Design Project Two: Audio Equalizer

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Abstract

This project created an audio equalizer with three different bandpass filters. Each band had an adjustable gain based on the project specifications. The project challenged us to apply our knowledge of operational amplifiers, sinusoidal signals, and active filters. The created design was tested and compared against theoretical results.

Through testing, it was observed that the experimental data roughly lined up with the theoretical data. More broadly, the experimental data followed the same trend and curvature as the theoretical data. Generally, however, as the frequency for each test was increased, the experimental data appeared to deviate more from the theoretical data. For example, when gain was tested at 2.5-2.5-2.5 for each bandpass filter, at 100kHz, the theoretical data outputted around -15dB while the experimental data outputted around 3dB however, at a lower frequency like 500Hz, the theoretical and experimental data were both around 5dB. It's important to understand though, that not every component will be ideal. Something like an operational amplifier might not behave in the exact way we expect it to or a resistor could have a higher or lower resistance than marked. Furthermore, obtaining the correct position for the potentiometer was hard so many times, it was estimated which could contribute to the differences in data. In the future, it would be best to move the potentiometer and test the output gain before measuring it to ensure that the potentiometer is in the correct position.

Introduction

In this project, an audio equalizer was created to apply our knowledge of operational amplifiers, sinusoidal signals, and active filters to a design of our own. Specifically, the goal of the project was to create an audio equalizer with three frequency bands (bass, midrange, treble) that would change the output of the music processed through the audio equalizer while also adhering to specific design specifications. To do this, calculations were performed to design the circuit, simulations were done to ensure the output of the designed circuit was as expected, and data of the implemented circuit along with music being played through the design was observed.

Theory

Multiple processes were involved in the design of the project. This can be broken down into two main theories: operational amplifiers and active filters.

Operational amplifiers (referred to as op amps from now on) are 8 pinned integrated circuits that are able to amplify weak electric signals. Each pin has a specific function. An enlarged diagram of an op amp can be seen in Figure 1 below which labels the important functions of an op amp's pins.

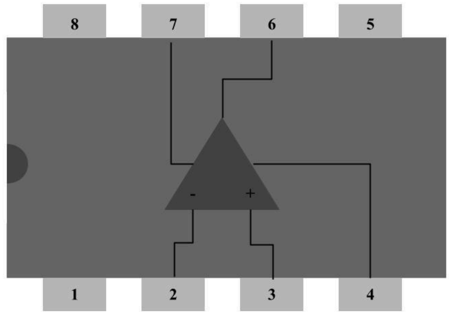


Figure 1. An enlarged view of an Operational Amplifier with Pins labeled.

From Figure 1, positive voltage (V_{CC}) goes into pin 7 and negative voltage (V_{EE}) goes into pin 4. Pin 2 is the inverting input (V_N) and pin 3 is the non-inverting input (V_P). Pin 6 is the output voltage.

An op amp can be in different states. Namely, an op amp can be in a saturated state or a linear state. In terms of saturation, it means the max output voltage that can be outputted by pin 6 is the positive or negative input voltage. An op amp's output voltage can never go past its positive or negative input voltages. In terms of linear operation, it means that the output voltage from pin 6 can be anywhere between the positive or negative inputted voltage. This acts linearly with respect to non-inverting input minus the inverting input. We can see this represented graphically in Figure 2 below.

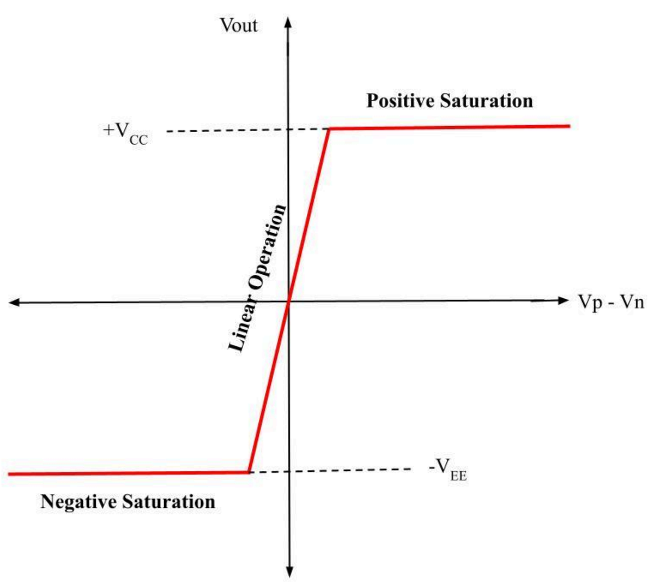


Figure 2. Diagram of Operational Amplifier Saturation and Linear Operation.

With op amp analysis, we can follow 2 rules assuming that the op amp is ideal: the infinite impedance rule and the virtual short rule. The infinite impedance rule is assumed regardless of the state of the op amp. This rule assumes that the current entering the inverting and noninverting pins is so small that it can be considered 0A. The virtual short only applies when the op amp is in linear operation. It assumes that the voltage going into the inverting pin is the same as the non-inverting pin. If the op amp is in saturation, we can't assume the virtual short and have to determine the inverting input based on the op amp's positive or negative saturation.

Filters are circuits that let certain frequencies through while rejecting others. There are two types of filters, active and passive, for our purposes, however, we use active filters which are a combination of active elements like op amps with passive elements like capacitors or resistors. There are certain types of active filters. Specifically, there are low pass filters, high pass filters, bandpass filters, and bandstop filters. For our purposes, we need to only focus on bandpass filters. A Bandpass filter's graph is akin to a bell curve which means as frequency increases, the gain will also increase. There will be a range of frequencies when gain is at its max and once that range is passed, gain will start to decrease as frequency increases. In Figure 3 below, we can see a graphical representation of this.

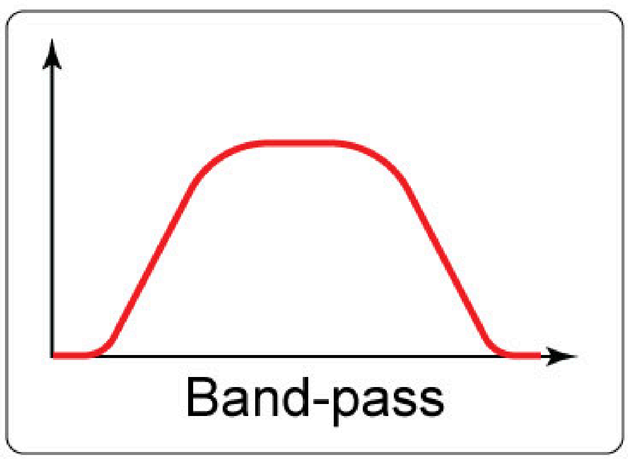


Figure 3. Graphical representation of Active Filter Types. (From Professional Active Filter IC Supplier - Rantle East Electronic)

For multiple-feedback bandpass filters, the center frequency, resonant gain, or gain at the center frequency, and the quality factor for the filter can be determined using the following equations.

$$f_o = \frac{1}{2\pi C\sqrt{R_1 R_2}} \tag{1}$$

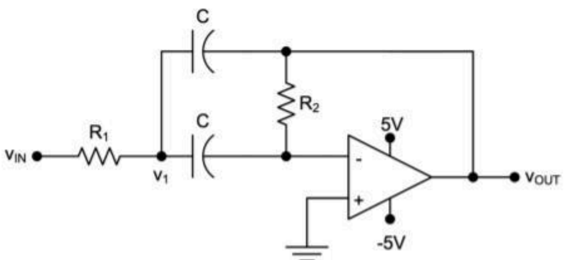
$$A_r = -\frac{R_2}{2R_1} \tag{2}$$

$$Q = \frac{1}{2}\sqrt{\frac{R_2}{R_1}} \tag{3}$$

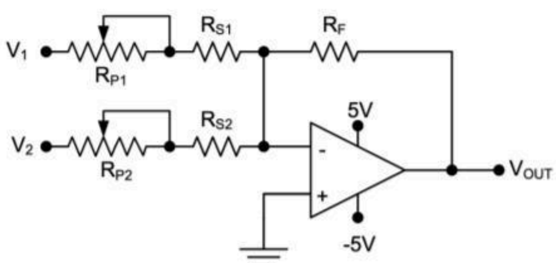
In the equations above, f_o is the center frequency in Hz, A_r is the resonant gain in volts, and Q is the quality factor with no units. Furthermore, R_1 and R_2 are resistors in Ω and C is capacitance measured in farads.

Design

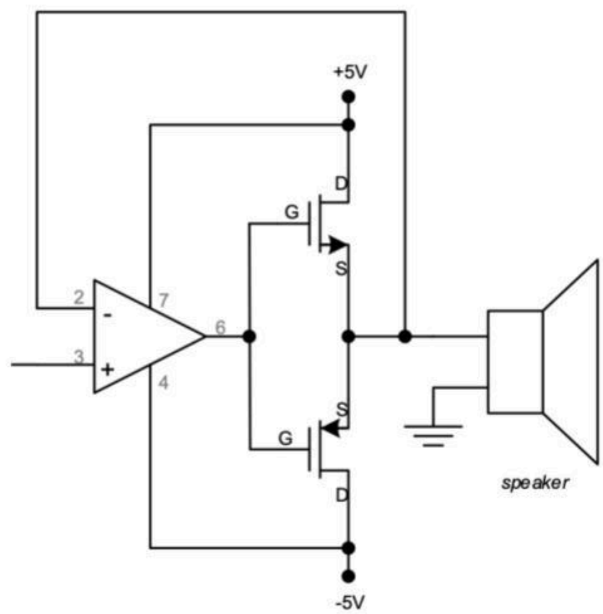
Our design consists of three basic sections: The bandpass filters, the variable gain summing amplifier, and the "push-pull" amplifier that outputs to the speaker. The basic "building blocks" of the design can be seen in Figure 4 below.



Multiple-Feedback Bandpass Filter (From Experiment 7: Active Filter Circuits ECE2001).



Variable Gain Summing Amplifier (From Design Project One: A Music and Microphone Mixer Circuit ECE2001).



"Push and Pull" Amplifier (From Design Project One: A Music and Microphone Mixer Circuit ECE2001).

Figure 4. Various "Building Blocks" implemented in Design

These "building blocks" were chosen to adhere to the specifications given to us. These specifications can be seen in Figure 5 below.

parameter	Value(s)
Supply Voltages	± 5V
Bass band	Center frequency[2] = 180 Hz ± 10% Quality factor[3] = 0.7 ± 10% Range of adjustable gain[4] = 0.5 - 10
Midrange Band	Center frequency = 1.0 kHz ± 10% Quality factor = 0.7 ± 10% Range of adjustable gain = 0.25 - 5
Treble Band	Center frequency = 5.6 kHz ± 10% Quality factor = 0.7 ± 10% Range of adjustable gain = 0.125 – 2.5
Application	audio, using the output of a computer sound card, a power amplifier, and an 8 Ohm speaker

Figure 5. Design Specifications (From Design Project Two: Audio Equalizer ECE2001)

Given these specifications, the center frequency and quality factor for each band implored us to use multiple-feedback bandpass filters and the range of adjustable gain for each band influenced us to use a variable gain summing amplifier.

The variable gain summing amplifier shown in Figure 4 was expanded upon to include another input. This resulted in the variable gain summing amplifier shown in Figure 6 below.

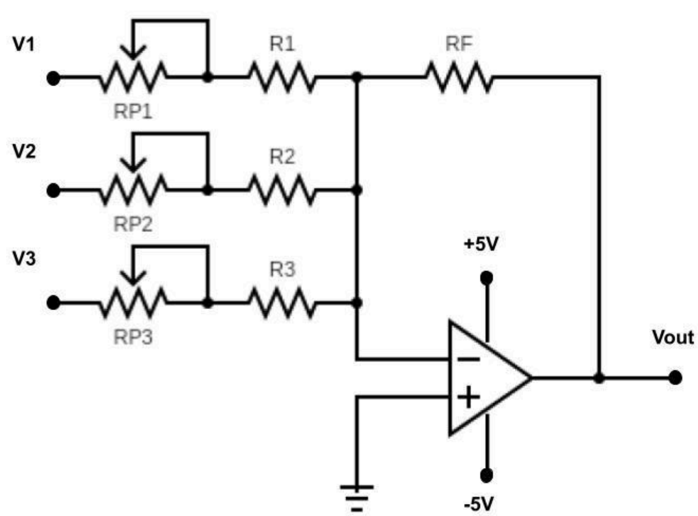


Figure 6. Modified Variable Gain Summing Amplifier

From Figure 6, V1, V2, and V3, are input voltages in volts, RP1, RP2, and RP3 are potentiometers with a max resistance of 10kΩ and a minimum resistance of 0Ω based on the positon of the potentiometer, RF is a resistor measured in ohms, and Vout is the output voltage of the summing amplifier in volts. Based on this, we can determine an equation for Vout.

$$V_{OUT} \;=\; -V_1\left(\frac{R_F}{x_1R_{P1}+R_1}\right) - V_2\left(\frac{R_F}{x_2R_{P2}+R_2}\right) - V_3\left(\frac{R_F}{x_3R_{P3}+R_3}\right) \tag{4}$$

If we were to look at each input voltage individually, we get the following 3 equations.

$$V_{OUT} \;=\; -V_1\left(\frac{R_F}{x_1R_{P1}+R_1}\right) \tag{5}$$

$$V_{OUT} \;=\; -V_2\left(\frac{R_F}{x_2R_{P2}+R_2}\right) \tag{6}$$

$$V_{OUT} \;=\; -V_3\left(\frac{R_F}{x_3R_{P3}+R_3}\right) \tag{7}$$

For equations 4, 5, 6, and 7, each of the variables is the same as stated above from Figure 6.

Using equations 1, 2, and 3, we can determine the resistors and capacitors needed to follow the design specifications. Furthermore, using equations 5, 6, and 7, we can determine the resistors needed to acquire variable gain based on the specifications. Given this, a full audio equalizer circuit was created as shown in Figure 7 below and the fully implemented circuit in Figure 8 below.

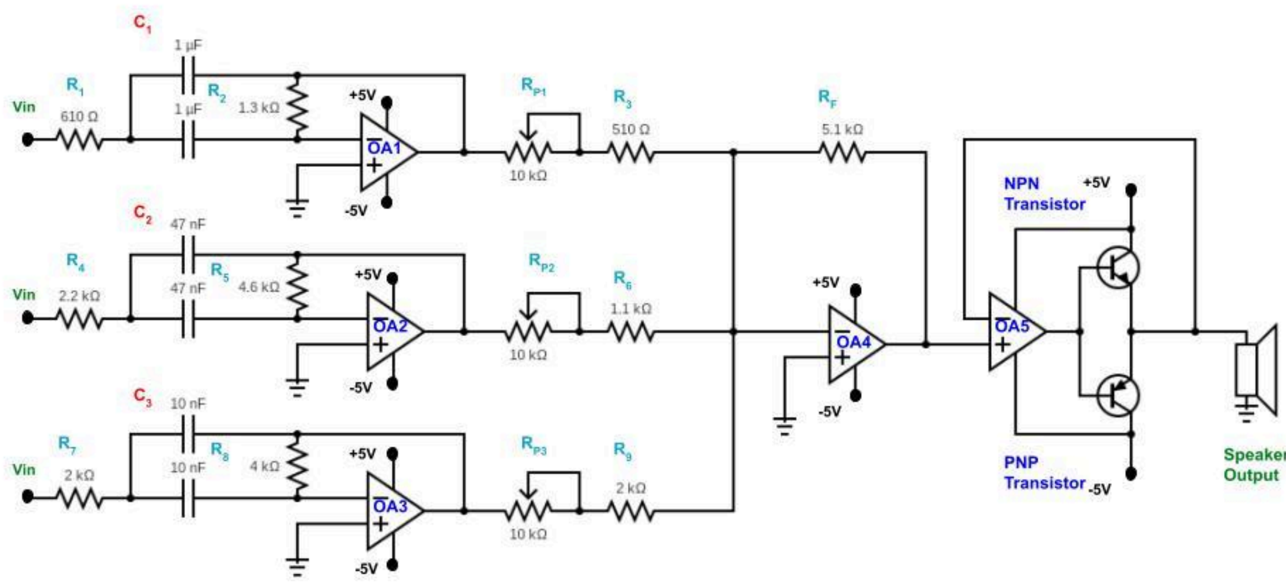


Figure 7. Full Audio Equalizer Design Circuit

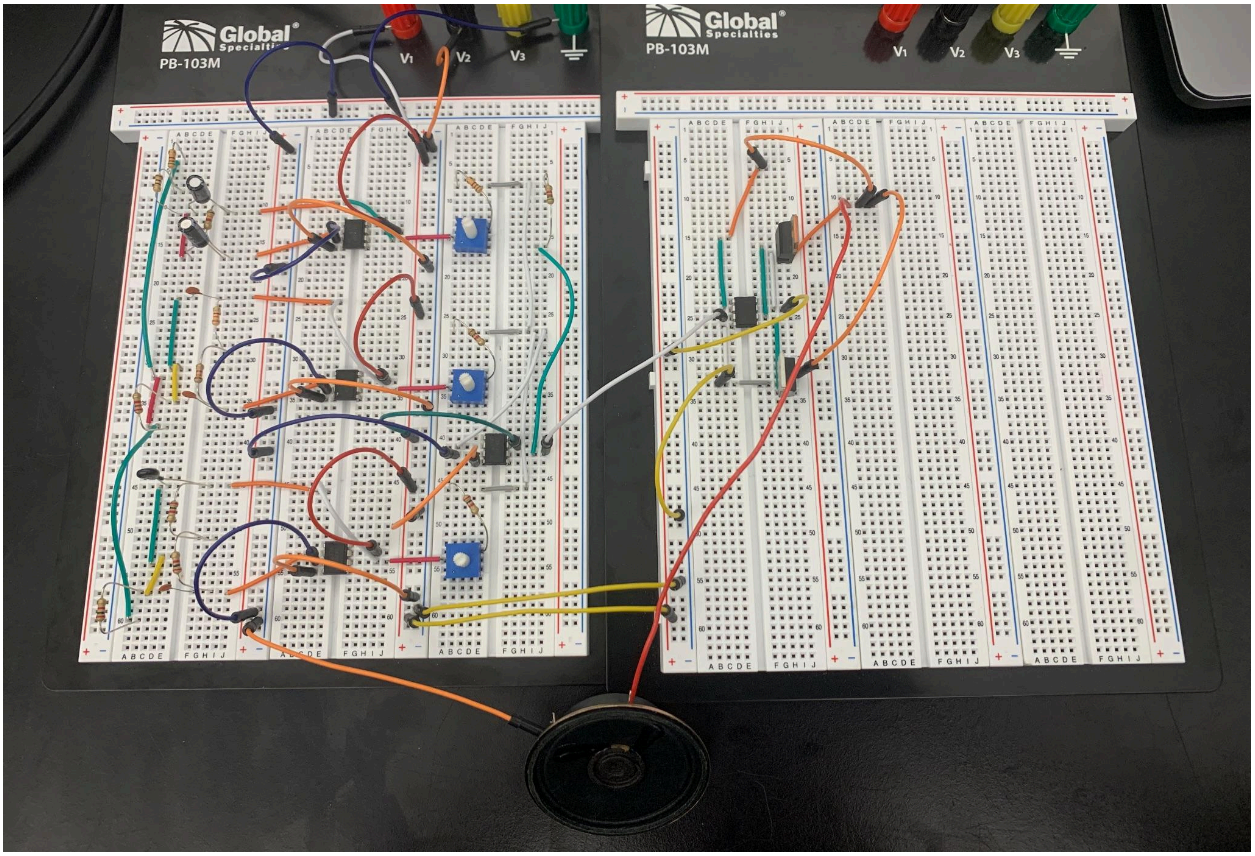


Figure 8. Audi Equalizer Design Circuit Implement on a Breadboard

Given the circuit from Figure 7, the design meets the following variable gain, quality factor, and center of frequency as seen in Table 1 below.

Table 1. Implemented Design Values

	Center Frequency (Hz)	Quality Factor	Minimum Gain (V)	Maximum Gain (V)
Bass Band	178.724	0.723	0.5168	10.65
Midrange Band	1064.467	0.723	0.4801	4.845
Treble Band	5626.977	0.707	0.425	2.55

The way our circuit was designed, center frequency, quality factor, and maximum gain were given the highest priority. As such, the minimum gain for the midrange band and treble band were higher than the specified minimum range of adjustable gain. However, all other values are within a 10% tolerance of the specified values.

Experimental Procedures

The design was implemented and tested using various components and materials. Various resistors, capacitors, and transistors were used along with op amps, potentiometers, a prototyping board, wires, and a speaker. Either music input or a signal was the input to the design. Music was generated through a computer and a signal was generated through a signal generator. An oscilloscope was used to view the output of the design. As our design needed a ground, the triple power supply was connected in a way that allowed for the setup of a ground. Two channels of the triple power supply were used where one channel's positive terminal supplied the positive voltage and the other channel's negative terminal supplied negative voltage. The remaining two were combined to form a ground.

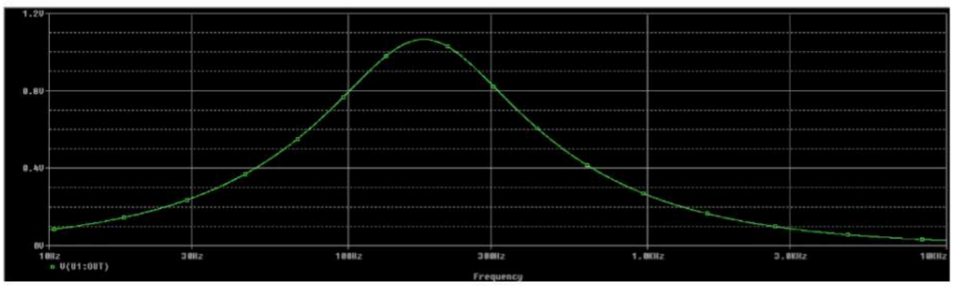
Two main experiments were conducted on the design. The first one measured the frequency response of the design. The second experiment was a qualitative test where music was played through the design.

For the first experiment, the frequency response of the design was tested with different gains for each of the bands. More specifically, three gain variations were tested for the bass band, midrange, and treble band respectively: 10-5-2.5, 2.5-2.5-2.5, and 0.5-1.0-2.0. One sine wave with a peak to peak voltage of 1V was used as input for each band. To obtain the respective gains for each band, the potentiometers were set to different x positions. For the 10-5-2.5 gains, all the potentiometers were set to x = 0. For the 2.5-2.5-2.5 gains, x = 0.166 for the bass, x = 1.03 for the midrange, and x = 0 for the treble. For the 0.5-1.0-2.0 gains, x = 1 for the bass, x = 0.422 for the midrange, and x = 0.05 for the treble. Simulations of each individual bandpass filter were conducted along with simulations of the overall design using the potentiometer settings from above.

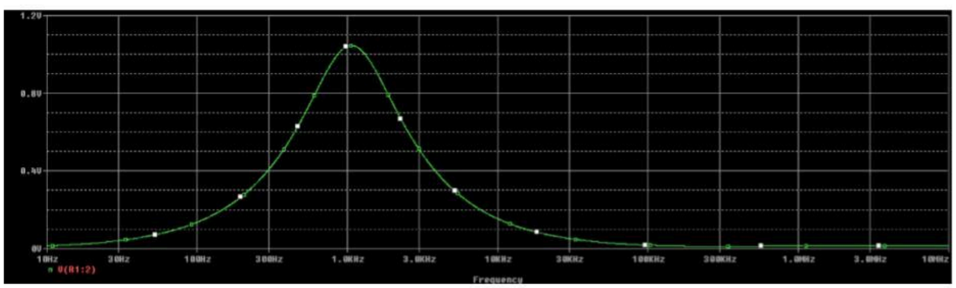
For the second experiment, a simple qualitative test was done where music was inputted into the design. As the music played, the potentiometers were moved to observe the impact it had on the music output through the speaker.

Results and Discussion

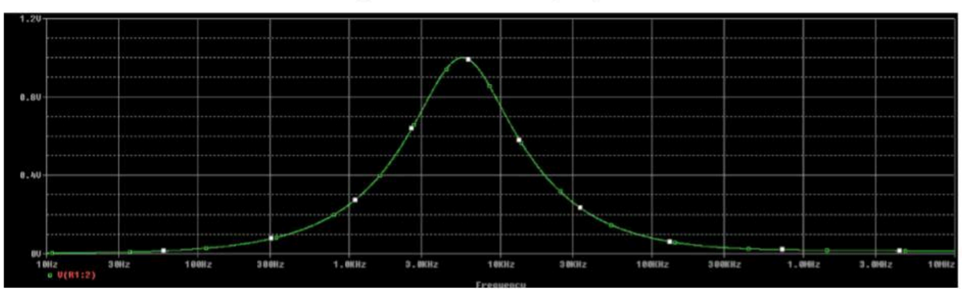
As mentioned, simulations of the individual bandpass filters were conducted. This was to ensure that our calculations output the expected behavior. Figure 9 below shows the output of each bandpass filter.



Bass Band with Center Frequency 180 Hz



Midrange Band with Center Frequency 1 kHz



Treble Band with Center Frequency 5.6 kHz

Figure 9. Graphs of each Bandpass Filter

For each of the simulations, the band filters were tested individually and a sine wave of 1V peak to peak was inputted. Figure 9 shows that the design of each filter meets the specified center of frequency.

For the 10-5-2.5 gain, experimental data can be seen in Table 2 below. Decibels were calculated using 20log(Vout/Vin) where Vin in all cases was 1V.

Table 2. Data from 10 - 5 - 2.5 Gain Settings

Part 1: 10 - 5 - 2.5				
x = 0 for all potentiometers				
Input V: 1Vpp				
Frequency (Hz)	**Vout (V) - Amplitude**	**Vout (V) - Peak-to-Peak**	**Decibels (dB)**	**Phase (Degrees)**
10	0.42	0.84	-1.514414279	80
50	2.16	4.32	12.70967494	70
100	4.4	8.8	18.88965344	49.7
150	4.8	9.6	19.64542466	20.6
180	4.8	9.6	19.64542466	6.5
200	4.8	9.6	19.64542466	0
250	4.8	9.6	19.64542466	-13.5
300	4.6	9.2	19.27575655	-20.3
500	3.04	6.08	15.67807159	-22.3
750	2.96	5.92	15.44643413	-20
1000-1k	2.8	5.6	14.96376054	-28.8
1500-1.5k	2.08	4.16	12.38186661	-34.6
2000-2k	1.52	3.04	9.657471672	-40
2500-2.5k	1.2	2.4	7.604224834	-25
3000-3k	1.06	2.12	6.526717219	-11.6
5000-5k	1.36	2.72	8.691378081	9.02
5600-5.6k	1.48	2.96	9.425834221	11.3
7500-7.5k	1.76	3.52	10.93085327	6.48
10000-10k	1.92	3.84	11.68662449	0
25000-25k	2	4	12.04119983	-35.3
50000-50k	1.36	2.72	8.691378081	-75.2
100000-100k	0.72	1.44	3.167249842	-115

Figure 10 below shows a plot of frequency versus decibel response on a log scale with the simulated and experimental results plotted on the same graph. Figure 11 below shows a plot of frequency versus phase shift on a log scale with simulated and experimental results on the same graph.

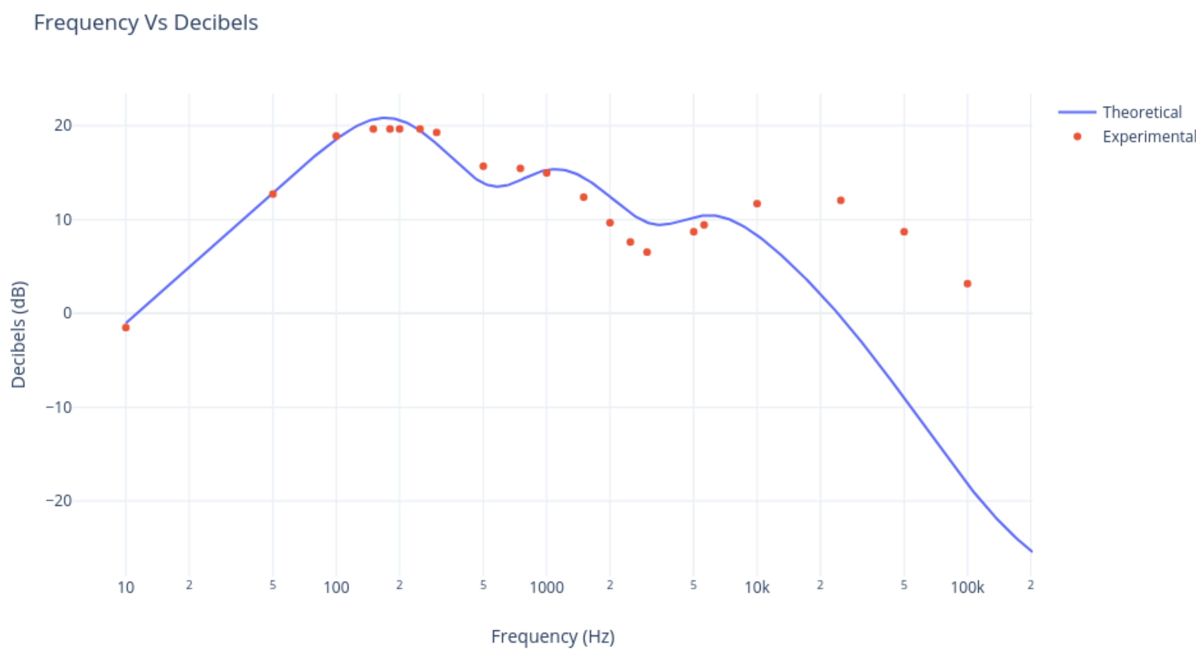


Figure 10. Frequency versus Decibel for 10 - 5 - 2.5 Gain

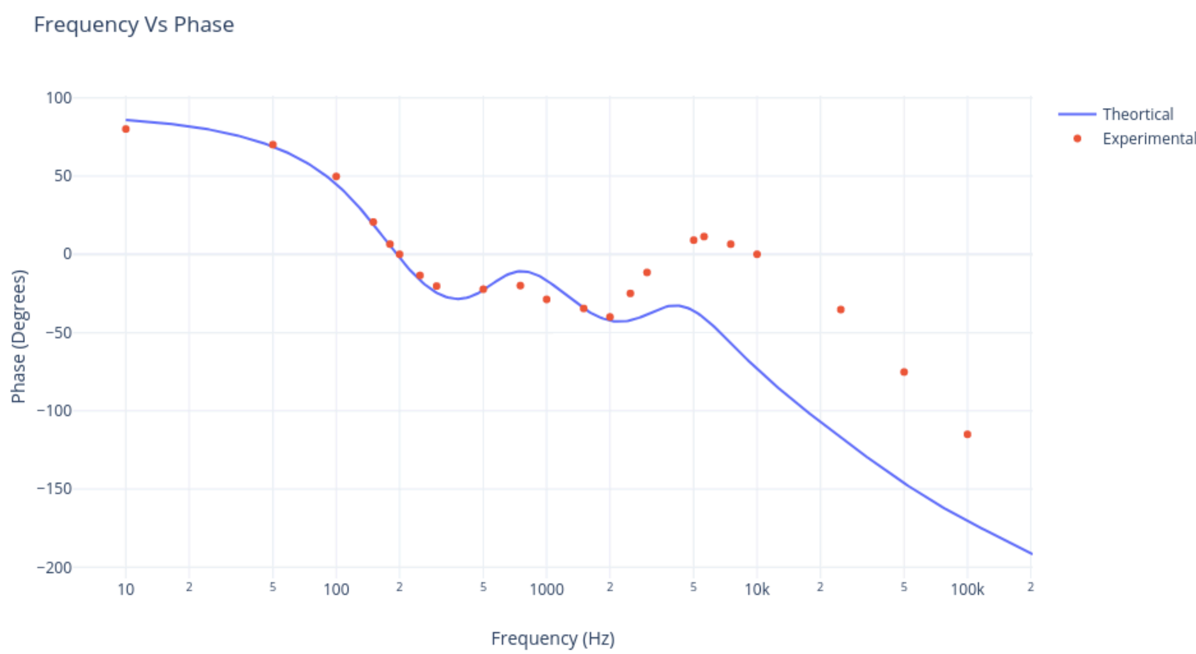


Figure 11. Frequency versus Phase Shift for 10 - 5 - 2.5 Gain

From the two graphs, our experimental results seem to generally follow the theoretical results. What's important to recognize are the relative maximums within the frequency vs decibel graph. It's expected that there will be a relative maximum at each of the centers of frequencies, 180Hz, 1kHz, and 5.6kHz. We can see from Figure 10 that there is a relative max at 180Hz but from thereafter, the relative max experimentally happens either before or after the center frequency. Similarly, not all the data lines up for the phase shift graph in Figure 11. Some point after 2kHz, the phase-shift greatly deviates from the theoretical phase shift. The differences in local maximums and deviations from the theoretical results can be attributed to multiple factors. This can include something like unideal components where resistors, op amps, capacitors, and wires might not act like we expect them to. Something like a resistor could have a lesser or greater resistance than the resistance marked or wires could add resistance to the circuit. Furthermore, there could be human error. Many times, when measuring the phase shift, the values would jump from one to another and at some point, we would settle on one of those numbers. Thus, we could have possibly recorded a slightly different value than the actual one.

For the 2.5-2.5-2.5 gain, the measured and calculated data can be seen in Table 3 below. As from before, decibels were calculated using 20log(Vout/Vin) where Vin in all cases was 1V.

Table 3. Data from 2.5 - 2.5 - 2.5 Gain Settings

Part 2: 2.5 - 2.5 - 2.5				
x = .166 for Bass | x = 1.03 for Midrange | x = 0 for Treble				
Input V: 1Vpp				
Frequency (Hz)	**Vout (V) - Amplitude**	**Vout (V) - Peak-to-Peak**	**Decibels (dB)**	**Phase (Degrees)**
10	0.14	0.28	-11.05683937	86
50	0.6	1.2	1.583624921	70
100	1.1	2.2	6.848453616	50
150	1.36	2.72	8.691378081	30.2
180	1.44	2.88	9.187849755	17.5
200	1.4	2.8	8.943160627	7.9
250	1.28	2.56	8.164799306	0
300	1.12	2.24	7.004960367	-5.91
500	0.92	1.84	5.29635646	-2.89
750	0.96	1.92	5.666024574	-3.24
1000	0.9	1.8	5.105450102	-10.1
1500	0.66	1.32	2.411478624	-6.49
2000	0.6	1.2	1.583624921	15.8
2500	0.68	1.36	2.670778167	27.8
3000	0.8	1.6	4.082399653	34.5
5000	1.28	2.56	8.164799306	34.4
5600	1.4	2.8	8.943160627	30.8
7500	1.6	3.2	10.10299957	25.9
10000	1.76	3.52	10.93085327	12.2
25000	1.92	3.84	11.68662449	-15.3
50000	1.6	3.2	10.10299957	-54
100000	0.74	1.48	3.405234308	-101

The data from Table 3 plotted alongside simulated results can be seen in Figures 12 and 13 below where Figure 12 plots frequency versus decibel from the experimental and theoretical data on a log scale and Figure 13 plots frequency versus phase shift from the experimental and theoretical data also on a log scale.

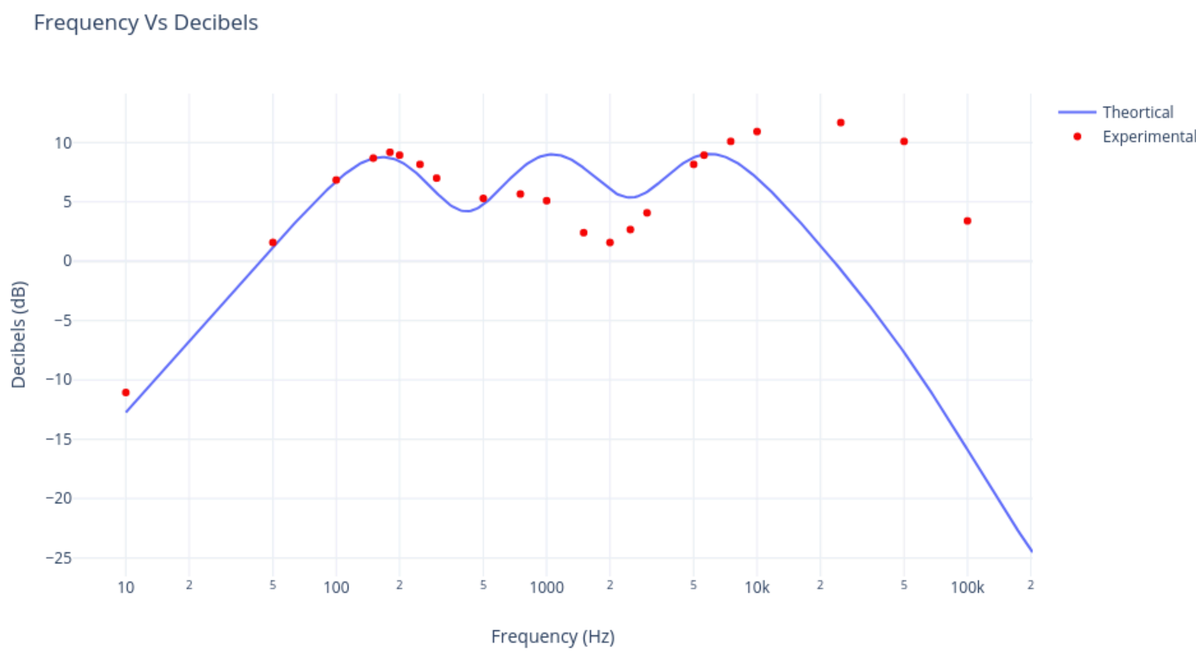


Figure 12. Frequency versus Decibel for 2.5 - 2.5 - 2.5 Gain

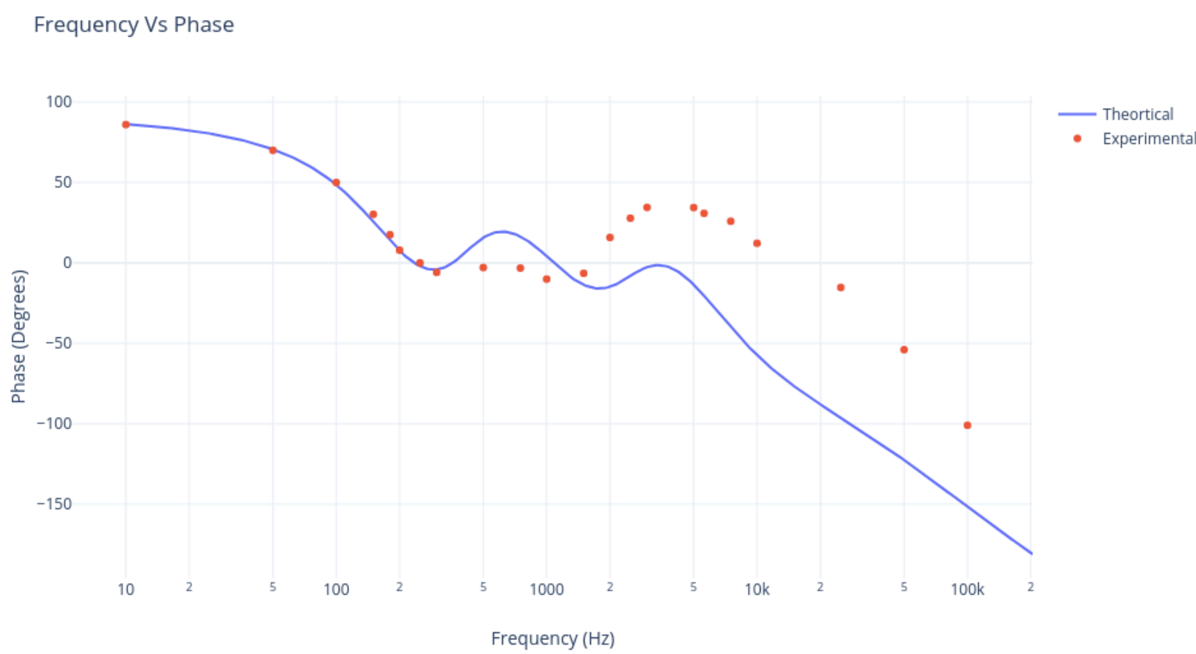


Figure 13. Frequency versus Phase Shift for 2.5 - 2.5 - 2.5 Gain

Similar to before, it appears that our experimental data roughly lines up with the theoretical data. As frequency increases, the experimental data deviates more from the theoretical data for both the frequency versus decibel and frequency versus phase. This is somewhat more expected for this gain setting than the previous gain setting. All the factors mentioned above can contribute to the differences however, something important to note here is that the potentiometers were moved to obtain these gains unlike with the previous gain settings. The potentiometers were moved based on estimation to output the specific gains so it's very likely that the potentiometer's positions weren't in the exact correct spot which could greatly contribute to the differences seen between the experimental and theoretical results.

For gain 0.5-1.0-2.0, the data can be seen in Table 4 below where decibels were calculated using 20log(Vout/Vin) where Vin in all cases was 1V.

Table 4. Data from 0.5 - 1.0 - 2.0 Gain Settings

Part 3: 0.5 - 1.0 - 2.0				
x = 1 for Bass | x = .422 for Midrange | x = 0 for Treble				
Input V: 1Vpp				
Frequency (Hz)	**Vout (V) - Amplitude**	**Vout (V) - Peak-to-Peak**	**Decibels (dB)**	**Phase (Degrees)**
10	0.08	0.16	-15.91760035	86
50	0.18	0.36	-8.873949985	81
100	0.28	0.56	-5.03623946	58.3
150	0.344	0.688	-3.248231235	41.5
180	0.352	0.704	-3.048546817	34.3
200	0.352	0.704	-3.048546817	31.1
250	0.328	0.656	-3.661923212	27
300	0.32	0.64	-3.87640052	30.1
500	0.42	0.84	-1.514414279	38.9
750	0.5	1	0	32.4
1000	0.48	0.96	-0.3545753392	33.1
1500	0.52	1.04	0.340666786	46.2
2000	0.64	1.28	2.144199393	53.2
2500	0.78	1.56	3.862491967	52.1
3000	0.92	1.84	5.29635646	52
5000	1.36	2.72	8.691378081	39.5
5600	1.44	2.88	9.187849755	39.7
7500	1.64	3.28	10.31747687	28.6
10000	1.8	3.6	11.12605002	20.1
25000	2.08	4.16	12.38186661	-9.02
50000	1.68	3.36	10.52678555	-51.6
100000	0.88	1.76	4.910253356	-95.4

Figures 14 and 15 below show the data from Table 4 plotted alongside theoretical data on a log scale.

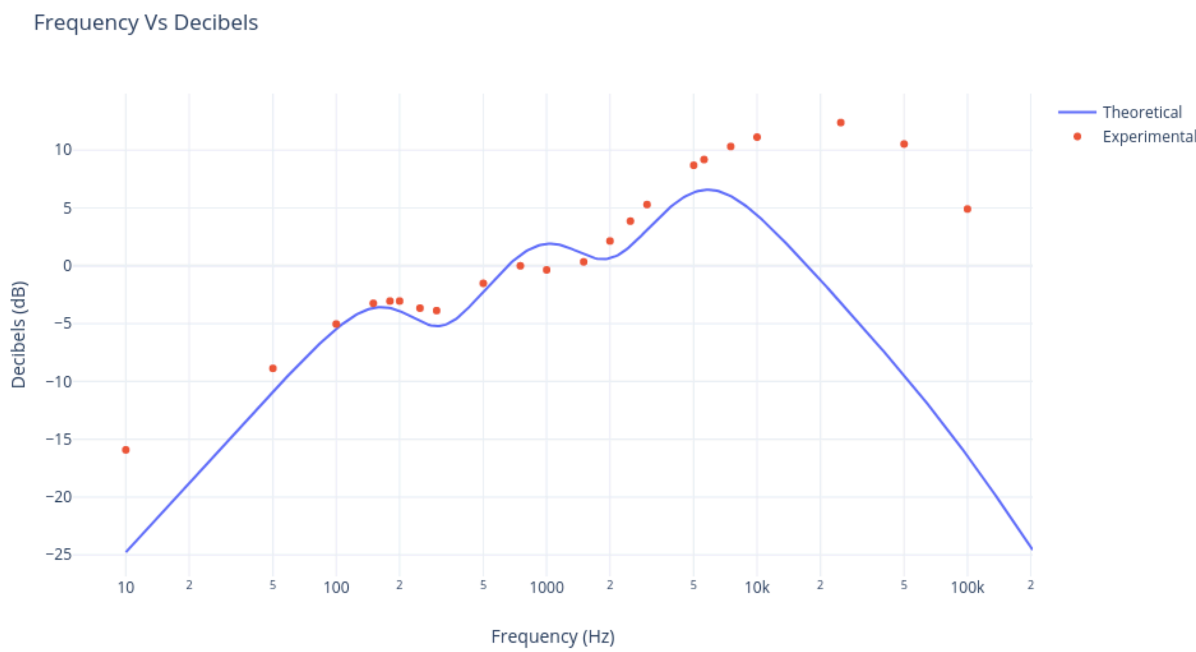


Figure 14. Frequency versus Decibel for 0.5 - 1.0 - 2.0 Gain

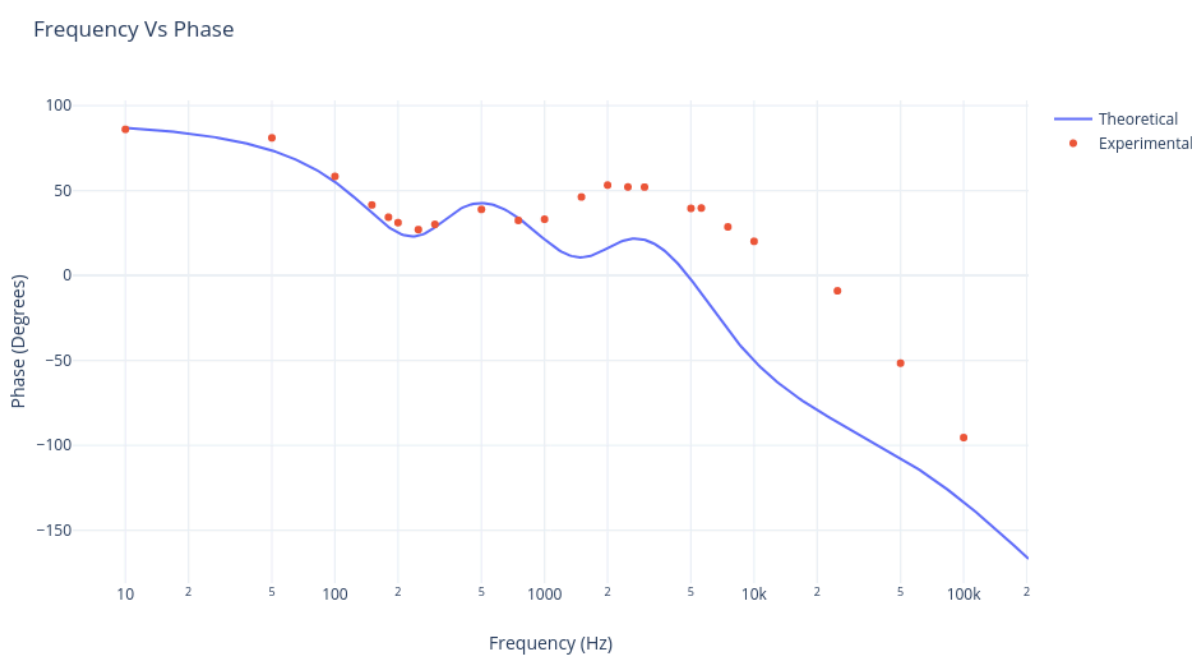


Figure 15. Frequency versus Phase Shift for 0.5 - 1.0 - 2.0 Gain

Similar to the previous gain settings, it appears that the theoretical and experimental results roughly line up with each other; however, as frequency passes around 2kHz, the data starts to deviate more. As before this could be attributed to things like unideal components however, a more likely explanation has to do with the potentiometer. As explained before, the potentiometer was moved based on an estimate meaning it's not certain the potentiometer was at the correct position. Furthermore, for something like the treble band, the gain settings required the potentiometer to be set at $x = 0.05$ which was done theoretically in the simulation however, experimentally, the potentiometer was just left at $x = 0$ as $x = 0.05$ was too hard to realistically implement.

For the second experiment, it was a qualitative test. After taking measurements, music from a computer was used as input. As the music was played, the potentiometers were moved to observe the impact on the music. It was observed that as the bass band filter's potentiometer was moved from a lower x position to a higher x position, the bass of the music increased. For the midrange band, as the potentiometer was increased, it appeared that the clarity and overall volume of the music increased. Finally, for the treble band, as the potentiometer was increased, it appeared that clarity and volume of the voice/ singer increased.

Conclusion

Overall, the project challenged us to create a design that would be an audio equalizer based on the knowledge we have learned. To do this, calculations were done to ensure specifications were met, the developed design was simulated, and then it was implemented on a prototype board and tested. When comparing theoretical and experimental data, the experimental data roughly lined up with the theoretical data. Generally, as the frequency increased, however, the experimental data deviated more from the theoretical data. This difference can be attributed to multiple things such as components being unideal or human error. Not every component will act ideally, that is, a resistor, for example, might not have the exact resistance as marked or a capacitor might not have the exact capacitance as marked. Even an op amp might not behave in the expected way. In terms of human error, this could mainly stem from having to turn the potentiometer. To obtain certain gains for testing, the potentiometer had to be turned and most of the positions the potentiometer was turned to were general estimates which could have contributed to the data being different. In the future, it would be beneficial to know exactly how much to turn the potentiometer. This could be done by testing the output voltage at the center of frequency for each bandpass filter. Furthermore, measuring the resistors and capacitors before implementing them into the circuit might be beneficial too.