

DIALING IN A DSP CROSSOVER: A TUTORIAL, PART 1

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PART 1 Dial-in DSP Crossover

Part 1 of this tutorial addresses the first needs of using DSP crossovers:

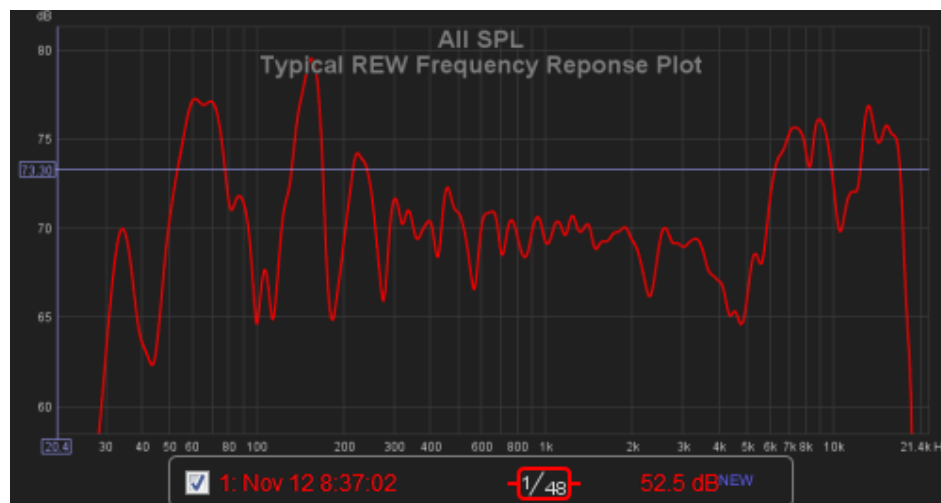
- a) Flattening overall SPL vs. frequency (i.e., frequency response)
- b) Selecting crossover filter types and filter steepness
- c) Identifying/setting correct crossover frequencies
- d) Setting channel delays to remove the effects of driver time misalignments (both physical misalignment and phase shifts due to chosen crossover filters)

(Part 2 of the tutorial addresses flattening overall phase response of loudspeakers using DSP crossovers and its subjective listening effects.)

1. Flatten individual driver/horn SPLs and set relative channel gains

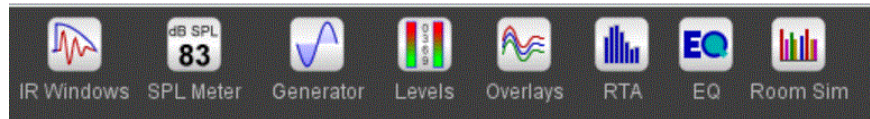
This step describes REW's EQ facility to help flatten the SPL (frequency) response of each driver in the loudspeaker—one driver/horn at a time.

Assuming that you have taken a measurement using REW ([Room EQ Wizard](#) freeware, also see appendices A-3 and A-4 of this tutorial), you probably have a plot that looks something like this:



What you have is a frequency response that's not quite as flat as you'd like. So for that particular plot, you'd like to find the DSP crossover equalization filters (PEQs) to make it flat—but without a lot of cut-and-try and doing a bunch of measurements (upsweeps) along the way.

So you look at the REW window and you see the function bar across the top of the window:



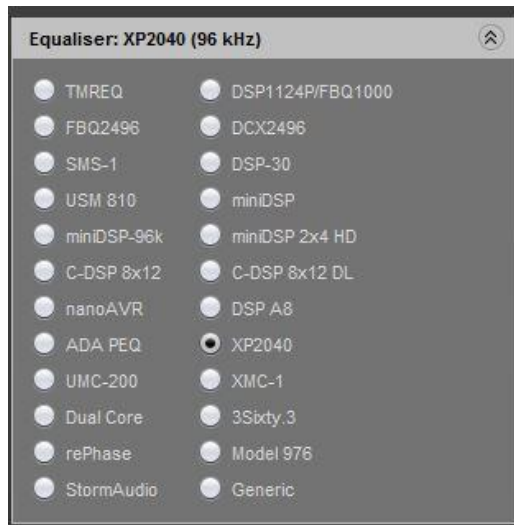
The next-to-last button is "EQ", which you push, and the following window pops up:



The window has two plot areas, top and bottom: SPL response (a.k.a., "frequency response) in the top plot--which we will use in this tutorial and a second plot area which can provide a waterfall, impulse, or pole-zero plot (which we won't use in this tutorial, but you are encouraged to explore using it). There are two arrows next to the left border of this bottom window. If you punch the arrow pointed down, the bottom plot is minimized and the upper plot size now occupies the space that both windows did. (Minimize the bottom window now to maximize the top SPL response plot.)

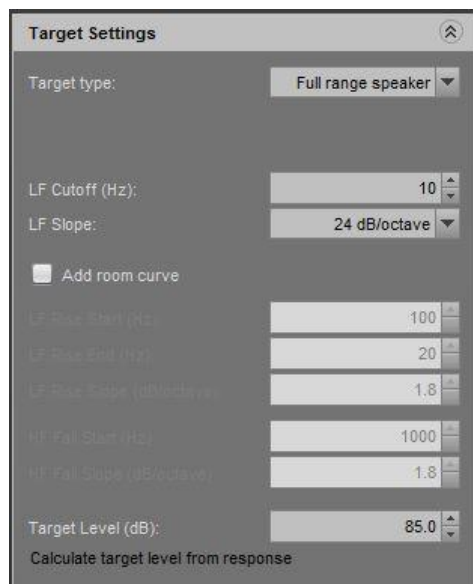
On the right top side of the window, there are 5 bars with text embedded in them, namely: Equalizer, Target Settings, Filter Tasks, Modal Analysis, and Resonances.

If you click on the first bar, it opens a menu downward and shows you a list of equalizers that are supported directly by REW.



If you don't see your equalizer model in that list, fear not. You can pick one that's close in terms of your options. For example, using the EV Dx38 or Yamaha SP2060, I used "miniDSP", and it works. If using Xilica, then select "XP2040", etc. If you sequentially choose each equalizer, and look at the PEQ filters (which we'll generate in a moment), you'll find one that give you the best fit to the features of your crossover. Once you select a crossover type, minimize the Equalizer sub-window by clicking on its top bar again.

Now click on the next menu bar - Target Settings:



The important settings here are "Full Range" if you're equalizing a full range loudspeaker. REW can also help you with subwoofers and "bass limited", i.e., HT surround loudspeakers that are intended to cross above a certain subwoofer crossover frequency.

LF slope indicates intended the slope of the roll-off of the low frequency end (12 or 24 dB/octave) corresponding to the crossover filter slope used between the loudspeaker and the subwoofer. If you look at the measurement plot that you are using to equalize, it will show you the lowest crossover frequency that you should use. For my Jubilees and K-402-MEH, that crossover frequency is ~ 30 Hz. If you're using home theater loudspeakers, it may be as high as 100 Hz. Set that LF roll off point to match your loudspeaker. The same settings are used for the high end.

All of these settings are settable to new default values within the "Preferences" menu on the top bar of the REW main window, which saves time having to set them again and again.

This is the place to put in your "house curve" if you choose to have non-flat loudspeaker response. (I generally don't recommend using house curves unless no other approach can compensate for poor loudspeaker directivity control vs. frequency, or for stereo recordings that retain as-mastered equalization curves before demastering is performed).

You can boost/cut highs or lows using the above controls, and the resultant curve is visible so that you can see what your "goal curve" is. I aim for flat response everywhere since I demaster my recordings, but you may choose to use a "one size fits all" house curve to compensate for non-flat EQ used during mastering on stereo recordings. In the context of home theaters, this is chiefly what a "[house curve](#)" is doing if the loudspeakers have good directivity control.

The last item on this menu is extremely important. It tells REW what SPL (loudness) to aim for overall when it optimizes the equalization PEQs. You want to get this right, because setting it too high or too low will create problems. You're aiming for mid-band SPL, not higher. For the measurement above, 70 dB is the right answer. If you use a higher value, REW will try to boost all frequencies below your target level. If you use a setting that's too low, REW will attenuate everything to make that lower SPL. **It's better to set this value 1 dB too low than 1 dB too high.** It's also better to attenuate using PEQs than to boost. Use the overall channel gain setting on your digital crossover to generally boost or attenuate one driver channel of your loudspeaker. It will be apparent when you need to change the gain of a channel.

After you have accomplished this, each driver/horn in the loudspeaker will be ready to integrate together and to set their relative delays. After you've flattened the response of each driver (using mostly attenuating PEQs), then run one sweep with all channels on to see their relative channel gains. You can set the approximate crossover frequencies and crossover filter types/slopes to see the relative SPL of each driver without having excessive overlaps between channels. Having the driver channels at the same relative gain will facilitate reading the spectrograms to see the delays more clearly.

Assuming you followed everything above, you should have something that looks similar to this after you select the next menu bar, called "Filter Tasks":



Note that the aqua-colored target curve on the left-hand plot is going right through the middle of the curve. Once you've set up your house curve preferences in the Preferences dialog, the only thing that needs to change from loudspeaker to loudspeaker is the "Target Level (dB)" setting after each PEQ optimization or measurement sweep. That value will reset itself after every measurement to 75 dB, and you'll have to manually set it to the right level after each measurement by looking at the measurement and deciding what the target level should be.

The next thing to look for is the "Match Range XXX to YYY Hz". If you're setting the bass PEQs first, then set these two frequencies to:

- 1) The lower frequency should be the lowest frequency that you want to try to equalize. I wouldn't set this more than an octave below where it naturally starts to fall off.
- 2) The upper frequency is set to about 1/3 octave below your woofer-midrange crossover frequency.

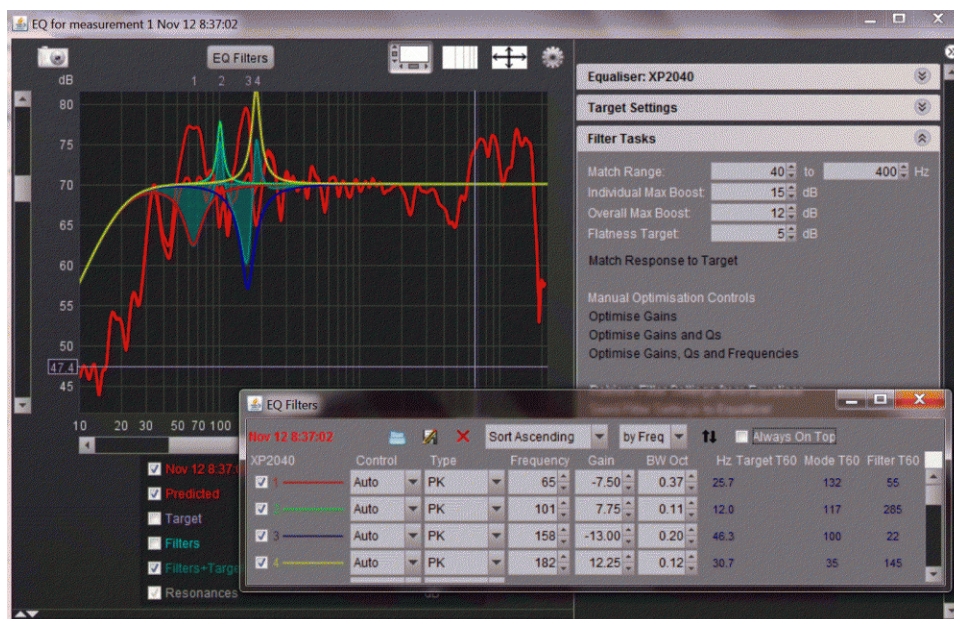
The next item is to set your flatness target. I generally use 4 dB (implying the optimizer will stop optimizing when the predicted response gets to plus or minus 4 dB from your target level, between the two frequencies that you selected above. Try setting it at 4 dB.

REW uses the smoothness settings for the measurement frequency response plot (that you set under the "Graph" menu) to determine the optimized PEQs. I recommend using "Psychoacoustic" smoothing before running the EQ filter optimizer, so that you're not trying to create a lot of high-Q (i.e., very narrow) filters that are probably trying to correct for high-Q room modes.

Now you are ready to generate the PEQs...

Punch "Match Response to Target" and watch the optimization. Filters are created, optimized, and consolidated before your eyes. The optimizer eventually stops.

When you do this for the example above, and then punch the "EQ Filters" button just above your plotted data, you should see something like this:



The corrected curve is shown, along with the individual PEQ curves and the overall PEQ correction curve (you can turn the curve traces on and off using the check boxes below the plot to see everything). The "EQ window" is selectable from just above the plot. This presents the PEQs optimized by the REW PEQ optimizer, and/or those that you can type into your active crossover settings. You just transfer these values to your crossover control software, or you can punch them in using the front panel of the crossover itself. For the EV Dx38 and the Yamaha crossovers, this is easy to punch them in the front panel. For Xilica, I recommend using the XConsole application on your computer. For miniDSP, you can export from the REW EQ facility directly into the miniDSP application running on PC or Macintosh.

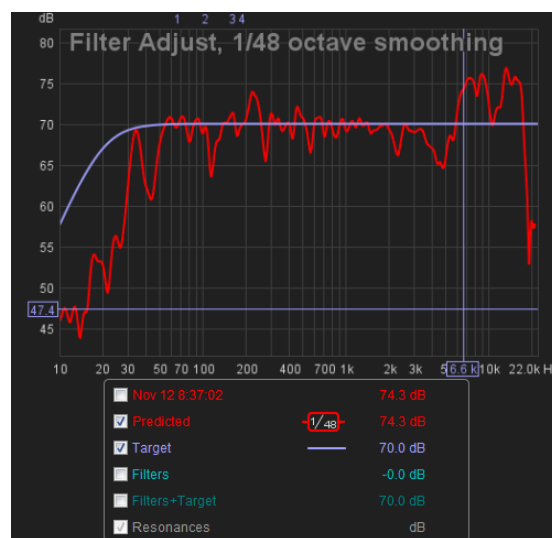
If you generated too many PEQs within REW's EQ facility, just hit "Reset Filters for Current Measurement" and "okay" the dialog box that pops up. Increment the flatness target by one or two dB upwards, then punch "Match Response to Target" again. If too few PEQs are generated (or none), decrement the flatness target by 1 dB then reset and run again.

Repeat this process until you get something that looks reasonable in both the predicted response curve and the number of PEQs. Once you transfer those new values into your digital crossover, run a sweep again to verify the results. I think that you'll be very close to your goal...in one try. If not, then go back to the beginning and change the gains of the channels, the target level used, and/or the flatness level, then try again.

If your resulting PEQs are outside the values allowable for your crossover, then go back up to the "Equalizer" type menu, and select a different equalizer type until you find one that results in the PEQs that you can use. The EV Dx38 has limitations on attenuation of PEQs that the Yamaha and the Xilicas do not. You'll find that having the ability to do big attenuating PEQs with your crossover will save having to double up PEQs to get enough attenuation. This is a big advantage of the newer digital crossovers - so you don't have to use shelving filters.

You'll notice that REW doesn't use shelf filters. If you want to use them, simply dial in what you want and take a sweep then use that sweep's data to continue to build your PEQs.

Here's a plot of the "predicted response" of using the PEQs above:



Repeat this process for your high frequencies (tweeter and/or midrange) and lay in your PEQs for those frequencies into the output channels of your digital crossover.

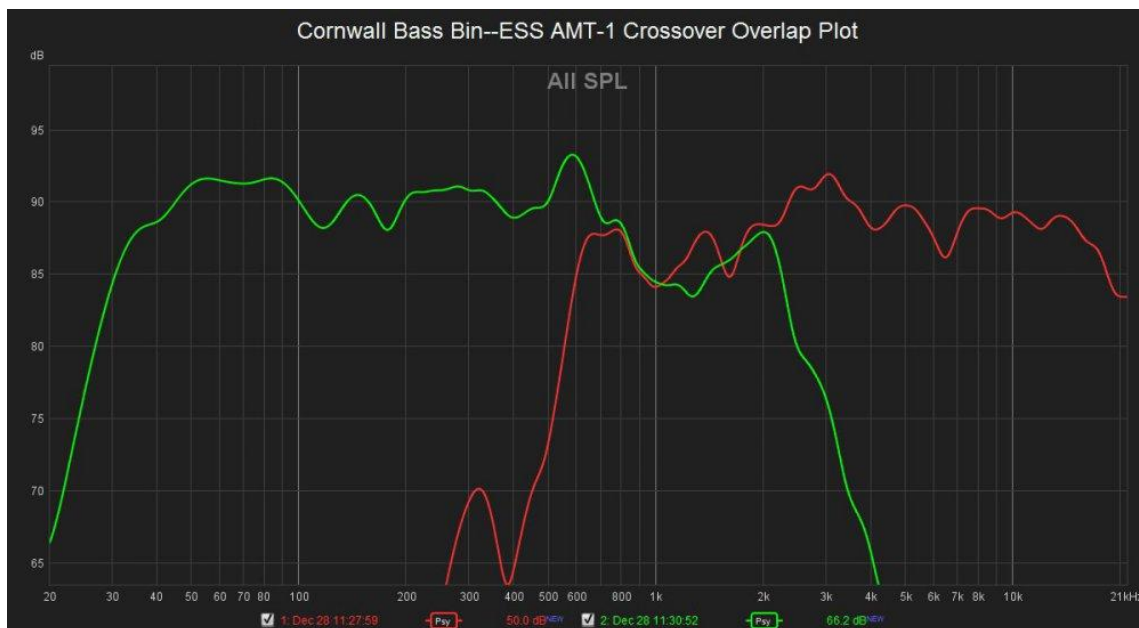
If you have frequency response issues in the crossover regions, then use the resulting PEQs from the optimization on the INPUT side of the crossover, or if you start to run out of PEQs on the output channels, switch to the channel inputs and keep going.

If you're dissatisfied to the flatness of the response after all is done, and you still have available PEQ filters not being used in the crossover, then you can run the optimization again using the updated equalization PEQs embedded to see what more you can add to flatten response.

2. Set crossover filter type & slope

Now that you have the individual drivers flattened in SPL and relatively close in terms of overall channel gains, now you can choose the preferred crossover slopes. Look at the driver overlaps in the frequency bands where you intend to cross them over by taking

individual sweeps of each driver channel and then looking at the combined frequency response plots within REW using the second button, called "All SPL" on the button bar just above the plot. The width of these natural bands of overlap will help to determine how steep or shallow a set of crossover filters that you should use. In the case of drivers/horns that can just barely make it to the crossover frequency on each side of the crossover frequency, you might be forced to use a steeper filter to bridge the crossover frequency region—the "crossover interference band". In the case of two drivers (and horns as the case may be), they can have significant natural response overlaps, such as the following plot of a Cornwall bass bin with an ESS Air Motion Transformer (AMT-1) on top, both showing their two-octave overlap in frequency response:



For the above case, it is possible to use a very low order crossover filter set, (high pass for the higher frequency driver, low pass on the lower frequency driver) because of the width of the crossover interference band—approximately two octaves from ~600 Hz to 2400 Hz. It's usually wise to choose somewhere in the center of the interference band as the center crossover frequency. If you wish to use higher order filters for other reasons than just driver interference band overlap width, that's certainly at the discretion of the person dialing in the loudspeakers. I personally recommend also trying out first-order filters (where possible) if using higher order filters to hear the difference, and set up two presets on your DSP crossover to be able to switch back and forth between them to hear the differences (if any).

I've already matched the relative gains of each driver channel so that the output frequency response is relatively flat across the full frequency spectrum of the drivers that you're crossing (~40-20,000 Hz in this case).

The next step is to set the crossover filter types and slopes within your DSP crossover (using the *XConsole* application in the case of Xilica crossovers). Note that you don't

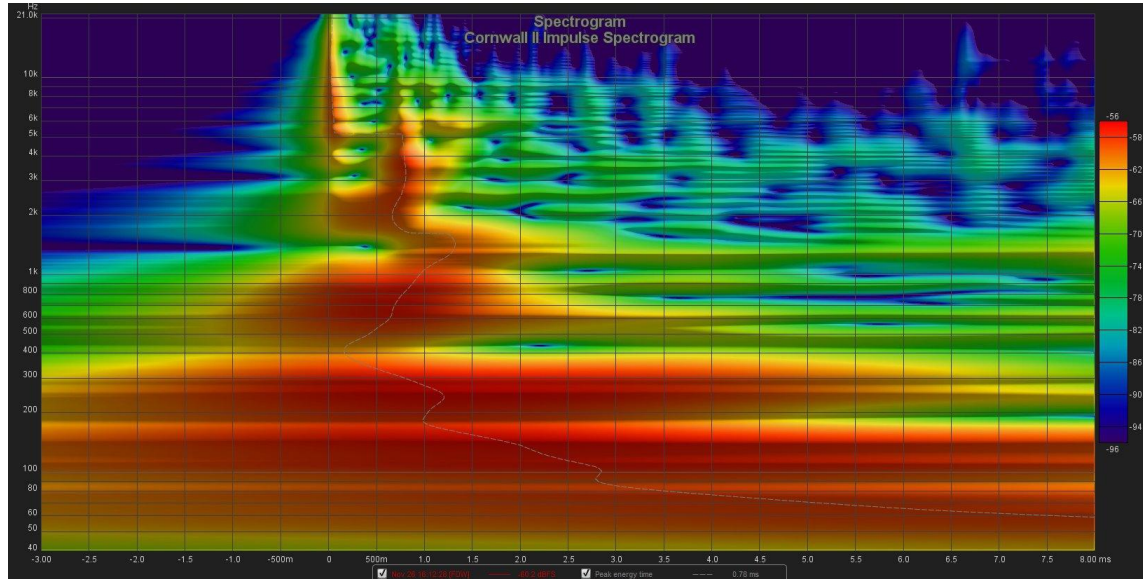
have to worry about calculating relative delays at this point if you wish to just take a REW upsweep measurement and look at the results of the measurement to directly measure the needed delays. Many people will select something like 24 dB/octave Linkwitz-Riley filters (i.e., fourth order). The Xilica can also go up to 8th order (48 dB/octave) filters—but at the expense of adding 720 degrees of phase shift, thus significantly degrading the resulting impulse response of the loudspeaker.

3. Take REW sweep with all drivers and chosen crossover filters

At this point, you've got all drivers relatively matched in gain, their frequency responses flattened, and the chosen crossover filters set, so it's time to run a full sweep using REW to see the relative delays of each driver/horn within the single loudspeaker. What you will see is uneven frequency response and phase curves within REW. This is normal and indicates that time alignment of channels is now needed.

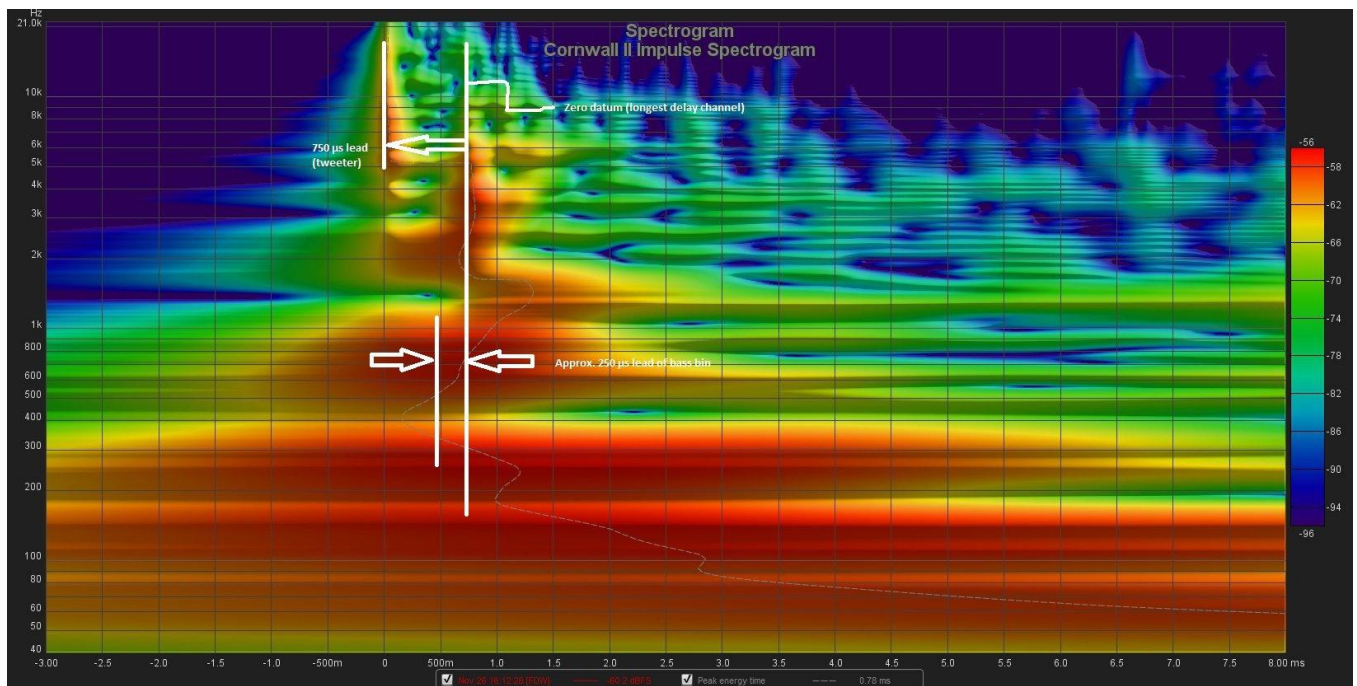
4. Read spectrogram relative channel delays

From the full sweep measurement with all drivers and crossover filters, select spectrogram view from the plot menu bar just above the plot (the next-to-last button on the plot menu bar). After setting the plot preferences (frequency vertical, dark background, and scaling zoomed in to 0.5 ms horizontal divisions), you'll see something like this:



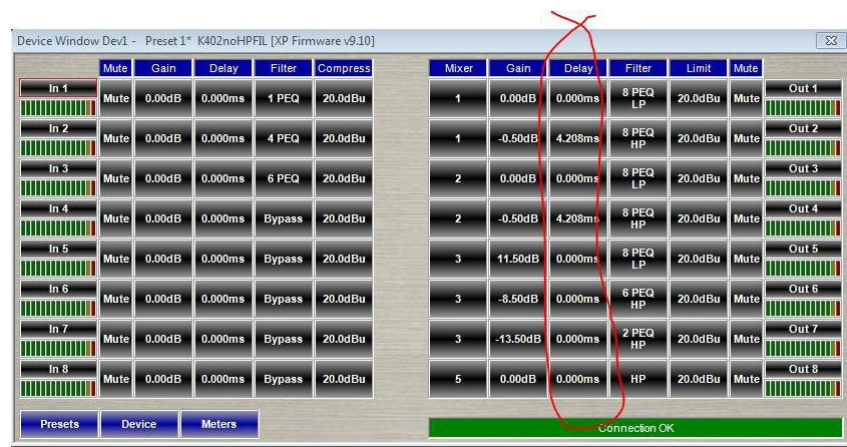
The above spectrogram is from a 1979 Klipsch Cornwall measurement, showing the time delay between the K-77 tweeter impulse energy (at the top of the plot), the K-55 midrange horn/driver at about 750 μ s behind the tweeter, then the direct radiating woofer at about 250 μ s behind the tweeter, and 500 μ s in front of the midrange (to find this, look at 400 Hz at the peak energy plot line for this particular measurement). So the channel delays must be set relative to the latest arriving channel—which is the

midrange in the Cornwall (and is usually the bass bin in other loudspeakers). So the tweeter channel will need to be delayed 750µs and the woofer by 250µs in this example.

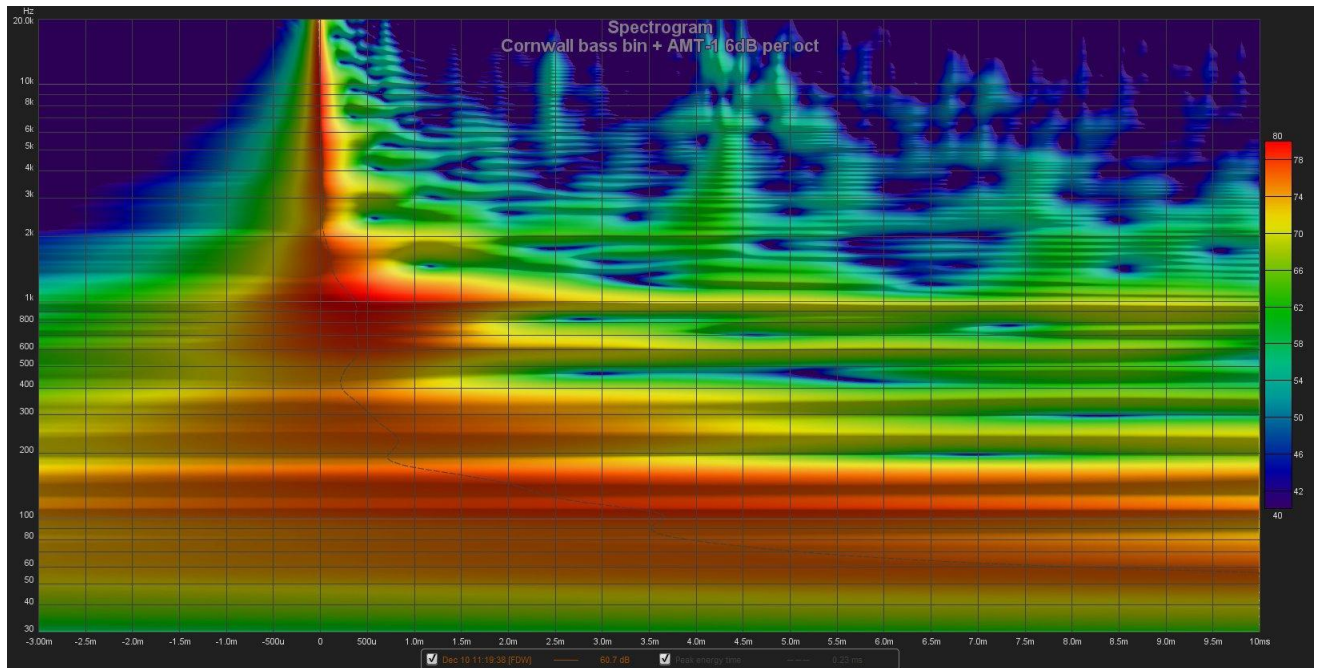


5. Set crossover delays and PEQs, rerun REW sweep

When those values of delay are inserted into the output channels:



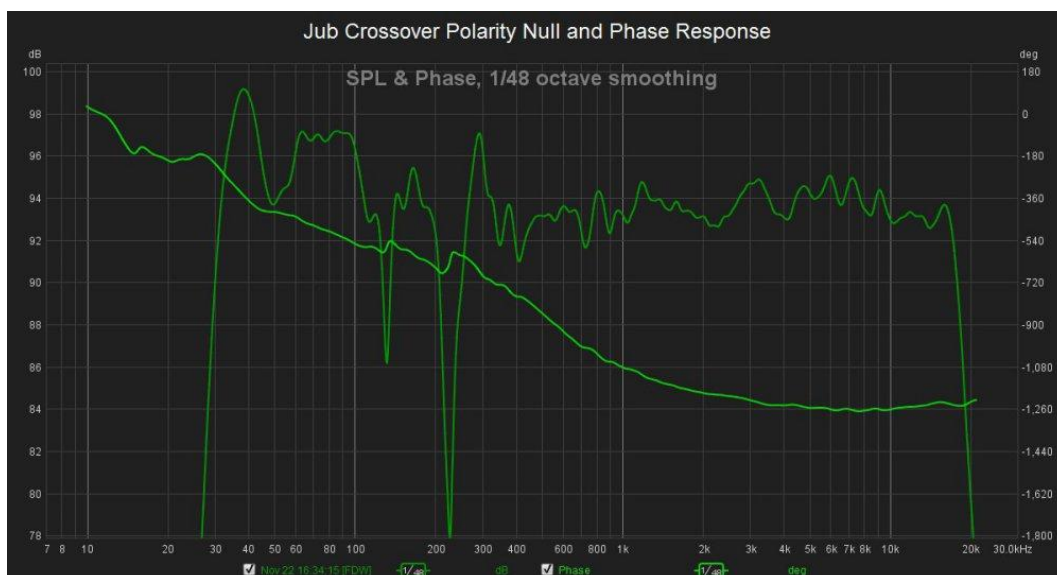
After those delays are set and another REW measurement upsweep is made, something like the following will be seen in the spectrogram plot:



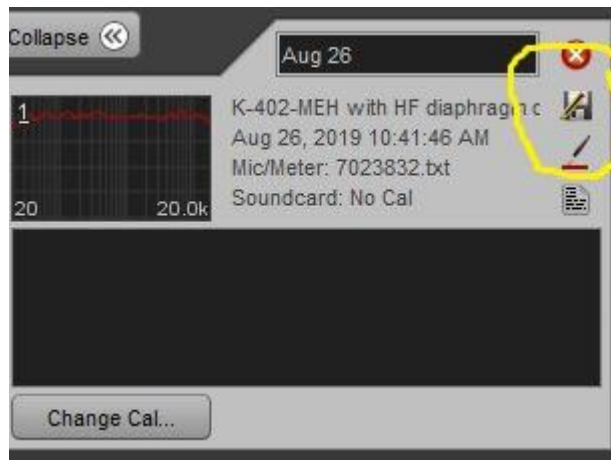
Note the gradually increasing lag of the peak energy time as the frequency decreases. This is the characteristic that is desired in order to get the best impulse response. The smoother the peak energy time curve, the better the impulse response.

6. Look at driver-to-driver crossover band FR & phase

After the delays have been set, take another upsweep using REW, then look at the SPL & Phase plot (the first selection button on the menu bar just above the plot).



If the resulting response looks acceptable, punch the save button to save the measurement within REW in the measurement dialog box (shown below) and name it descriptively when the “save measurement” dialog box appears.



7. Check step/phase/group delay plots & update delays/filters

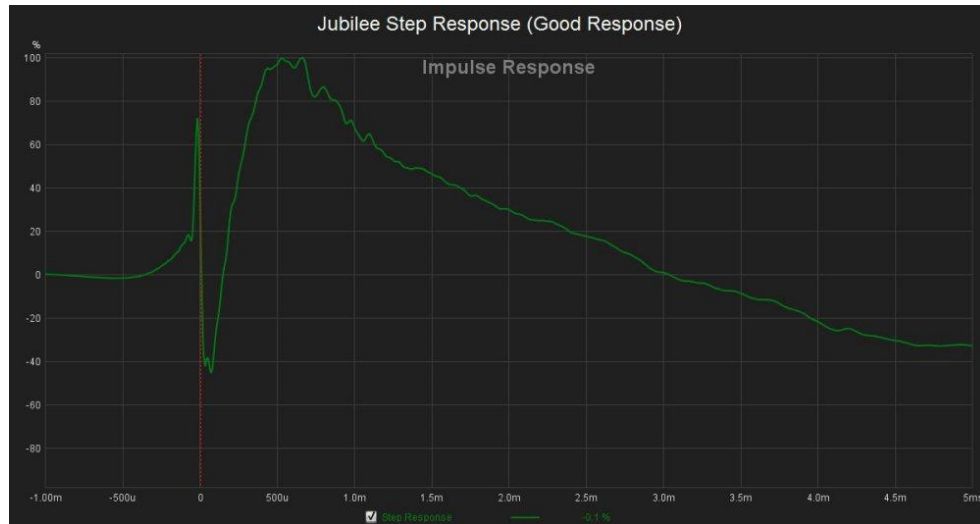
Note the big nulls at ~125 Hz and ~215 Hz. These are caused from drivers not being time aligned. This may indicate a need to flop the polarity of one of the channels using XConsole. This will depend on the crossover type and steepness that is chosen. If you change the crossover type or slope, this will need to be revisited and adjusted again.

The phase curve also goes through a tight little zigzag at these nulls. This also shows up on the group delay plots (i.e., the first derivative of phase with respect to frequency):



These are indicators that something is still not dialed-in properly. Go back and re-set the time delay on the leading drivers until the frequency response, phase, and group delay plots are smooth.

The step response (the plot is found under the Impulse Response [IR] plot window) should look something like this:



...and not like this:



The difference between these two step responses is the correct setting of channel delays between the higher and lower frequency “ways” of the loudspeaker, as well as the choice of lower order crossover filters (second order in this case) and use of Bessel type filters in order to minimize the phase and group delay growth through the crossover interference bands.

Once you've completed the adjustments in frequency response and delays for the first loudspeaker, now it's time to move to the other loudspeaker and complete steps 1-7. The [XConsole](#) (the Xilica PC/Macintosh control application) in particular provides a "Copy" function under the "Setup" menu to copy settings from one channel to another which are user selectable. This saves a lot of time, and allows you to clone your settings to the other loudspeaker channels without keying mistakes.

8. Summary of Methods Used and Results

In general, the methods above proceed via:

- 1) first take raw (non-PEQ correction) measurements of each driver in the loudspeaker
- 2) correct the SPL response of individual drivers to flatten their standalone SPL response by using output channel PEQs. Usually the biggest PEQ corrections are the most important, and smaller PEQs using higher "Q" filters can wait until later.
- 3) Run full-range sweep measurements of all drivers in one loudspeaker (generally 10 Hz→ 20 kHz), then correct the time alignment of the drivers while simultaneously flattening SPL response further through the crossover interference band(s). This is accomplished by using either a spectrogram view or excess group delay, then referring back to the SPL/phase response views.
- 4) Check the SPL response through the crossover interference band, and if attenuations/cancellations are apparent, try flipping the polarity of the lower frequency channel and measure again.
- 5) Repeat the procedure while refining the channel delay(s) and SPL flattening via PEQs until the desired level of SPL flatness and group delay/spectrogram verticality is achieved.
- 6) Generally, improvements in SPL and phase/group delay flatness are audible until ± 1.5 dB SPL flatness is achieved across the audible frequency spectrum (-3 dB bass cutoff point up to the tweeter/high frequency driver -3 dB mass rolloff frequency). The sound quality keeps getting better and better.

Part II of this tutorial includes specific techniques to reduce the overall phase growth of the loudspeaker under test until the total phase response is within ± 90 degrees of phase from 100-200 Hz up to the high frequency mass rolloff frequency (generally above 15 kHz). Unless your room and loudspeakers are set up to curtail early reflections, much of the benefits of phase flattening usually cannot be discerned. It is strongly advised that extreme near-field absorption treatments be used in the listening room, and proper loudspeaker positioning is achieved before attempting phase flattening procedures so that the results can be heard.

APPENDICES

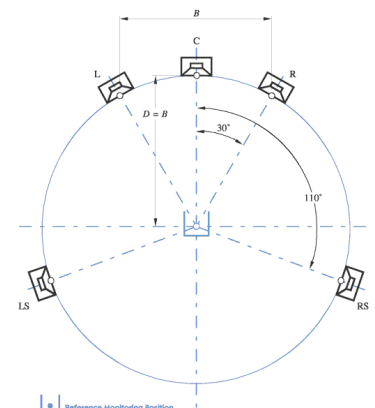
A-1. Select and Acquire DSP Crossover (# I/O Channels and Connections)

The first task is to select and acquire hardware and software for your stereo or multichannel sound system which at least matches the fidelity achievable with your present equipment and the needs of the loudspeakers in terms of DSP crossover noise floors and gain ranges.

With the knowledge that a DSP crossover can provide transformational performance in loudspeakers in listening rooms, it is also true that the quality of the DSP crossover also determines the ultimate performance achievable. By that, it is also seen that there are DSP crossovers that are available in the marketplace that are below the level of hi-fi needs, particularly in the areas of gain range, signal to noise ratio, quiescent noise levels, external connection noise, poor fidelity of analog output circuitry, and insufficient digital word bit depth which controls the apparent sound transparency of the unit. (The models that are not recommended include Behringer, some entry-level miniDSP models including 2x4 and the OpenDRC models, entry-level dbx DriveRack, and many other off-brands on the market, most of which are priced below \$200USD.)

Considering higher quality level and price points in DSP crossovers, all aspects of these units actually exceed the needs of home hi-fi, including sound quality transparency, quiescent noise levels and gain range. The home hi-fi enthusiast first considers this better level of DSP crossover performance before succumbing to the “lowest price syndrome” that many seem to fall for. Recommend DSP crossover brands include Xilica, Yamaha, ElectroVoice, and the higher priced models from miniDSP (including the 2x4 HD and the 4x10 HD). The use of balanced external connections reduces common mode noise (power line noise) significantly, and availability of AES3 (AES/EBU) digital inputs also significantly reduces noise levels.

For our example, consider a Xilica XP-8080 crossover, consisting of one box which has the equivalent of four stereo biamping crossovers (bi-amping all loudspeakers), or tri-amping two front loudspeakers and bi-amping a center loudspeaker. One XP-8080 can cross an entire 5-channel system, with an AVP providing the separate subwoofer channel outputs to DSP



front-end subwoofer amplifiers that do the EQ and amplifying for the two subwoofers.

A-2. Install DSP Crossover

A-2.1 Acquire USB calibrated microphone and calibration file, cables TBS



Figure 1 - Xilica XP-8080 DSP Crossover

As an example of a commercial DSP crossover, a Xilica XP8080 can accept up to 8 inputs and up to 8 outputs and is meant to be placed between your preamplifier's outputs and your amplifiers' inputs to replace your loudspeaker passive crossovers in order to directly couple your amplifiers' outputs to your loudspeakers' woofers and

midrange and tweeter drivers. Other Xilica crossovers in this series include the XP-2040, XP-3060 (now discontinued), and XP-4080. These crossovers are 96 kHz sampling/24 bit units that I'd classify as "hi-fi" in terms of their sound quality.



The XP-8080 is the only one in this series to use the Phoenix (a.k.a., "Euro") bare wire screw-down terminals, shown above in green and to the right, disconnected:

These terminal blocks are removable from the back of the unit for ease of assembly with the connecting cables.

The XLR "microphone cable" connections are used on most quality multichannel preamp/processors to reduce

line noise.



These cables are also available with RCA "unbalanced" connections on one end (resulting in male and female cables) for those that have preamplifiers and amplifiers with RCA connectors only.

Note that you can buy XLR—bare wire cables and save the cost of the extra connector.

For a 5.1 multichannel system, there are five input channels plus one of subwoofer channel. When using Euro (Phoenix) connectors, the wires are stripped down and inserted into the respective terminal blocks, then screwed down.

A-2.2 Connect DSP crossover between preamp and amps

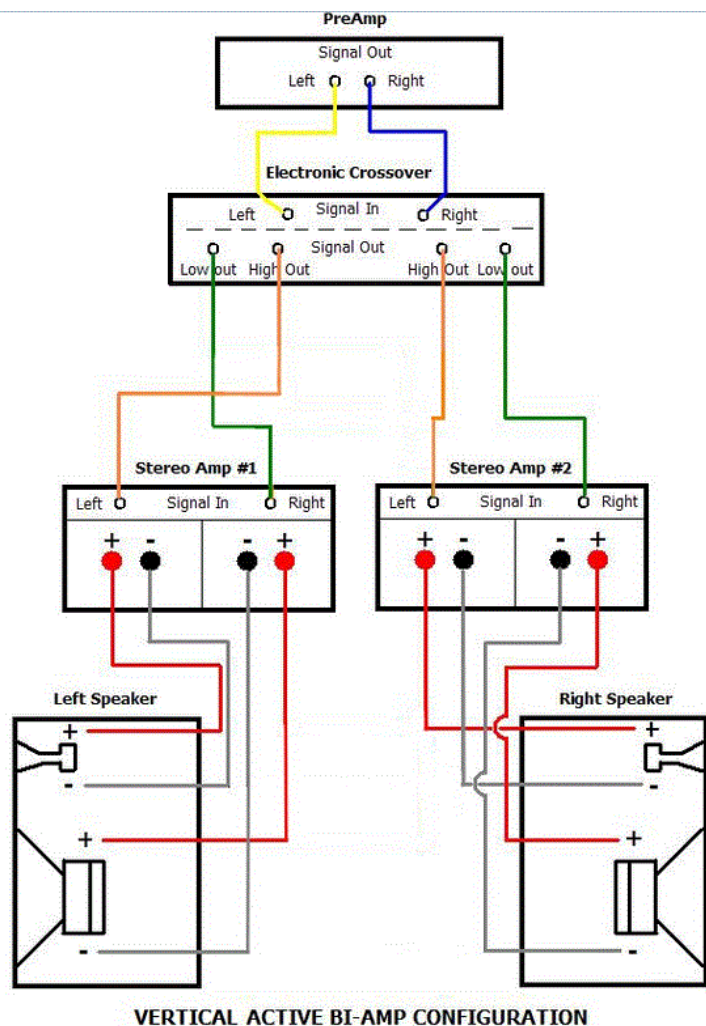
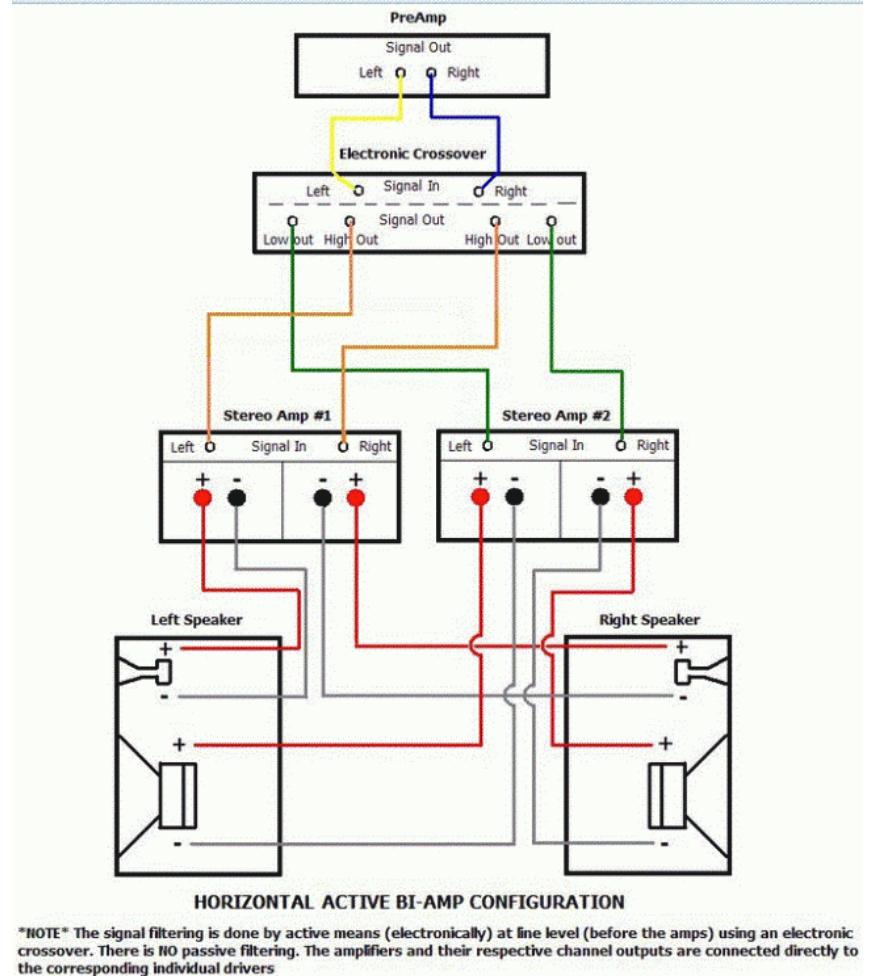
The connection of the amplifier outputs to the loudspeakers is your choice: you can assign input channels to output channels in software, as shown below. If you're using different types of amplifiers (tube and SS) the figure to the right is a typical connection for "horizontal bi-amping" for two channels.

In similar fashion, vertical bi-amping can be used to spread the load from the bass bin channels, assuming that the two+ power amplifiers are identical in performance.

After making connections for the XP-8080's channels and connected their other ends to their respective preamp outputs (5 channels), and downstream amplifiers and USB microphone, you're ready to program the DSP crossover using their supplied software (XConsole in this case, available for Windows and Mac OSs).

For our example, Xilica provides a [YouTube training video for using their XConsole software](#). Here are the "gotchas" that I found using the XConsole PC application that is freely [downloadable from Xilica](#):

1) Assigning input channels to output channels: that is done by selecting the on-



screen button from the "Mixer" column for the output channel of interest (on the right-hand block of buttons). Make sure that you move the gain slider to the top of its range for the input channel that you wish to connect to the output channel.



2) Transferring equalization settings from other crossovers: Xilica uses "bandwidth" for the sharpness or coverage width of each filter, instead of "Q" used in, for instance, ElectroVoice and Yamaha digital crossovers. The table to the right converts Q to bandwidth (and back). However, the Xilica window also calculates and shows the "Q" just below the "Oct" inputs for each "PEQ" filter shown below. You can iterate the BW input that you use to see the equivalent Q for each filter, so you don't have to use the table to the right.

3) Shelf filters: the Xilica shelf filters interpret the bandwidth much differently than the "slope" (i.e., 6 or 12 dB/octave) of the EV and Yamaha crossovers. One suggestion is to iterate the slope's bandwidth value from 1.5 to 2 octaves so that the resulting curve shown in the plot area is smooth and sloping correctly, i.e., not creating a "S" in the middle of the response curve. The workaround that I used was to abandon the use of shelf filters until I can find documentation from Xilica explaining the "bandwidth" parameter relative to how EV and Yamaha use that parameter.

Q	BW (Oct)
0.5	2.54
0.55	2.35
0.6	2.19
0.65	2.04
0.667	2
0.7	1.92
0.75	1.8
0.8	1.7
0.85	1.61
0.9	1.53
0.95	1.46
1	1.39
1.1	1.27
1.2	1.17
1.3	1.08
1.4	1.01
1.414	1
1.5	0.945
1.6	0.888
1.7	0.837
1.8	0.792
1.9	0.751
2	0.714
2.15	0.667
2.5	0.573
2.87	0.5
3	0.479
3.5	0.411
4	0.36
4.32	0.333
4.5	0.32
5	0.288
5.5	0.262
6	0.24
6.5	0.222
7	0.206
7.5	0.192
8	0.18
8.5	0.17
8.65	0.167
9	0.16
9.5	0.152
10	0.144
15	0.096
20	0.072
25	0.058
30	0.048
35	0.041
40	0.036
45	0.032
50	0.029

Controls Window Dev1 Out Ch1

Frequency Response Plot (Dev1 Out Ch1):

- Y-axis: +20, 15, 10, 5, 0dB, -5, -10, -15, -20, -25, -30
- X-axis: 20, 100, 1k, 10k, 30k
- Plot shows a flat response at 0dB with three marked points: a yellow circle at ~400Hz, a green dot at 1k Hz, and another yellow circle at ~2k Hz.

Name	Delay	Mixer	PEQ1	PEQ2	PEQ3	PEQ4	PEQ5	PEQ6	PEQ7	PEQ8	X-Low/HPF	X-High/LPF	Limit
Gain	0.000ms	Off	Off	Off	Off	Off	Off	Off	Off	Off	None	None	20.0dBu
0.00dB	0.00 m	Off	Off	Off	Off	Off	Off	Off	Off	Off	1000Hz	1000Hz	0.3ms
0.00dB	0.00 ft	Off	Off	Off	Off	Off	Off	Off	Off	Off	6dB/Oct	6dB/Oct	2X
+ Polarity	Mute	Off	Off	Off	Off	Off	Off	Off	Off	Off	FIR-L/HP	FIR-H/LP	
			PEQ Reset	Bypass	Bypass	Bypass	Bypass	Bypass	Bypass	Bypass	1000Hz	1000Hz	

Change Channel: [Dropdowns]

Plot Options: [Dropdowns]

5) "How To" manual for each setting: I recommend using the [EV Dx38 manual](#) as a training guide for each parameter in the settings, since Xilica apparently hasn't provided a user manual explaining each parameter in similar detail. [Note that the Dx38 manual shelf filter discussion on "Q" will not apply to Xilica crossovers.]

TBS.

See [“Getting Started with REW”](#)

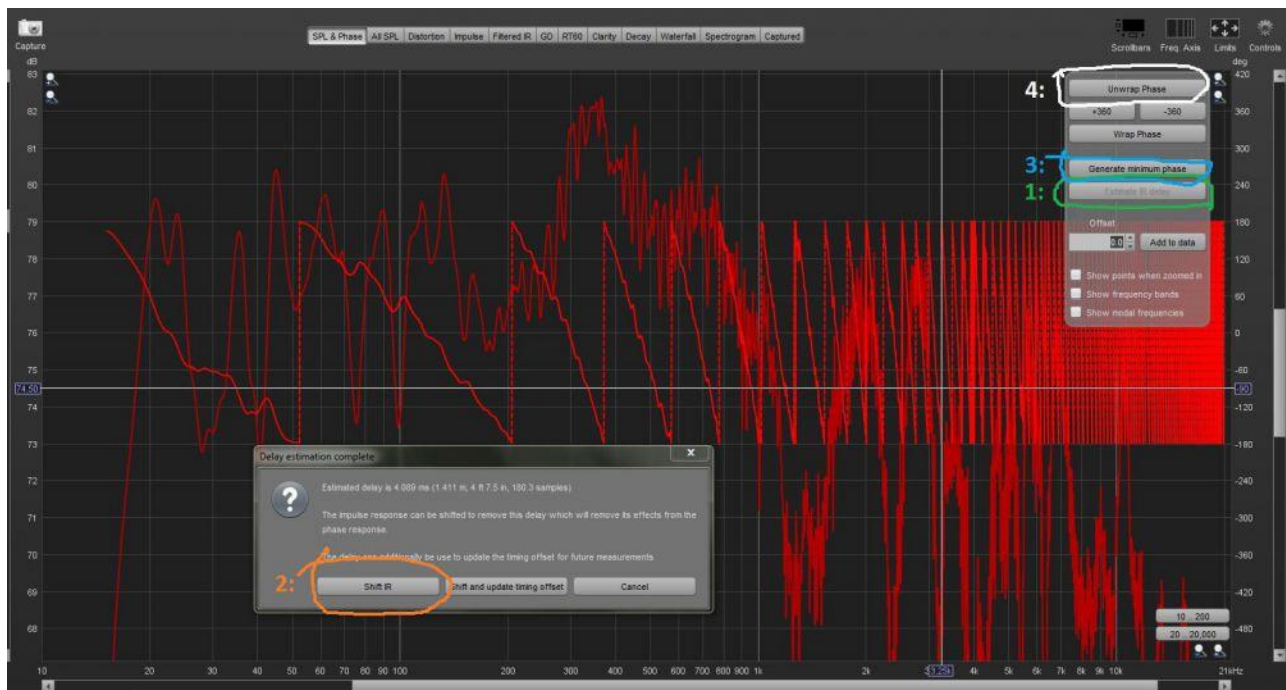
Set the microphone at listening height (for the Jubilee, this is about at the top of the bass bin, one metre in front of the front baffle of the Jubilee bass bin) in order to minimize in-room reflections which show up in the group delay plot. For loudspeakers that are over 6 feet tall, you will have to move back to the listening position and use as much absorption and time gating as you can to suppress first reflections obliterating the phase/group delay data.

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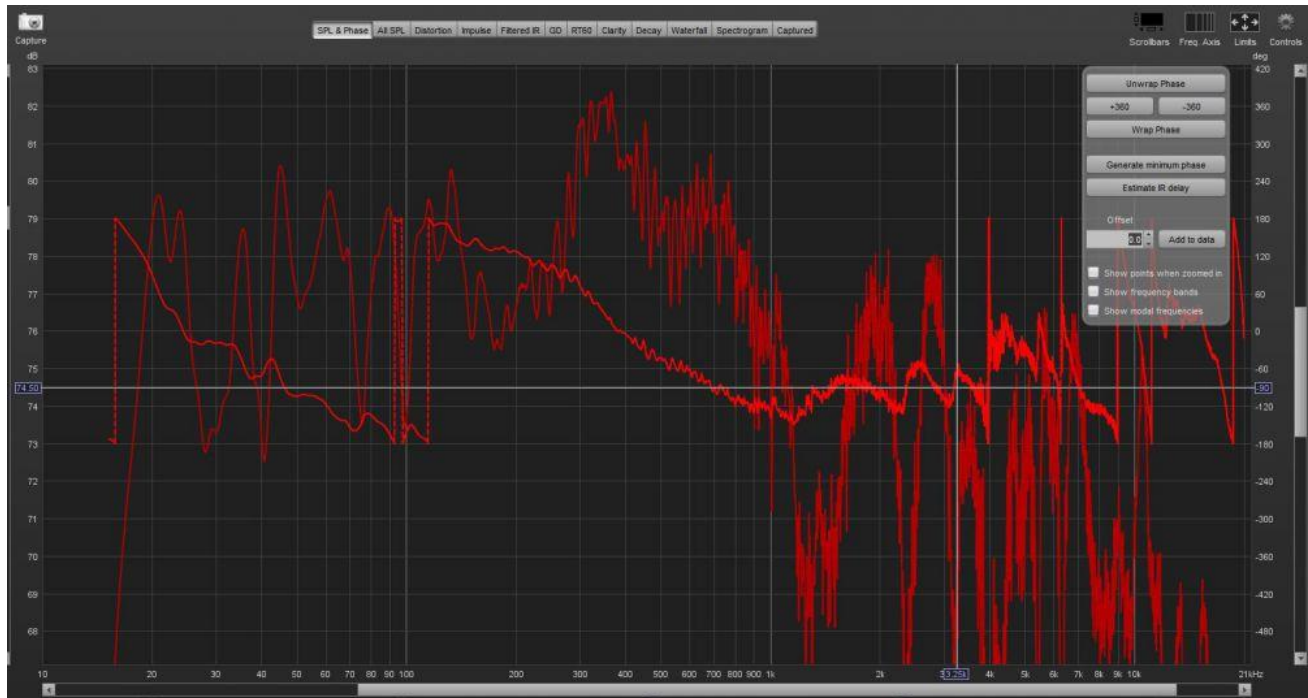


After a REW upsweep measurement has been taken, the following calibration of the impulse response datum for the measurement is required:

Below you will see an SPL/phase measurement just after the upsweep has been taken, with the Controls menu turned on to the right. There is a button on the Controls menu marked "Estimate IR Delay" ("IR" = impulse response). You need to push that button, then a window will pop up called "Delay estimation complete", and you push the "Shift IR" button on that menu.



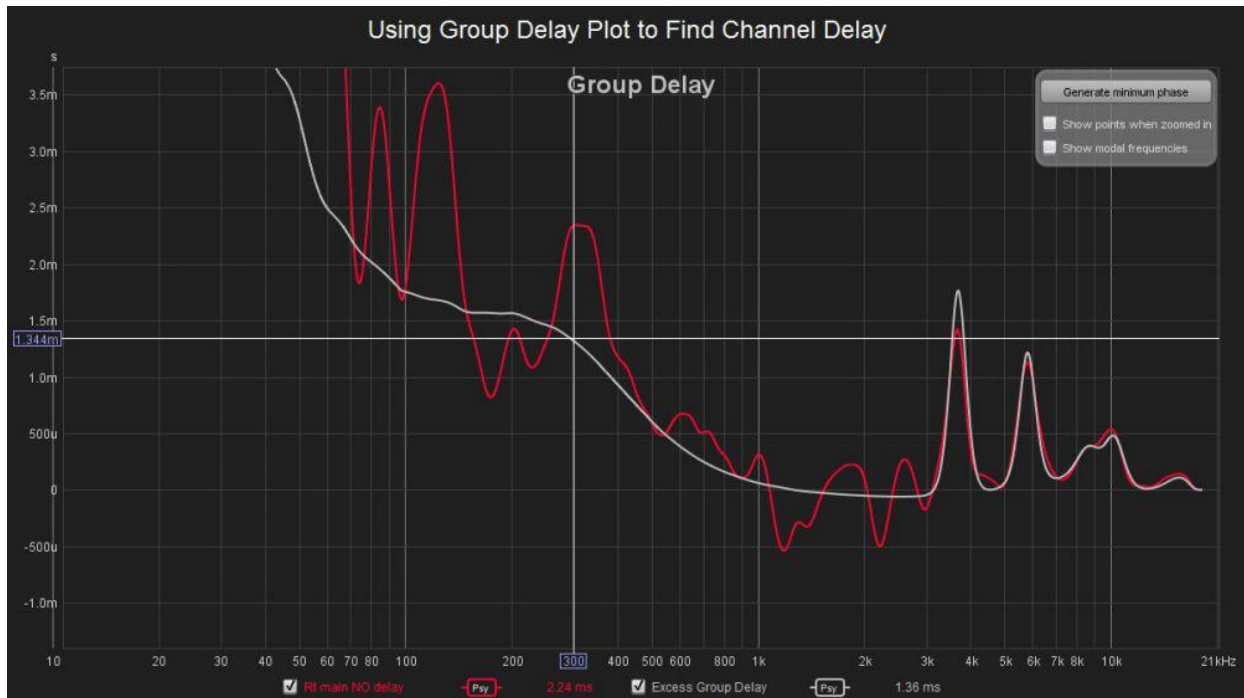
Then you return to the Controls menu and push the "Generate minimum phase" ("3") button, then the "Unwrap Phase" button ("4"). Then you will see something like this:



Notice that the SPL vs. frequency curve is unaffected, but the phase curve has been transformed. If you then save the measurement, these settings (all except "Generate minimum phase") will be saved with the measurement so you don't have to push these buttons again for this measurement. The "Generate minimum phase" button will have to be pushed after every time that you reopen a saved measurement.

Now your phase and group delay plots will be useful for setting channel delay (time alignment).

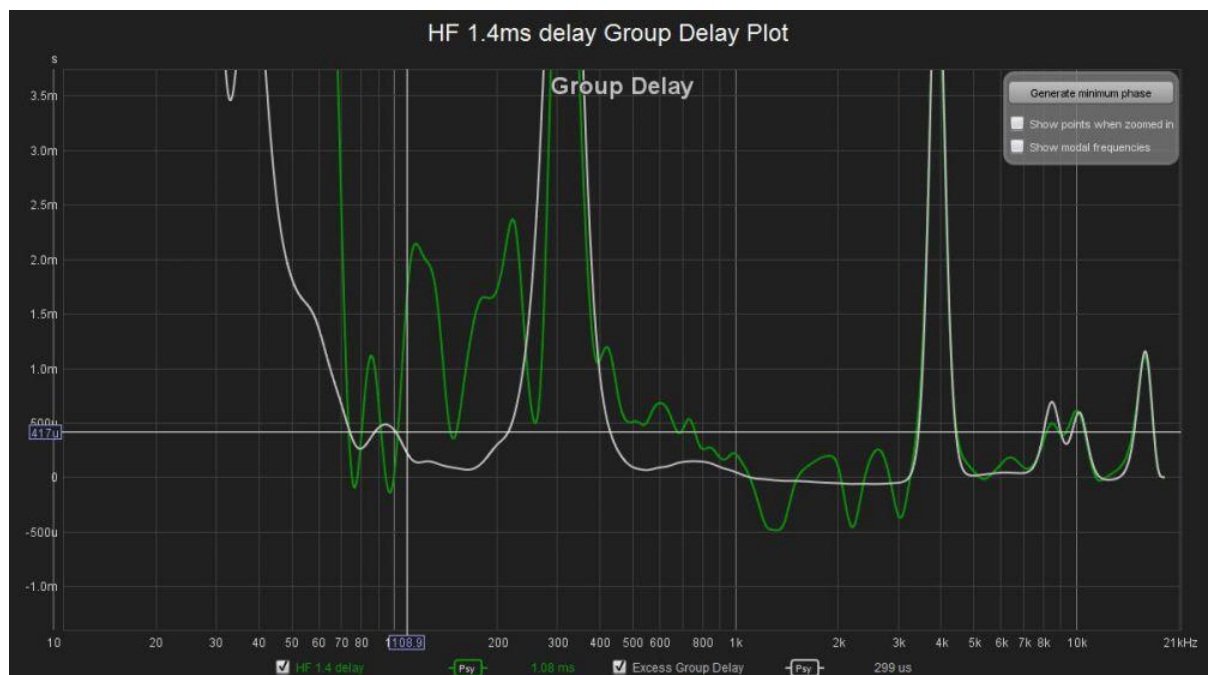
After accomplishing the above calibration of the measurement's IR datum, calculating the minimum phase (and at the same time, the minimum group delay, excess phase and group delay curves) and unwrapping the phase curve, the following plot will show the correct value for HF channel delay (see cursor position at the 300 Hz crossover frequency in this example):



So the correct value of channel delay to use in your next measurement is shown by the vertical axis value of the excess group delay curve at the crossover frequency (300 Hz): 1.344ms in this case.

The above shift in delay also required a polarity reversal on the bass bin so that it would be in-phase at the crossover interference band. The reason for this is due to the 1.4 ms delay, which is just about 1/2 a wavelength at the 300 Hz crossover point.

Here is the group delay plot (including excess group delay) without the polarity flop on the bass bin, but also showing the correction of the excess group delay curve back towards



zero. The obvious polarity issue at the 300 Hz crossover point yet to be addressed, as shown by the spike of the group delay curve at that point.

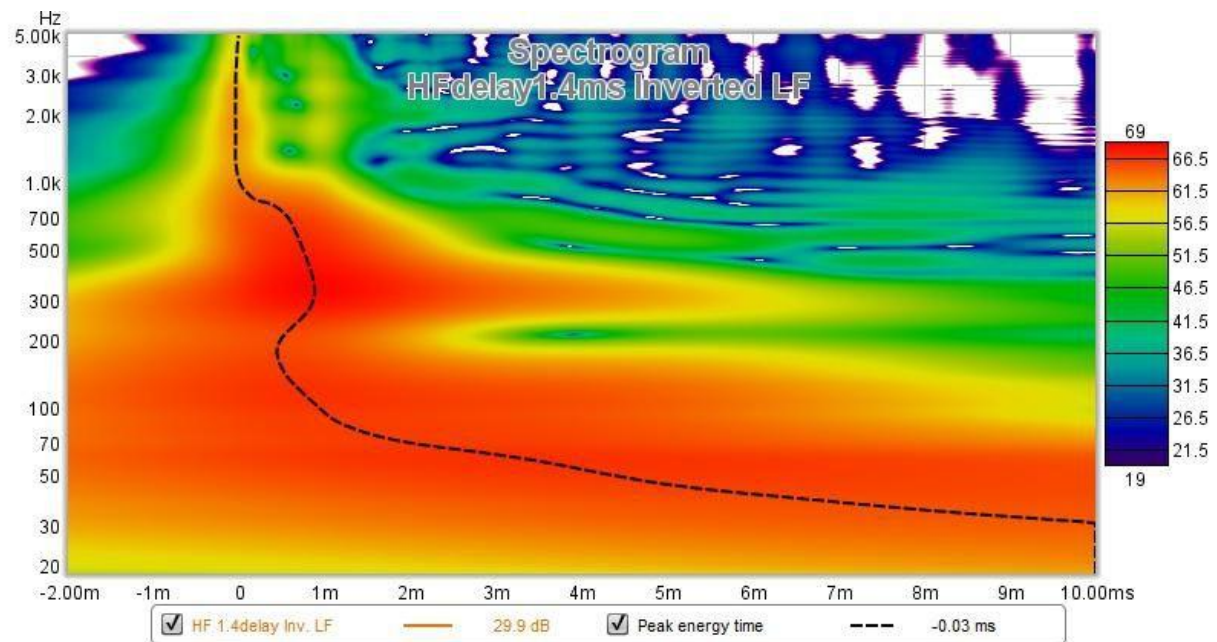
By the way, the above shift in delay also required a polarity reversal on the bass bin so that it would be in-phase at the crossover interference band. The reason for this is due to the 1.4 ms delay, which is just about 1/2 a wavelength at the 300 Hz crossover point.

Here is the group delay plot (including excess group delay) without the polarity flop on the bass bin, but also showing the correction of the excess group delay curve back towards zero. The obvious polarity issue at the 300 Hz crossover point yet to be addressed, as shown by the spike of the group delay curve at that point.

And below you will find the spectrogram after the bass bin polarity is flopped:

Also note that the HF driver itself experiences a lagging impulse (and phase) response below 1 kHz, which shows up as the broader portion of the impulse column between

~180-1000 Hz. This is inherent in the driver itself. However, the overall impulse response has been moved toward the left, which is the objective of this activity.



[End of Part 1]

A-6 Basic aspects of using Digital Signal Processing (DSP) crossovers in bi-amping and tri-amping operation

DSP crossovers imply bi-amping/tri-amping, which is defined as the use of one amplifier for low frequencies, and one or more amplifiers for mid-to-high frequencies. This is done via the use of an active DSP crossover unit which is inserted between the preamplifier and power amplifiers.

A-6.1 The advantages of active bi-amping/tri-amping

- 1) It provides much greater [driver control](#) than a passive crossover/full-range-loaded amplifier configurations.
- 2) It provides a [better load](#) for your [amplifiers to drive](#), and an effective gain in each amplifier's effective output. It will provide lower amplifier-originated [intermodulation distortion \(IMD\)](#).
- 3) It provides much [greater protection](#) of your tweeter/midrange drivers under clipping/overload conditions.
- 4) It provides the ability to use [less expensive amplifier designs](#) for each driver.
- 5) It provides for [time alignment of drivers](#) within a single speaker (a "must have" capability)
- 6) It provides for [better crossover performance](#) in both amplitude AND phase in the crossover region for smoother crossover performance, including more stable soundstage imaging vs. frequency.
- 7) It provides [stability of crossover performance](#) relative to passive crossover drift during and immediately after under high-load speaker output conditions, i.e., it maintains electrical output linearity under heavy load conditions.
- 8) It [requires lower-quality wire/connectors](#) than a similarly configured passive crossover/full-range amplifier configuration.
- 9) It allows [on-the-fly changes](#) in crossover frequency, EQ and channel gain settings to support changes in your setup configuration, i.e., facilitating the fine-tuning use of tools like [Room EQ Wizard \[REW\]](#), replacing individual drivers, speaker position changes, and adding channels for playback (2.0, 5.1, 7.1, etc.).

A-6.2 Disadvantages of active bi-amping/tri-amping

- 1) It requires two (biamping) or three (tri-amping) power amplifier channels per speaker, with associated wires/connectors.
- 2) It requires an active crossover unit.

A6-3 DSP crossover functional description

- 1) An [active crossover](#) provides separation of frequencies of the incoming pre-amplifier output signals, breaking each upstream channel into two (bi-amping) or three (tri-amping) downstream channels: a woofer channel and mid-range/tweeter channel).
- 2) It provides [higher-quality equalization \("EQ"\) capability](#) for each channel.
- 3) Digital crossovers [typically provide for delay](#) to allow for time alignment of the drivers within a single speaker. (This is a similar function to an AV Processor that time aligns speaker-to-speaker in a 5.1/7.1 array.)

A-6.4 Disconnecting a speaker's passive crossover from the drivers

DSP crossover integration into the system requires at least the woofer (or low frequency driver) that must be disconnected from the passive crossover to permit bi-amping. If your speakers are 3-way (i.e., woofer, midrange, tweeter in each cabinet), then you may retain the passive crossover between the midrange and the tweeter if using bi-amping (...but for tri-amping, all drivers must be disconnected from the passive crossover networks)

A-6.5 Passive bi-amping

[Passive bi-amping does not bring the benefits of active bi-amping](#), only the disadvantages of extra cables and connectors. Generally, it is not worth the expense of the extra amplifier. In particular, passive bi-amping does not provide for delay adjustment or filter/EQ parameter flexibility.

A-6.6 DSP crossover brands/units

Many manufacturers make DSP crossovers, including ElectroVoice, dBX, Yamaha, Ashley, Behringer, Xilica, Lake, etc. Prices go from \$100USD to many thousands of dollars. There are also lower-cost alternatives, such as miniDSP. Price is generally commensurate with sonic performance.

A-6.7 DSP crossovers used in configurations other than an active crossover box

The "powered subwoofer" channel found on most AV Receivers/Processors is a limited example of a for-purpose active crossover channel (i.e., mono bass channel). Usual features include gain control (at the integrated subwoofer/power amplifier unit), user selectable crossover frequencies, and sometimes GEQ/PEQ (graphical and parametric equalizer) filters built into the AVR/AVP.

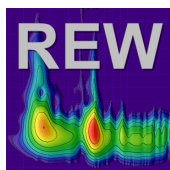
Delay adjustment for each speaker channel is usually included in AVR processor functionality to correct for speaker distance room placement variances. Additionally, an "Audyssey"-like feature on some AVRs/AVPs features a built-in real-time analyzer (RTA) to help the user set up their speakers in a room environment.

A-6.8 Why use a DSP crossover (and not passive crossovers)?

DSP crossovers avoid the shortcomings of passive crossovers:

- Very high quality parametric equalization,
- Digital delay/time alignment, and
- Fast power limiting
- Can replace the need for “room correction software” (i.e., Audyssey, YPAO, Dirac, etc.)
- Dial-in your sound system to correct the deficiencies of your listening room acoustics and loudspeaker acoustic output.

Even more so than any other single piece of electronics in the signal chain, the DSP crossover provides the ability to dial in loudspeaker performance, both in terms of objective measureable performance and subjective listening engagement. It is the full attainment of that hi-fi capability which is the subject of this tutorial. The DSP crossover brings with it the ability of the hi-fi enthusiast to fully adjust the sound system to accommodate listening room acoustics and nonlinear system response in ways never before dreamed possible, leading to breathtaking performance levels.



The use of in-room acoustic measurement tools, such as Room EQ Wizard (REW) complements the needs of dialing in loudspeakers using a DSP crossover to achieve the highest performance achievable.

Even though I have used DSP crossovers in sound systems for many years, it has been recently that additional DSP crossover dialing-in techniques (found in Part 2 of this tutorial) have been added which have significantly improved the subjective listening performance of

the entire multichannel or stereo system—without requiring the use of finite impulse response (FIR) filtering (see figure 3) capability in the DSP crossover (something that adds significant computational demands and higher hardware costs). These newly discovered techniques are found here in a detailed how-to format which address phase and group delay growth minimization, as well as zero-phase growth crossover filters (i.e., so-called “fractional order” crossover filters) to the usual tasks of achieving flat SPL vs. frequency. When these phase/group delay techniques are used with loudspeakers having full-range-directivity (i.e., fully horn loaded), a new level of hi-fi performance is achieved.

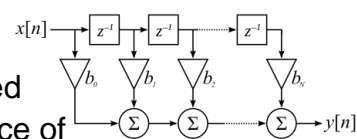


Figure 2: FIR Filtering Introduces Significant Cost and Complexity



Figure 3 - A Klipsch Jubilee DSP Crossover Installation