How to Build a DSP X-Over Preset
from the specifications
in the owners manual

Part One: Initial Setup

By Marty McCann

Bartman has been asking for the presets for his system that consists of QW-1, QW-215, and QW-218. Back when Boomer did most of the initial presets, we did not have a QW system on hand to measure. I had some presets from when I went to both Nigeria and Vietnam, but they are in a laptop that has crashed. I will have access to a QW system in a few weeks and will measure it and come up with a specific preset, but for now let me use this as a teaching opportunity based on the published specifications of these products. Below are the specs for the QW-1:

Loudspeaker Spec Summary

QW-1F

Frequency response, 1 meter on-axis,
swept-sine in an anechoic
environment:
200 Hz to 18 kHz (±3 dB)
Usable low frequency limit
(-10 dB point):
150 Hz
Power handling:
Full Range:
600 Watts continuous
1200 Watts program
2400 Watts peak

Mid Frequency Section:
600 Watts continuous
1200 Watts program
2400 Watts peak
High Frequency Section:
100 Watts continuous
200 Watts program
400 Watts peak

Recommended Amplifiers:
Mid Frequency Section:

CS-1400	Bridge Mode	1400 Watts
CS-4000	Stereo	1350 Watts
PRO 8200	Stereo	1450 Watts

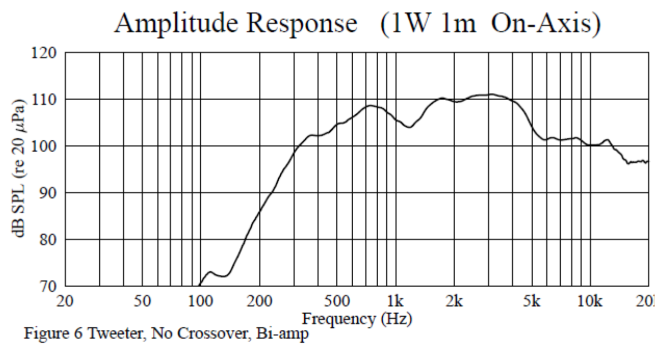
High Frequency Section:

CS-1400	Stereo	330 Watts
CS-2000	Stereo	495 Watts
PRO 5200	Stereo	290 Watts

Sound pressure level, 1 Watt,
1 meter in an anechoic environment:
Full Range:
101 dB SPL, (2 Volt input)

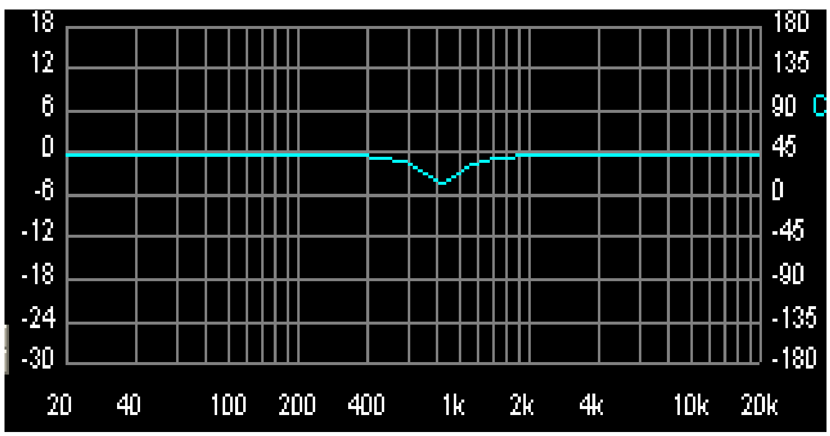
Mid Frequency Section:
101 dB SPL, (2 Volt input)
High Frequency Section:
111 dB SPL, (2.83 Volt input)
Maximum sound pressure level
(1 meter):Mid Frequency Section:
129 dB SPL continuous
135 dB SPL peak
High Frequency Section:
131 dB SPL continuous
137 dB SPL peak

Time Offset:
Mid Frequency: 0 ms
High Frequency: delay 0.813 ms

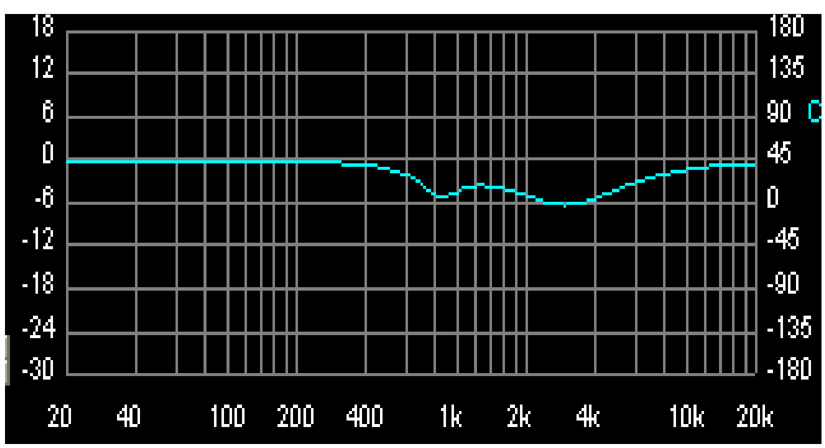


QW-1 High Frequency Horn

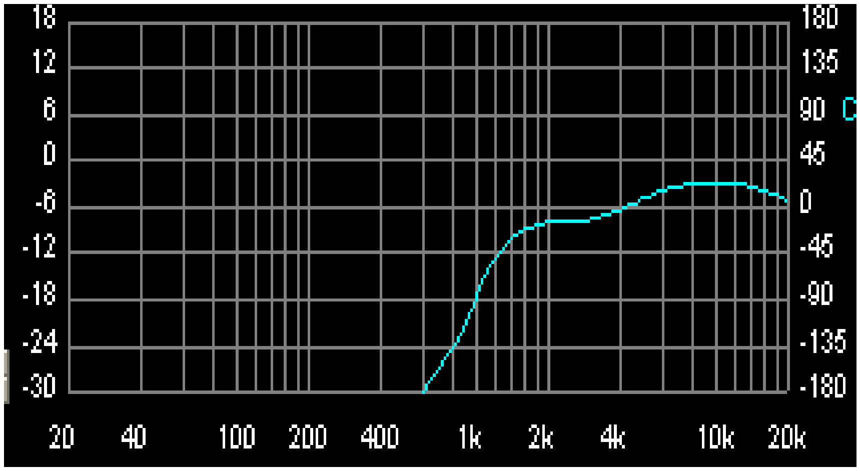
So we will begin by looking at the above raw response of the 44XT compression driver mounted on the 60 x 40 CH642qt horn in the QW-1 to decide on how to program the high frequency section of the DSP that will crossover the QW-1. Even though I am going to cross over the horn at around 1.6 - 2 kHz, I observe a bump between 500 Hz and 1200 Hz that I would like to remove. An anomaly within an octave of a crossover point can effect the response particularly in the phase domain. So to center the PEQ, if I subtract 500 from 1200, that is 700 Hz, ½ of which is 350. So now I add 350 to 500 to get a center frequency of 850 Hz. I will initially program in a PEQ centered on 850 Hz with a bandwidth of 0.577 and -4 dB of cut. See below:



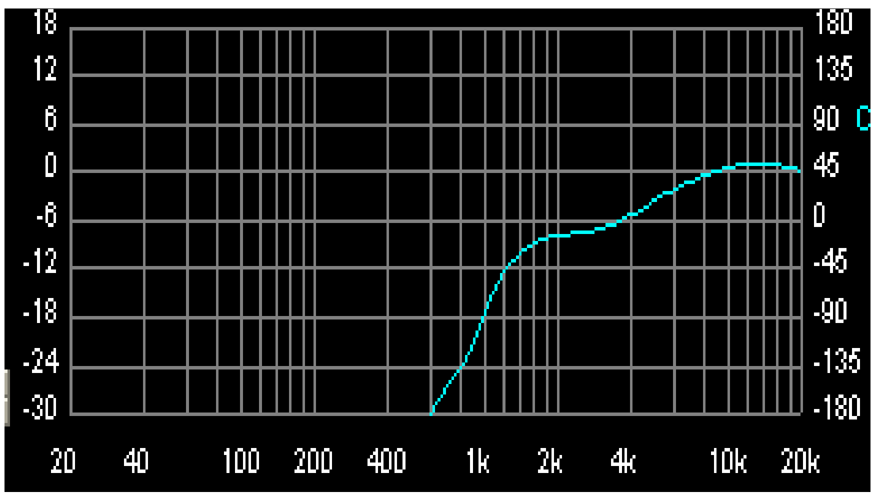
Next I will invoke a PEQ to reduce the next wide bandwidth bump from about 1500 Hz to 4500 Hz. 4500 – 1500 = 3000. Divided by two and added to 1500 equals a center frequency of 3 kHz. So the next PEQ will be at 3 kHz with a bandwidth of 1.5 and -6 db of cut. See below:



Next I will go ahead and engage a 2 kHz X-Over point with a 4th order -24 dB per Octave Bessel Filter as a High Pass (Low Cut) and I will also invoke a 2nd Order -12 dB per Octave Low Pass (High Cut) at 18 kHz. See below:



Next we bring in a High Self filter at 16 kHz with +7 dB of boost (this may need to be increased later). See below:



We will leave this for the moment and go to the bandpass below the high frequency horn in the QW-1.

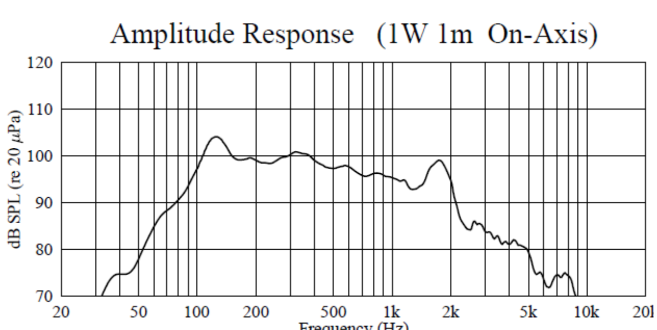
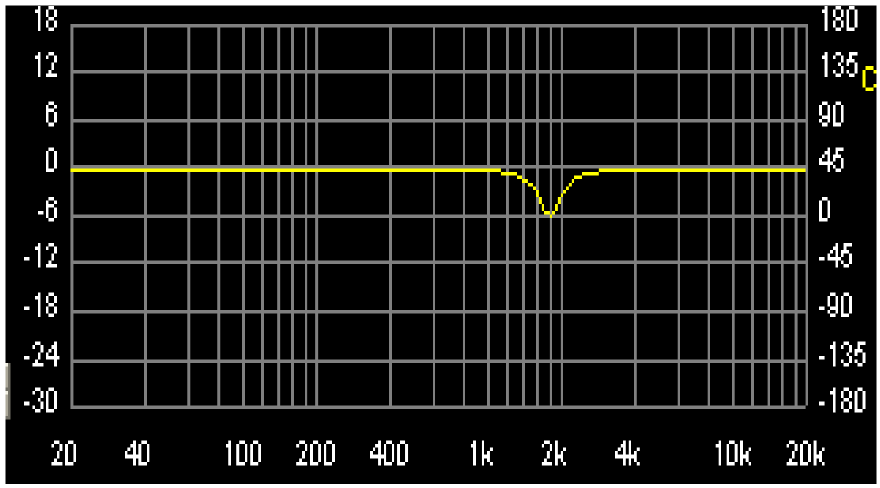


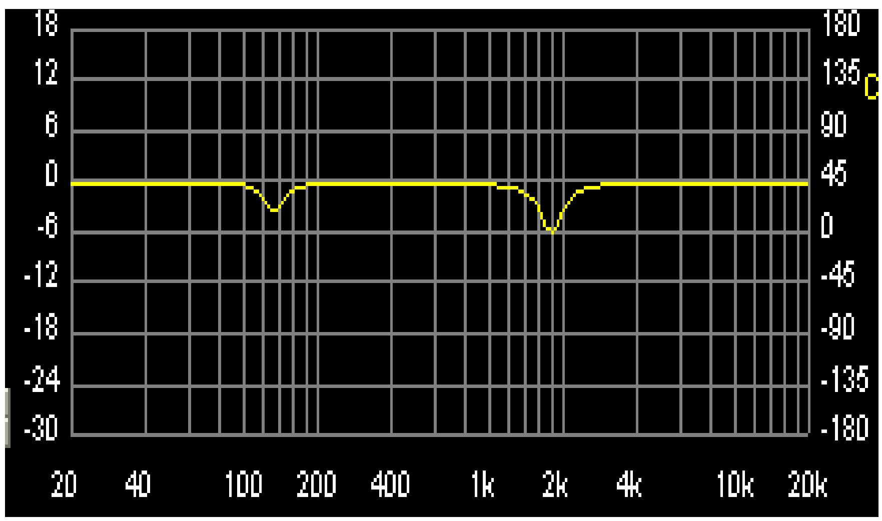
Figure 8 Woofer, No crossover, Bi-amp

QW-1 Horn Loaded dual 10” BW's

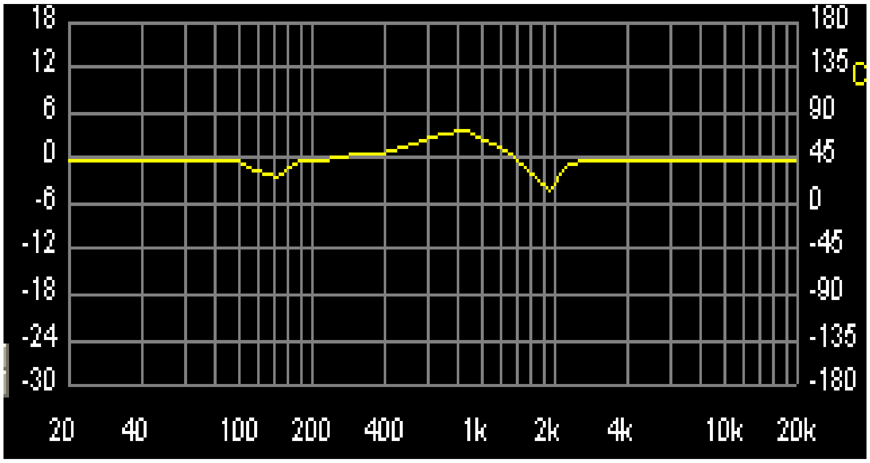
In examining the above raw frequency response of the dual horn loaded10 inch Black Widows in the Mid section of the QW-1, we can see two bumps that we would like to remove. Again they are above and below the X-Over points, but never the less can affect the response. So the first PEQ looks like it should be centered on 1.8 kHz, with a bandwidth of 0.250, and -6 dB of cut. See below:



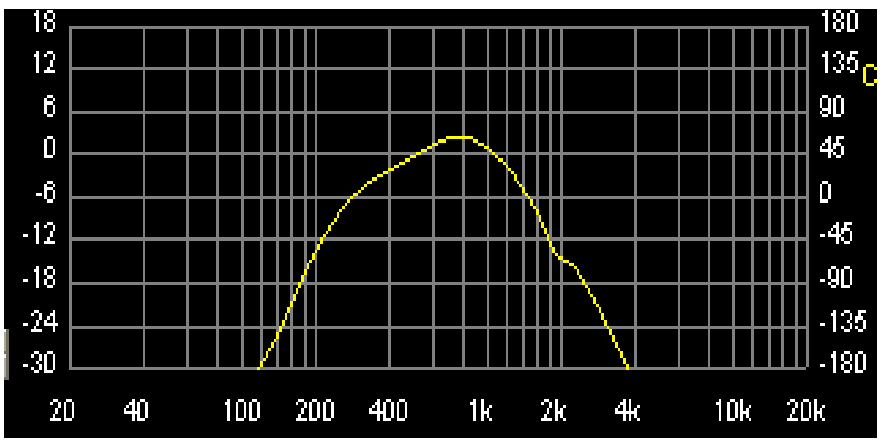
And the next PEQ will be dialed in at 135 Hz with a bandwidth of 0.250 and -3.5 dB of cut. See below:



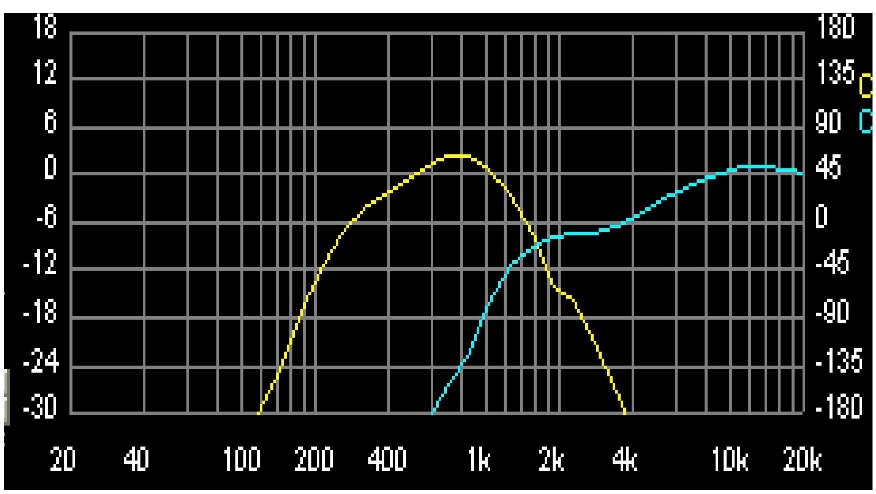
Next we are going to lift the region above 500 Hz a little. The center frequency of this dip looks to be at about 750 Hz, but I am going to shift it to 800 Hz because the area to the right of that is down a little more.



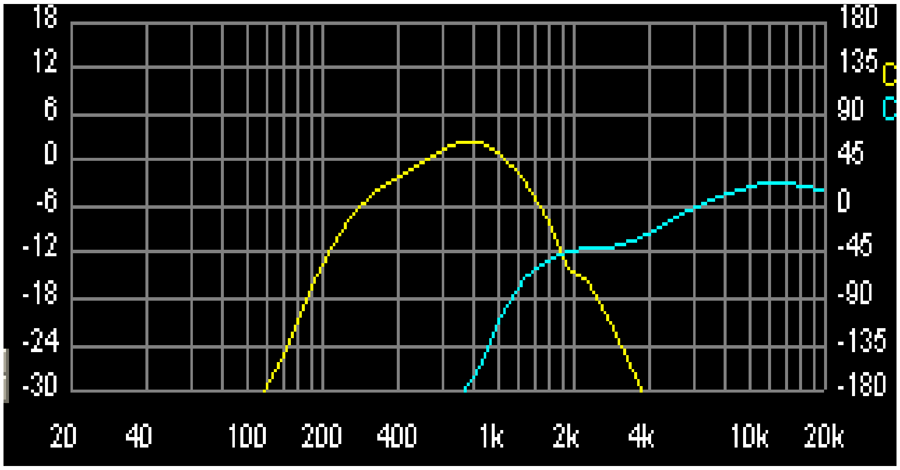
Next we will invoke the X-over filters. Since this is going to be a four way system, we will crossover the dual 10 inch BW's with a High Pass (Low cut) at 400 Hz employing a 4th Order Bessel filter function, and hand off between the 10's and the 44XT with a Low Pass (High cut) at 1600 kHz (for now) utilizing a 4th Order Linkwitz/Riley filter function . See below:



Here are the low pass and high pass bands together:



Now in looking at the 1 Watt at 1 Meter specifications of the two bandpass transducers, we can see that the Mids have a sensitivity of 101 dB at 1 meter, while the mid-range section of the 44XT has a rating of 111 dB. We have already reduced the sensitivity of the mids by -6 dB with a PEQ centered on 3 kHz. So I will reduce the output of the X-over module by an additional -4 dB:



Now we take into account the specifications of the QW-215F enclosure:

QW-215F Spec Summary

Frequency response, 1 meter on-axis, swept-sine in an anechoic environment:
47 Hz to 1.5 kHz (±3 dB)
Usable low frequency limit (-10 dB point):
34 Hz
Power handling (both woofers driven):
1200 Watts continuous
2400 Watts program
4800 Watts Peak

Sound pressure level, 1 Watt, 1 meter in an anechoic environment (both woofers driven):
100 dB SPL, (2.0 Volt input)
Maximum sound pressure level (1 meter):
131 dB SPL continuous
137 dB SPL peak

Recommended Amplifiers:

CS-2000	in Bridge Mode	(2150 Watts)
CS-4080HZ	in Stereo	(2040 Watts)
CS-4080HZ	in Stereo w/1 Spkr per channel	(2500 Watts)
PRO 9200	in Stereo	(2200 Watts)
PRO 7200	in Bridge Mode	(3300 Watts)

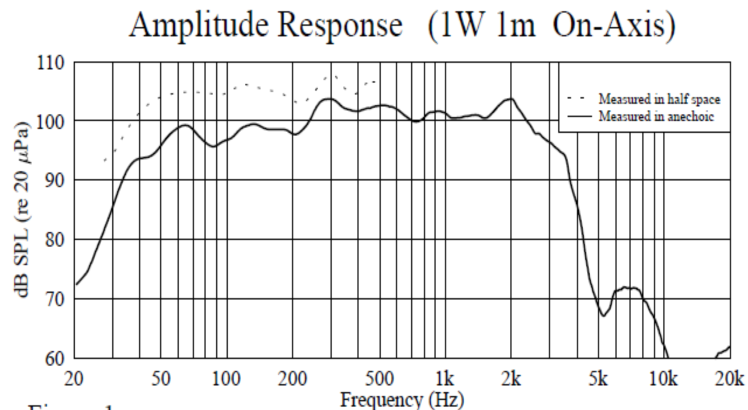
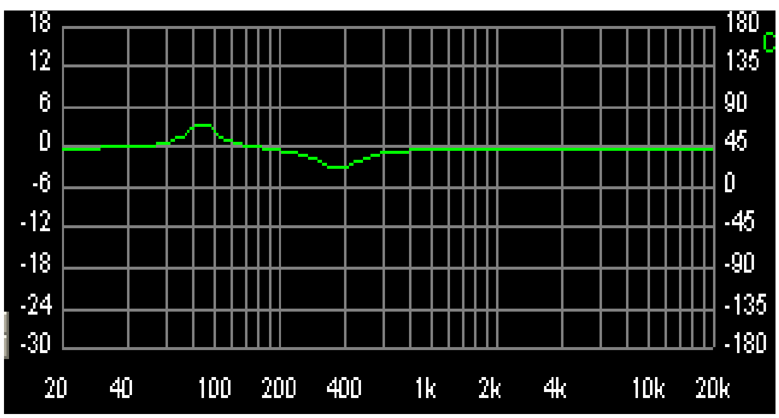


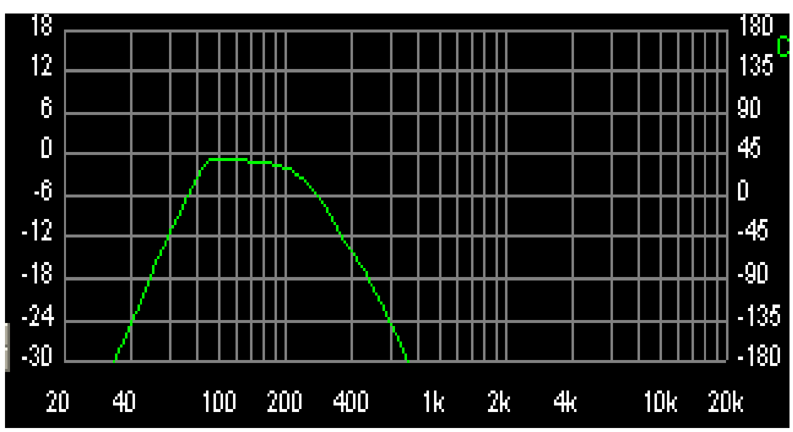
Figure 1

QW-215F

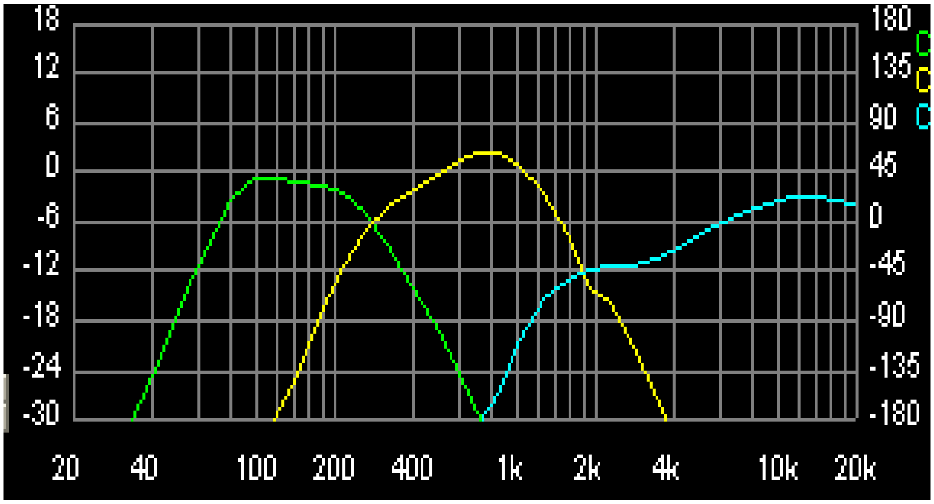
We are going to cross over this enclosure as a Low Mid bandpass. Now since this 215 enclosure will probably be on top of the QW-218's, we will go by the solid line, as it won't be in the immediate presence of the floor. So I want to first remove a bump above 150Hz. Since the upper frequency bump meets the 100 dB line at the 250 Hz and 500 Hz points, I will center the next PEQ at 375 Hz with a bandwidth of 0.707 and -3 dB of cut. I will then add a small boost at 88 Hz with a bandwidth of 0.5 octave and +3.5 dB of boost. See below:



Next we invoke the X-Over module with 4th Order -24 dB per octave Linkwitz/Riley filters at 80 Hz and 315 Hz:



Now let's turn on all three bandpasses:



Next we go to the QW-218 and examine this enclosures specifications:

QW-218 Spec Summary

Frequency response, 1 meter on-axis,
swept-sine in anechoic environment:
Anechoic: 47 Hz – 1.5 kHz (±3 dB)
Half-space: 40 Hz – 1.5 kHz (±3 dB)
Usable low frequency limit
(-10 dB point):
Anechoic: 34 Hz
Half-space: 30 Hz
Power handling:
(Both woofers driven)
1600 W continuous
3200 W program
6400 W peak

Recommended Amplifiers:
CS-3000 in Brdige Mode (3100 Watts)
CS-4000 in Bridge Mode (4000 Watts)
PRO 7200 in Bridge Mode (3300 Watts)

Sound pressure level, 1 Watt, 1 meter
in anechoic environment:
Anechoic: 100 dB SPL, (2 V input)
Half-space: 106 dB SPL
Maximum sound pressure level
(1 meter):
Anechoic:
132 dB SPL continuous
138 dB SPL peak
Half-space:
138 dB SPL continuous
144 dB SPL peak
Box tuning frequency:
36 Hz

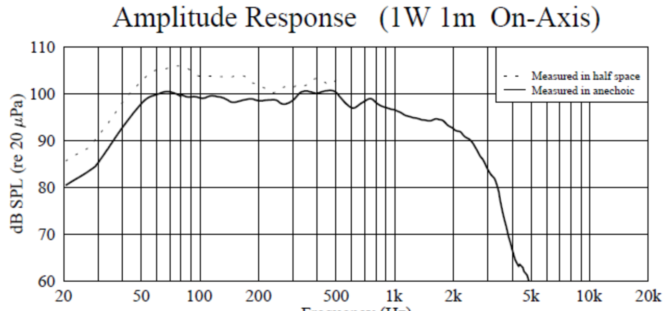
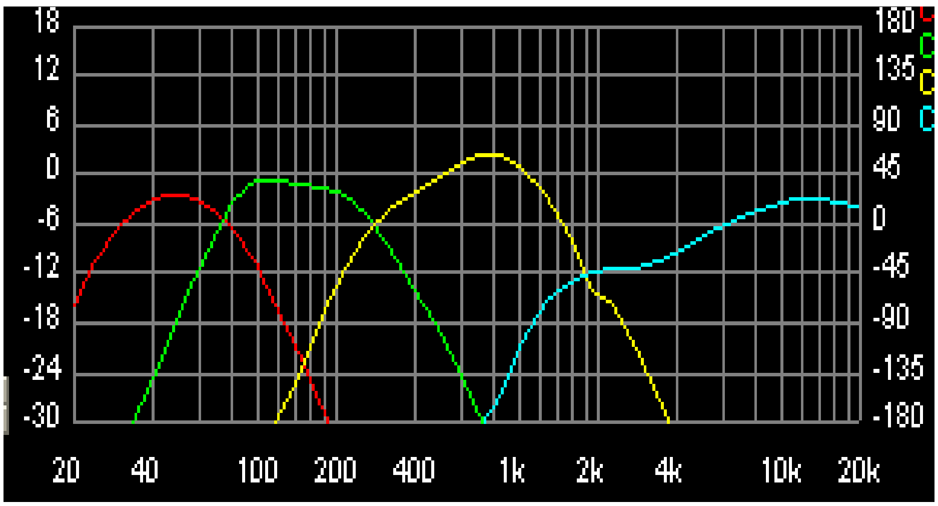


Figure 1 Q-Wave 218

The above frequency response looks pretty smooth, so we will simply invoke two filters using a 4th Order Linkwitz/Riley at 30 Hz and 80 Hz:

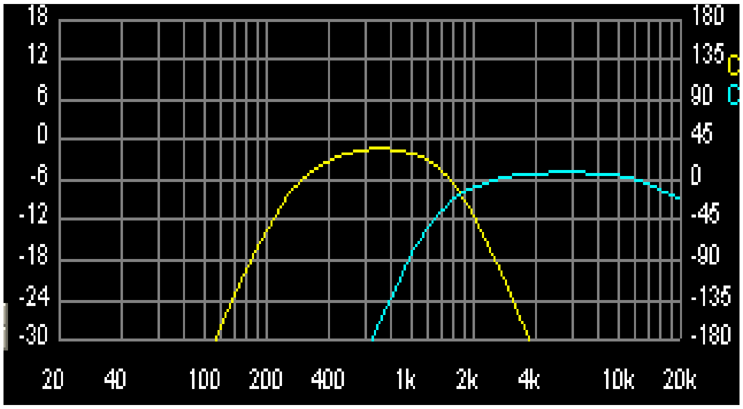


Now it looks as if the QW-218 low frequency enclosure is down, but remember that this enclosure will probably be on the ground where it will couple and play louder. Also, I recommend 2 x QW-218 (actually 2 x VR-218's) to give this system the ability to perform at the best possible level for the low frequencies.

Now the only remaining parameters that I can NOT deduce from looking at the specifications is the necessary delay setting to align the acoustic centers of the transducers. In looking at the QW-1, the Mid-range horn is longer than the length of the 60 x 40 CH-642qt horn that has the 44XT compression driver mounted on it. So that tells us that the high frequency bandpass (Output 4) is going to need some added delay. Now above in the specifications for the QW-1, the **time offset** is listed as 0.813 milliseconds. However that spec is obtained from an ETC (Energy/Time Curve) of the impulse response of the transducers in each bandpass (Horn loaded 10" BW's and 44XT on Quadratic Throat CH-642qt). This tells us that in the raw transducer state, the Highs are 0.813 mSec ahead of the Mid's in propagation time. However the actual delay setting will be more than this due to the group delay added by the X-Over module filters and the PEQ's. More on this in Part Two, where I will show you a method that will allow you to align the acoustic centers.

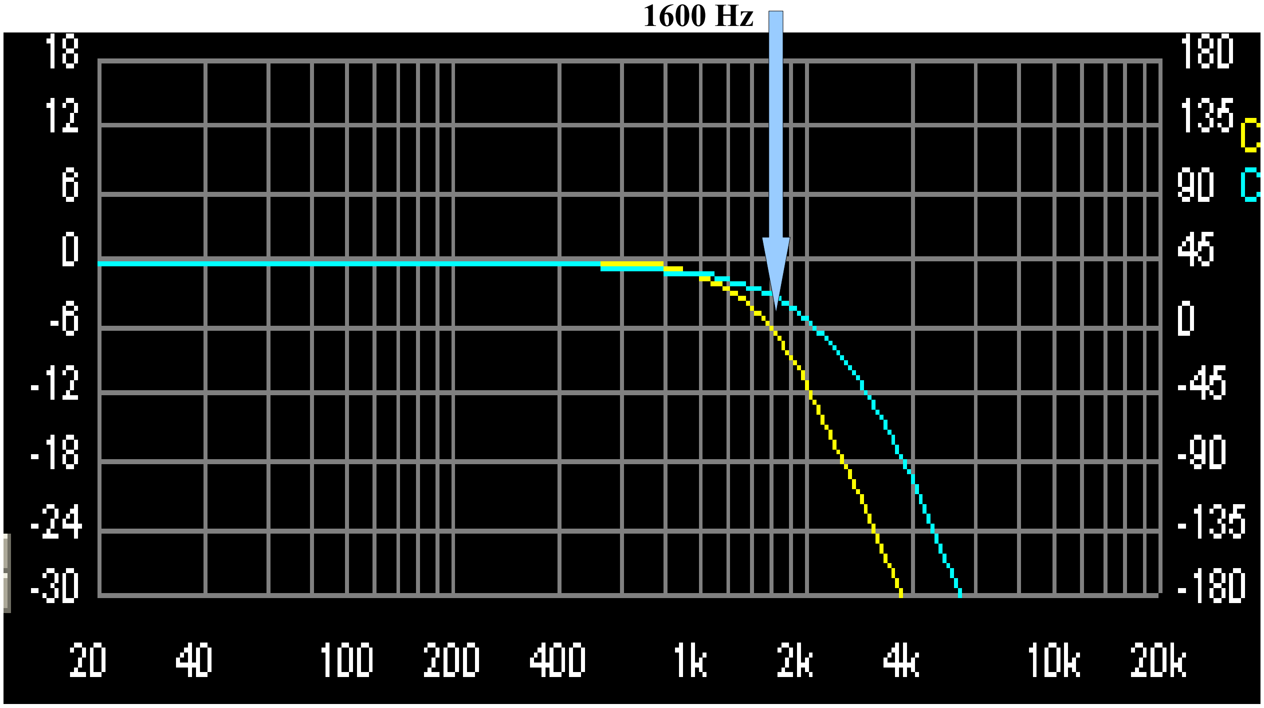
PART TWO Acoustical Center Alignment

In the graphic below it is difficult to tell what the actual electro-acoustical point is due to the under-lapped X-Over points of 1600 Hz and 2000 Hz between the Mids and Highs. In looking at both bandpasses with the PEQ's bypassed we see:



However I believe now is a good time to explain further the differences between the Bessel Filter and Linkwitz/Riley filter functions. So I will change the settings we came up with at this point to show more details of the two filter functions and their respective differences. In order to do this I will show a generic Low Pass X-Over of 1600Hz using a both a 4th Order (-24 dB/Octave) Linkwitz/Riley filter and a 4th Order Bessel filter.

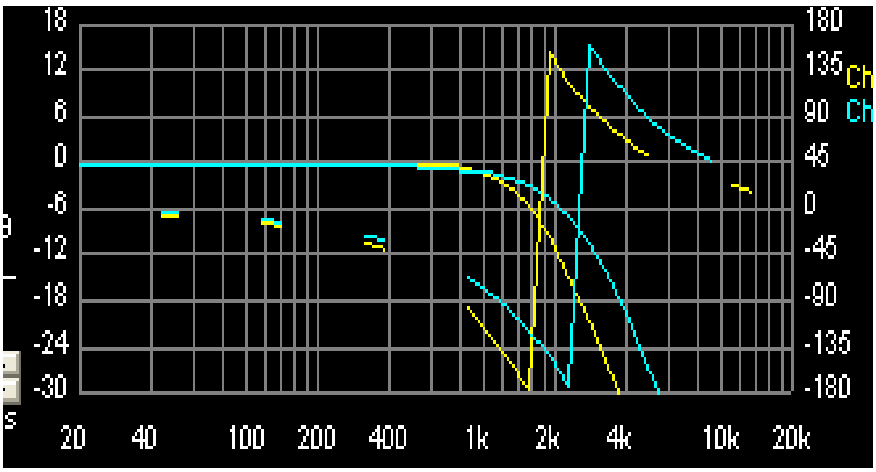
1600 Hz



I cropped and enlarged the above graphic so you could better see the display and intersection points. Notice the decibel scale on the left. The Yellow trace is the 4 th Order Linkwitz/Riley Low Pass filter set for 1600 Hz. The Blue trace is the 4 th Order Bessel Low Pass filter also set to 1600 Hz.

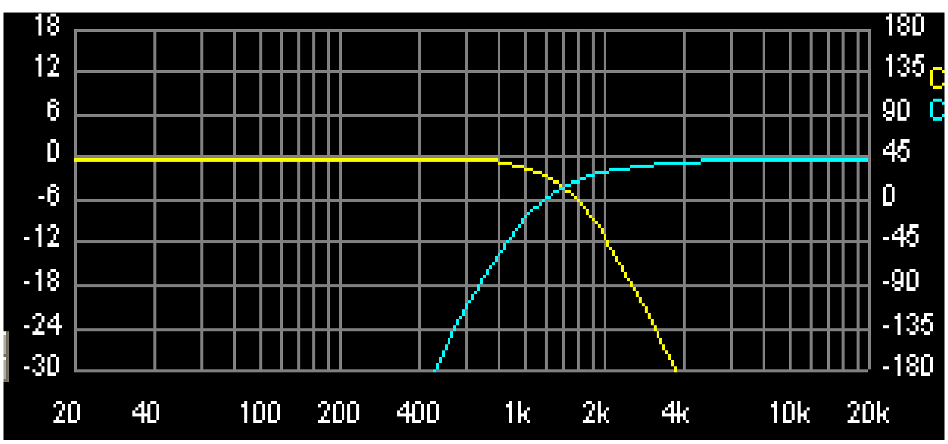
The definition of the X-Over point when using Linkwitz/Riley filters is that the crossover frequency is at the -6 dB down point, as illustrated above. A Bessel filter is a very special animal, in that unlike Butterworth or Linkwitz/Riley filers that essentially have the same component values repeated for each pole of the filer (each pole is also referred to a an Order), the Bessel filter has differing values of components in each pole, resulting in a staggering of the filter poles. For our purposes the Bessel filters have been "normalized" for the X-Over frequency to be defined as the -3 dB down points. The Blue trace of the Bessel filter shows the -3 dB down point at 1600 Hz. The knee of the initial roll off is also a little gentler, but if you look further out, you can see that it does fall into the -24 dB per octave roll off slope.

Now we will look at the phase response of these two filters:



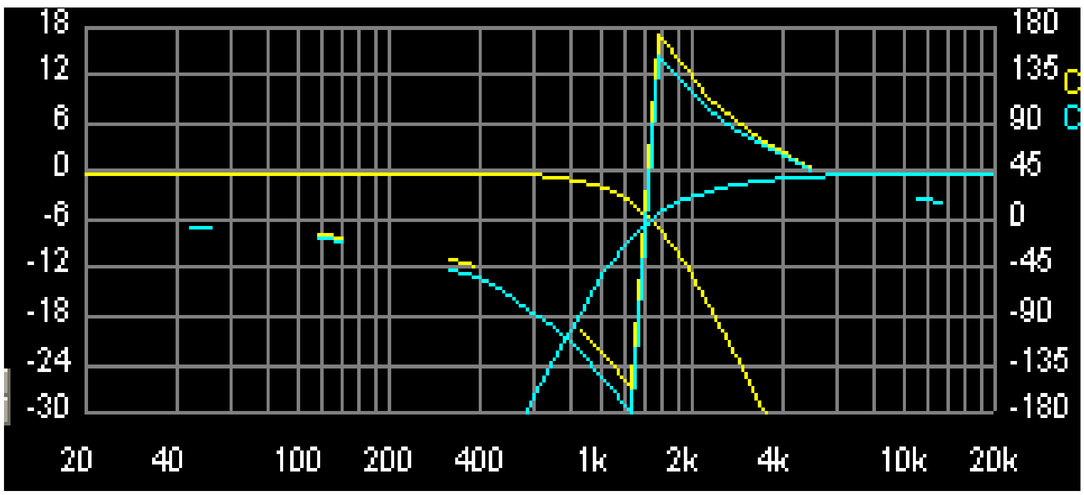
Notice above that the Blue trace indicates that the Bessel filter function also has less phase shift within its bandpass. Where the Yellow trace of the Linkwitz/Riley shows -180 degrees at about 1600 Hz, the Blue trace of the Bessel reaches -180 degrees further out, at about 2200 Hz.

Now let's make the Bessel filter a High Pass at 1600 Hz:

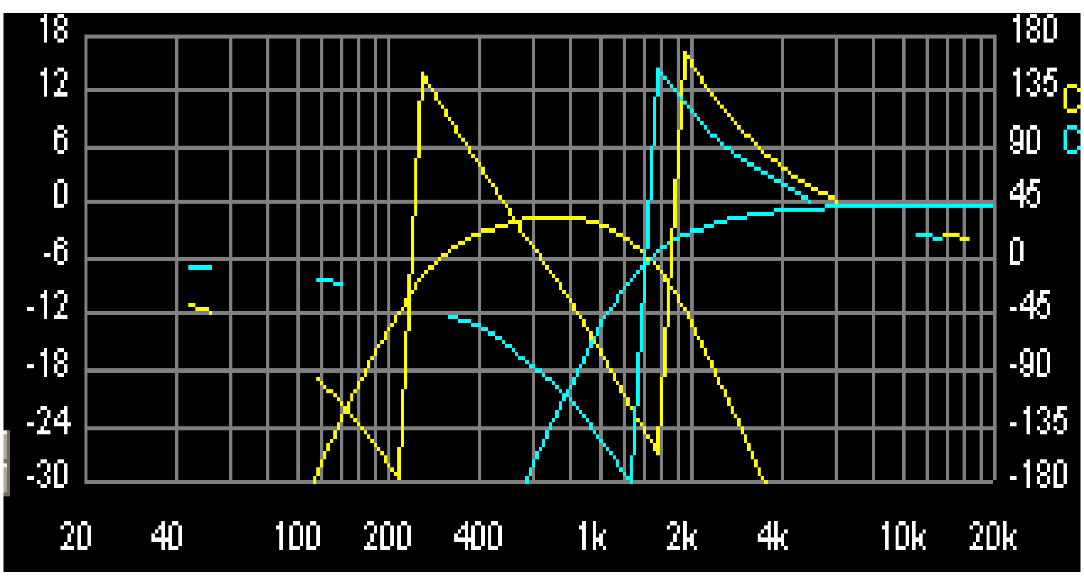


Notice that the Bessel is -3 dB down at 1600, and it is -6 dB down at 1200 Hz, while the L/R is of course down -6 db at 1600 Hz.

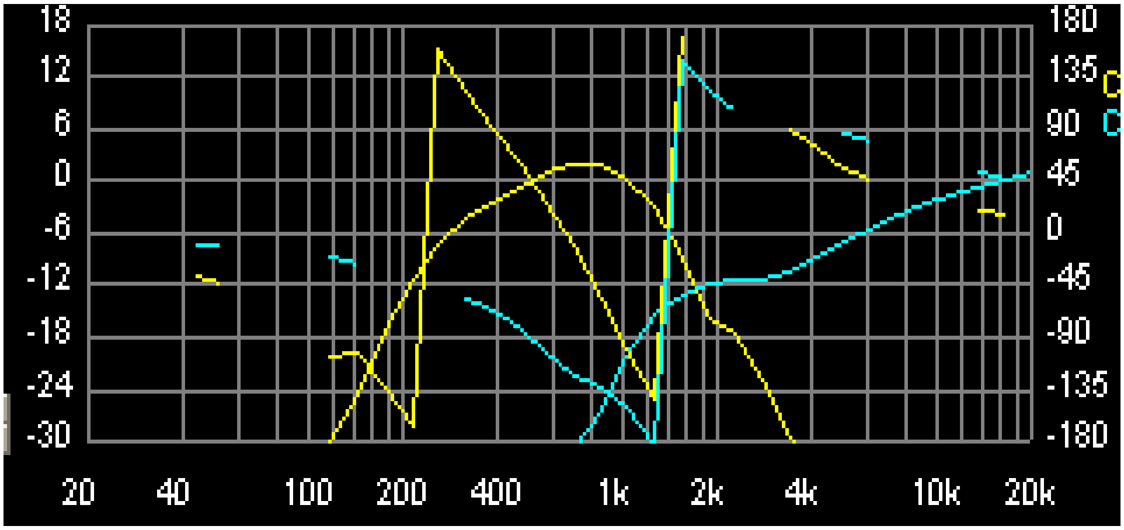
Now when you align the acoustic centers within the DSP you will then have a coincident source for both transducers. When two sources of sound are absolutely coincident, they sum together to produce a +6 dB increase in SPL. When two transducers are not aligned as far as their acoustical points of origin, then they are said to be non-coincident. Two non-coincident sound sources will only add to produce a +3 dB increase in SPL. Since we are going to align the acoustical centers via the delay capabilities in the digital signal processor, we want to under-lap the these two X-over points (by their definitions), so that they then intersect at the -6 dB down points. So I will move the High Pass Bessel filter up to 2 kHz, and the Linkwitz/Riley Low Pass filter down to 1500 Hz, to allow them to overlap at the -6 dB down points. See Below:



Note above that the -6 dB down points are now intersecting at 1500 Hz, and also notice that the phase is also very close at this intersection point. So the X-Over is pretty much in alignment in the electronic domain. But remember, as pointed out earlier, we also know that we have very different acoustical points of signal origin with the much longer Mid-range horn than High frequency horn in the QW-1F. However we now have to restore our High and Low cuts, but I will hold off for now on the PEQ's, and sensitivity settings. See Below:

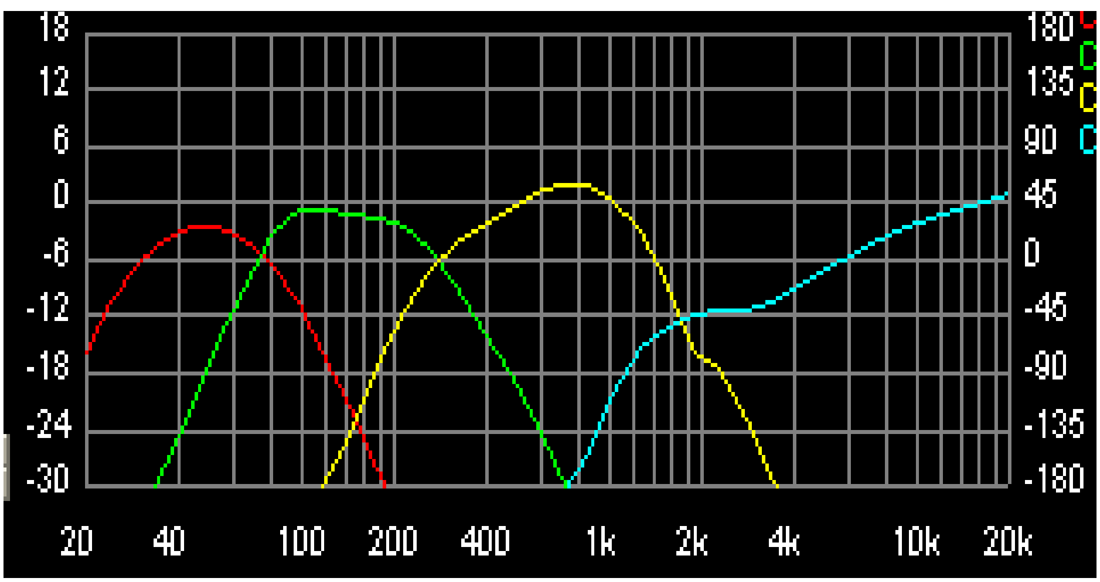


Now with just the Low and High cuts restored in each bandpass, notice above that we have moved off of our phase alignment. Now I will re-invoke the PEQ's and level settings:



Notice above that the phase is now better. It is very important to understand that every DSP function that you introduce in each output will affect the response in the phase domain. Did I not tell you earlier that addressing these bumps in the response, even though they were beyond the X-Over points, would improve the response in the phase domain? Now if I were able to measure the actual transducers involved, this would be easier. I really do love to teach this stuff.

So now I will turn on all of the the other X-Over filters, and their PEQ's:



So all of the above was the procedure to come up with the base line X-Over filters and PEQ settings. The next quest would be to come up with the proper alignment of the acoustical centers for each of the transducers in the system. Now there are several measurement tools available some of which include TEF, MLSSA, and Smaart Live. I have experience in using the TEF measurement system (Time/Energy/Frequency). I have used the TEF-10, TEF-12, and TEF-20 systems. All of these measurement systems employ dual channel FFT (Fast Fourier Transform) computer measurement systems. The TEF program is also referred to as Time Delay Spectrometry. The TEF is a great tool in the research lab, but it is a bit time consuming to implement in measurements in venue environments. The Smaart Live measurement system has the ability to make relatively fast measurements and properly implemented, can make measurements independent of the program material. In other words even with music, it can use the dual channel FFT function to display something called a Transfer Function. The **transfer function** allows you to see how the signal that left the mixing console differs from the signal that arrives at a measurement microphone some distance away. The Smaart Live measurement system makes the lowly RTA (Real Time Analyzer) simply **obsolete**.

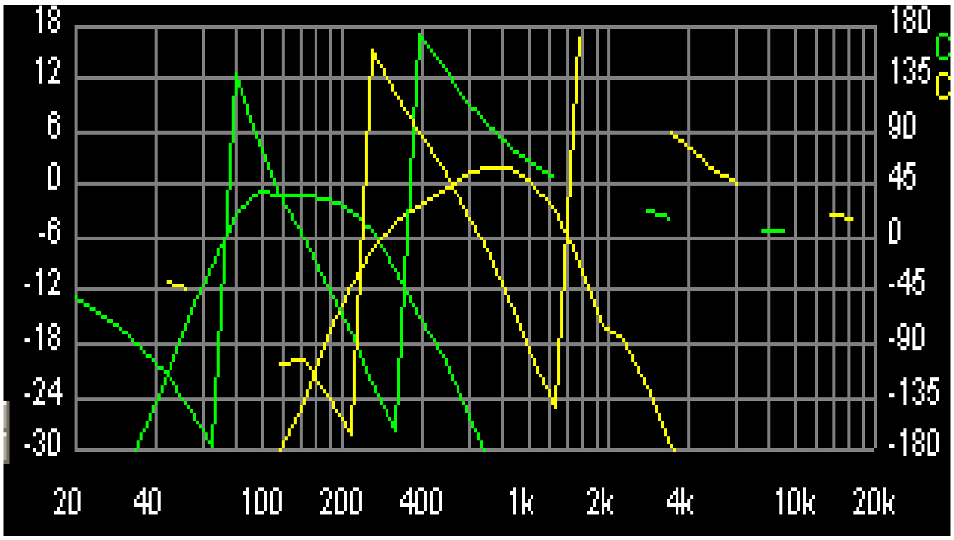
Now I realize that many band or regional sound providers do not have access to these measurement tools. However I highly recommend that you check out and consider the Smaart Live system. They make the software available **free** for 30 days. Here is a URL: http://software.informer.com/getfree-download-sia-smaart-live

With that said, I will now tell you how you can align the acoustic centers of this system using a sinewave generator. If you do have a function generator that does not include a frequency counter I would suggest that you also have a frequency counter. I also can recommend the Loftech TS-1 test system, that is a sinewave generator, a frequency counter, and decibel reading meter all in one.

Those of you with VSX-26/48's or a Digitool MX can use the internal function generator to obtain the test signal. I would suggest simply assigning the generator directly to an output that you are not using and connecting it to a channel input of your mixer. You would then set the oscillator to generate a 1.5 kHz tone and input it into your system. You only want to measure the left or right side. You can do it somewhat by ear by standing on axis and listening, but it is best to use a measurement Mic on axis. Caution; do not try and use your ears to judge the levels if you are NOT directly in front of and on-axis to the system. Due to the nature of standing waves, you can be fooled if you are not listening directly on-axis. This is also why a measurement microphone is best. The measurement Mic (such as the Peavey PVR-2) should be connected to a channel of you mixer that is NOT assigned to any outputs; use that channels PFL function to read the level of this test Mic. Place the Mic about 6 Ft away from the QW-1F directly between the Mid and High frequency horns; this will help get the polars aligned as well. If you have an oscilloscope, this can also help you see the null with a greater degree of accuracy, by driving the oscilloscope from the insert out of the mixer channels preamp that has the measurement Mic plugged into it. We will then tell you to listen to or measure Output 4 first. Next **mute Output 4** and **unmute Output 3**. The levels should be close to the same. Move the Mic forward or backwards a small distance to find a peak in the standing wave.

Now go to the **delay** section of **Output 4** and **reverse** the polarity, and also **unmute Output 4**. Now while either listening or measuring on axis begin to **add delay** to Output 4. There will be a point where the signal will diminish or get soft, and going further it will begin to get louder. Well I know from experience that I believe you will get a null in level with just a little delay, but if I remember correctly the proper delay setting will be more than the initial null, as remember the Mid horn is much longer, and if you null too soon you will be nulling the 2nd cycle of 1.5 kHz from the Mid horn to the 1st cycle of 1.5 kHz from the High frequency horn. So go for the 2nd null. One you find the 2nd null or minimum amplitude, simply make the **phase** of **Output 4 normal**, and you have the correct calibration.

Since you delayed the High frequency horn to the longer Mid horn, you must then align the QW-215F to the Mid of the QW-1. Here is a trace of the response and phase of Output 2 & 3:



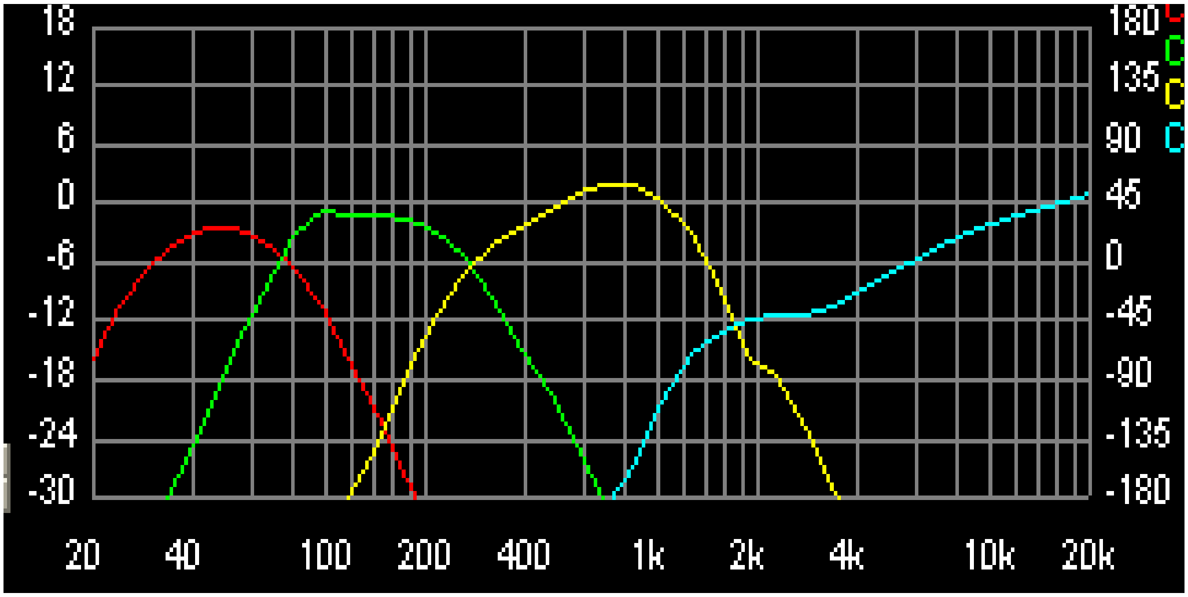
I also moved the Low Pass (High cut) X-Over point of Output 2 down to 300 Hz to coincide with the -6 dB down point of Output 3, which I chose to be a Bessel filter which is only down -3 dB at the designated X-over frequency of 400 Hz, and we now have the -6 dB down points intesecting at 300 Hz. I also re-inserted the PEQ's. Since the phase is fairly evenly spaced, we will use the intersection frequency of the two bandpasses at 300 Hz.

So the next thing that we are going to do is introduce a 300 Hz tone into the system. The measurement or listening test as before, will be measured on-axis and between the two enclosures at a distance of 10 Feet (we want to be a little more than 2 wavelengths distance away from the enclosures so as to measure the Far Field spherical wave, and not the particle velocity that is predominant in the Near Field). Since we are referencing everything to the longer Mid-horn of the QW-1, we will want to delay Output 2 or Low's that are going to the QW-215F. So **reverse** the polarity to **Output 2** and proceed with the null test. When you have determined the null point change the polarity of **Output 2** to **normal**. The null point should be greater than the delay that you dialed into Output 4.

To conduct the null test between the QW-218 Sub and the QW-215 Low Mid, use an 80Hz sinewave, as these two both use L/R 24 X-Over filters at 80 Hz. So we now want to delay the Subs in Output 1 to the QW-215F. Next, **reverse** the **polarity** and **mute Output1**. **Unmute Output 2.** Place the measurement Mic on the floor at a distance of 20 to 30 Ft away, and then move the Mic forward or backward a few feet to find the peak. Now **unmute Output1,** and begin to add **delay** to **Output 1** until you get the minimum null. Then make the **polarity** of **Output 1 normal.**

Now you should be pretty well calibrated. In few weeks when I get a chance to actually measure a complete QW system, I will publish the presets, and we can see how close we came.

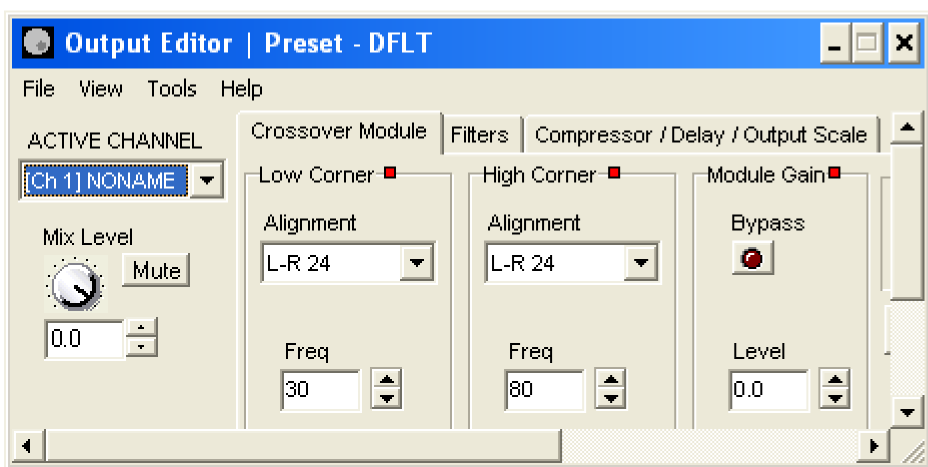
Here is the graph of the complete X-Over design for a DSP:



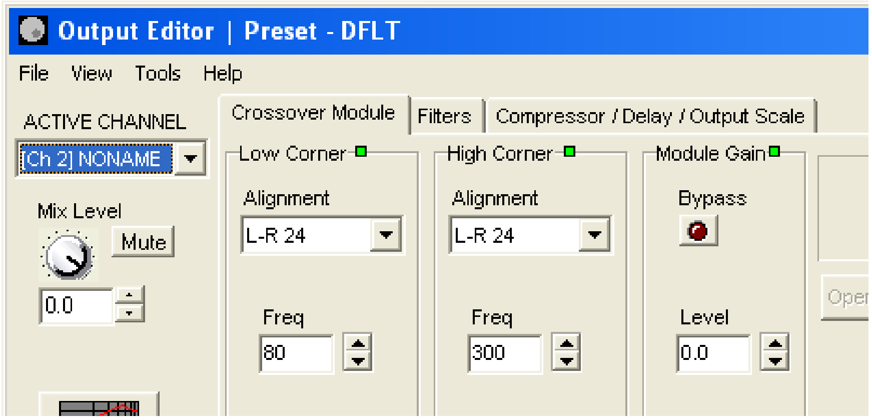
Of course I used the Digitool MX loudspeaker management system as the DSP device for this exercise.

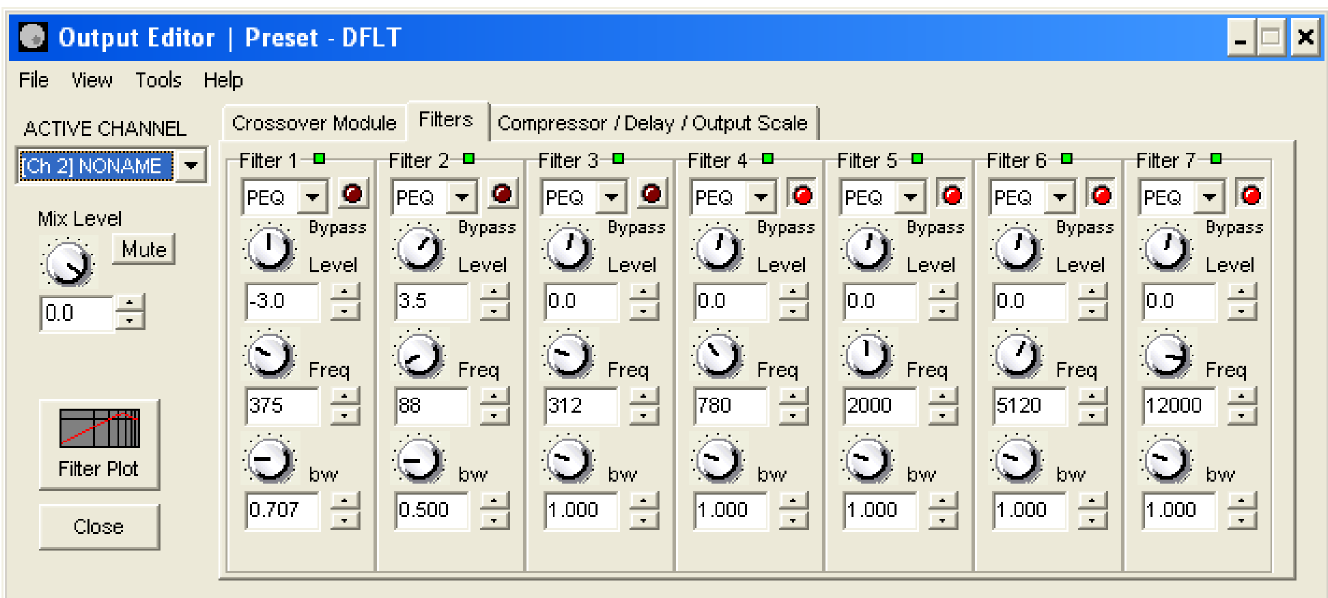
Below are the final X-Over and PEQ settings:

Output One X-Over Module (No PEQ settings):

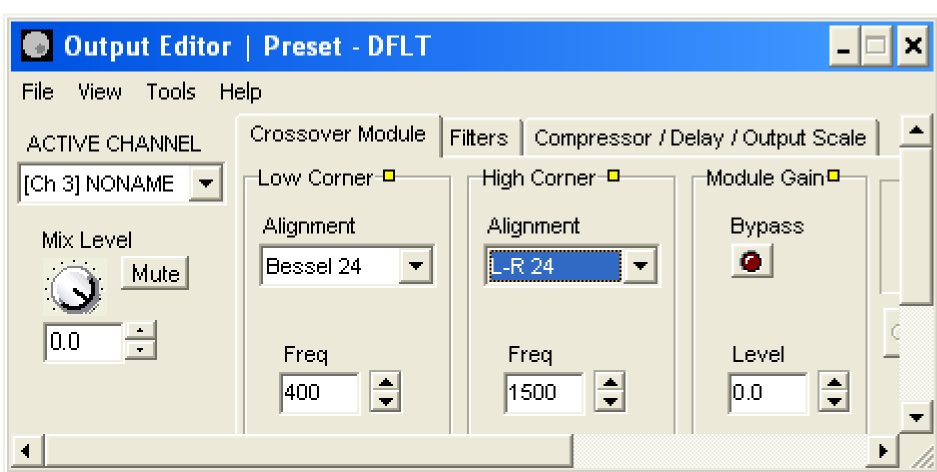


Output 2:





Output 3:



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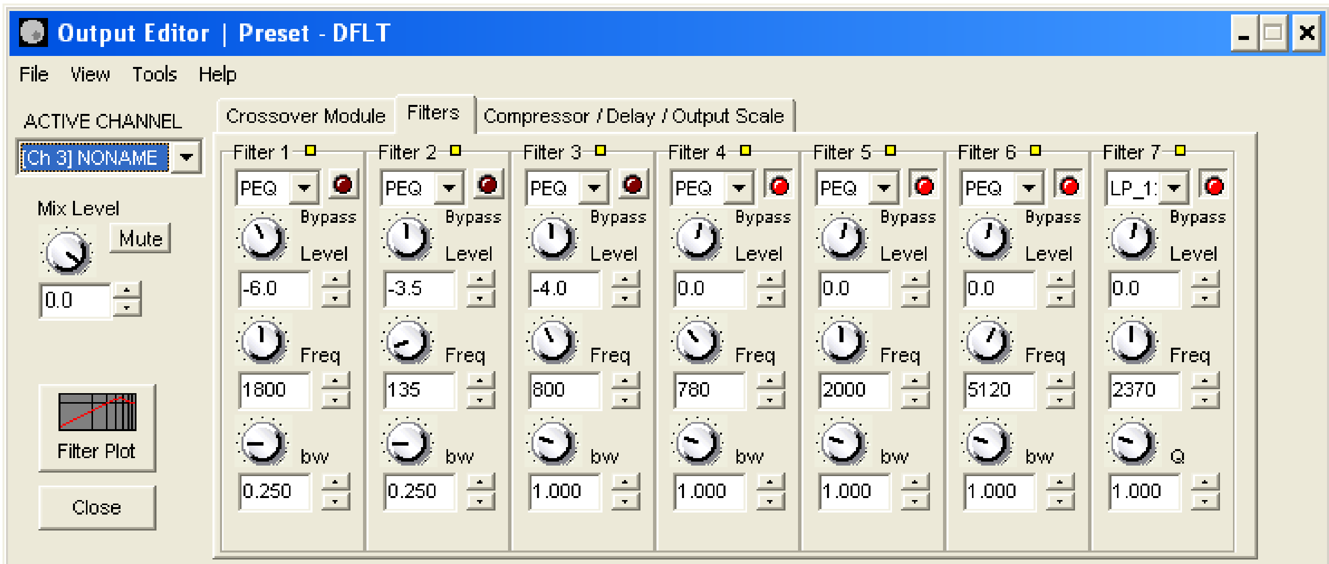

Output 4:

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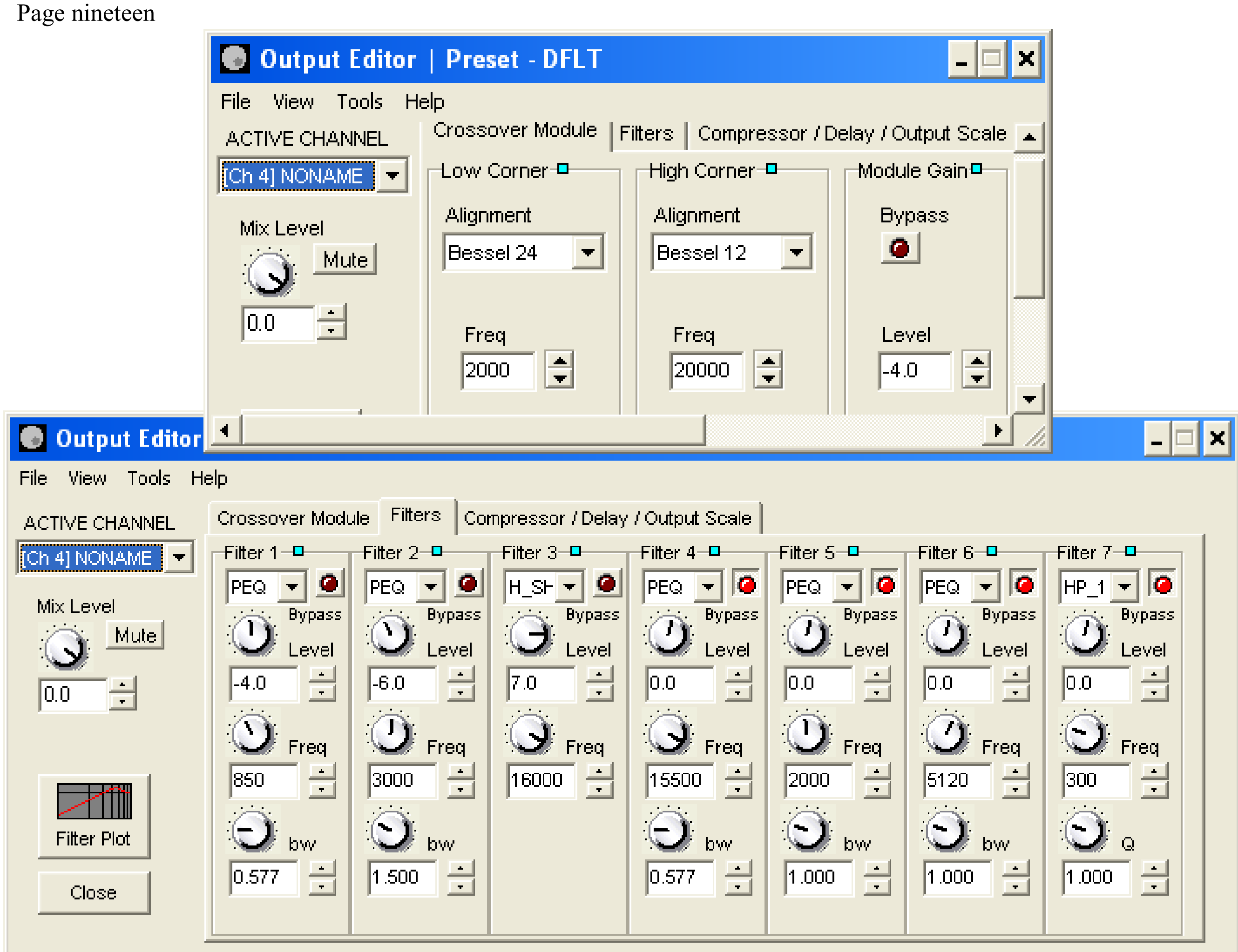
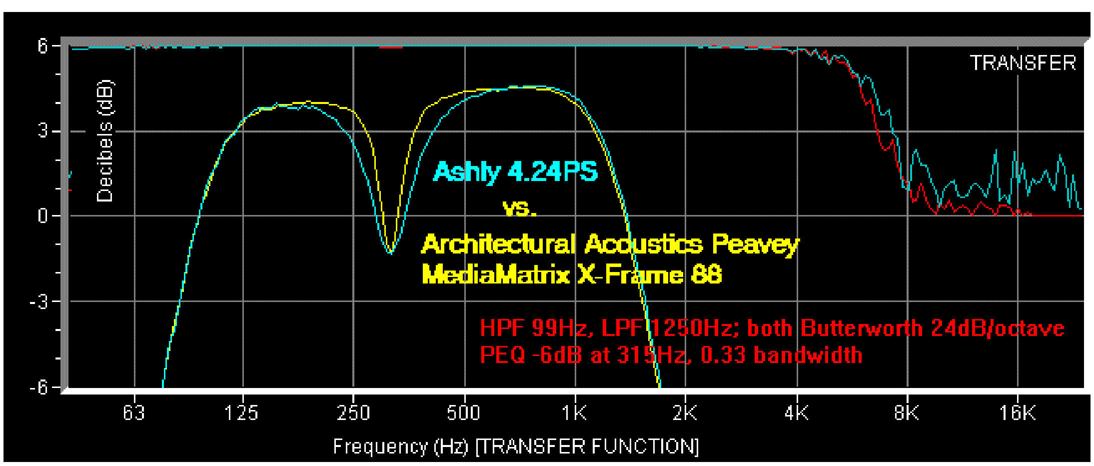


One again professional sound reinforcements system do need to be **"engineered"** in order to get the best possible performance. Part of that engineering also means that in order for a published DSP preset to work, all amplifiers in the system need to have the same Voltage gain. I have another white paper on power amplifier gain that establishes the need and importance of having matched gain.

I would also like to again encourage those of you who are trying to be the best and looking for ways to grow and learn even more about this wonderful world of sound, to look into the measurement systems available to us. The Smaart Live program is very affordable, less than a decent outboard quality stereo reverb. If you have the measurement tools and the skill and experience to employ the measurements when programming your choice of DSP product, you will have an unprecedented opportunity to calibrate your sound reinforcement system to a degree of precision and accuracy never before possible. Factory settings and presets are available from Peavey for our products and from other manufacturer's for their products. However there are still differences in the tolerances of the transducer devices themselves, and we now have the unparalleled unique and novel opportunity to make up for the variances in these product tolerances themselves.

This paper was intended to cover the basic process of using the information obtained from thoroughly documented loudspeaker model specifications to program a digital signal processor loudspeaker management system. However I feel that I must point out at this time that DSP presets are generally not truly or equally compatible on different DSP engine platforms. Why? It would take a very long and involved explanation to thoroughly explain this. Much has been written on this subject in the Pro Audio industry. So once again; depending on the DSP device and platform that you try and program with another platform's preset, you may or may not get the same performance as you would with say a Peavey VSX-26, VSX-48, or Digitool MX. In other words YMMV (Your Mileage May Vary). Although you should be in the ballpark, there is a huge difference between just being at some arbitrary place in the ballpark, than being precisely on the 1st Baseline, Section A, Row 12, Seat 5.

As just one example; below is a graphic that shows the transfer function of two different DSP platforms, programmed with the same values;



One of the problems with incompatibility between different platforms, has to do with a technical fact, and that is that some electronic engineers and digital programmers can not agree on the definition of the specification "**Q**". In physics Q is defined to be 2pi * ratio of total energy stored, divided by the energy in one cycle. In filter technology a Q less than (<) 0.5 is considered overdamped and a Q equal to 0.5 is critically damped, and a Q greater than 0.5 is underdamped. Since a filters ability to resonate is dependent on it's Q, an underdamped filter will “**ring**” or resonate with a greater amplitude than one that is overdamped

Some of the other variables and unknowns when dealing with different digital platforms are; different A/D and D/A converters, differences in sample and bit rates, Fixed Point Arithmetic versus Floating Point, Infinite Impulse Response (IIR) filters versus Finite Impulse Response (FIR) filters. There are more complex issues that vary due to what DSP engine is under the hood. Specifications vary with chip sets. Is the engine under the hood, a Motorola M50006, an Analog Devices SHARC or Black Fin, or some other? All of these considerations contribute to the overall differences, but the most significant difference comes down to the code itself. Some code writers are better than others and thus some code is more eloquently written. Code can be written in Assembly, 'C', or even BASIC. Many program functions are written as subroutines in Assembly and then programs written in 'C' may access the smaller Assembly code. So there are Hybrids. Digital Audio has made many magnitudes of improvements since it was first introduced to the Consumer and Pro Audio markets.

We recently had a poster complain in another thread in a section of this website, how he hated digital audio. I believe that perhaps this individual has not moved into the 21st century and is stuck in the 1980's mentality regarding digital technology. Today's digital processors while not perfect, are far better than their predecessors.

The future of digital audio is in the software itself. In recognizing the limitations of earlier digital circuits and program renditions, they are now writing software with implementations within the code itself that overcomes problems recognized in earlier digital products. There are new techniques and terminology that combine analog circuits with DSP algorithms. We now have single bit multi-rates, Delta Modulation, Delta-Sigma converters. Who would have thought in the beginning that by adding additional noise to a signal (dithering), that you could actually obtain a better sample. Today's digital engines apply convolutions of impulse trains of unity amplitude and zeroth-order hold (1/sinc[x]), combining the use of digital low pass filters and decimation, as well as cultivated techniques involving subtractive dithering. (And I thought a Dither was an Appalachian musical instrument, and that Group Delay meant that the band would be late). We have moved far beyond the Paleolithic period of digital audio technology. Either wake up and plug-in the digital coffee pot, or simply put the other foot in the analog tar pit.