.7.

Play with filter component values to adjust the Q to reach the desired response. Increasing the F3 spread of the offset sections decreases pass band ripple, but you'll have to reduce the notch attenuation to compensate, or move the notch center frequency.

This starting point will give you an initial roll off rate of around 50 dB per octave, and the bounce stays at least 55 db down. It should work pretty well for a 27TDFC at 1500 Hz. The slight bump in low end the high pass section takes care of the slight droop in the tweeter response and the high end rise compensates for some of the top end roll off. There is some rise in the low pass section's bottom end, so less baffle step compensation is required.

To set this topology up on the board:

1. Configure your input section as desired

[1] There Ain't No Such Thing As A Free Lunch

2. Assemble the filter and EQ sections
3. Jumper JP13 positions 1 and 2
4. Connect JP8 position 1 to the input of EQ2 (pin 2)
5. Connect the output of EQ2 (pin 2) to JP8 pin 2
6. Connect JP15 position 1 to the input of EQ1 (pin 2)
7. Connect the output of EQ1 (pin 2) to JP15 pin 2
8. Optionally, if you do not need to adjust the gain of the low pass section you can take the output from EQ1's output, since it is buffered.

## 3.5 The Linkwitz Transform

The board includes provisions for a Linkwitz transform, also known as a biquad filter. This is another shelving filter, with a second order slope. It can be configured as either high or low pass.

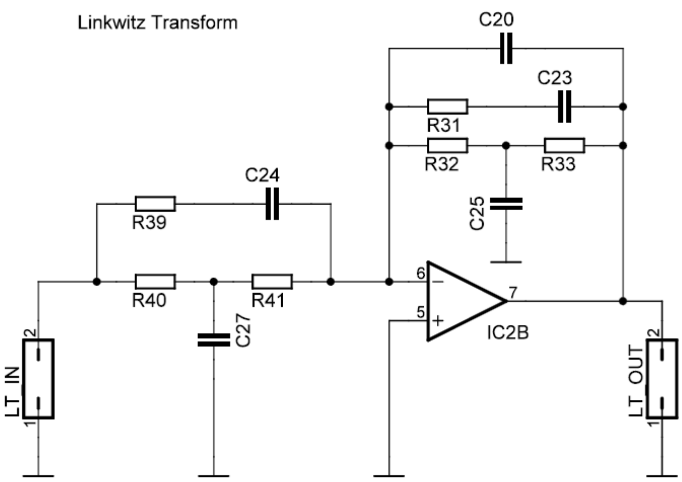



**Figure 5 – THE LINKWITZ TRANSFORM**

The most common audio use of this filter is to extend the low frequency response of a sealed subwoofer. In this application the filter effectively cancels the high pass filter of the subwoofer's response and substitutes another lower one. Be careful that you don't ask too much of the woofer. Using this EQ can tax the woofer's power handling and excursion capability.

One way to see if your driver is up to the task is to model your proposed system in WinISD and apply a Linkwitz Transform to get the desired response. Display the driver excursion, and increase the signal level until the driver reaches Xmax at your target F3. Check the Amplifier Apparent Load to see if you are exceeding the driver's limits. If not, check the SPL display to see if it will be loud enough for your goal.

There's no replacement for displacement. If you want loud and low, be prepared to buy large drivers with lots of Xmax and amplifiers with plenty of power. For example, ignoring room gain, a single Adire Maelstrom can produce 126 dB at 100 Hz, but "only" 103 dB at 20 Hz without exceeding Xmax.

If you run out of excursion before you run out of power handling, decrease the box size. Similarly, if you run out of power handling before you run out of excursion increase the box size. It is the same old tradeoff – you can have extension, small box size, and efficiency. Pick any two. Keep in mind that usually extremely low frequency signals in typical program material are of short duration so you can exceed the driver's thermal limits somewhat. Just be sure that your amp can actually produce the power required.

To accurately implement a Linkwitz transform, you should measure the actual F3 and Q of your subwoofer. Often they are not quite what modeling programs predict. A spreadsheet to calculate component values is on Linkwitz site at http://www.linkwitzlab.com/pz-eql.xls and a version useful for just inputting your driver's Theile-Small parameters and box volume is here: http://sound.westhost.com/download.htm#lxfrm.

### 3.5.1 Shelving High Pass

Other uses of this circuit include a shelving high pass circuit to compensate for a tweeter's second order roll off below resonance. For example, to maintain a fourth order acoustic roll off, the filter would need to be fourth order above the tweeter's resonance and second order below. This filter section combined with a standard second order section would accomplish this. Linkwitz did this at one point then apparently decided the effect of just using a standard crossover was not audible enough to warrant the complexity. See his site for further reading.

### 3.5.2 Dipole Equalization

The Linkwitz Transform section of the board can also be used for Dipole Eqalization in the Bridged-T configuration. Jumper C23 and Omit R32, R33 and C25. See http://www.linkwitzlab.com/filters.htm for design information.

With a little creativity with jumpers, this section can also be used to create the dipole EQ section depicted in "The Edge," a free program from www.tolvan.com. This handy program that allows you to input your baffle dimensions and driver compliment/placement and computes both dipole losses and diffraction effects. It then allows you to enter parameters for compensation networks and displays the resulting frequency response. Its predictions are quite accurate.

To create the filter used in The Edge, omit R39, R31, C27, C24 and C25. Jumper R40 and place a jumper from the C23 end of R31 to the R32/33 end of C25. Use the board spaces labeled R41 and R32 for The Edge's R1. The R33 spot is used for R2, and C23 for C. C20 is our now familiar 100-220 pf bandwidth limiter.

**Special note about this filter section:**

This filter section is **inverting**. You will have to account for this elsewhere in the system. The simplest solution is reverse the connection to the driver. The electrons don't care which terminal is positive, the signal is AC after all. Driver terminal marking is just a convention to enable correct phasing. If the voltage applied to the positive terminal is positive compared to the negative terminal, the driver moves forward. If negative, the driver moves backwards (except some older JBL Pro Drivers.) If you invert the signal in the Linkwitz Transform (a positive going signal in becomes a negative going signal out) and then invert it again with your driver connection (apply a negative going signal to the negative driver terminal, the cone moves forward.

If you insist on inverting the signal again in the crossover to allow you to connect the red terminal on your amp to the red terminal on the driver, you can either build an inverting buffer on perf-board, or if you don't use the all pass section it can be configured as an inverting buffer without too much difficulty. We will leave it to the reader to determine how to do this.

## 3.6 The All Pass Filter

The board provides a first order all pass section to allow for phase correction. This performs a function similar to a delay or sloped or offset baffles. Some claim it is not the time difference but the rapidly changing phase through the crossover region that impacts the sound. Time delay vs. phase correction arguments often get heated using the same words to describe different phenomena. Although perhaps a bit overstated, Linkwitz describes an active crossover that does not include phase correction as "marginally useful."

The goal of this circuit is to account for the distance between the driver's acoustic centers and the phase difference between the drivers caused by both the crossover filters and the drivers' own inherent phase shift.

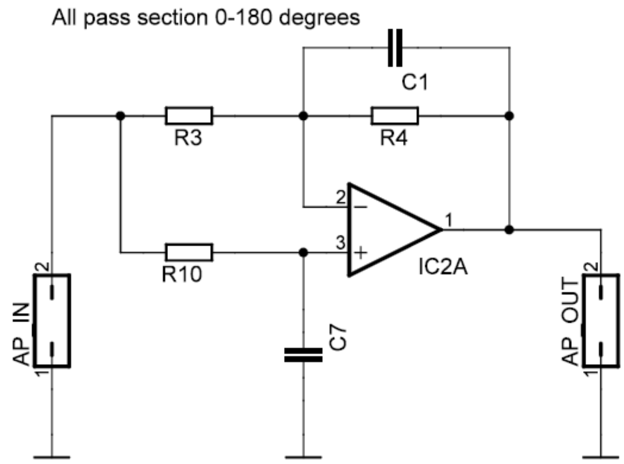



**Figure 6 – THE ALL PASS FILTER SECTION**

In this circuit, R3=R4 and can be any value consistent with noise and impedance matching concerns as mentioned in the discussion of the input buffer. Using the same or similar value as used for R10 is reasonable and helps minimize output DC offset. All resistors should be 10K ohms or less. C1 is a bandwidth limiter and should be in the range of 100-220 pf.

Even Linkwitz gives up on the math after a while and says that final values for exact phase compensation should be determined experimentally. A good starting point is compensating for the difference in the measured acoustic centers of the drivers with their crossovers. Don't have this? The distance between the voice coils is usually fairly close. Divide this distance by the speed of sound to determine the time needed.

The phase shift and group delay are determined by R10 and C7. The equations are:

$$t = \frac{2R_{10}C_7}{1 + (2\pi R_{10}C_7 f)^2}$$

$$f_0 = \frac{1}{2\pi R_{10}C_7}$$

Choose a value of C and R to give you the required delay at the crossover frequency with an eye on $f_0$. To ensure a fairly consistent delay through the crossover region, $f_0$ should be higher than the crossover frequency. In various writings Linkwitz has used "much greater than" and ">>" to describe how much greater. In the example on his filter page, however, $f_0$ is not quite twice the crossover frequency. Choosing R10 and C7 so that $f_0$ approaches an octave higher than the crossover frequency is probably sufficient. Note that you may not be able to get the required delay with a single section.

The spreadsheet posted in the construction thread allows input of R10 and C7 and displays the delay at various frequencies along with $f_0$.

## 3.7 Output Buffers and Gain Adjust

The output buffers provide a low impedance source to drive your amplifier through reasonable interconnects. A potentiometer provides a means of adjusting the overall signal level for each filter section.

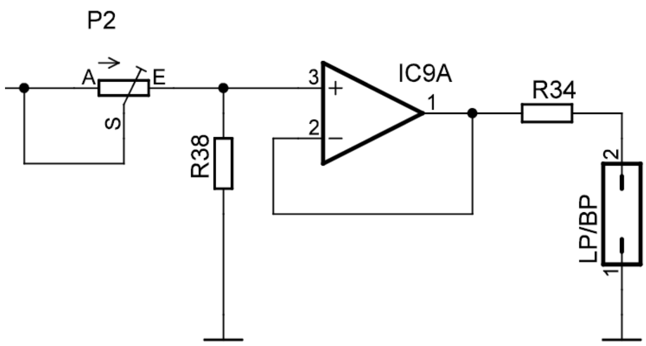



**Figure 7 – THE OUTPUT BUFFER**

R18 and R34 should be 100 ohms, and provide stability in the case of reactive interconnects.

P1 / R21 and P2 / R38 form voltage dividers used to adjust the signal level of the section. Any reasonable values can be used. If P1/P2 are set at 20K and R18/R38 are 10K, the attenuation range is 0-9.8 dB, which should be suitable for most applications. Potentiometers should be multi-turn varieties for ease of use accurately setting the level.

If no attenuation is desired, such as a two way woofer section when the tweeter is more sensitive, omit R38 (R18) and jumper pins 1 and 3 of the potentiometer.

# 4. Design Examples

The purpose of the following examples is to show how to design each section of the filter board and integrate them into a system. The examples are just a run through preliminary design.

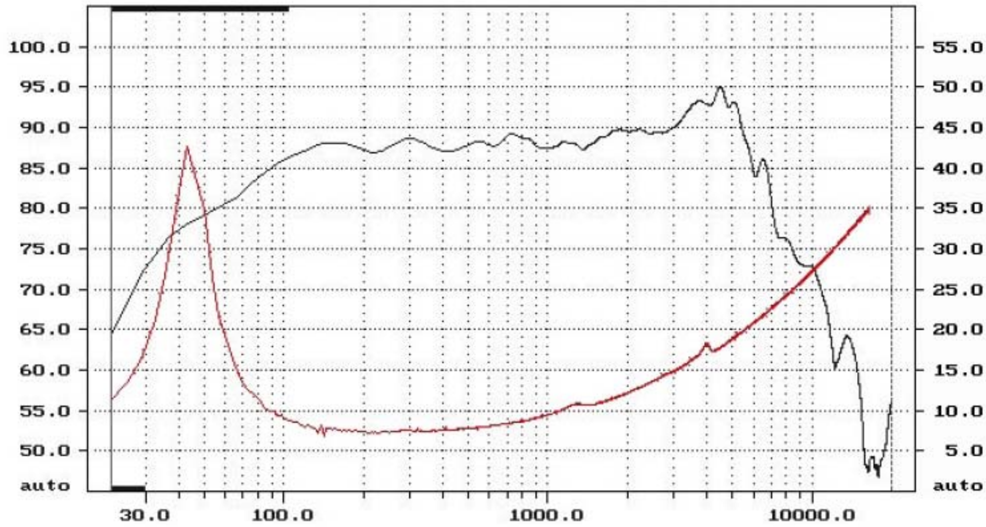
For best system performance, measurement of your drivers in your baffles is required. With response measurements in hand you can then design a filter and measure the resultant system response to determine if changes are required.

## 4.1 Two way

To begin a filter design you must know the transfer function that you need based on measurement of your drivers in your enclosures. For this example we will use the published data for a project done a while ago, for which measured data is no longer available.

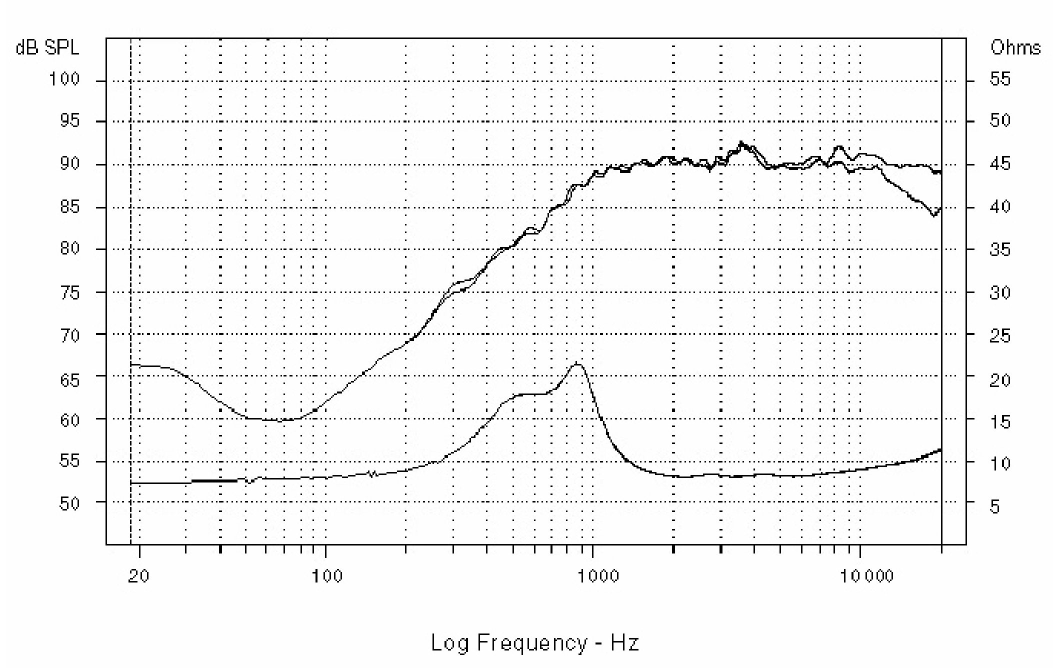
The drivers:

**Woofer – Focal 6W4254, formerly used in the Utopia series.**




Woofer Frequency response

**Tweeter – Focal TC120-TD5**




**Tweeter Frequency response**

## The Enclosure

The enclosure is a ported box, .8 cubic feet tuned to 42 Hz. A rough cut was made on phase alignment by tilting the woofer portion of the baffle, making the project look something like the JM Lab Micro Utopia, which was the inspiration for this project.

Looking at the frequency responses, and based on JMLab's crossover frequencies, I decided to use a fourth order Linkwitz-Reilly electrical filter at 2500 Hz. The measured response was surprisingly close to the published numbers, with a bit of a diffraction dip at 1100 Hz.

You can see that the woofer needs a notch filter to take care of the rising response at 5KHz.

The tweeter also has a slight bump in its response centered around 3500 Hz that needs to be addressed. Measurements will determine whether the tilted front panel compensates for woofer-tweeter phase differences adequately.

The baffle is 10" wide and has 1.25" roundovers on the upper three sides. Simulating the baffle in The Edge (www.tolvan.com) using a width of 9" matched measured response. The speakers will be positioned well out in the room, so a full 6dB of baffle step compensation is needed.

To begin the filter design, we're working with a single ended source capable of driving anything over 600 ohms to 2V peak to peak. Don't forget to jumper the –in connection to the ground pad right next to it.

**Input Buffer and Baffle Step Compensation.**

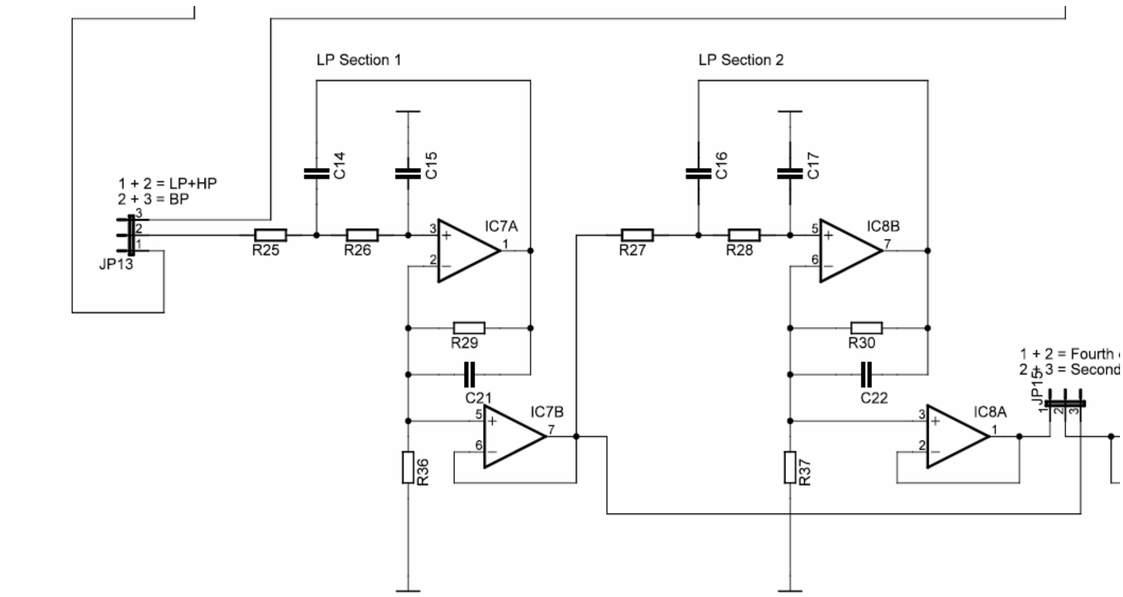
The input buffer will be used to provide baffle step compensation. To limit the bandwidth and susceptibility to RFI, we'll use a 150 KHz low pass filter on the input, setting R11 at 100 ohms and C9 at 10 nf. We'll use 10K for R13 setting the input impedance close enough to 10K for practical purposes. For the absolute minimum DC offset, Set R13 equal to the parallel equivalent of R12 and R16. Practically, I have measured only 4 mV of offset using 10K for R13 when the opamp was an NE5532.

Unlike passive baffle step compensation, you don't loose woofer sensitivity when you go active. You get a boost in the low end, rather than cutting the midband. This means that you have to be careful about power handling and excursion limits, even if midband response should be within limits. When modeling the woofer's output maximums in WinISD for example, run the signal up to maximum cone excursion and then subtract the amount of baffle step compensation you have from the midband output to find the maximum SPL.

Plugging our 9" (23 cm) baffle width into the active filter design spreadsheet gives us 4518 for R12 and 4596 for R16. Use the nearest standard value – 4.95K. We now have an input buffer that has 6 db of baffle step compensation centered at 500 Hz. and a 150 KHz low pass filter.

**Fourth Order Linkwitz-Reilly High and Low Pass Filters**

There are two ways to achieve a fourth order Linkwitz-Reilly electrical transfer function with this board. The first is the form you are most likely to have seen – the unity gain Sallen Key Topology.




Using the low pass section as an example, simply jumper R29 and R30, omit R36, R37, C21 and C22.

Rather than reinvent the wheel, I just use the calculator that Rod Elliot posted on his website, www.sound.westhost.com. Then C15 and C17 are C, while C14 and C16 are 2C. Using 9.1K for R, C of 5 nF gives a crossover frequency of 2473 Hz, close enough to our target. For this one I used 5 and 10 nF polystyrene caps from Mouser, measuring and picking the closest ones. They are cheap enough you can buy 4 times as many as you'll need to use. You'll need that many since they have a 10% tolerance. They don't exactly fit the layout of this board, but should fit without too much difficulty.

One trouble with this approach is the difficulty finding tight tolerance capacitors in suitable values. Another is that you now have the buffer stages that are not required.

The second way to get a fourth order Linkwitz-Reilly transfer function is to use the topology as presented and set each section to a Q of 0.7. See www.linkwitzlabs.com/filters for the Q requirements if you want to use other than a fourth order electrical filter.

Back over to the spreadsheet we go, plugging in 2500 Hz and Q = .7 and your choice of capacitor value. When choosing the capacitor, try to keep it over 10 nF and the resistor values between 1K and 100K. For the sake of argument, let's choose 22 nF for C14-17, which gives 2894 ohms for R25-28. Practically, you can use any value between 10 and 47 nf, so you can use whatever you can find in 2% tolerance or better that fits the 4.5 mm x 7.5 mm 5 mm pin spacing footprint.

Now on the spreadsheet's Q tab, dial in .7 and we come up with NNNN for R29 and 30, YYYY for R36 and R37. You'll notice that with anything other than Q=.5 you have some gain. If the following stage can accept the signal at the new level, you would not need the interstage buffer. On this board the output of each stage is taken from the negative input, where the feedback network has attenuated the signal to unity gain. A buffer is used between stages to eliminate the possibility of the following stage loading the filter and impacting the response.

These calculated resistor values work for the high pass sections as long as you use the same capacitor value. The only reason to use the same value is convenience. If you have a few of some special cap that you want to use in the high pass section, there is no reason that the high and low pass sections must use identical component values.

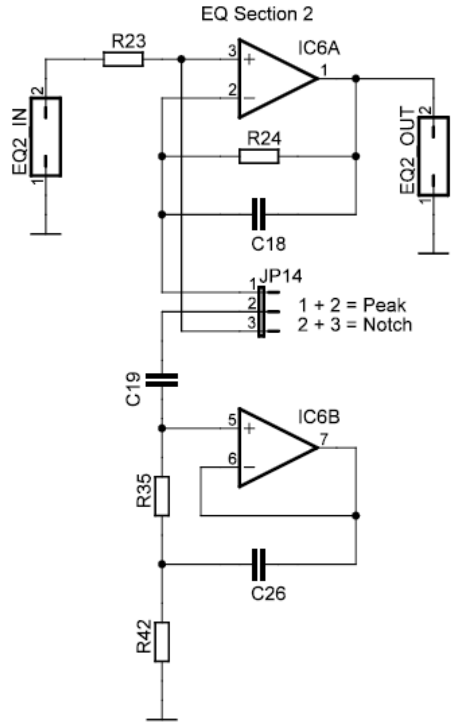
**Notch Filter Design**

In the case of the woofer notch, we make a connection from JP15 pin 1 (low pass output) and the input (pin 2) of the EQ section we choose. The output of the notch returns to JP15 at Pin 2 if you need to pad down the woofer output. If you will run the woofer without attenuation, you can simply use the output buffer of the notch filter as your output to your amp.

Look at the woofer frequency response peak. The 8 dB peak is centered around 4700 Hz, with 3 dB down points of 3600 and 5200. This required some eyeball smoothing of the curve, but you use the frequencies where the response is 3 dB below the peak to determine the required notch Q.

Using EQ section 2 as an example, plug the numbers into the notch filter design tab along with your choice of C19. The bigger you make C19, the bigger C26 must be. Try to keep C19 at least 5 nf, and have a practical value for C26. You may find that you need to add another cap on the bottom of the board to reach the value that you need. You'll also want to keep an eye on R23, to ensure that it is a practical value.

In my crossover I used 4.7 nF for C19, which gives a 162 nF target for C26. 150 nF is a standard value, combined with a 10 nF on the bottom of the board, got me right on, since I found two 150 nF caps that measured 152 nf.




Jumper pins 1 and 2 of JP14 for the notch. You will also jumper R24 and omit C18.

The tweeter notch is similar. In this case, extend the basic shape of the peak to find the 3 dB down points. The center is 3500, the 3 dB down points 2800 and 4700, resulting in a Q of 1.8.

I used a 2.2nF cap for C19 and got 30 nF for C26, R 42 and R35 are 5610, R23 is 5610. I bought a bunch of 33 nF 10% caps and found a pair that were on the low end of the tolerance.

Midband sensitivities of the drivers are close, I expected to need ~2 dB of padding on the tweeter section to match levels. Instead, I used the extra tweeter sensitivity to partially fill in the hole at 1,100 Hz mentioned earlier. This results in the acoustic crossover being a bit lower than the electrical crossover. Measurements showed I got lucky, and don't need to add any phase compensation. That will be covered in the next section.

Overall, I am satisfied with the system. If I did it again, I'd add a 3 db notch around 850 Hz to bring the 750-1000 Hz range down a bit since on reflection, it seems to be a diffraction peak there rather than a hole at 1,100 Hz.

I built a similar system using Fountek JP-3 tweeters. The difference is subtle, with neither tweeter having a clear advantage. Each has sounds more realistic on certain instruments than the other. After reading Zaph's test results, I doubt I'll go ribbon again.

## 4.2 Three Way Dipole

This example is hypothetical three way using a Seas 27TDFC tweeter in a waveguide (as shown on www.zaphaudio.com), and a pair of Dayton RS225s in a dipole baffle similar to Linkwitz's Phoenix sitting atop a JBL 2254 woofer/subwoofer. The design is based on published and calculated data – no measurements of the proposed system have been made. It should be close, but is in no way an optimized system.

The waveguide increases the tweeter sensitivity in the low end, enabling the use of a very low crossover. The arbitrary crossover target is 1500 Hz, using a Cauer-Elliptic crossover that approximates an $8^{th}$ order Linkwitz-Reilly initial rolloff to limit the tweeter stress. We will also need to compensate for the frequency response of the tweeter in the waveguide. A shelving high pass filter should work well.

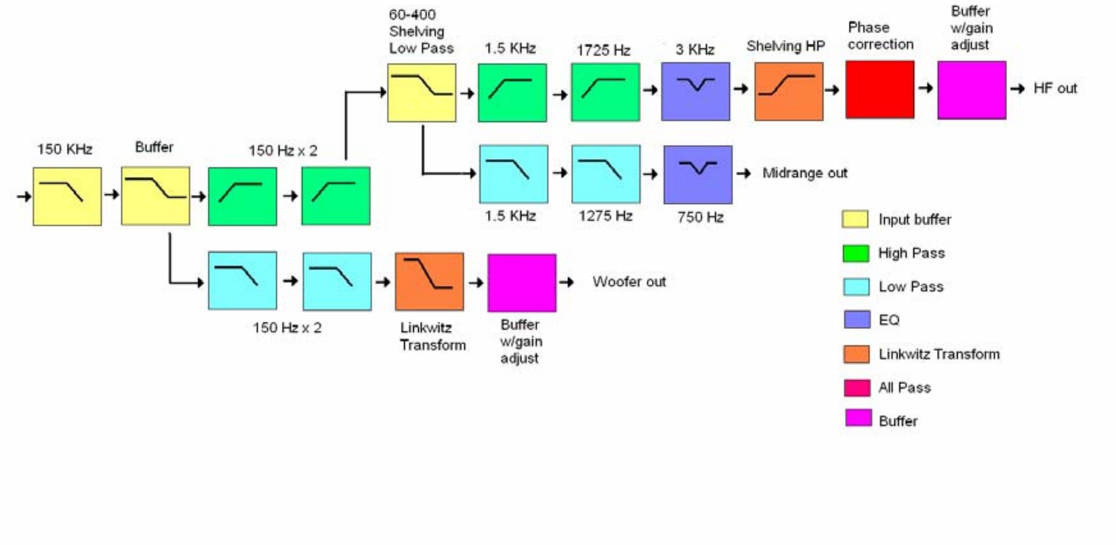
The RS225s will be rolled off hard, well below where breakup modes are an issue, so no separate notch filter is required there. At the low end of their range, we have the dipole rolloff to deal with, requiring at least one shelving low pass filter.

The 2245Hs will be in a sealed 8 cubic foot box, using a Linkwitz transform to extend their response to 15 Hz with a system Q of .5. The crossover to the dipoles will be at 150 Hz, to limit the demands on the RS225s. To dipole experts, the choice to cross over to a sealed sub at such a high frequency borders on sacrilegious. Why not use a couple of Peerless CSX-12s per side and get the real dipole advantage? There are two main reasons for this choice. First, I chose this for simplicity. We will be able to demonstrate many of the possibilities this board offers without overwhelming the less experienced reader. Secondly, I happen to have the drivers on hand, want to try a waveguide and four CSXs don't fit in my budget at the moment.

We will use two boards for this crossover, making available two all pass sections, two Linkwitz transform sections, four notch/peak EQ sections and two shelving low pass sections in addition to the basic high and low pass filters.

There are two approaches to filter design in 3+ way filters. The tweeter needs a high pass filter, the midrange a bandpass filter and the woofer a low pass filter. One way is to simply feed each filter the same input and have the midrange high pass section feed into its low pass, and the boards provide for this with a jumper. The second approach which will be used here is to start with the woofer to mid crossover and feed the high pass signal to the midrange-tweeter crossover. This means that the tweeter signal passes through additional opamps, but has the advantage of maintaining the midrange-tweeter phase relationship. Some argue that this is more important than any degradation of the signal by passing through additional opamps. Linkwitz uses this arrangement, so that's enough evidence for me.

A block diagram of the filter:



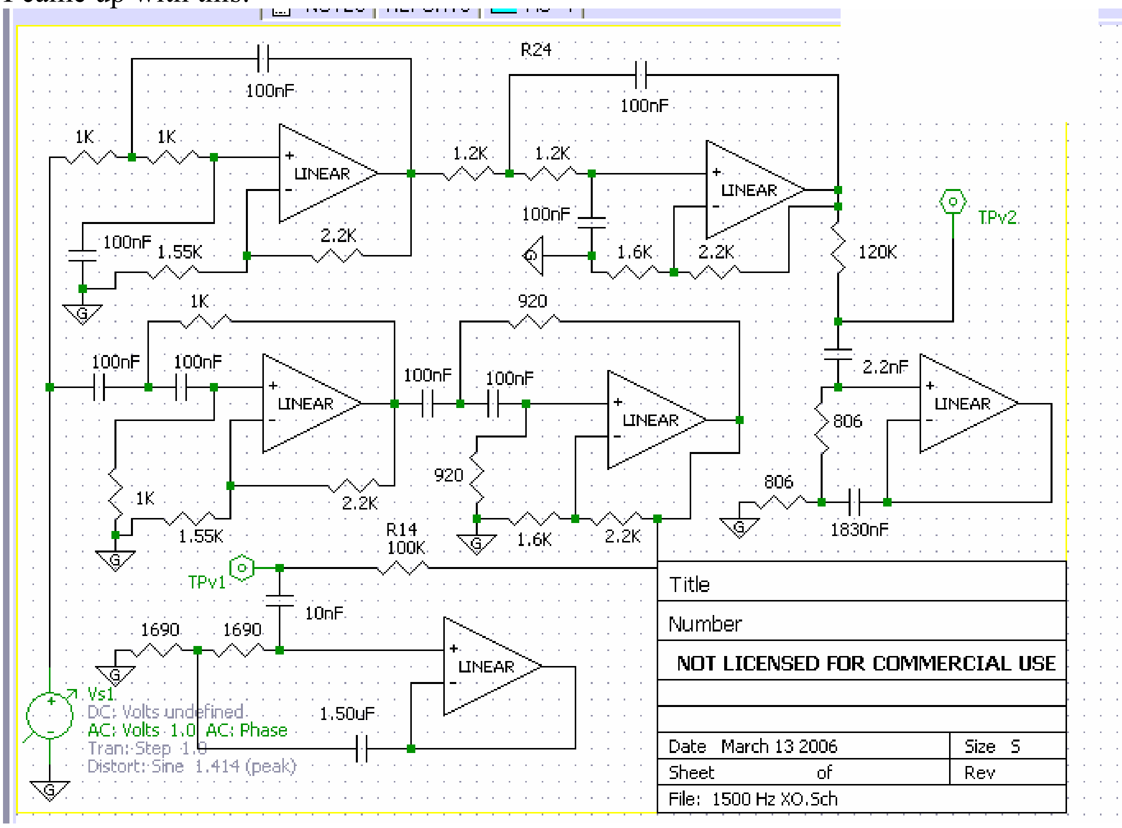

We'll start detail design with the midrange to tweeter crossover. Refer to the section earlier for quick and dirty Cauer-elliptic filter design. We need two high and two low pass sections, and a notch filter for each.

For the high pass filter we need one section with a cutoff frequency of 1500 Hz and Q of 1.7 and a second with a cutoff frequency of 1725 Hz, and a Q of 1.6. Now add the notch filter, centered at 3KHz Q=6.5 and total cut of 34 dB.

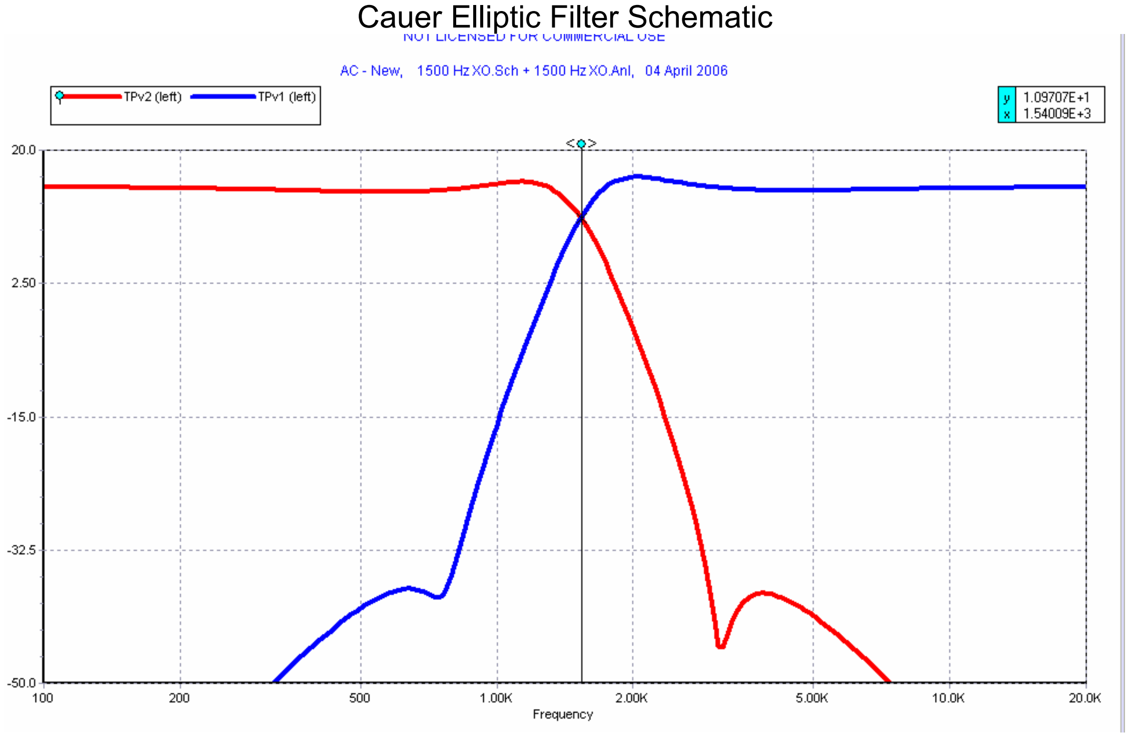
For the low pass side there is one section at 1500 Hz, Q=1.7, one at 1275 Hz Q =1.6 and a 34 dB Q=6.5 notch at 750 Hz. Go to the spreadsheet to determine component values.

Plug all of this into your handy dandy circuit simulator and check the response. Since the tweeter has a slight droop on the top end, we'll increase the notch depth to both flatten response in the crossover region and provide a couple of dB increase in upper end response. Doing the same on the woofer side will result in a bottom end rise, which will come in handy when we address baffle loss compensation. Now adjust component values to the nearest standard values and check that the response hasn't changed significantly.

I came up with this:



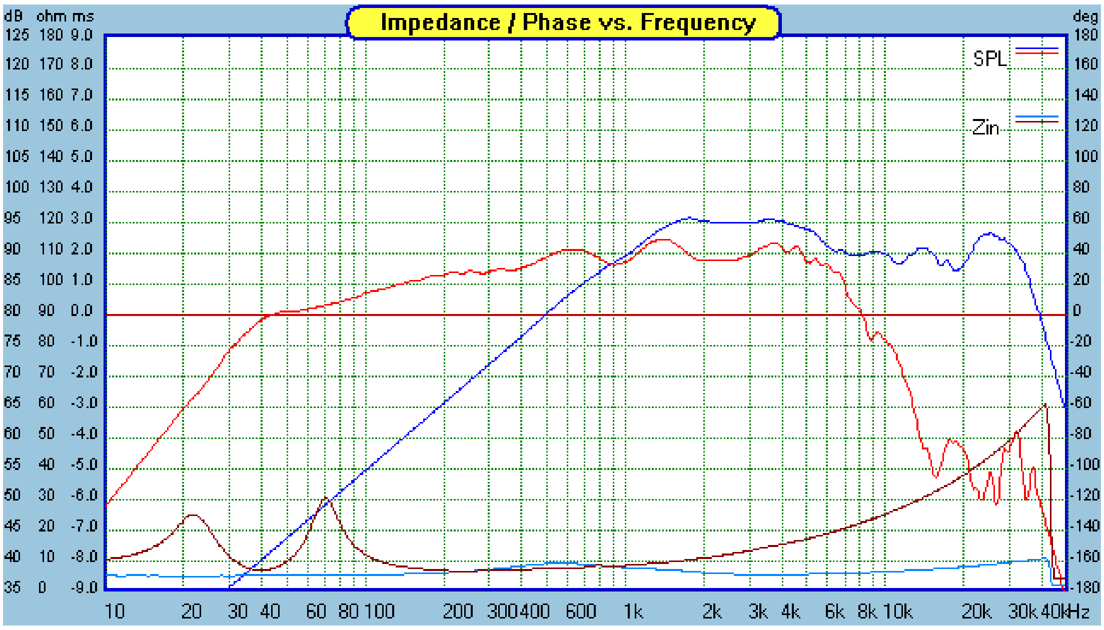

Cauer Elliptic Filter Schematic



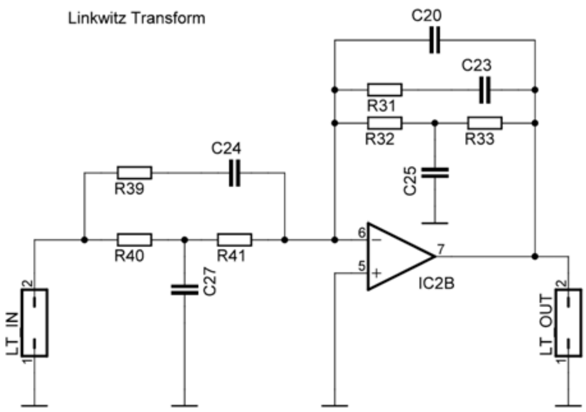

Cauer Elliptic Filter Response

The ripple is ~1dB, the bounces are >50 dB below the pass band response[2]

Now let's look at the overall tweeter/waveguide response courtesy of [www.zaphaudio.com](http://www.zaphaudio.com) Obviously, the blue curve is the one we are interested in.




Notice that the response rolls off at ~6 dB/octave for ~6 dB at 4 KHz as the waveguide unloads. A first order shelving high pass filter will compensate for this nicely. The Linkwitz transform section can be configured as an inverting shelving high pass filter quite easily. Another option is to use one of the notch sections on the other board reconfigured slightly. We'll use the Linkwitz transform section for simplicity.




[2] There wasn't room in the simulator to include the gain reduction and buffers on the board, so the pass band response is +15 dB, reflecting the gain in the two filter stages.

If you jumper R33 and R41 and omit C23, C25, C27 and R31, you are ready to design a shelving high pass filter using the spreadsheet's shelving filter section.

The Midrange panel is a 33 cm x 70 cm dipole, and has an expected 6 dB per octave roll off starting at 700 Hz. There is also a 2.5 db peak at 750 Hz due to basket resonance and the cavity resonance of the U shaped baffle. We'll make a notch filter to compensate for the peak and use the midrange-tweeter board's input buffer to compensate for the dipole roll off.

Designing the notch is straightforward, following the same procedure as the previous example. The midrange is the least sensitive driver in the bunch and I will fix its output. Therefore, connect the filter output to the notch input and use the notch output buffer as the filter's output buffer.

To design the shelving low pass filter for dipole compensation look at the frequency range over which compensation is required. An excellent tool for dipole design is "The Edge", available at www.tolvan.com. Using it you can model your baffle and get a graphical representation of the dipole roll off and the compensation required.

Response roll off begins at 700 Hz. To get reasonably flat response through the crossover region, The Edge suggests compensation from 60 to 390 Hz, and 15.9 dB of boost. Enter the boost required and the frequency where half of this boost (~8 dB) is required along with your choice of capacitor value (nf) into the spreadsheet to determine resistor values. Using 220 nF for C10, R12 is 2156 and R16 is 11,292. For practical values use 2.15K and 11.3K (it took a few iterations with different C7 values to hit this). Remember, simulation does not replace measurement. Use the same approach with measured data, equalize, measure again and repeat as needed.

Imaginary measurements[3] determine that the acoustic center of the tweeter is 20 mm ahead of the midranges at crossover. We will use the group delay of an all pass filter section to compensate.[4] Linkwitz tells us that we should strive to stay on the flat part of the all pass group delay curve. While 20mm of delay at 1500 Hz (.06 msec) is achievable with a single stage using R10 = 1.5K and C7 = 22 nf, this combination puts the crossover slightly down the group delay curve. Since we have two all pass sections available on two boards, use them both and set R10 = 750 and move well into the flat section of the group delay curve.

Connect the tweeter high pass output to the shelving high pass (in the Linkwitz transform block), then to the two all pass sections before returning it to the tweeter adjustable gain

[3] I have not had a chance to build this system. The tweeter offset distance is simply a number chosen to demonstrate the principle of using an all pass section for phase compensation.

[4] Newcomers to the hobby may be wondering why add group delay when so often we hear that ported subwoofers sound bad because of their high group delay. Here it is used as a tool to compensate for non-ideal tweeter/midrange alignment. Group delay by itself is not particularly audible but rapidly changing group delay may be. There is also the difference in magnitude. Ported enclosure group delays are on the order of tens of milliseconds, here we are talking about hundredths of a millisecond (10-2 msec).

buffer. Remember that the shelving high pass section is inverting, so you'll need to reverse the tweeter connections to the amp to keep everything in phase. If you use a capacitor in the tweeter-amp connection (highly recommended for tweeter protection) it should be at least 30 uF to keep it from affecting the overall tweeter response.

If you are a non-believer in the value of the all pass phase correction section(s) I suggest that you try using a pin header and connector pins such as Mouser P/N 571-1021282 to enable bypassing the all pass sections. Listen to a cymbal crash with and without the all pass sections. If properly designed, you will notice that the phase corrected version sounds much more realistic, with more impact. I have a test CD that has isolated cymbals for just this purpose. When I first tested the concept, my then 12 year old son asked what I did to mess it up and told me to "fix it" when I removed the phase compensation.

Moving on to the woofer crossover I will use a 150 Hz second order electrical Linkwitz Reilly filter. This is possible since the midrange panel is equalized flat to well below crossover, and the woofer is flat to well above crossover. To achieve a Linkwitz-Reilly second order filter, the Q must be 0.5. Therefore the buffers are configured as unity gain followers (Jumper the resistor from output to –In, omit the other resistor in the feedback path. Using the spreadsheet, we come up with the following values:

Capacitors = 100 nf
Resistors = 10.6K (use 10.5K)

Other component values work, but this one keeps component values reasonable.

The woofer will be equalized to give response to 15 Hz with a Q of .5 using the Linkwitz transform (LT). Why did I choose 8 cubic feet for this driver? Mainly because the boxes are already built as ported 6$^{th}$ order systems. The ported system has group delay peaking around 51 milliseconds. I can reach my SPL target blocking the ports and adding a LT with only 11 msec of group delay at 15 Hz. I could reduce the size of the box if I was to rebuild. Going to 6 cubic feet the woofer would run out of linear excursion and power handling at the same time.

Sealed with a LT, each woofer can deliver >102 dB down to 20 Hz at less than 0.1% THD. This is an obsolete pro driver so it is rather sensitive – 94 dB/watt. I will be able to reach 102 dB with only 130W. I'll have 800+ per channel available just to take care of movies with lots of LFE. These drivers turn up fairly regularly on EBay if you are interested. The current extended bass driver is the 2242H. Its Fs is 35 Hz, so will need more EQ, but it also has higher power handling ability.

It is time to plug the numbers into one of the Linkwitz transform calculators and adjust the inputs until you come up with reasonable component values. I used:

C23, C24 – 39 nf
R40, R41 – 27.4K
R32, R33 – 213K

C27 – 494 nf
C25 – 64 nf

We are almost finished with this design. Now to look at the input buffer. In many systems you will want some gain here to maximize the signal to noise ratio. In this system, however, there is a lot of boost in both the Linkwitz transform and the dipole compensation. To prevent overloading the following stages and ensure plenty of headroom, we will use a unity gain input buffer, with a 150 KHz. low pass.

This is easily accomplished by jumpering R12 and omitting R16 and C12. The 150 KHz. low pass filter can be made with 100 ohms and 10 nF as mentioned earlier.

Since the two notch filters on this board are not used, there is no need to place their opamps. Similarly, there is no need to place IC5 or IC8 in the unused filter sections.

Connect the High pass output, JP8 pin 3 to the +input of the midrange-tweeter board. Jumper R21 to prevent oscillation or noise generation by IC9B.

Jumper JP15 pins 2 and 3 and connect the buffer output to your woofer amplifier.

# 5. Construction considerations

## 5.1 Power Supply

This circuit is not particularly sensitive to power supply issues if good quality opamps are used. The TI OPA2134 has a power supply rejection ratio in excess of 100 dB, so you are not likely to have any issues with any reasonably stable, quiet supply. A supply consisting of 470 uF of your favorite electrolytic capacitor bypasses with your favorite film capacitor before and after a 3 pin regulator such as the 7815/7915 should be sufficient. We offered the scalable PSU as an alternative that can be used for a variety of applications, including this one.

Each OPA2134 draws approximately 10 mA at idle, so if all positions are used, each filter board will draw 90 mA. Since the loads seen by the opamps are relatively high, there should be no significant increase in draw with large signals. When driving multiple boards from a single power supply, consideration should be given to proper heat sinking of the pass devices. Most likely no heat sinking will be required if you are driving two boards, but it is up to you to determine the requirements. With the Scalable PSU/regulator as shown with the 15V parts kit, you will need heat sinking if you drive more than two fully stuffed filter boards (or 18 opamps on multiple boards)

Similarly, size your power transformer by adding up the dissipation in the regulator(s) and the board(s), and using the standard 2x safety factor or audiophile 3x+. There is little price difference between a 15VA transformer and a 50, so go big if you have the space.

## 5.2 Bypass capacitors

The board provides for local bypass in the form of 10.5 mm diameter, 5 mm pin spacing electrolytic capacitors at the board power input and a 2.5 mm x 7.5mm x 5 mm pin spacing capacitor at each opamp. As with the power supply, use your favorite brand.

The board level bypass can be any value that you can fit, from zero on up. 220 uF should be plenty.

For the local bypass, you can use any brand and tolerance 100 nF polyester film caps. For those needing more guidance, start with Panasonic FC and Wima MKT2. Both are "audiophile approved" types.

## 5.3 Resistors

The Sallen Key filter is fairly sensitive to component variations. For practical purposes, 1% tolerance is fine. The board spacing for resistors is 2 mm and the leads are 10 mm. Standard ¼ watt resistors are fine, 1/8 watt is plenty. You can use designer resistors, but they may not fit readily.

## 5.4 Filter capacitors

The same tolerance considerations apply to capacitors as resistors. Use the tightest tolerance capacitors that you can find/afford for the 5.5 x 7.5 mm package with 5 mm pin spacing. Polyethylene film types should be fine, although many swear by polypropylene, polystyrene and film and foil types.

## 5.5 What opamp?

This is largely a personal preference choice. Most opamps in a DIP8 package should work, if you pay attention to the gain/bandwidth product. Depending on your system and tastes you may be satisfied with the TL072.

You will see the Gain Bandwidth Product on the first page of many datasheets calling it an X MHz device. The OPA2134 is an 8 MHz chip. This means that with a signal of 1 MHz you can get a gain of 8 with this chip, or unity gain at 8 MHz. As a rule of thumb, choose an opamp with a GBP at least 10 times higher than your maximum frequency of interest. If you want to use all of the 150KHz of your input buffer's low pass, you should choose at least a 1.5MHz chip.

A caution with an extremely high speed chip is that it becomes more susceptible to oscillation due to parasitic reactances.

You will find all sorts of recommendations if you search the web. Some people swear at chips others swear by. You pay your money and you take your chances. However you aren't likely to be disappointed by some of the usual suspects, such as NE5532, OPA2134 or LM6172.

### 5.6 Sockets or not?

I didn't come out and recommend a chip, did I? If you are worried about your choice, definitely socket your chips. It is not a bad idea anyway, since it removes the possibility of overheating and destroying the chip.

### 5.7 Potentiometers

After much frustration trying to get a single turn pot set to the proper value, I have officially banned them from my workbench. You can choose whatever brand strikes your fancy, but make it a multi-turn.