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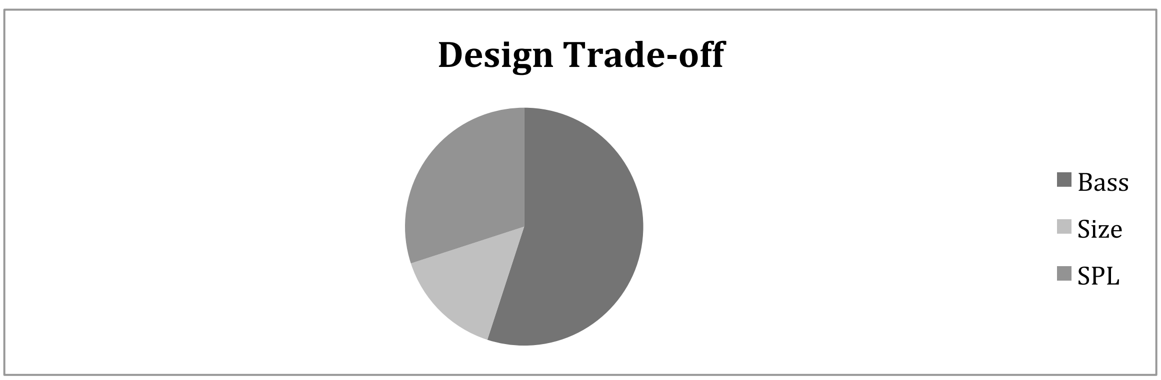
Design Statement

The overall objective for my design was to create high integrity studio monitors at a reasonable cost. I decided to go with a 2-way system that should be relatively portable. There are multiple factors that went in to the final design objective that will be discussed. In order to best organize and prioritize my design objects, I weighed out which characteristic of my speakers would be most necessary for a studio monitor system. The main trade-off is between system sensitivity, size, and frequency response[1]. For studio monitor applications, it is more important to have an extended frequency response with high integrity, meaning I should be able to make compromises with the size of my enclosure and with my sensitivity.



Because my size has least importance in my design trade-off, I can look into transmission-line enclosures and vented enclosures. While size is the least important factor in the trade-off, I'd still like to have a reasonably portable system. This would make a transmission-line impractical. A vented enclosure seems the most practical because I'm able to extend my bass response while maintaining

[1] Murphy, *Intro to Loudspeaker Design,* 56

efficiency[2]. The compromise being made with a vented enclosure is the accuracy of the transient response.

After establishing a foundation for prioritizing my system, I need to specify the needs of the system. Starting with my most important factor of frequency response, I did research to determine where specifically I could make a compromise in frequency response without loosing all low-end information. . The human audible range spreads from 20 Hz to 20 KHz. Because I'm designing a 2-way system, I won't have the added reinforcement of the subwoofer to extend my frequency response. This means I have to determine a reasonable roll off frequency for the woofer I want to use. I used Logic Pro and a mix I was working on and set an equalizer to the stereo output of the mix. Here I was able to determine which frequencies would be the best compromise for a roll-off without loosing important information for mixing. I determined the absolute highest frequency for the woofer roll off should be 45 Hz. Anything above 45 Hz and I hear noticeable loss in the low end of the output. While John Murphy states in Introduction to Loudspeaker Design that in general studio monitors can have a bass roll-off around 30 Hz, I'm taking into consideration budget constraints and practicality limitations.[3]

As far as the upper range of the system, it's easier to achieve closer to 20 KHz and would be ideal for this system. Overall, my frequency response should range from less than 45 Hz to 20 KHz.

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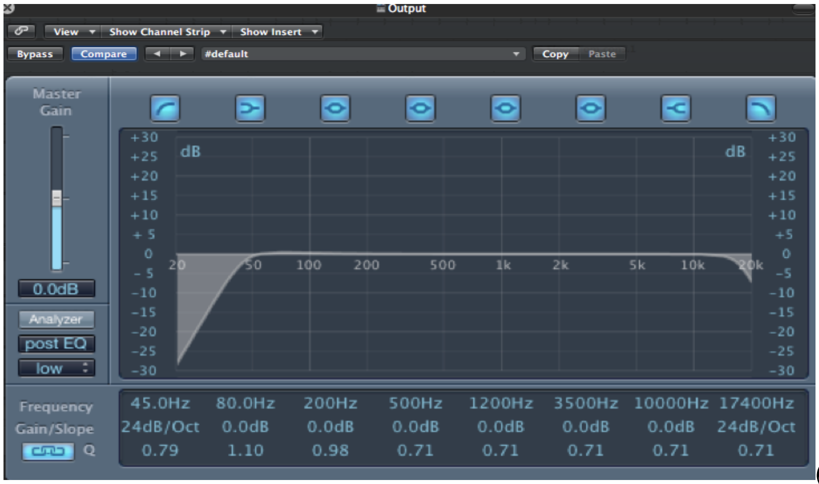
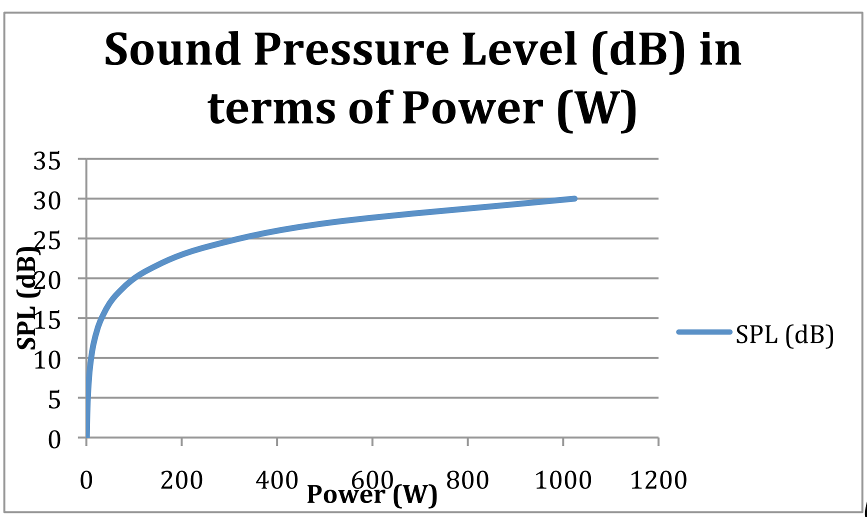
[2] Murphy, *Intro to Loudspeaker Design,* 27
[3] Murphy, *Intro to Loudspeaker Design,* 40



(Logic Pro output equalization meant to simulate ideal system frequency response)

After defining my frequency response, I need to define the sound pressure level sensitivity of my system. Here, I determined that my general mix level is at about 85 dB at 2 m. The 2 m distance results in a 6 dB drop. In order to balance for this loss, a power increase is required. Each doubling of the power supplied results in a 3 dB increase.[4]



(Inverse Square Law)

[4] Plummer, January 19, 2011

Ideally, my system should have a sensitivity around 90 dB 1W/1m. This means that at 2 m, the level is at 84 dB. In determining the power required I must also consider headroom. Generally speaking for most popular music, headroom of 14 dB is adequate. [5] This means I need to reach 99 DB (15 dB difference) making my power requirement 32 W.

As far as the size of the speaker enclosure, I initially set parameters to limit the size from becoming less portable. However, my finalized dimensions will be defined by how the woofer responds to the modeled enclosed volume. For now, it is just important to understand the ratio of height, width, and depth that best removes standing waves inside the enclosure. This ratio is known as the golden ratio and is defined in Introduction to Loudspeaker Design by John Murphy.[6]

V(B) Cubic FT	Height (In)	Width (In)	Depth (In)
0.05	7.15	2.73	4.42
0.1	9.01	3.44	5.57
0.15	10.32	3.94	6.38
0.2	11.36	4.34	7.02
0.25	12.23	4.67	7.56
0.3	13	4.96	8.03
0.4	14.31	5.46	8.84
0.5	15.41	5.89	9.52
0.6	16.38	6.26	10.12
0.75	17.64	6.74	10.9
0.9	18.75	7.16	11.59
1	19.42	7.42	12
1.25	20.92	7.99	12.93
1.5	*22.23*	*8.49*	*13.74*
1.75	23.4	8.94	14.46
2	24.46	9.34	15.12
2.25	25.44	9.72	15.72
2.5	26.35	10.07	16.29
2.75	27.2	10.39	16.81
3	28	10.7	17.31

[5] "Level Practices: The K-System."
[6] Murphy, *Intro to Loudspeaker Design,* 88-89

While at this point I don't know the exact box volume that would optimize the system, I do know that I want them to be relatively portable. This means I would like to keep my internal box volume at or under two cubic feet.

As far as enclosure type, there are many options available each offering different responses to low frequency.

The first option, a sealed box, is basically an extension on the idea of an infinite baffle[7]. The baffle is where the driver is mounted and when used alone is designed to separate the sound pressure generated in the front of the driver from the back. This idea would remove some of the cancellations that would occur if the driver were in free space. An infinite baffle would remove all these cancellations. Because the box is sealed, an air constraint is created causing the driver to be stiffer. This causes a higher resonant frequency of the driver. If the driver free air resonant frequency is matched to the air constraint of the cabinet, it's easier to generate an accurate transient response.

The quality factor, or sharpness of the resonance, is represented by the electrical-magnetic properties, the actual mechanical stiffness of the driver, and the air load of the driver[8]. The Qts, which is the total system quality factor as mentioned above, greatly influences the transient response of a sealed box system. As this quality factor increases, as does the transient decay time. As the quality resonance is increased, the transient resonance will as well as allowing a longer energy response time. This can be seen graphically below.

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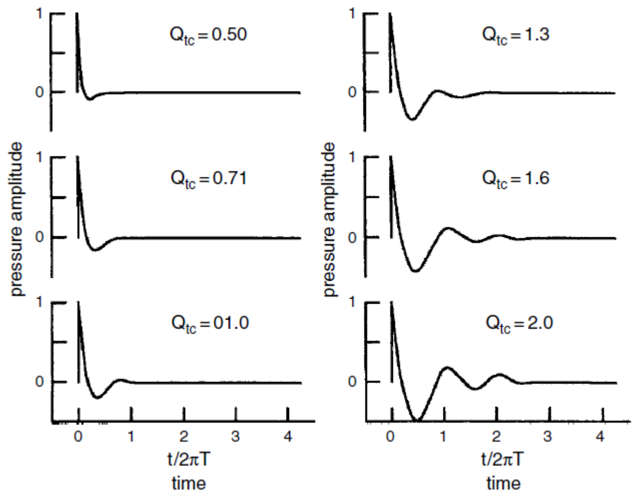
[7] Newell, *Loudspeakers*, 67
[8] Newell, *Loudspeakers*, 69-70.



(after small[2]) [9]

A Bass reflex design uses a port, or hole in the cabinet, to tune the cabinet's resonance. The port's resonance allows for the driver to use less travel distance and energy and still maintains a flat response[10]. When the frequency drops below that resonance, the port is designed to radiate the lower frequencies giving low-end support.

The graphs below show the frequency response of driver mounted in an open baffle, a sealed box, and a reflex box. Part A is optimized for the open baffle and part B is optimized for the reflex. With a properly optimized system, each of the three options can produce a fairly similar frequency response creating a design trade-off depending on my own preferences.

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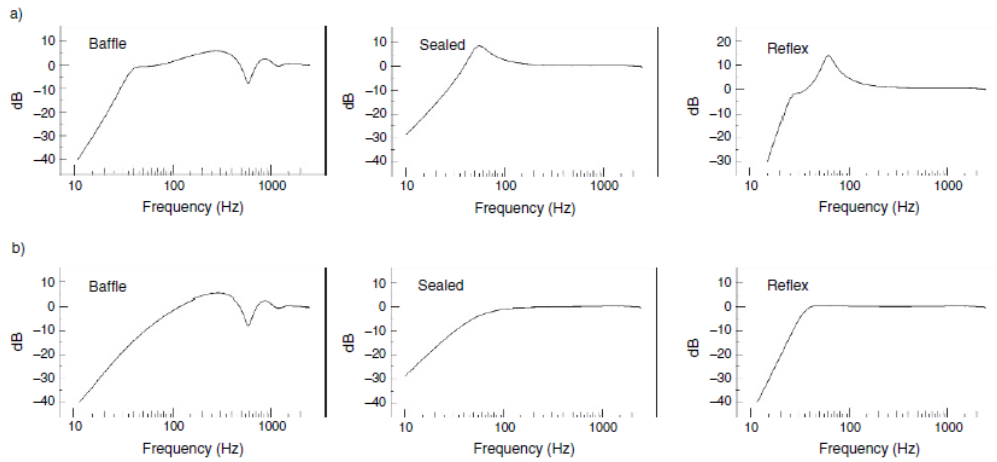
[9] Newell, *Loudspeakers*, 69.
[10] Newell, *Loudspeakers*, 71, 73.



[11]

A Transmission line is basically a pipe within the cabinet treated with acoustic absorbing material that is vented out or closed and is loaded at the rear of the driver[12]. A closed transmission line for low frequencies almost acts as sealed box adding air constraint to the driver. When open-ended, the transmission line tends to act like a reflex by reducing driver excursion and again allowing for a low end SPL increase by resonating.

The use of passive radiator entails using an active driver and a cone with no voice coil. The passive driver is used to tune the box through change in mass. The idea is similar to using a port in that it increases the SPL of lower frequencies through the radiation produced from the back of the driver. The passive radiator produces the same advantages as using a port, but with a sharper resonant frequency roll-off.

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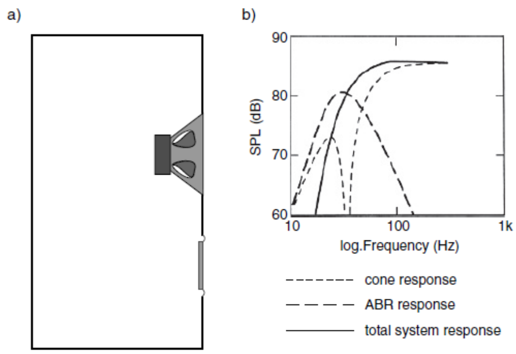
11 Newell, Loudspeakers, 74.
12 Newell, Loudspeakers, 76.



[13]

The design trade-off between these designs is essentially based off the resulting transient response due to change in total system quality factor, how the sound pressure level changes as frequency changes, and construction. Taking these factors into consideration, I originally planned for a sealed box given the sharper and more precise transient response. However, after modeling multiple drivers in Winspeakerz in 4^{th} order vented enclosures, I found that I could extend my bass response well beyond what I would get with a sealed box without sacrificing an adequate transient response.

I understand that using a transmission line would give me even more low-end response, but I also do have to factor in the size increase that would result from such a design. I still want to maintain a reasonably sized enclosure. Overall, my best option for this system would be a vented enclosure.

The other main issue with cabinet enclosures deals with diffraction. With each enclosure there is a fluctuation of sound pressure level in terms of frequency. The idea is to minimize this fluctuation, or ripple, in order to produce a flat frequency response. I chose to go with a rectangular enclosure with a beveled edge as shown below:

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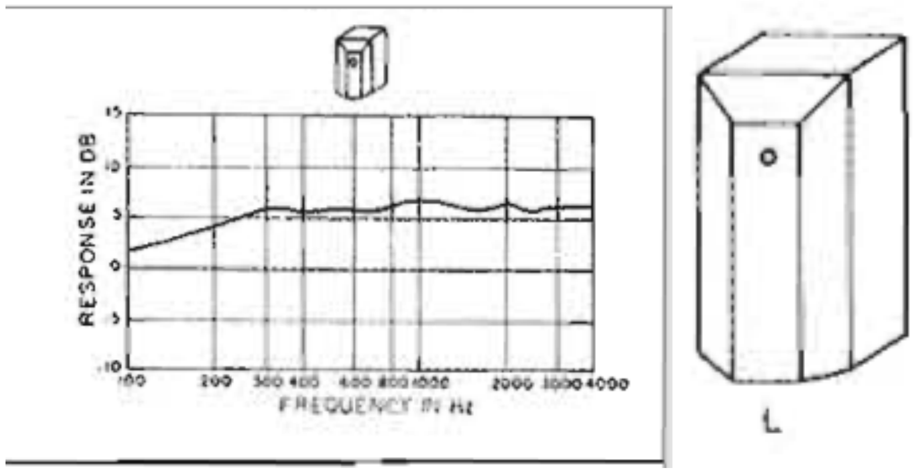
[13] Newell, *Loudspeakers*, 82



This design is able to flatten the overall response and reduce the losses of frequencies due to the reflective properties of certain frequencies. The design of the enclosure has such heavy importance because the interaction of the higher frequencies and lower frequencies result in loss in bass response. [14] This idea is known as diffraction. Essentially, high frequencies tend to be projected forward radiating in half space (relative to the baffle). Half space radiation resembles a hemispheric pattern. Low frequencies on the other hand radiate in full space, or a full circle relative to the baffle. Because the low frequencies are radiated in free space, the density of the energy will be lower than the higher frequencies. This causes a perceived loss of bass in the overall response.

A speaker radiating in half space will have a 6 dB higher sound pressure level than a speaker radiating in full space.[15] This 6 dB transition from higher to lower frequencies is known as a baffle step. The frequency this transition is based around depends on the size of the baffle.

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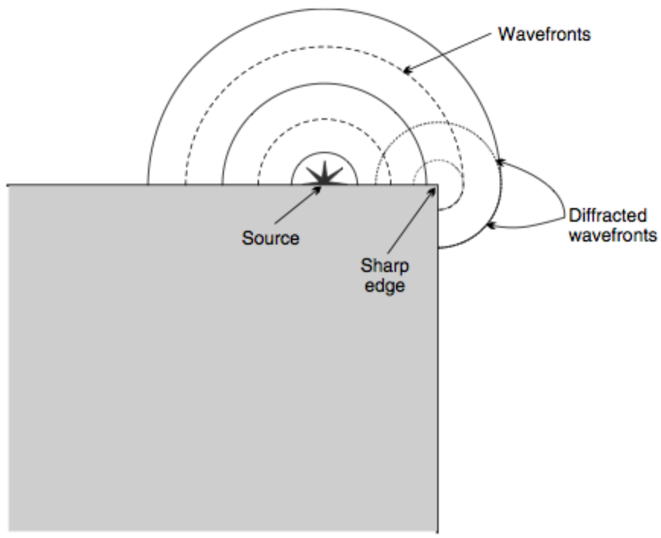
[14] Murphy, *Intro to Loudspeaker Design,* 68
[15] Murphy, *Intro to Loudspeaker Design,* 69

In addition to diffraction with high and low frequencies, the mid-range frequencies also have a unique response to enclosure shape. The sudden expansion of the radiating pattern around a sharp edge of an enclosure causes wave refractions.[16] These refractions act as a second source and can cause comb filtering in middle frequencies.



By going with a beveled edge, the sound waves are less likely to be projected off the sharp corners presented with just a simple rectangular design ultimately making overall on-axis and off-axis frequency responses flatter.

The actual materials used for the enclosure are just as important as the shape. I decided to layer my speaker wall with multiple sheets to avoid excess sound leakage. The outermost wall will be ¾" Baltic birch plywood. This will be sealed against a ¾" sheet of MDF. On the internal portion of the enclosure, I will be using a ½" sheet of sonic barrier, a dampening material comparable to black hole. This configuration will basically disallow sound energy from causing excess vibrations in

[16] Newell, *Loudspeakers*, 90

the speaker walls. These vibrations will eventually cause issues with the overall frequency response, as this will be considered lost energy.[17]

I based my initial woofer selection off of a couple main characteristics: Cost, free air resonant frequency, sensitivity, and frequency behavior near crossover points. As I narrowed my selection, I then modeled the mostly likely candidates to determine the lowest possible F3, or woofer roll off frequency. I then used a data analysis spreadsheet to narrow my selections down even more.

Speaker Decision Analysis

Woofers:

Spec W/ unweighted rating (out of 10):	**Total Weighted Rating**	**Sensitivity**	**VB**	**Cost**	**Modeled F3**
SEAS CA18RLY H1217	**7.25**	88 dB: 7	52.59 L: 6	$71.55: 7	35 Hz: 8
Scan-Speak Classic P17WJ-00-08	**6.8**	88 dB: 7	42.15L: 7	$65.55: 8	40 Hz: 6
Hi-Vi D6.8 Bass-Midrange	**7.1**	85 dB: 6	23.67 L: 8	$74.00: 6	37 Hz: 8
Tang Band W6-1139SI	**7.25**	83 dB: 6	11.84 L: 9	$61.42: 8	39 Hz: 7
Fostex FW 167	**6.55**	89 dB: 7	15.41 L: 8	$64.00: 8	49 Hz: 5
Audax HM170C0	**5.4**	90 dB: 8	12.93 L: 8	$99.10: 5	60 Hz: 3
Scan-Speak Discovery 6.5"	**6.55**	88 dB: 7	14.47 L: 8	$67.40: 8	52 Hz: 5
SEAS Prestige U18RNX/P (H1571)	**6.15**	88 dB: 7	14.59 L: 8	$89.55: 6	50 Hz: 5
SB Acoustics SB17NRXC35	**6.8**	92 dB: 8	15.83 L: 8	$61.25: 8	52 Hz: 5

Specification	*% Weight*
F3	*40*
Sensitivity	*25*
Box Volume	*15*
Cost	*20*

Based off of my decision analysis, I narrowed my woofer selection to three choices: SEAS CA18RLY H1217, Tang Band W6-1139 SI, and HI-Vi D6.8 Bass-Midrange. The SEAS and the Hi-Vi have slightly better bass response, but larger box volumes in addition to being slightly more expensive than the Tang-Band. While the Hi-Vi's box size is much smaller than the SEAS', the Tang-Band has a breakup around 1 KHz possibly causing maybe an issue for crossovers. In addition, the Tang

[17] Plummer, February 18, 2011

Band's frequency response plot may not be as accurate as I would hope. The Hi-Vi is pretty similar to the SEAS, but the CA18RLY has a smoother and lower frequency response with higher sensitivity. I feel by choosing the SEAS CA18RLY, I can still maintain an acceptable budget, produce the lower frequency response I want, avoid larger issues with the crossover, and still keep the box size manageable.

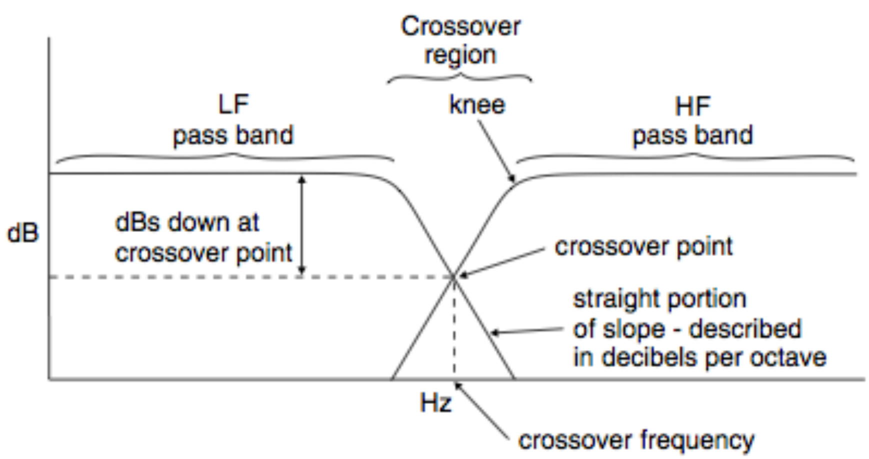
Tweeters:	**Total Weighted Rating**	Sensitivity	Cost	Frequency Behavior*	Fs
SB Acoustics SB25AC-C000-4	**7**	91 dB: 8	$49.60: 5	8	700 Hz: 7
Scan-Speak Discover D2606	**6.05**	91.4 dB: 8	$36.16: 6	5	850 Hz: 6
Vifa XT25BG60-04	**7.4**	92.4 dB: 8	$48.77: 5	8	630 Hz: 8
Tang Band 28-1177SB	**7.8**	90 dB: 8	$21.00: 9	8	775 Hz: 7
Dayton Audio RS28F-4	**7.4**	88db: 7	$49.75: 5	7	530.9 Hz: 9
Vifa NE19 VTA-04	**7.45**	89.4:7	$28.20: 8	8	775 Hz: 7
Tang Band 28-537SH	**6.8**	91 dB: 8	$40.53: 6	8	900 Hz: 6
Scan-Speak D2604	**7.7**	94.06 dB: 9	$55.50: 5	7	475 Hz: 9
SEAS Prestige 27AFNC/G H1397	**6.6**	90 dB: 8	$33.35: 7	8	1170 Hz: 5
SB Acoustics SB26STCN-C000-4	**7.25**	92 dB: 8	$31.90: 7	9	950 Hz: 6
Vifa XT25SC90-04	**7.45**	91.7 dB: 8	$25.50: 8	9	837 Hz: 6
SEAS Prestige 27TDFC H1189	**8**	90 dB: 8	$42.40: 6	8	550 Hz: 9

**How Flat Response is in tweeter range*

Specification	*% Weight*
Fs	*40*
Sensitivity	*15*
Cost	*20*
Freq Behav	*25*

As far as tweeters, there were many that seemed to be able to match up with the CA18RLY. My top 4 choices were the SEAS Prestige 27TDFC H1189, the Scan-Speak D2604, the SEAS Prestige 27AFNC/G H1397, and the Tang-Band 28-1177SB. After matching up the frequencies with the CA18RLY, I narrowed it down to the SEAS H1189 and the Tang-Band 28-1177SB. With the Tang-Band, I like the low resonant frequency of 775 Hz, but the seemingly flat frequency response seems a little sketchy. It appears to be flat within plus or minus 4 dB, which would be decent, but the graph quality isn't good enough to tell for sure. The 27TDFC clearly has an exceptionally flat response from 2 KHz ~ 15 KHz within plus or minus 2 dB. The resonance at 550 Hz is pretty awesome as well. While they are quite a bit more expensive, I can be assured the frequency response will be closer to my expectations set by the specifications.

Now that I've selected my woofer and tweeter (SEAS Prestige CA18RLY and SEAS Prestige 27TDFC, I have to figure out how to get optimal response from them. In any multi-driver system, the frequencies should to be divided up between the drivers. This is done through the circuiting of a crossover. The crossover filters out certain frequencies for a driver allowing it to only get signal from a certain bandwidth of frequencies. The graph below displays how a crossover is implemented. [18]



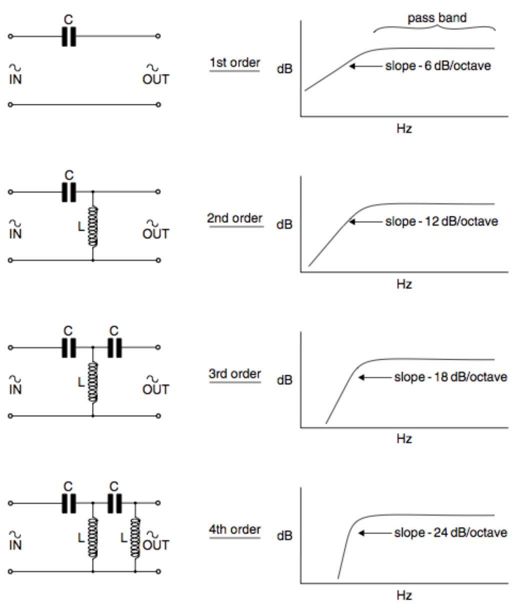
[19]

In dividing the frequencies, it is important to match the response of the driver to crossover. Mismatching these properties can result in an unwanted

[18] Newell, *Loudspeakers,* 124
[19] Newell, *Loudspeakers,* 125

frequency dip.[20] In applying this to my 2-way system, I would have a low-pass filter on my woofer and a high-pass filter on my tweeter.[21] The idea is to find a frequency where these drivers can 'meet' and base the roll off of the filters in the crossover around where both drivers have a solid response.

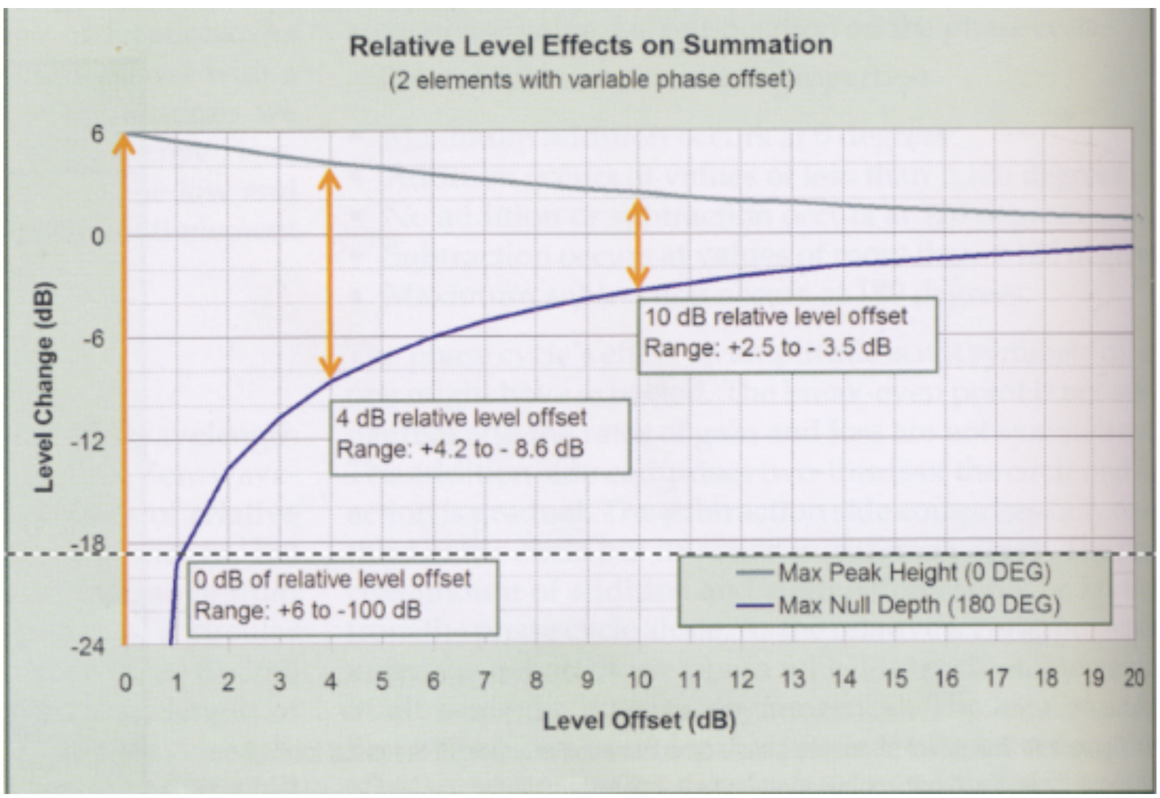
The filters in the crossover, as seen above, roll off as it approaches the crossover point. The slope of this crossover can be manipulated in order to better apply it to a system. Typically, the filters will have a slope of 6, 12, 18, or 24 dB per frequency octave.[22]This means that with each octave change of frequency will result in a change in the output dB by a factor of the given slope. The slope of the crossover is determined by the circuiting of the filter. Crossovers can be 1st-4th orders and affect the crossover as seen below:[23]



[20] Newell, *Loudspeakers*, 128
[21] Murphy, *Intro to Loudspeaker Design,* 98
[22] Murphy, *Introduction to Loudspeaker Design,* 99
[23] Newell, *Loudspeakers*, 132

A crossover point at 2 KHz seems like the best solution to matching the speakers. With a resonance at 550 Hz, the 2 KHz crossover point would steer clear of any issues with the tweeter. The response at 2 KHz for the woofer is pretty flat from about 800 HZ – 3 KHz, so a 2 KHz crossover point would be practical here as well.

I used the chart below to determine the summations with different crossover slopes between the speakers. This chart can be found in Bob McCarthy's Sound Systems: Design And Optimization. [24]



By using 3rd Order (18 dB/ Octave) crossovers on both the woofer and tweeter, I can get a balanced summation over the crossover point.

Therefore, I'll be using two 3rd Order crossovers at 2 KHz.

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[24] McCarthy, Sound Systems, 68

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