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ESE 215 - Introduction to Circuit Theory
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Final Project: Audio Filter with Power Supply

Introduction

The purpose of this lab was to design, build, and test an inexpensive audio docking station which would filter music based on its frequency. The content was separated into treble, 7.9kHz to 9.9KHz, and bass, 150Hz to 350Hz. In addition to the filters, the docking station included an AC/DC power supply that converted $120V_{RMS}$ to approximately +/- 15 V. In order to achieve this aim, we had to familiarize ourselves with an infinite gain bandpass op-amp, and Bode diagrams. These concepts were combined with the use of a transformer, a rectifier, filters, and a voltage regulator to create the final product.

Background

The final project was broken into three parts: a power supply subsystem, a filter subsystem, and an optional amplifier subsystem.

The first component was a power supply which took in an input AC voltage of $120V_{RMS}$ at 60Hz. It converted this input into two regulated DC outputs of approximately positive and negative 15 volts. Each output had an LED as an indicator that the outputs were on. This power supply was used to power the later-built filters. This component allowed our project to be able to work anywhere without the need for the DC Triple power supply. We built this component by rectifying the output of a center-tap transformer, and using a smoothing filter to make the signal DC. To find the proper values for the most effective smoothing filter, we followed hand calculations specified in section 1 of the lab report, and simulated the result on CircuitLab before building the physical supply and testing the result on the oscilloscope.

Besides the power supply, a filter subsystem was built to take in an unamplified audio signal, such as music, with a voltage range of 0-300 mV_{RMS} and a frequency range of 20Hz-20kHz. This system used infinite gain bandpass op-amps to separate the signal to produce two output channels. One channel was treble, outputting music only in a high frequency range of 7.9kHz-9.9kHz. The second channel was bass, outputting music only within a low frequency range of 150Hz-350Hz. To find the proper values to fit this bandpass range, we followed hand calculations specified in section 2, making slight modifications after simulation in CircuitLab, in order to account for errors associated with cascading filters. After a few iterations of calculations, we found the proper values, that we used in the final project. This system was initially powered by the DC Triple Supply for testing before being supplied by the power supply subsystem.

The optional amplifiers would have taken the two separate outputs of the treble and bass filters and amplified those signals at a center frequency of ¼ Watt. The amplifiers' loads were the speakers, each with impedance of 8 ohms. With such a low impedance, maximum voltage transfer would be required from the filters. To achieve ¼ Watts, a transistor amplifier, such as a MOSFET or BJT, would have to be designed. Our final design did not include this system.

The testing devices used in this lab are as follows:

- Oscilloscope: (Agilent Technologies, MSO7034B)
- Waveform generator: (Agilent Technologies, 33521A)
- Multimeter: (Agilent Technologies, HP34401A)
- Triple output power supply: (Agilent Technologies, E3631A)

Part 1: The Power Supply

Pre-building hand calculations:

The first part of our power supply was a transformer, which consists of two inductors. The transformer converts an input voltage into a secondary voltage based on its turn ratio. The turn ratio of our transformer was calculated with the following equations:

$$\frac{Vp}{Vs} = \frac{Np}{Ns} = \sqrt{\frac{Lp}{Ls}}$$

Where P stands for primary coil, S stands for secondary coil, V is for voltage, N is for number of turns, and L is for inductance. The primary voltage of our transformer was 120V while the secondary voltage was 23V. That is, the voltage and turn ratio was 120:23, which when reduced is 5.22:1 by using the top to bottom wires. The power ratio of an ideal transformer is 1:1 where no power is lost to the transformer core. Based on this ratio, the expected output of our transformer when inputting $120V_{RMS}$ was $23V_{RMS}$ split in half for the top and bottom wires. That is the final expected outputs was $11.5V_{RMS}$, or approximately 16V.

We used four 1N4004 diodes to build a bridge rectifier. A bridge rectifier, or full-wave rectifier, was chosen to replicate both the positive and the negative half cycles of the input waveform. The forward-bias voltage V_F across the diodes are approximately 0.93V, so given that our expected waveform input is about $16V_p$, the drop after the two diodes should approximate to ~$14.15V_p$.

The purpose of the smoothing filter is to take the rectified input, and turn it into a relatively steady DC output. To do so, we want to maximize τ_{dn} the amount of time taken to "come down" from the peak, and minimize the amount of time τ_{up} taken to reach the peak. This up-and-down is the cause of "ripple voltage." We accomplish this by placing a capacitor in parallel with the load, and calculating a capacitor value that minimizes ripple voltage. We noticed that using a larger cap value made $\tau_{dn} = R_L C$ larger, and thus the signal smoother, but a larger capacitor value caused the current running through the circuit to be too high, so we found an optimized value at $220\,\mu F$. Finding this capacitor value relies partially on the values used in our voltage regulator. R_L varies by load, so the larger the load, the smoother the filter. To make the most effective power supply, we work with the lowest R_L value expected: $200\,\Omega$.

The voltage regulator involves two components: a series resistor, and a parallel 15-volt zener diode to pull the output to 15 volts. The zener diode has a resistance of 16Ω, and using a resistance of $47\,\Omega$ for the series resistor, we end up with the following calculation for ripple:

$$T_{rect} = \frac{1}{frequency} = \frac{1}{120} = 8.33ms$$

$$R_z \parallel R_L = \frac{16*200}{200+16} = 14.8\Omega$$

$$V_R = \frac{[V_{s1} - 1.86 - V_z]T_{rect}}{R_s C} * \frac{(R_z \parallel R_L)}{R_s + (R_z \parallel R_L)} = \frac{[16-1.86-15]8.33*10^{-3}}{47*220*10^{-6}} * \frac{14.8}{47+14.8} =- 0.166V$$

$$Ripple\ fraction = \frac{(V_r/2)}{V_z} = \frac{0.166/2}{15} = 0.55\%$$

The ripple voltage represents a relative variation of about 0.55%, which is well within the desired range.

To calculate power budget, we assume that our filter consumes a maximum of 50mA (25mA per op amp). We do not have an amplifier, so we do not need current for that. The power supply provides 15V, and according to Ohm's law, our filter appears like a thevenin equivalent of

$\frac{15V}{50mA} = 300\Omega$. Power $= VI = 15V * 50mA = 0.75W$, so we need to account for a power budget of 0.75W.

The following Figure 1 depicts the block diagram of complete system:

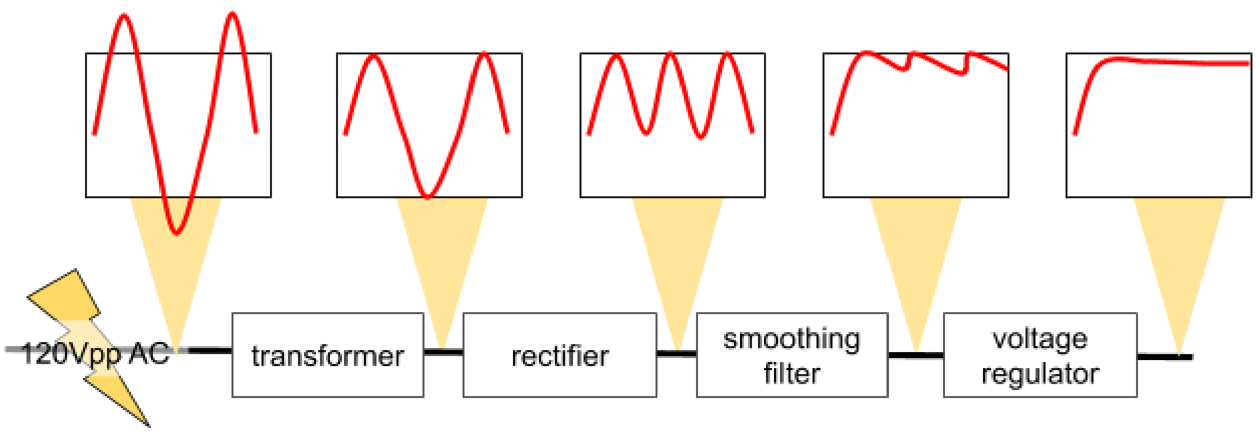


Figure 1: Block diagram of a basic DC power supply

<u>Simulations:</u>

Time Domain Simulation of each sub-component separated:

Before experimental setup, the different components were simulated on CircuitLab as seen below with a 12V 60Hz voltage source:

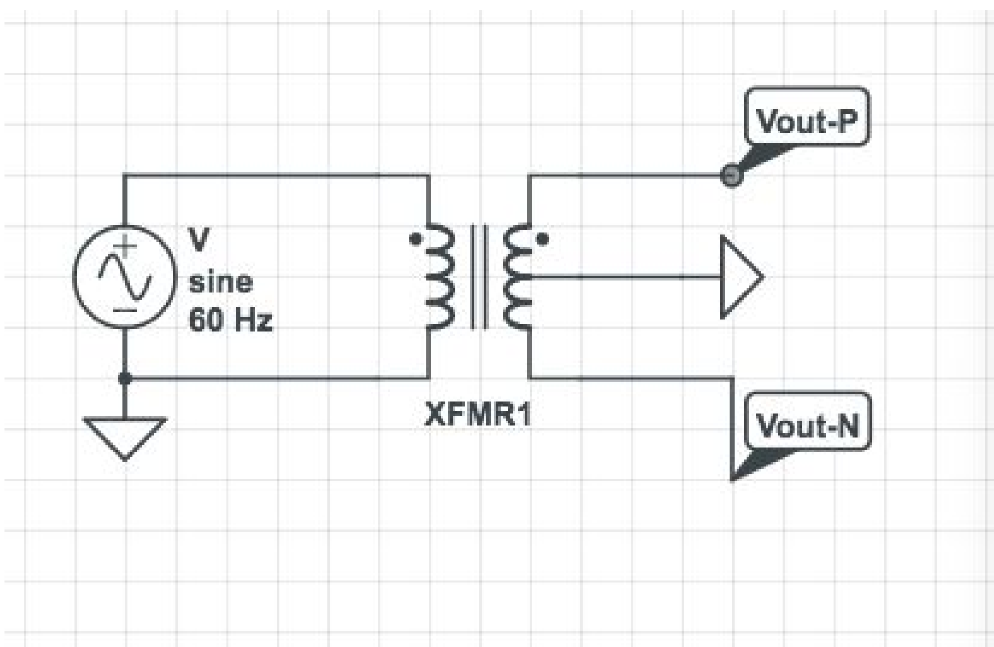


Figure 2: Transformer Simulation Setup

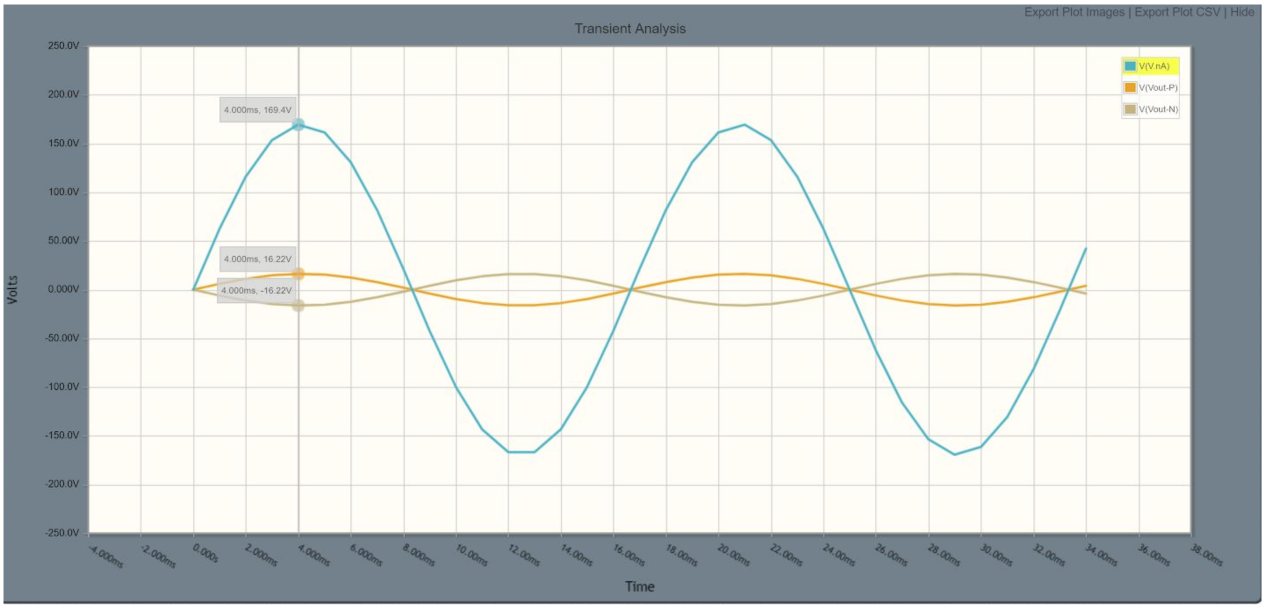


Figure 3: Transformer Simulation

Figure 3 plotted the time domain output of the wall voltage and nodes Vout-P and Vout-N, as shown in Figure 2 above. The wall voltage peaked at $120V_{RMS}$, Vout-P peaked at positive 16.2V, or ~$11.5V_{RMS}$, and Vout-N peaked at the opposite -16.2V. By grounding the center tap, Vout-P and Vout-N became 180 ∘ out of phase.

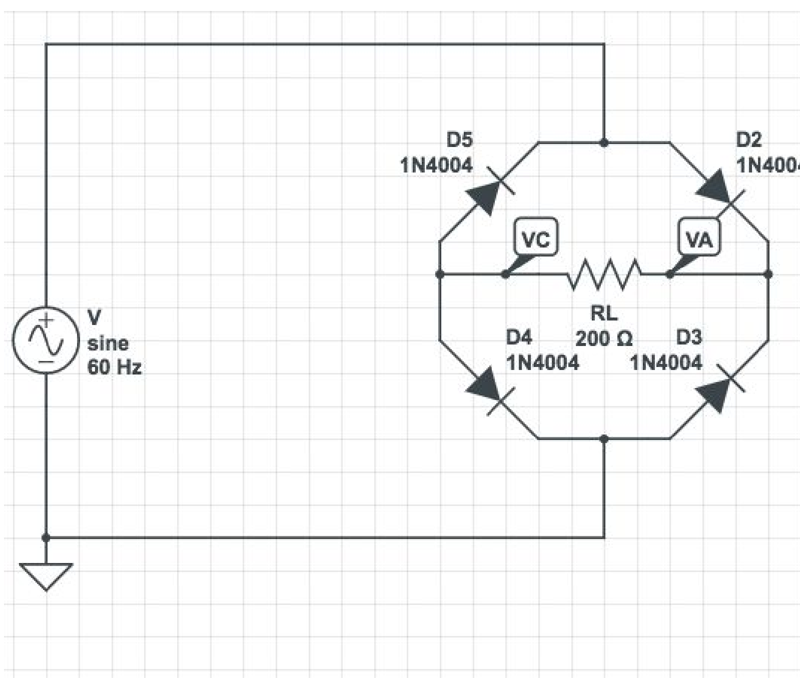


Figure 4: Rectifier Simulation Setup

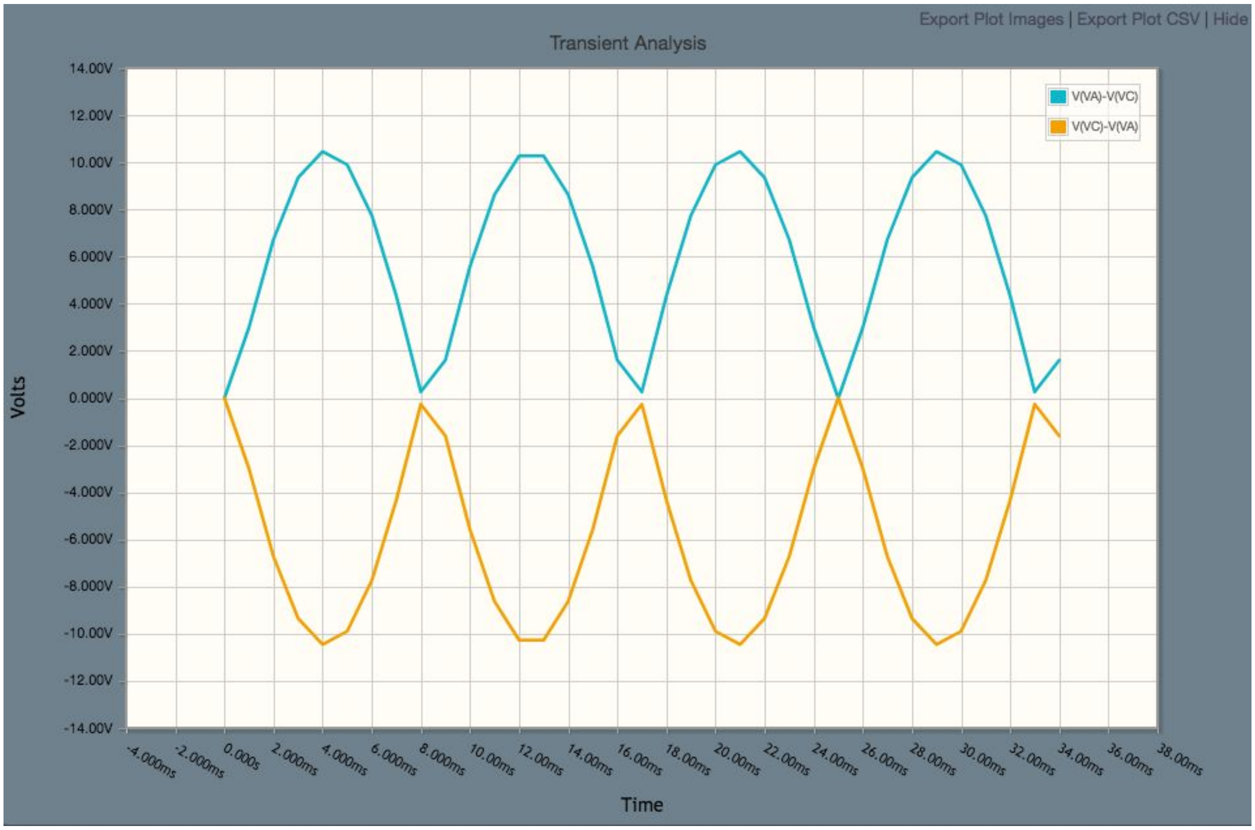


Figure 5: Rectifier Simulation

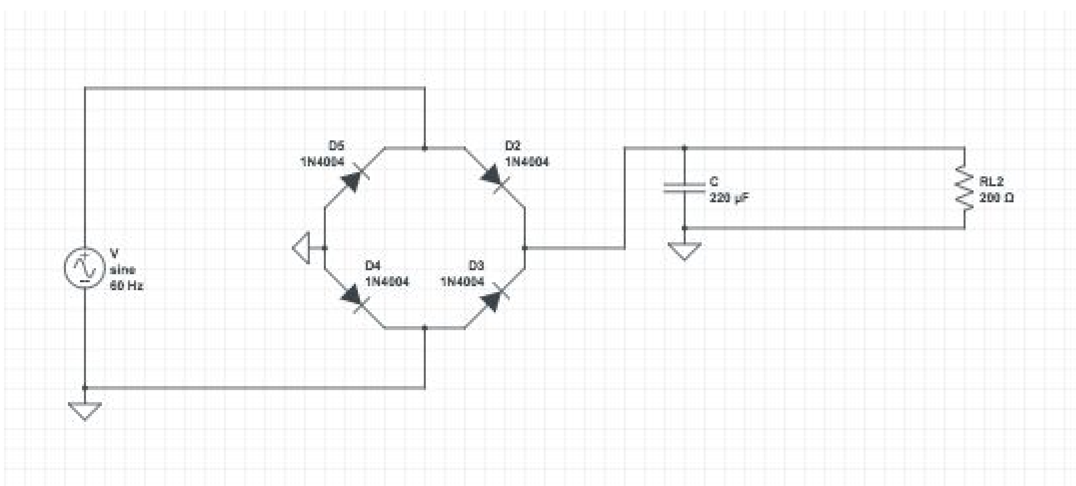


Figure 6: Smoothing filter Simulation Setup

Figure 7: Smoothing Filter Simulation

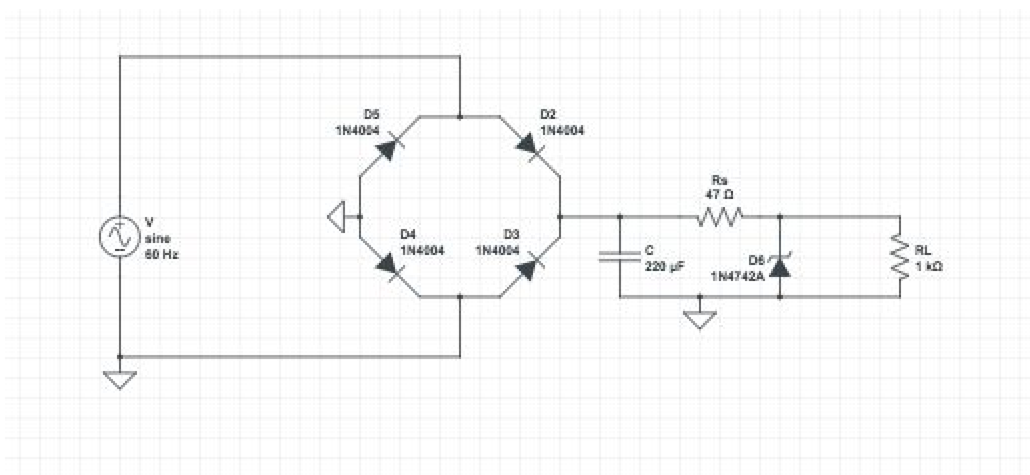


Figure 8: Voltage Regulator Simulation Setup

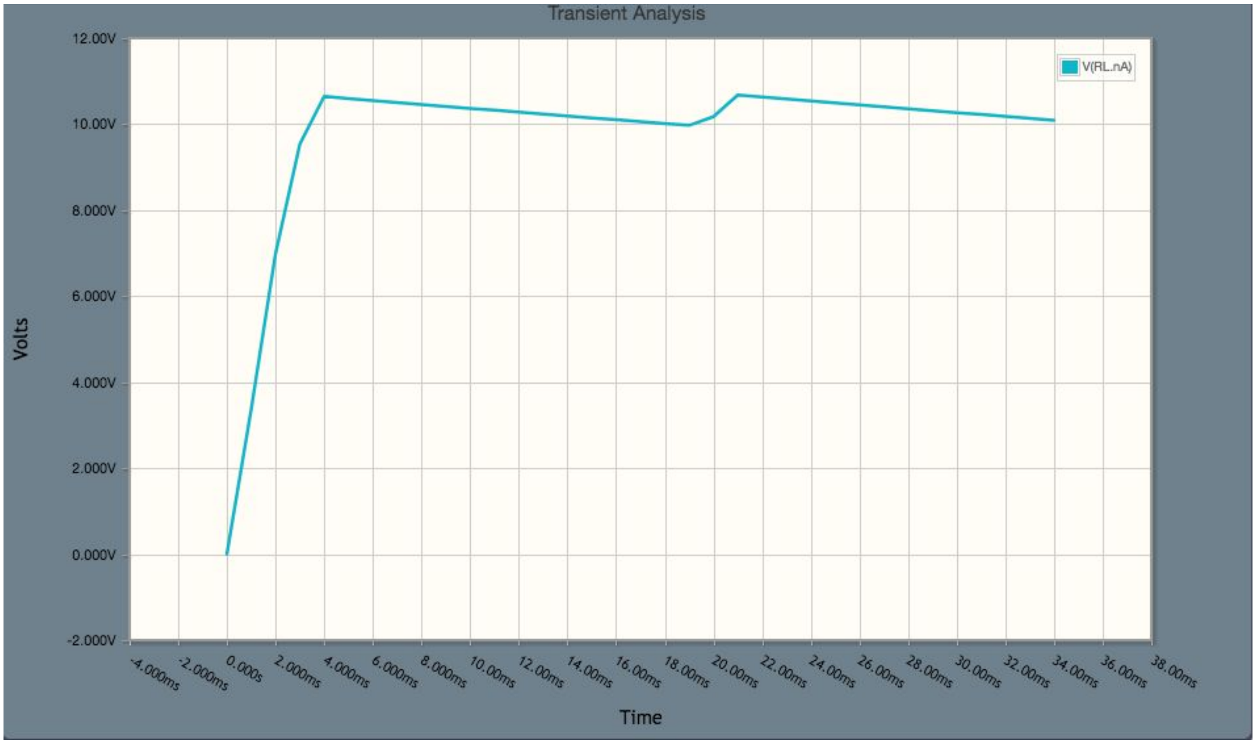


Figure 9: Voltage Regulator Simulation

Time Domain simulation of entire power supply:

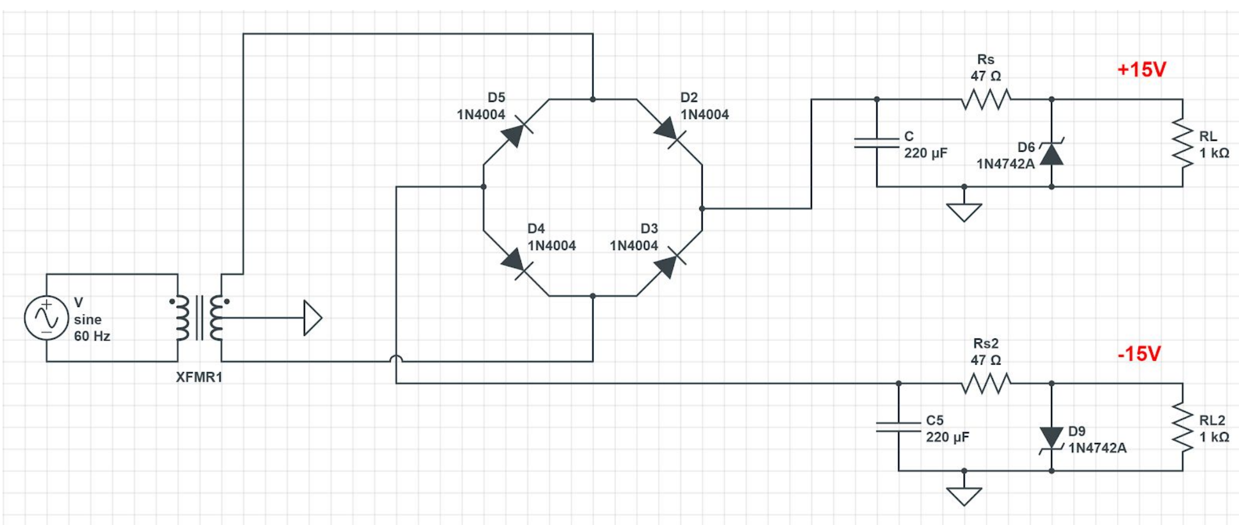


Figure 10: Schematic of Entire Power Supply

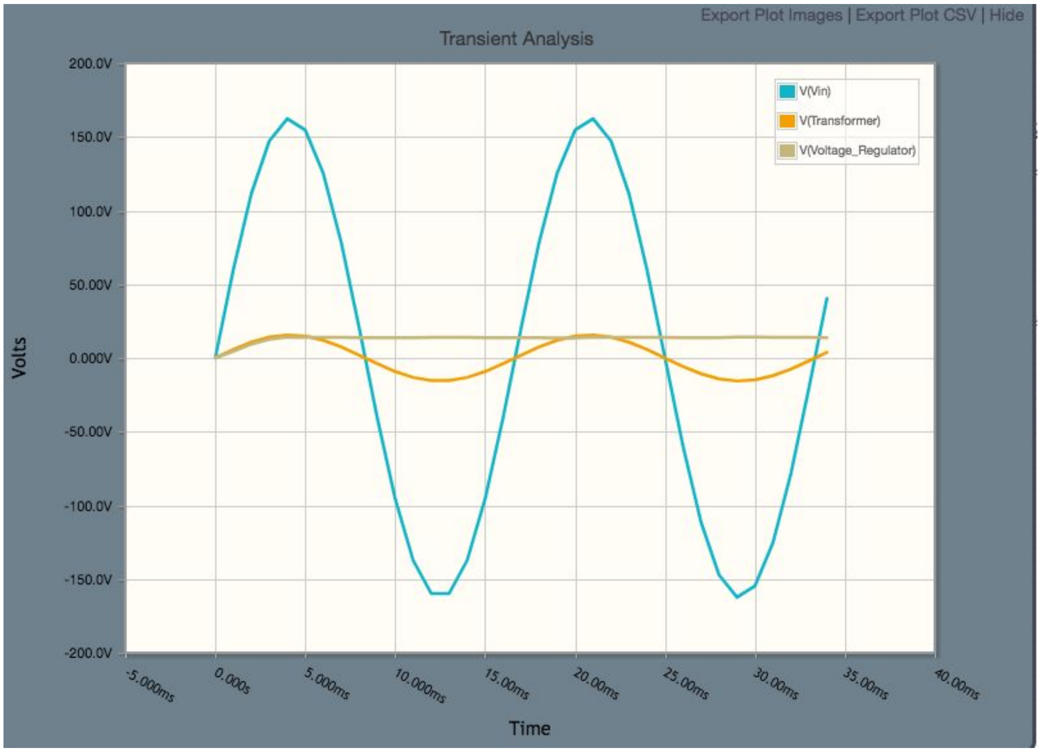


Figure 11: Simulation of Power Supply Showing Each Stage's Positive Output

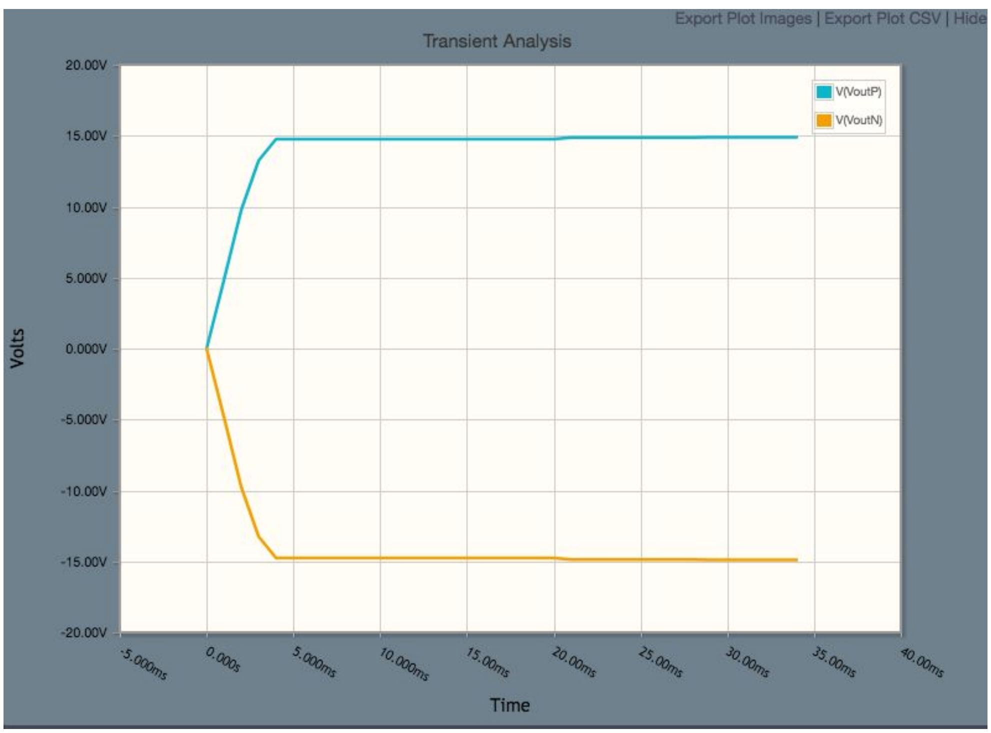


Figure 12: Simulation of power supply showing 15V open circuit voltage

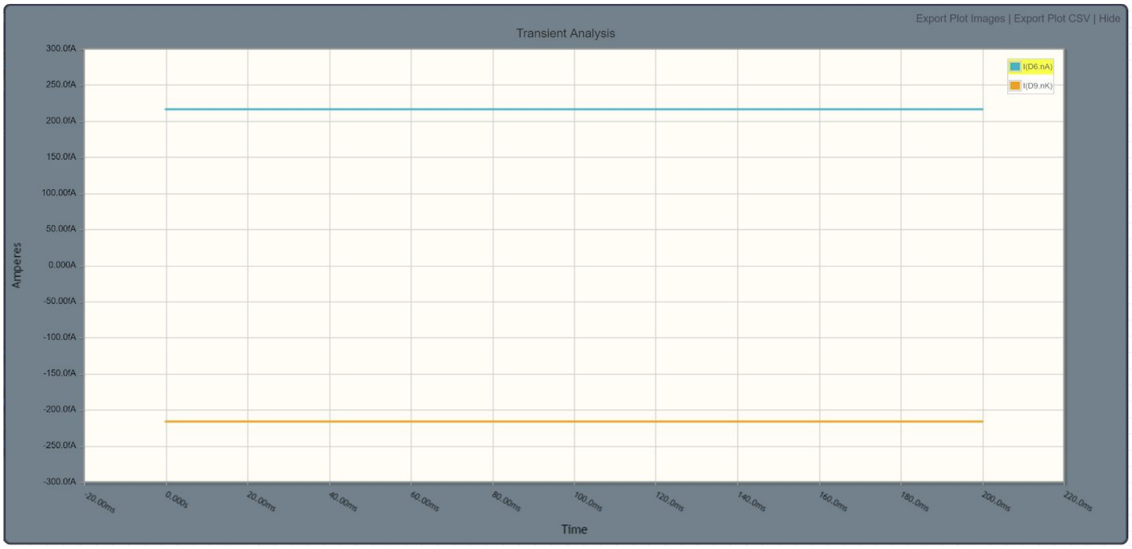


Figure 13: Simulation of power supply showing 15V short circuit current

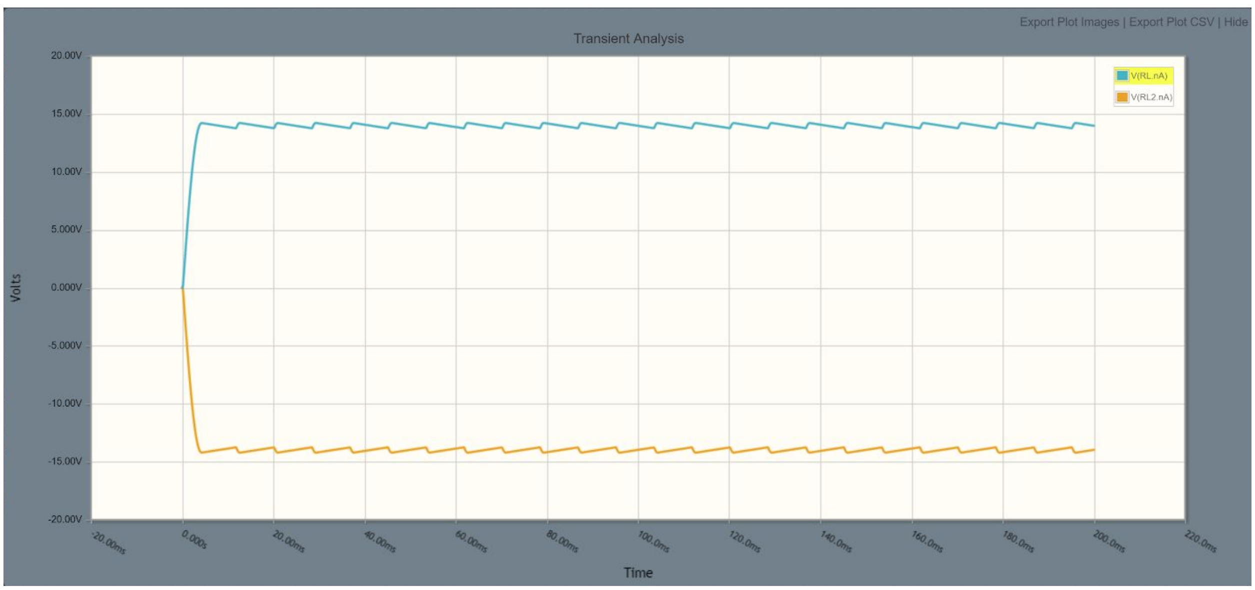


Figure 14: Simulation of 15V delivered to 1kOhm load resistor

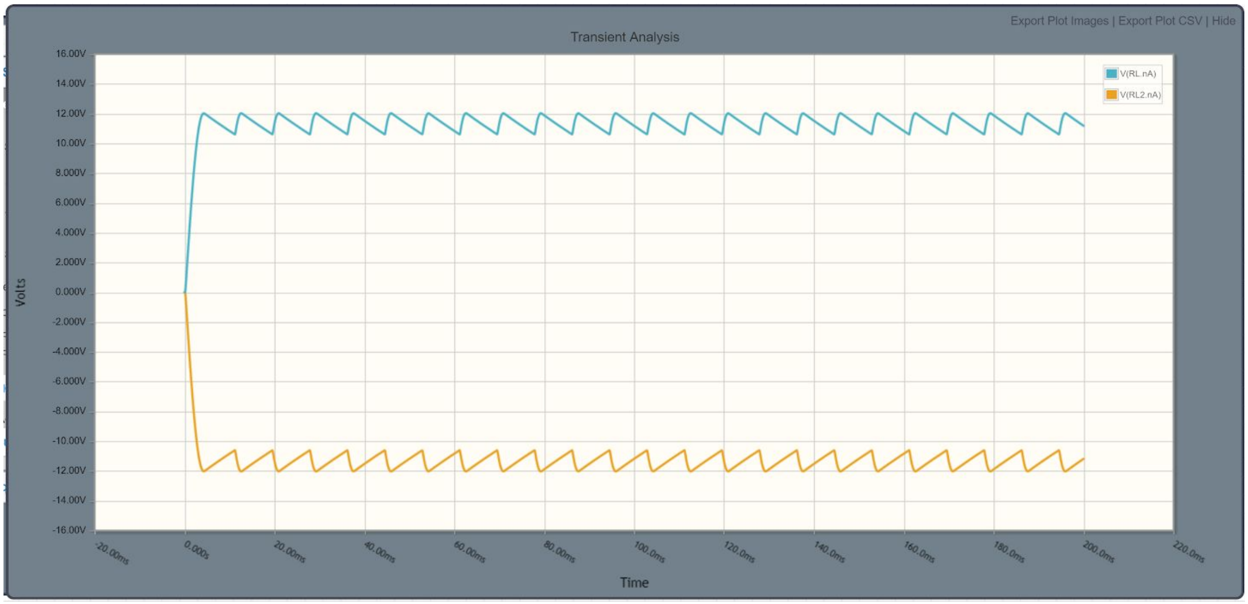


Figure 15: Simulation showing 15V delivered to a load equal to equivalent resistance of other subsystems (200 Ohm)

**We could not find the right zener diode in CircuitLab, so the ripples on the simulation are not quite what came through on the oscilloscope.

Setup:

We used an oscilloscope to examine our power supply output. After plugging the transformer in to the wall outlet, we used BNC-to-grabber cables to probe the power supply, and examined the waveform on the oscilloscope. We used an oscilloscope rather than a multimeter because we wanted to see how output changes over time. For example, we could use the oscilloscope to see the output waveform fluctuate after the bridge rectifier. Then even with our DC output, we could use the oscilloscope to examine the ripple voltage. The oscilloscope allows us to look at the peak-to-peak voltage of our output response, otherwise known as V_r, our ripple voltage. By dividing the ripple voltage by the average output, we can get the ripple fraction to qualify how effective our smoothing filter and voltage regulators are. Each sub-component of the power supply is verified by measuring the output on the oscilloscope. We expect the output to be as specified in figure 1.

Experimental Procedure:

All testing of the power supply was done by examining output on the oscilloscope. We probed at the following locations on Figure 16:

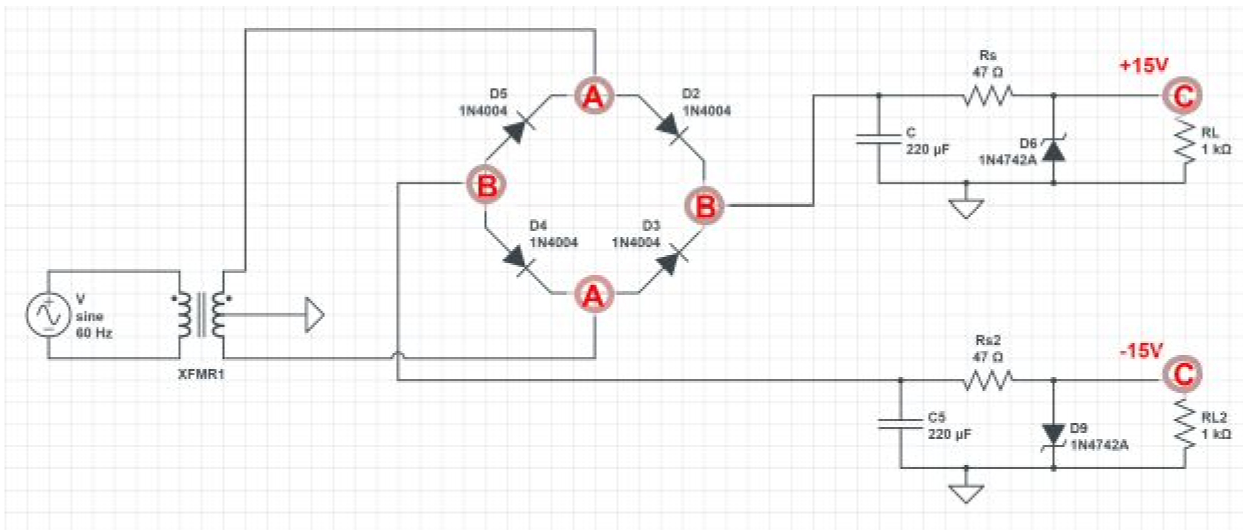


Figure 16: Probing locations on the power supply

We probed at points A to examine output after the transformer, points B to examine rectified output, and points C to examine our desired DC output. On the oscilloscope, we used the measuring options to look at peak-to-peak voltage at each stage. For points A, we expect the voltage to be a sine wave with magnitude 16V. At points B, we expect the outputs to be rectified sine waves, with a voltage drop to have magnitude of about 14.15V. Each rectified "wave" should either be all positive values, or all negative values. At points C, the output should be a relatively steady DC voltage, with minimal ripple voltage.

Experimental results:

At point A (after transformer):

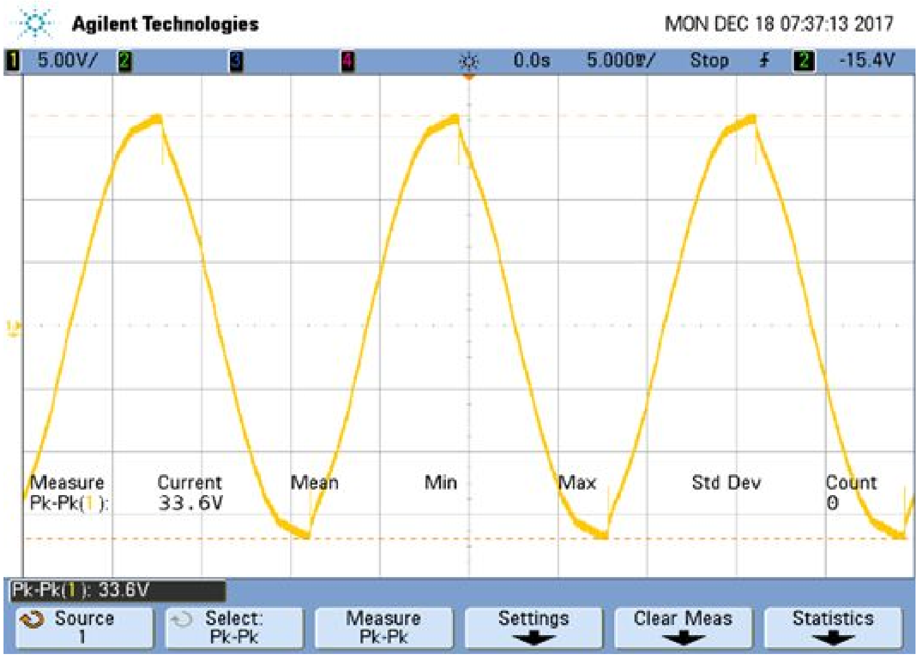


Figure 17: Transformed waveform.

The peak-to-peak voltage is about 33.6V, which is appropriate for our ~16V amplitude wave desired after the transformer.

At point B (after rectifier):

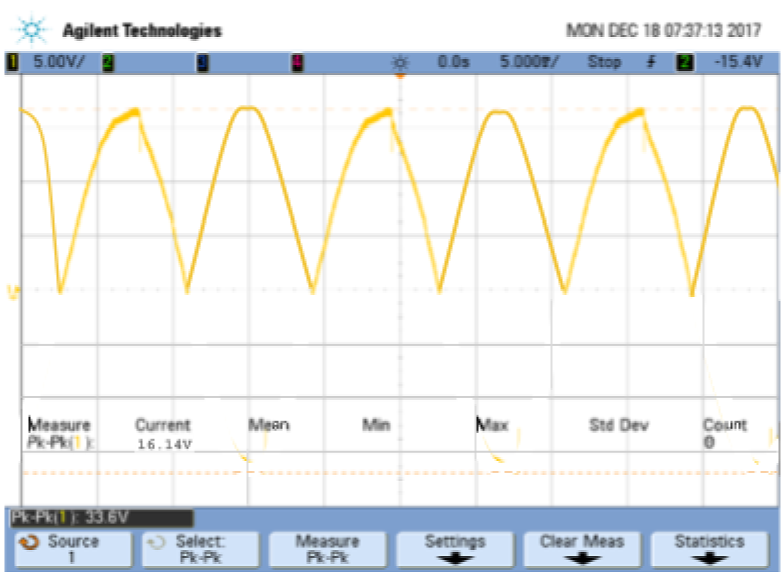


Figure 18: Rectified input

We can see that the rectified input has a peak-to-peak voltage of 16.14V, slightly less than half of the non-rectified input's peak-to-peak.

At point C (after smoothing filter and voltage regulator):

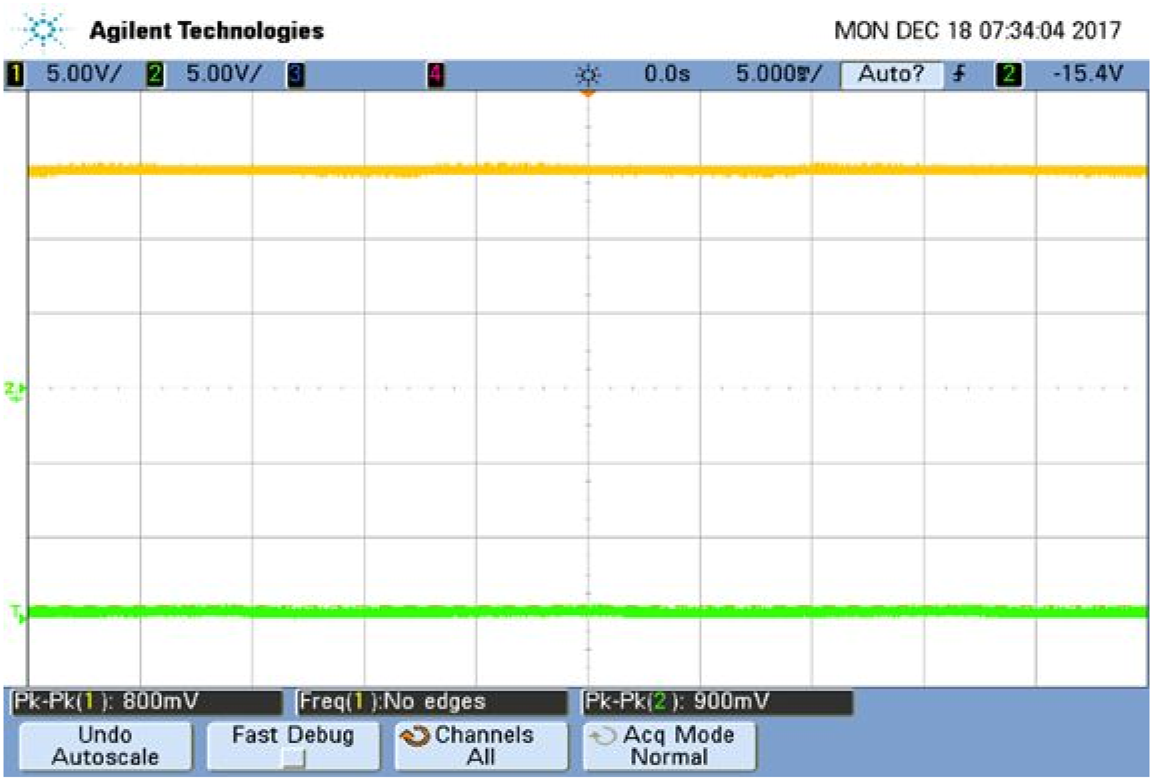


Figure 19: Power supply after smoothing filter and voltage regulator, zoomed out.

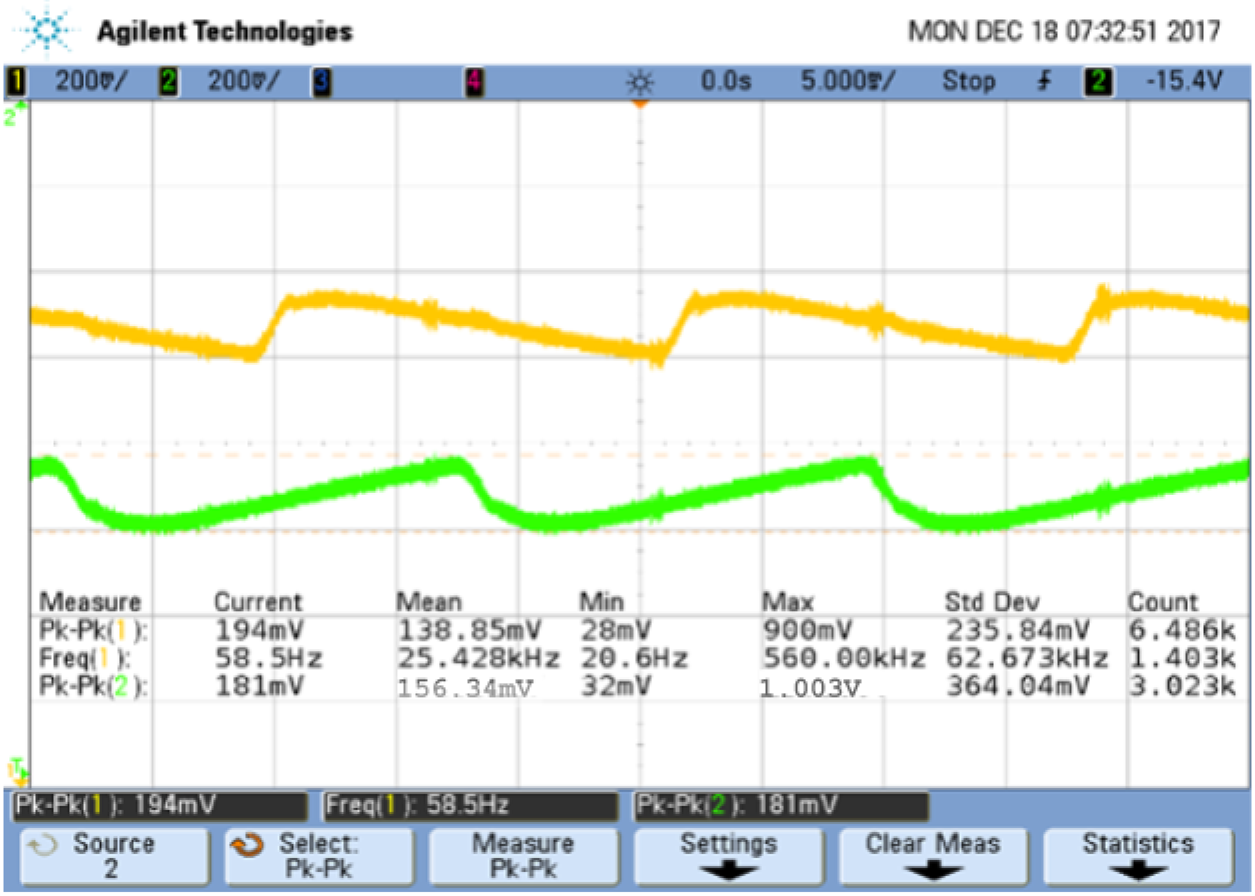


Figure 20: Power supply after smoothing filter and voltage regulator, zoomed in to see ripple.

After the smoothing filter and voltage regulator, the voltage sits at a relatively steady 15V DC. Zooming in, we can see that both positive and negative 15V have a ripple of 0.92% and 1.04%, respectively. Both of these are well within the desired range for ripple.

<u>Error analysis:</u>

Our hand calculations yielded ripple voltages of 0.55%, simulations yielded ripple voltages of 9.3%, and testing yielded ripple voltages of ~1%. Simulation was more than 800% off of testing voltage, which we attribute to the fact that circuitlab did not have the proper zener diode for us to use in simulation. Hand calculations and testing ripple voltages, however, differed by about 45%, which is not too bad for how small the values are. The ripple voltage in practice was likely higher because actual load values differed from expected.

Section Conclusion:

Our resulting power supply output was a reliably steady constant voltage. The higher the load resistance value, the steadier the voltage, which was something we needed to account for when making the supply. At first, we found capacitor and resistor values that worked perfectly with a load resistance of $2\,k\Omega$, but as soon as we reduced load to $1\,k\Omega$ or $200\,\Omega$, the ripple became too large. Once we accounted for the lowest reasonable load resistance, we got our final working power supply. The only other change we made while building the power supply was our zener diode, which we did not realize until after the first round of testing, was a 5V zener. When it began to get much too toasty, we replaced it with the 15V zener given to us.

Part 2: Filter Component

Pre-lab hand calculations:

We used the multiple feedback passband filter as our model for both our treble and bass filters.

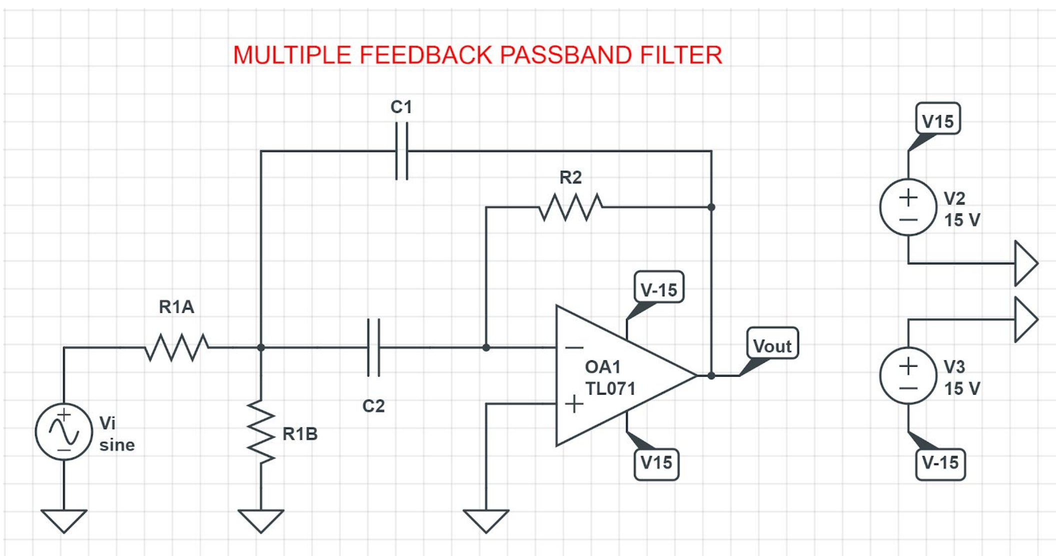


Figure 21: Multiple Feedback Passband Filter Model

Transfer function hand calculations:

$$Let\ A = voltage\ across\ R_{1B};\ B = V_n;\ s = j\omega$$

$$\frac{A-V_i}{R_{1A}} + \frac{A-V_{OUT}}{1/j\omega C_1} + \frac{A}{R_{1B}} + \frac{A-B}{1/j\omega C_2} = 0$$

$$\frac{B-A}{1/j\omega C_2} + \frac{B-V_{OUT}}{R_2} = 0$$

$$V_p = V_n\ ;\ B = 0$$

$$A(\frac{1}{R_{1A}} + j\omega C_1 + \frac{1}{R_{1B}} + j\omega C_2) + B(-j\omega C_2) = \frac{V_i}{R_{1A}} + \frac{V_{OUT}}{1/j\omega C_1}$$

$$A = (Bj\omega C_2 + \frac{V_i}{R_{1A}} + V_{OUT} j\omega C_1)(\frac{R_{1A}R_{1B}}{R_{1B}+R_{1A}+R_{1A}R_{1B}j\omega(C_1+C_2)})$$

$$-sC_2(\frac{R_{1A}R_{1B}}{R_{1B}+R_{1A}+R_{1A}R_{1B}s(C_1+C_2)})(\frac{V_i}{R_{1A}}) = \frac{V_{OUT}}{R_2} + sC_2(\frac{R_{1A}R_{1B}}{R_{1B}+R_{1A}+R_{1A}R_{1B}s(C_1+C_2)})(sC_1 V_{OUT})$$

$$\frac{V_i}{V_{OUT}} = (\frac{1}{R_2} + \frac{sC_2 sC_1 R_{1A}R_{1B}}{R_{1B}+R_{1A}+R_{1A}R_{1B}s(C_1+C_2)})(\frac{R_{1B}+R_{1A}+R_{1A}R_{1B}s(C_1+C_2)}{-sC_2R_{1B}})$$

$$= \frac{R_{1B}+R_{1A}+R_{1A}R_{1B}s(C_1+C_2)+R_2(s^2C_2C_1R_{1A}R_{1B})}{R_2(-sC_2R_{1B})}$$

$$\boxed{\frac{V_{OUT}}{V_i} = \frac{-s(R_2C_2R_{1B})}{s^2(R_2C_2C_1R_{1A}R_{1B})+s(R_{1A}R_{1B}(C_1+C_2))+(R_{1B}+R_{1A})}}$$

Based on the the transfer function, we can see that the filter is first-order because it is a constant * zero at origin * quadratic pole. The zero at origin has slope of 20 dB/decade, and the quadratic pole has slope -40 dB/decade, making for a total slope of -20 dB/decade, corresponding to a first-order filter.

We used the following equations to calculate values for resistors R1A, R1B, and R2, and for capacitors C1 and C2:

$$C1 = C2$$

$$Q = \frac{\omega_o}{bandwidth}$$

$$G = gain$$

$$R_{1A} = \frac{Q}{G(2\pi\omega_o C)}$$

$$R_{1B} = \frac{Q}{(2Q^2-G)(2\pi\omega_o C)}$$

$$R_2 = \frac{2Q}{2\pi\omega_o C}$$

<u>Treble filter:</u> $\omega_o = 8.9kHz$, G = 1.5, C = 1nF

$$Q = \frac{\omega_o}{bandwidth} = \frac{8.9 * 10^3}{2 * 10^3} = 4.45$$

$$R_{1A} = \frac{Q}{G(2\pi\omega_o C)} = \frac{4.45}{1.5*2\pi*8.9*10^3*10^{-9}} = 53k\Omega \rightarrow 51k\Omega$$

$$R_{1B} = \frac{Q}{(2Q^2-G)(2\pi\omega_o C)} = \frac{4.45}{(2*(4.45^2)-1.5)*(2\pi*8.9*10^3*10^{-9})} = 2k\Omega \rightarrow 2.2k\Omega$$

$$R_2 = \frac{2Q}{2\pi\omega_o C} = \frac{2*4.45}{2\pi*8.9*10^3*10^{-9}} = 159k\Omega \rightarrow 160k\Omega$$

<u>Bass filter:</u> $\omega_o = 250Hz$, G = 1.5*, C = 0.22 μF**

$$Q = \frac{\omega_o}{bandwidth} = \frac{250}{200} = 1.25$$

$$R_{1A} = \frac{Q}{G(2\pi\omega_o C)} = \frac{1.25}{1.5*2\pi*250*0.22*10^{-6}} = 1.8k\Omega$$

$$R_{1B} = \frac{Q}{(2Q^2-G)(2\pi\omega_o C)} = \frac{1.25}{(2*(1.25^2)-1.5)*(2\pi*250*0.22*10^{-6})} = 3.3k\Omega$$

$$R_2 = \frac{2Q}{2\pi\omega_o C} = \frac{2*1.25}{2\pi*250*0.22*10^{-6}} = 8.2k\Omega$$

We chose a gain of 1.5 because we wanted any minor static produced during the filtering process to be masked by the increase in volume. Also, since a bandpass filter tends to push the gain down, we wanted to artificially raise the gain to "make up" for the loss. The capacitance values were chosen after some hand calculation trial-and-error because it yielded the best resistor values available in Detkin lab.

The expected input of these filters are waveforms of various frequencies, and the output should be the waveforms either attenuated, if it is outside the bandwidth, or with the appropriate gain, if within the bandwidth. The output load is the 8Ω speaker.

We require a DC supply of 15V to rail our op-amps. We also need a maximum of about 25mA current to run each op-amp. Power is current times voltage, so we need to budget 15V * 50mA = 0.75W of power.

<u>Pre-lab simulation:</u>

To create the filters, we use a TL071 op-amp, the specified resistors, and two audio jacks: one for input, and one for output. The 15-volt DC voltage delivered to the Op-Amp rails are provided by our power supply, detailed in section 1 of the lab report.

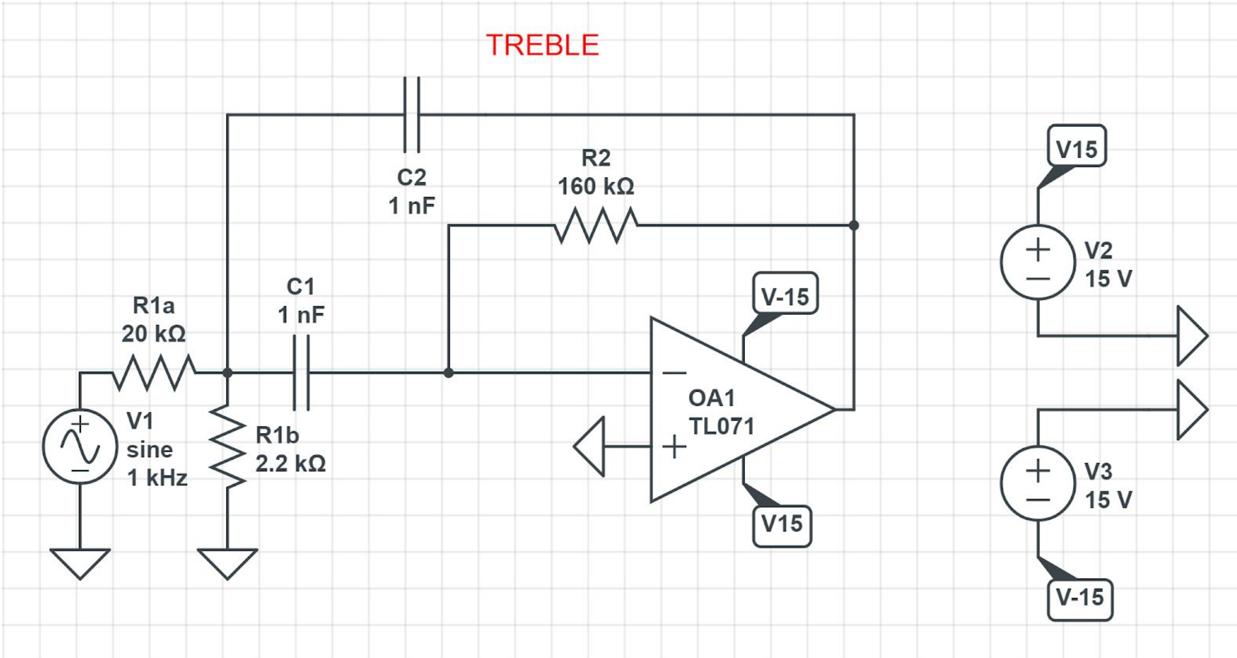


Figure 22: Schematic for Treble Filter

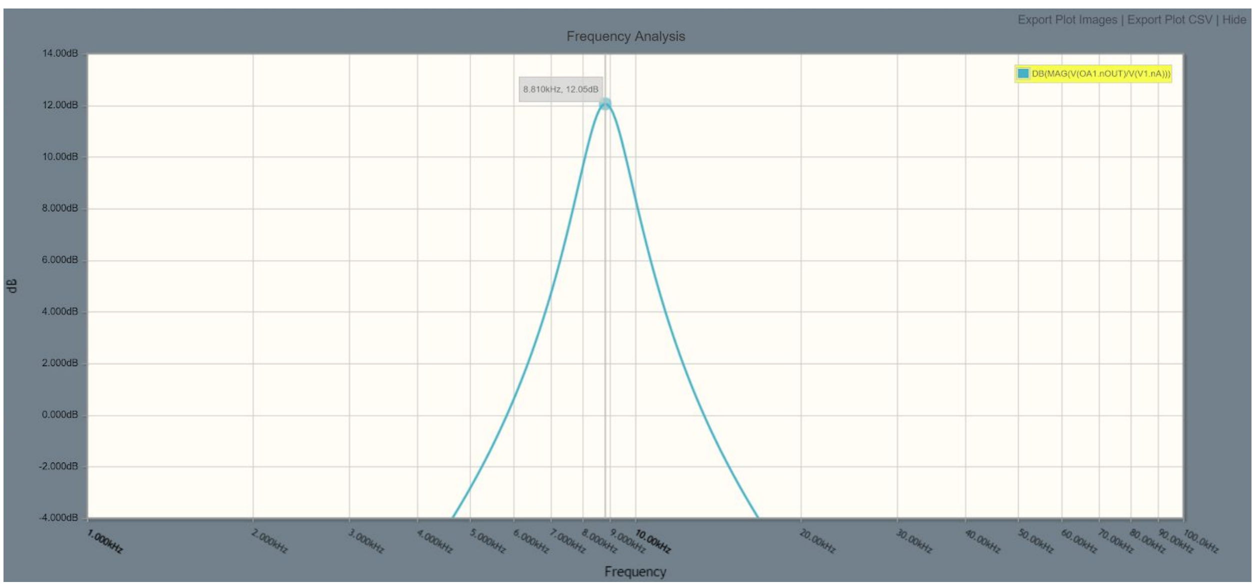


Figure 23: Treble Bode Plot with Center Frequency Label

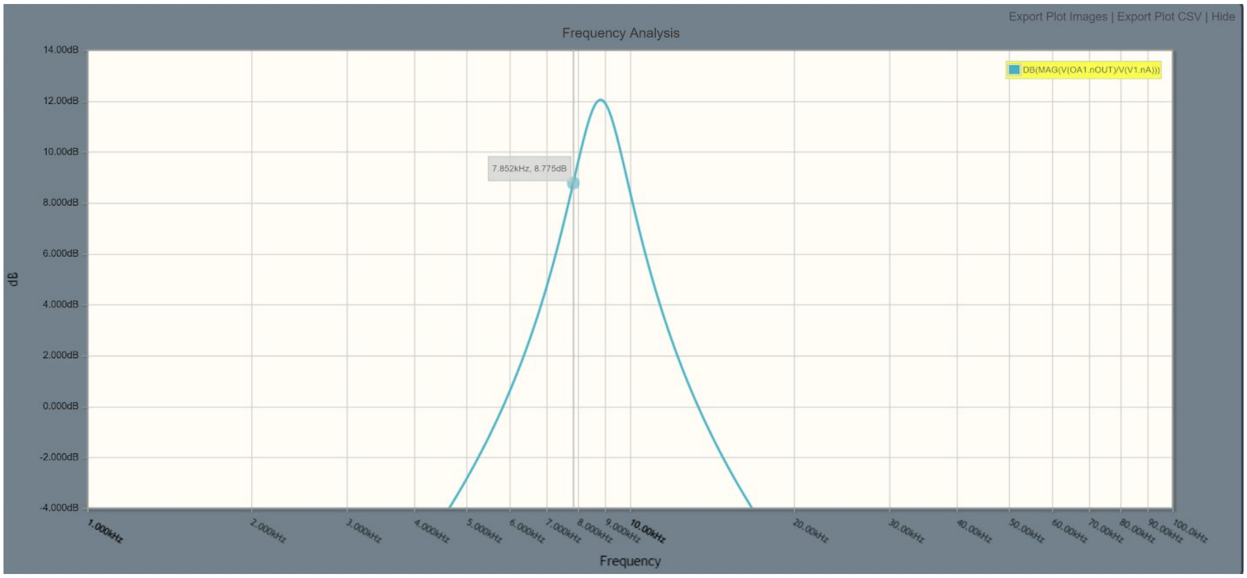


Figure 24: Treble Bode plot with low cut-off labeled

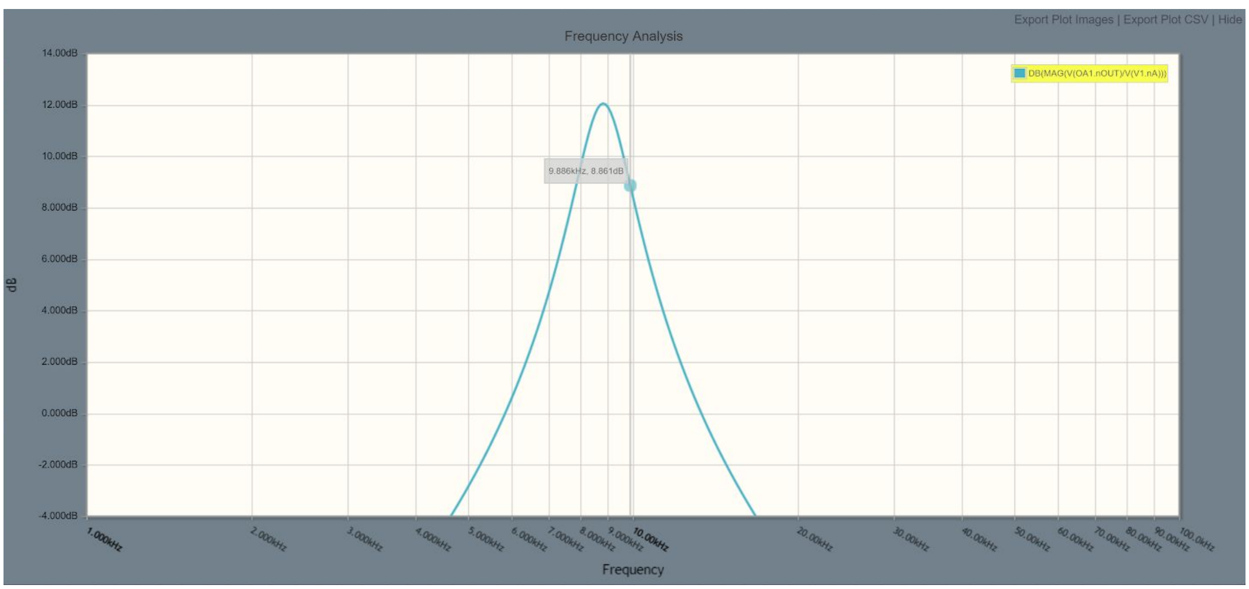


Figure 25: Treble Bode with high cut-off labeled

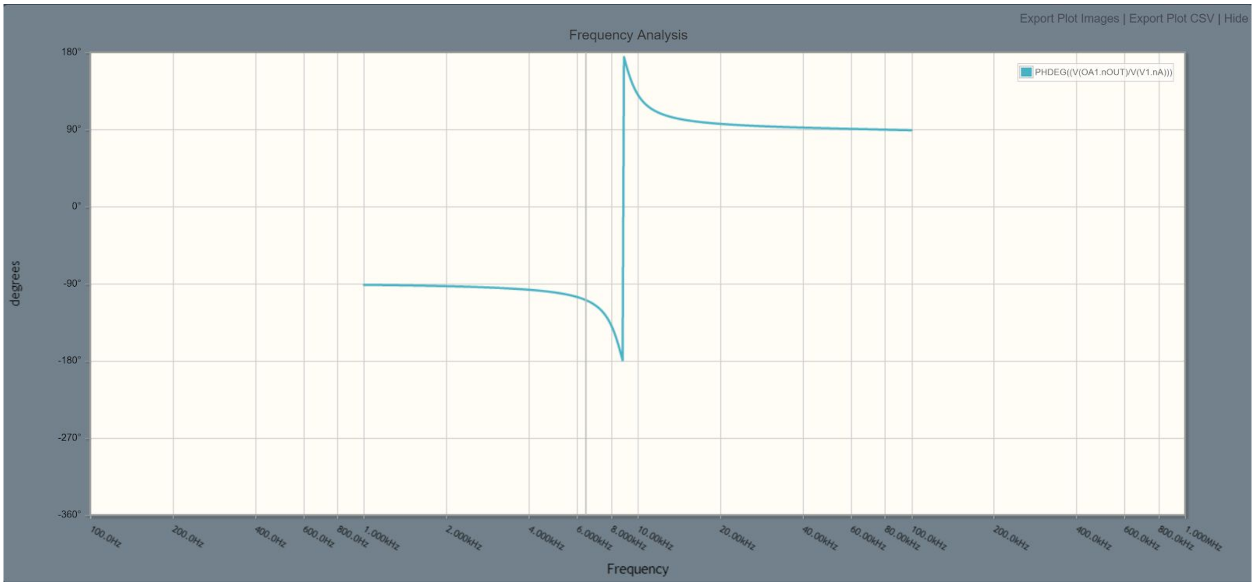


Figure 26: Treble Bode plot for phase

Slope, and thus roll-off rate, of bode plot is -20dB/decade, so it is a first-order filter.

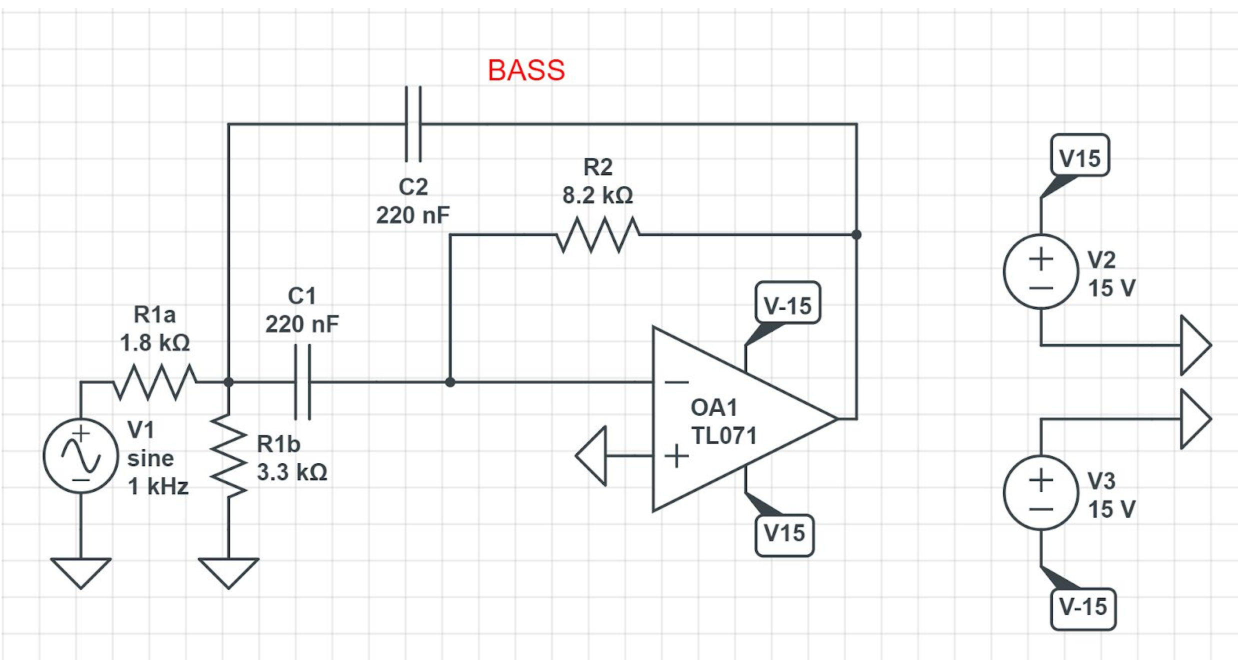


Figure 27: Schematic for Bass filter

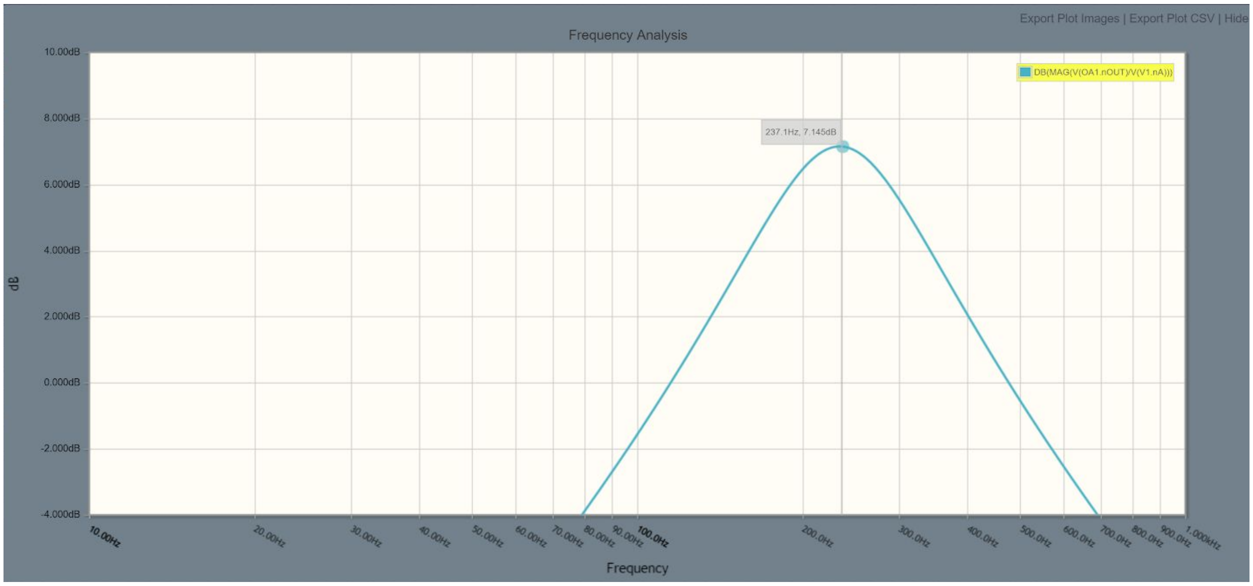


Figure 28: Bass Bode plot with center frequency labeled

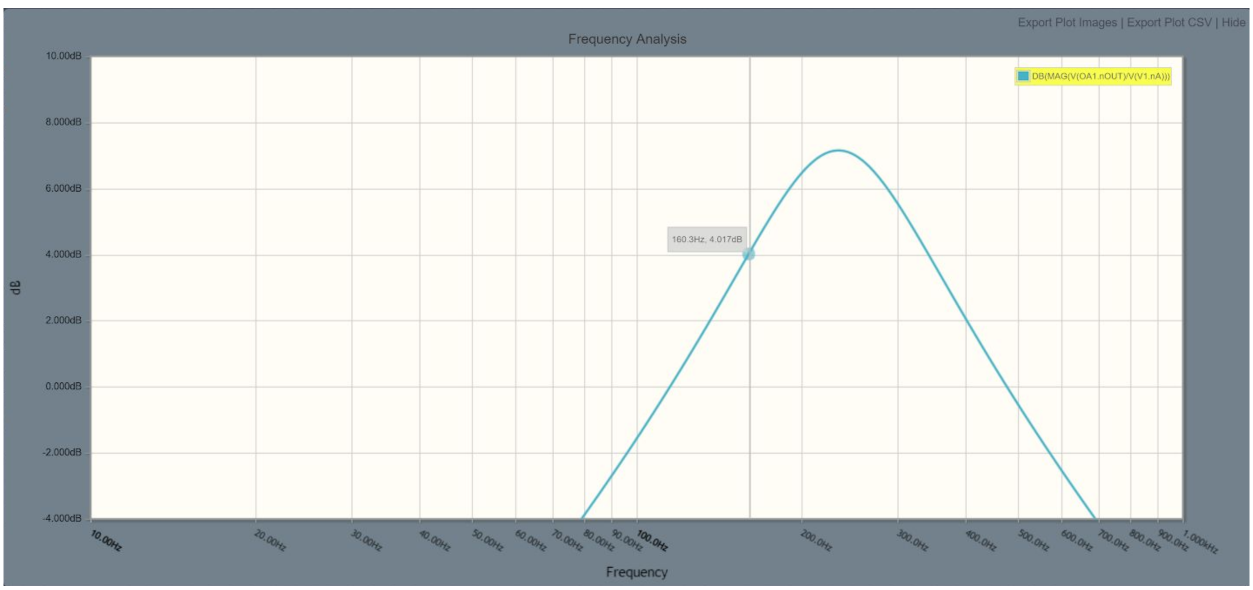


Figure 29: Bass Bode plot with low cut-off labeled

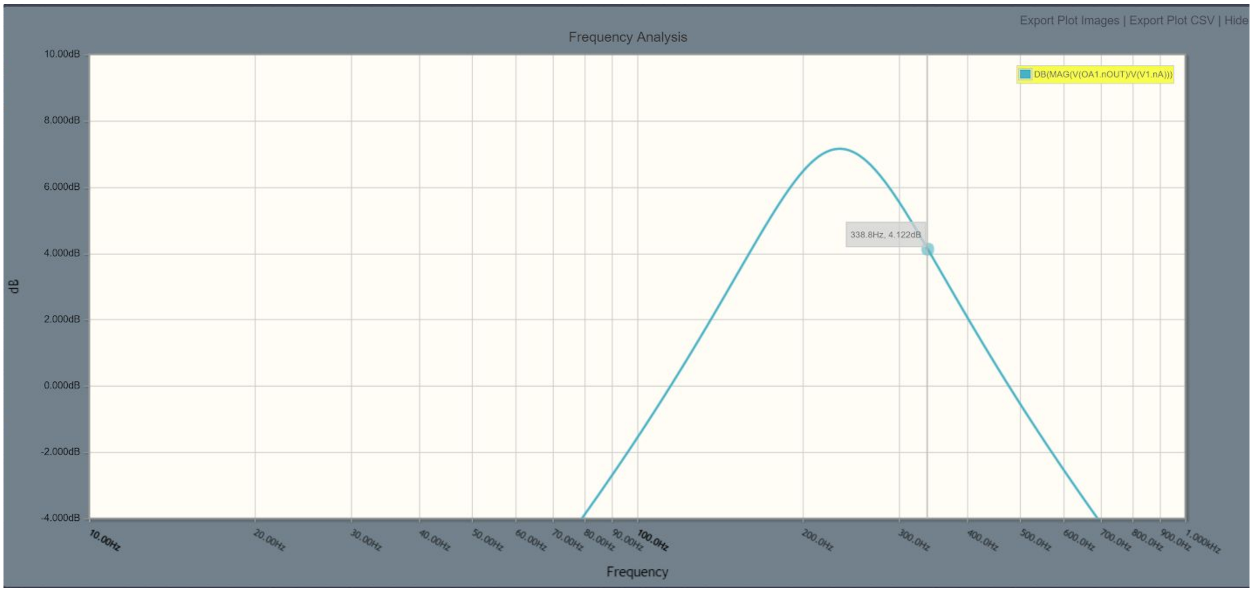


Figure 30:Bass Bode with high cut-off labeled

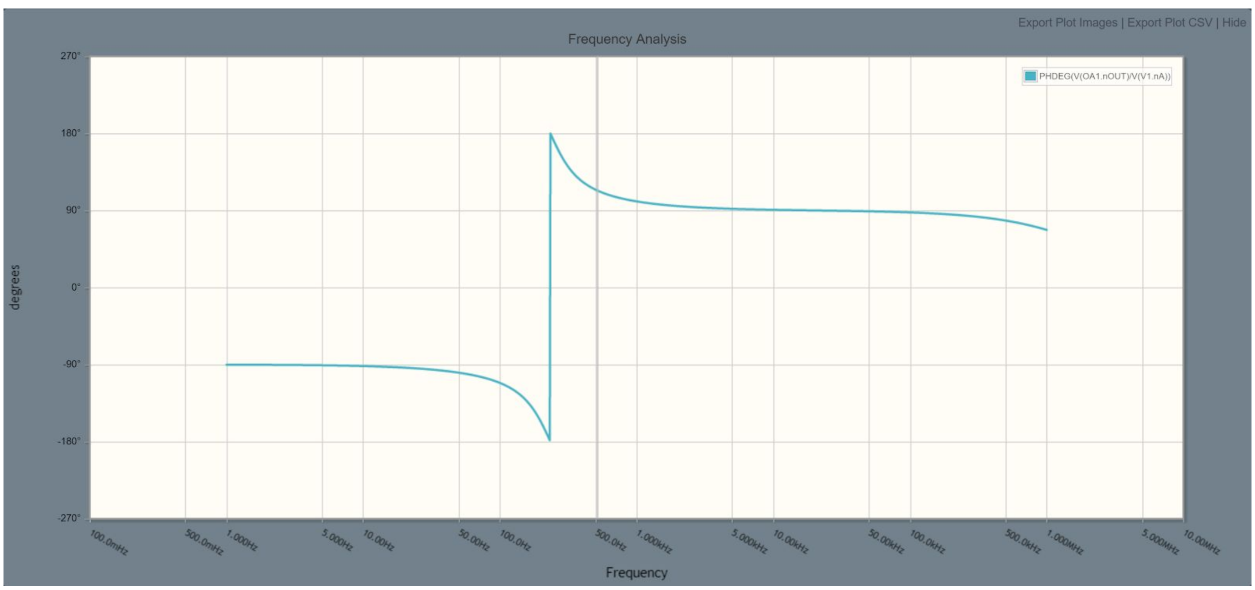


Figure 31: Treble Bode plot for phase

Slope, and thus roll-off rate, of bode plot is -20dB/decade, so it is a first-order filter.

Input/output impedance of each subcomponent shown:

The impedance of each resistor is simply its resistance in Ohms. The impedance of each capacitor is 1/jwC Ohms.

Time domain simulation at center frequencies:

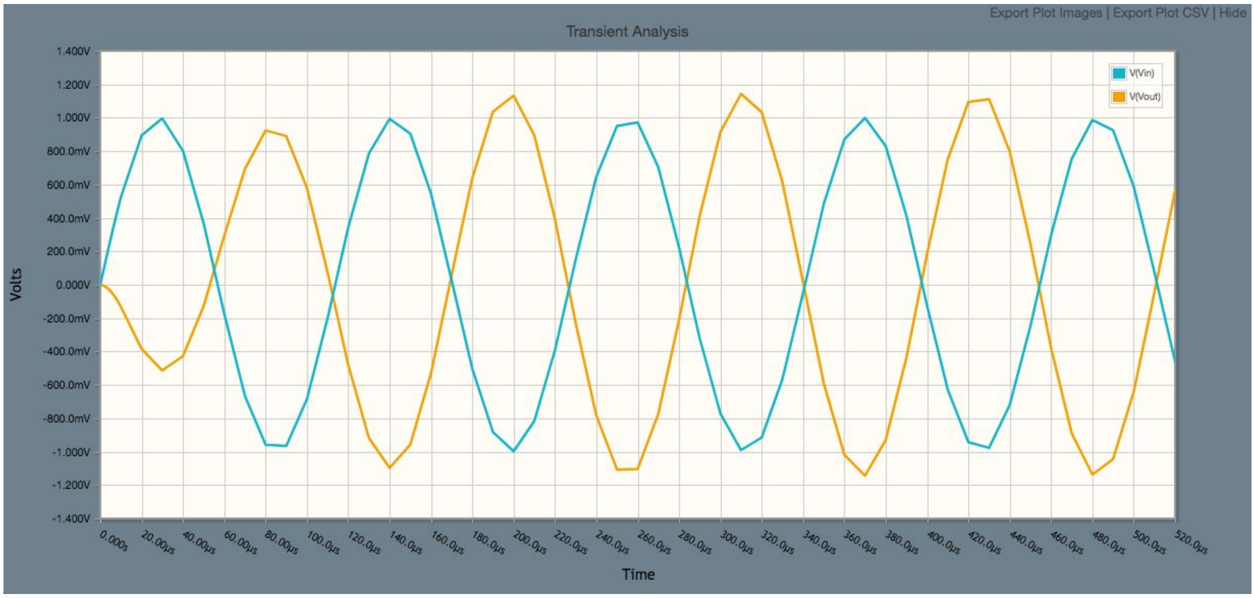


Figure 32: Time Domain Simulation of input/output Treble at center frequency (8.8kHz)

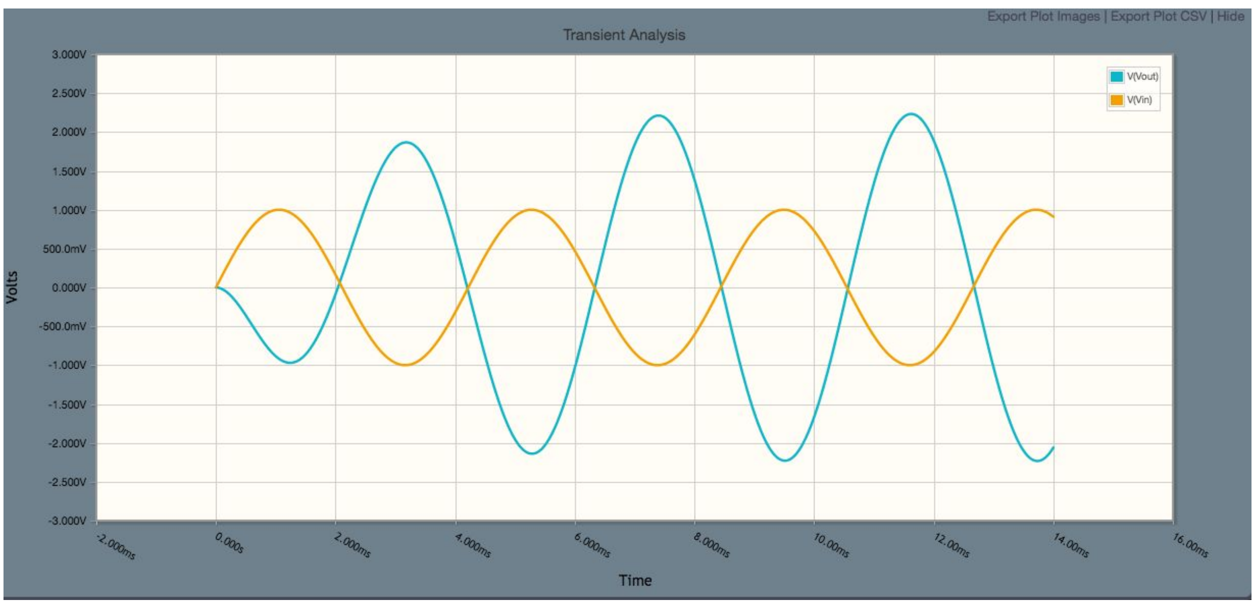


Figure 33: Time Domain Simulation of input/output Bass at center frequency (237Hz)

Time domain simulations at cutoff frequencies:

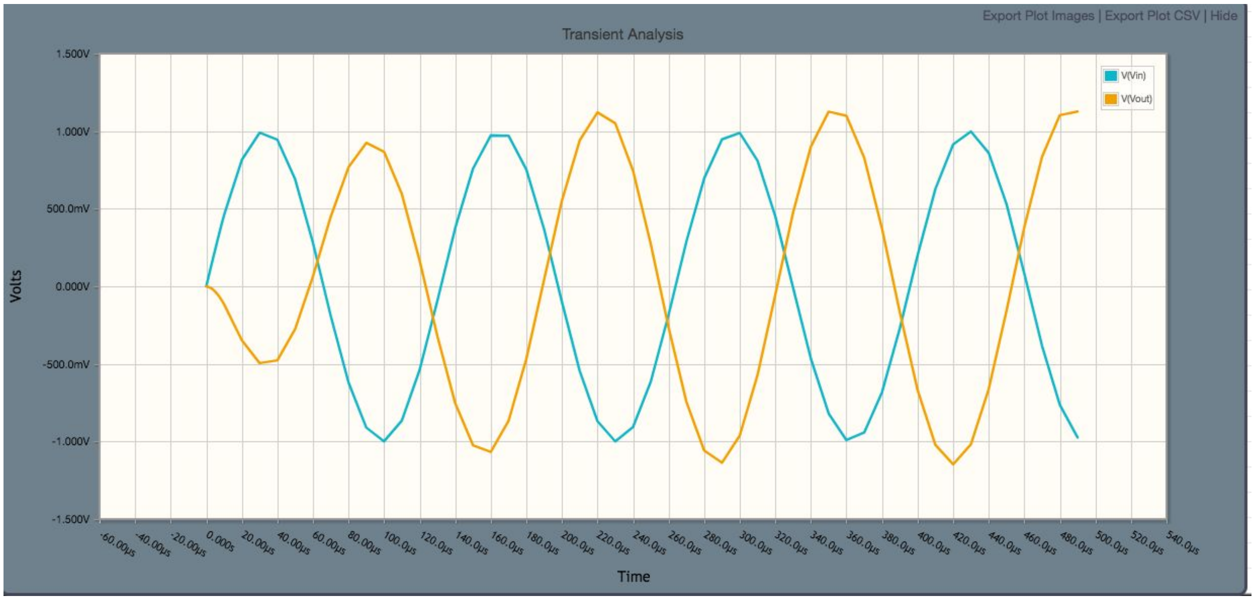


Figure 34: Time Domain Simulation of input/output Treble at low corner frequency (7.58kHz)

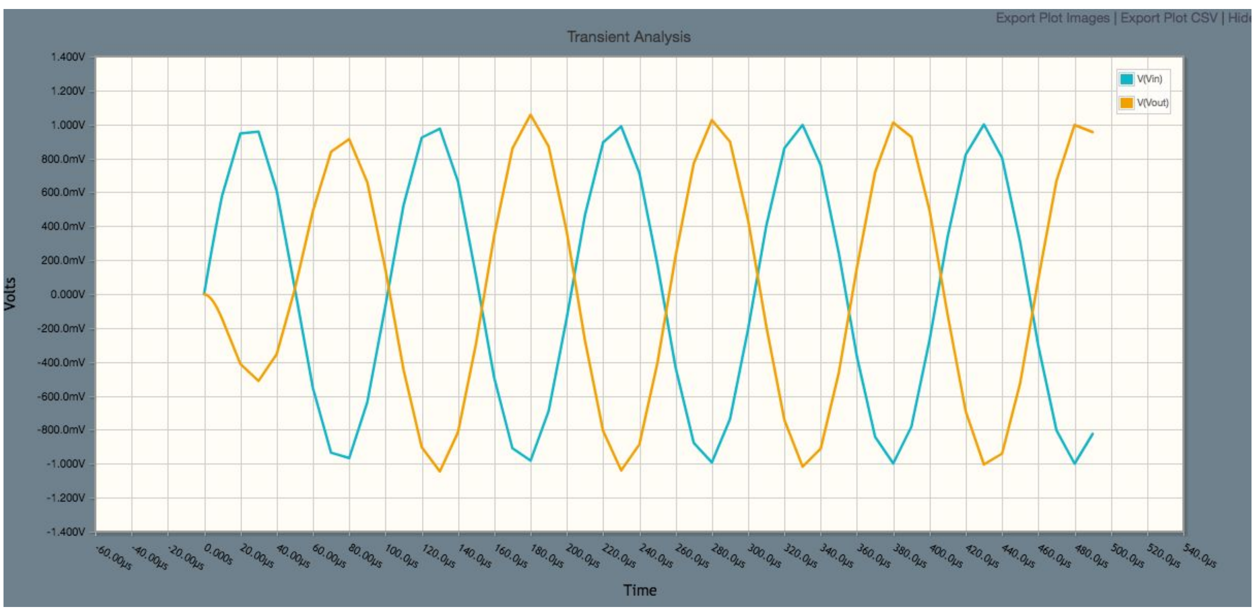


Figure 35: Time Domain Simulation of input/output Treble at high corner frequency 9.89kHz)

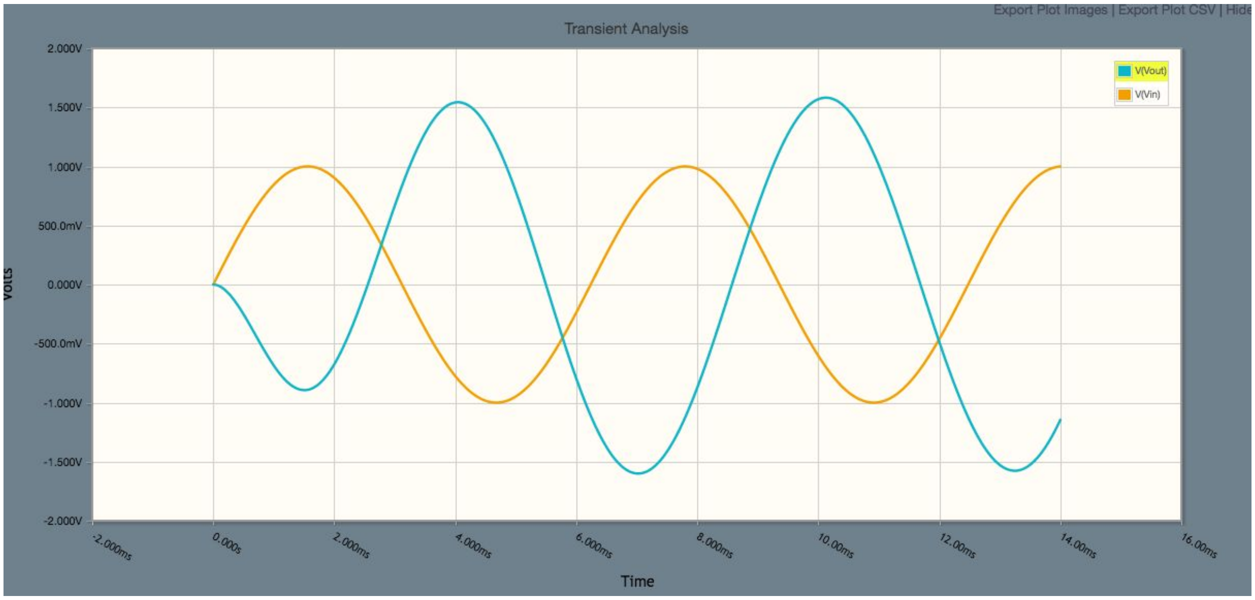


Figure 36: Time Domain Simulation of input/output Bass at low corner frequency (160Hz)

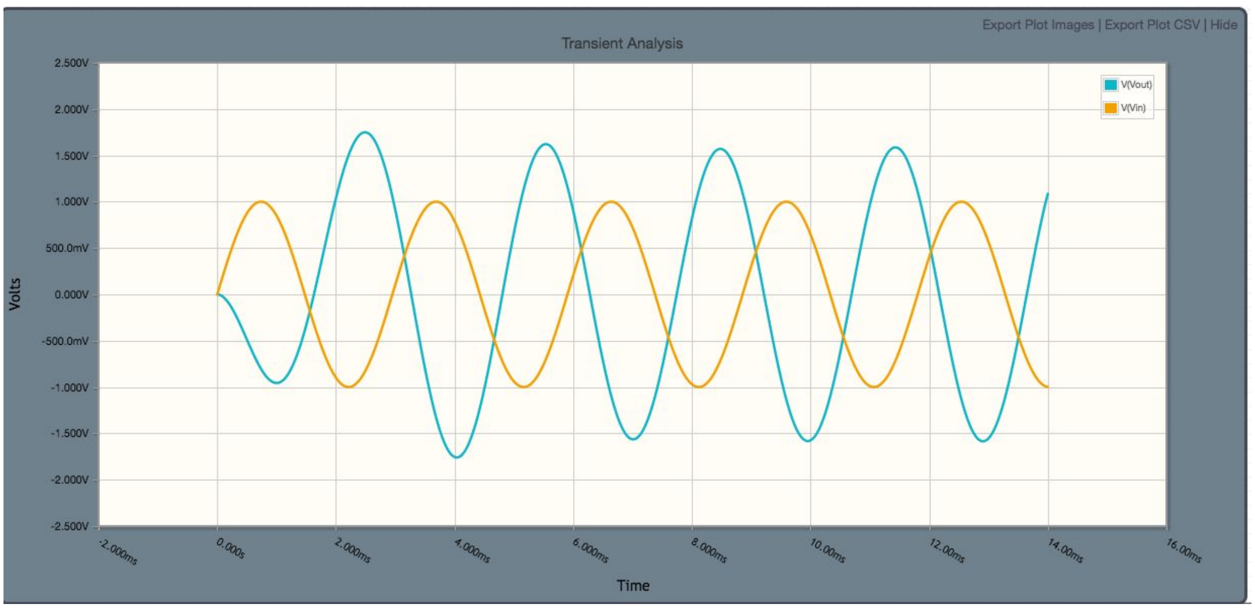
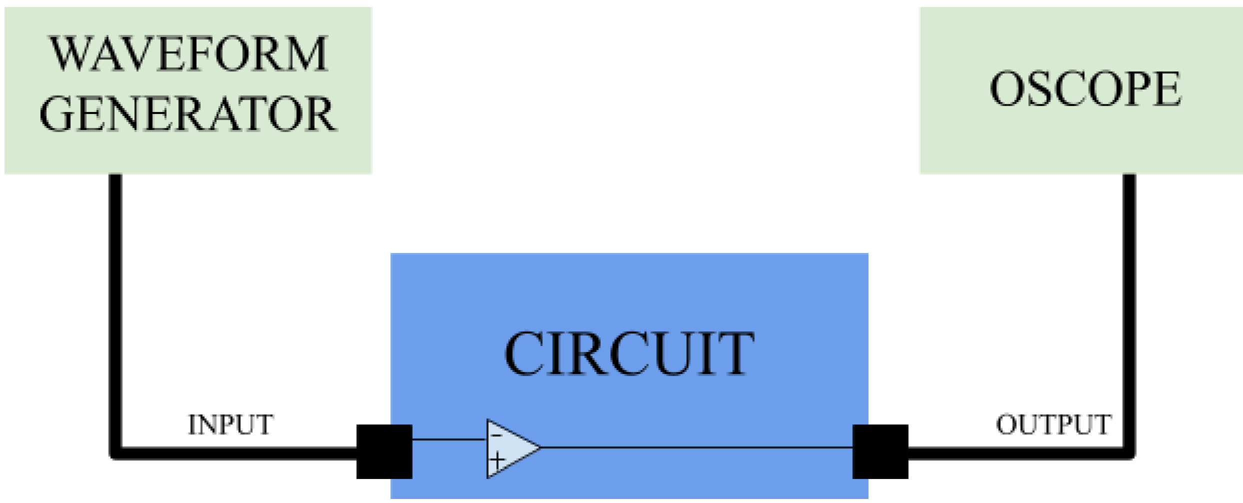


Figure 36: Time Domain Simulation of input/output Bass at high corner frequency 338Hz)

We require a DC supply of 15V to rail our op-amps. We also need a maximum of about 25mA current to run each filter's op-amp. Power is current times voltage, so we need to budget 15V * 50mA = 0.75W of power. In actuality, our op amps will pull much less than 25mA. (See figure 12 for current pulled.)

<u>Experimental Setup:</u>

We use an oscilloscope and a waveform generator to test our filter. On the waveform, we set a sine wave at 300mV, then examine the output on the oscilloscope at various set frequencies. We want to use an oscilloscope rather than a multimeter because it allows us to observe frequency response, which would be too difficult to gauge on a multimeter. Observe the schematic below for how we connected our circuit to the waveform generator and oscilloscope:



To test the treble filter, we set the frequency to 6kHz (below the cutoff), 7.9kHz (lower cutoff), 8.9kHz (middle of range), 9.9kHz (upper cutoff), and 11kHz (above the cutoff). To test the bass filter, we set the frequency to 100Hz (below the cutoff), 150Hz (lower cutoff), 250Hz (middle of range), 350Hz (upper cutoff), and 400Hz (above the cutoff). At or between the cutoffs, we should get a voltage within max voltage and max voltage divided by $\sqrt{2}$. Below and above the cutoffs, we should get an output less than max voltage divided by $\sqrt{2}$.

Experimental Procedure

In testing the filter, we generated a waveform, twiddled the frequency until we hit a max amplitude, and declared that frequency the center frequency. Then to find the corner frequencies, we took the max amplitude and divided by $\sqrt{2}$. Then again, we twiddled the frequency again until we got two different frequencies, one less than and one greater than the center frequency, that gave gain equal to the max amplitude divided by $\sqrt{2}$.

From the power supply, we require a DC supply of 15V to rail our op-amps. We also need a maximum of 50mA current to run the two op-amps. Power is current times voltage, so we need to budget 15V * 50mA = 0.75W of power.

Experimental results

Treble:

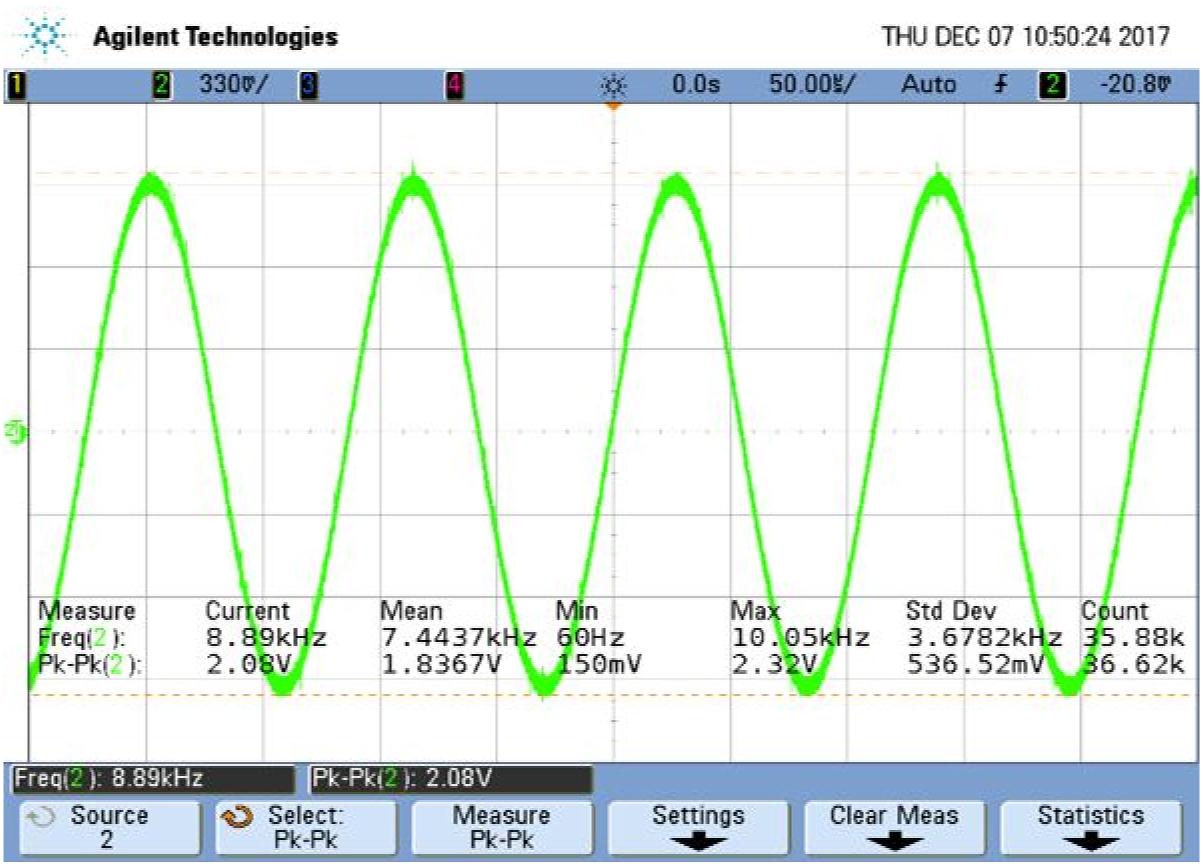


Figure 37: Time domain output at Treble center frequency 8.89kHz

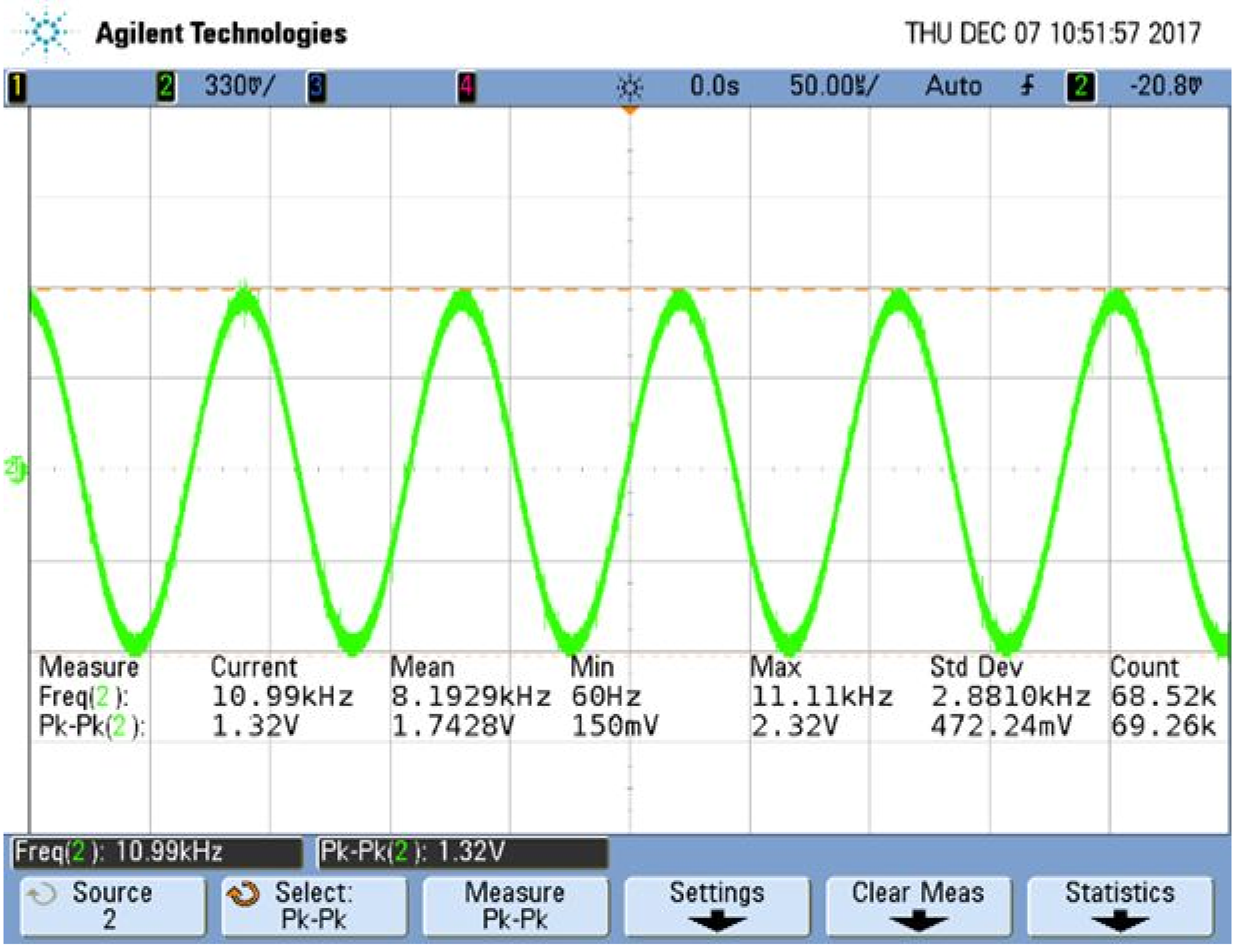


Figure 38: Time domain output at Treble high cutoff frequency 10.99kHz

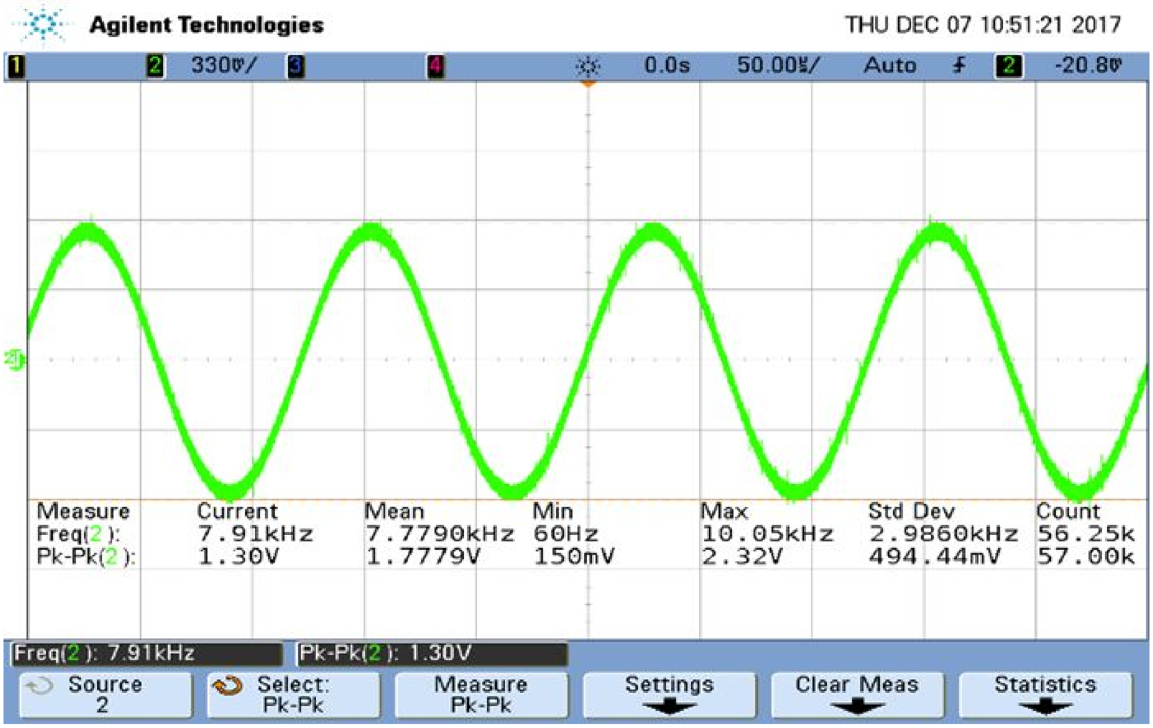


Figure 39: Time domain output at Treble low cutoff frequency 7.91kHz

Bass:

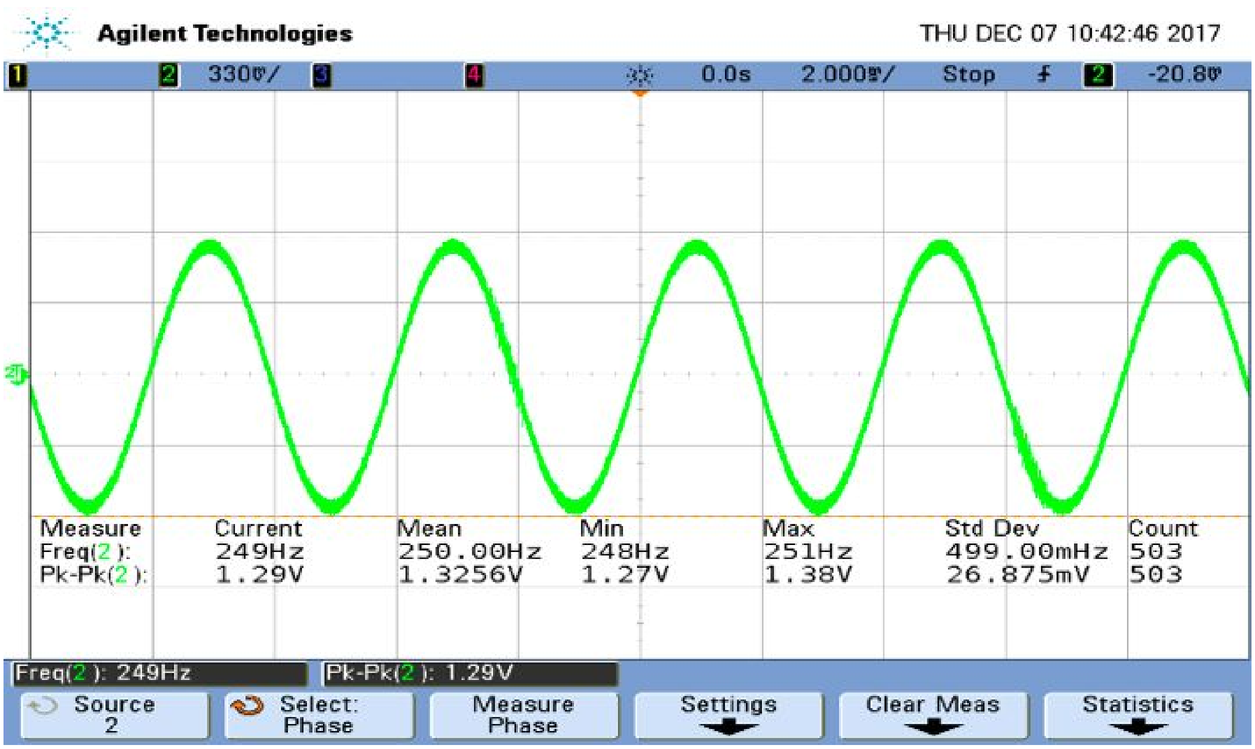


Figure 40: Time domain output at Bass center frequency 249 Hz

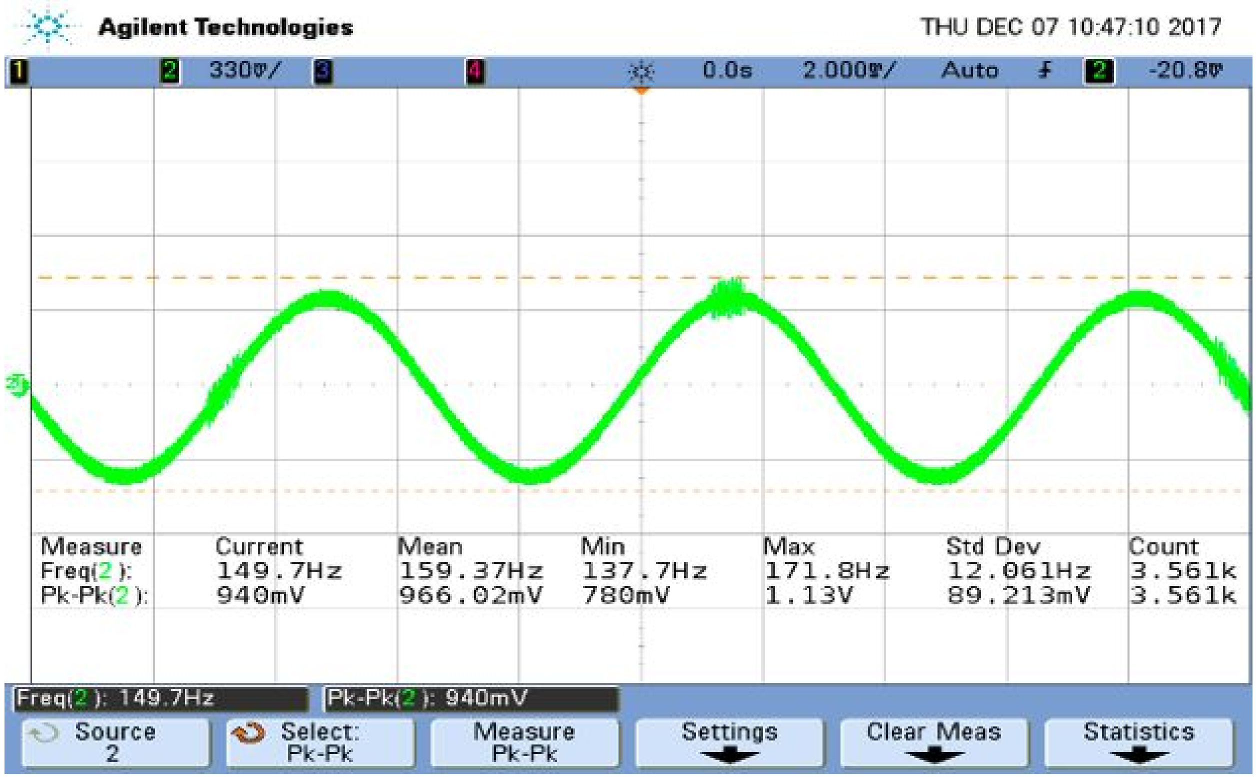


Figure 41: Time domain output at Bass low cutoff frequency 149.7Hz

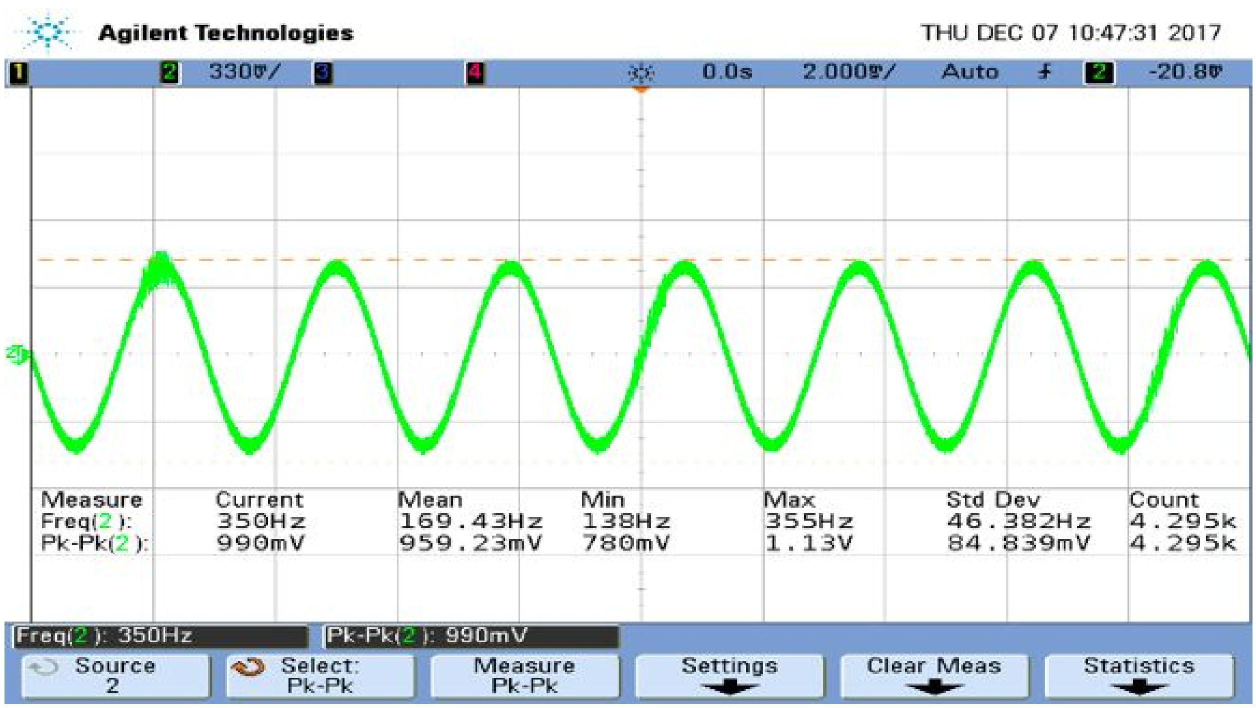


Figure 42: Time domain output at Bass high cutoff frequency 350Hz

Notice that the cutoff frequencies have peak-to-peak value of approximately the center frequency peak-to-peak value divided by $\sqrt{2}$.

$V_{center} = 2.08V$

$V_{cutoff} = \frac{2.08}{\sqrt{2}} = 1.4V$

The cutoff frequencies of 7.91kHz and 10.99kHz have the cutoff peak-to-peak voltage. Figures 37 through 42 show the difference in magnitude for various frequencies.

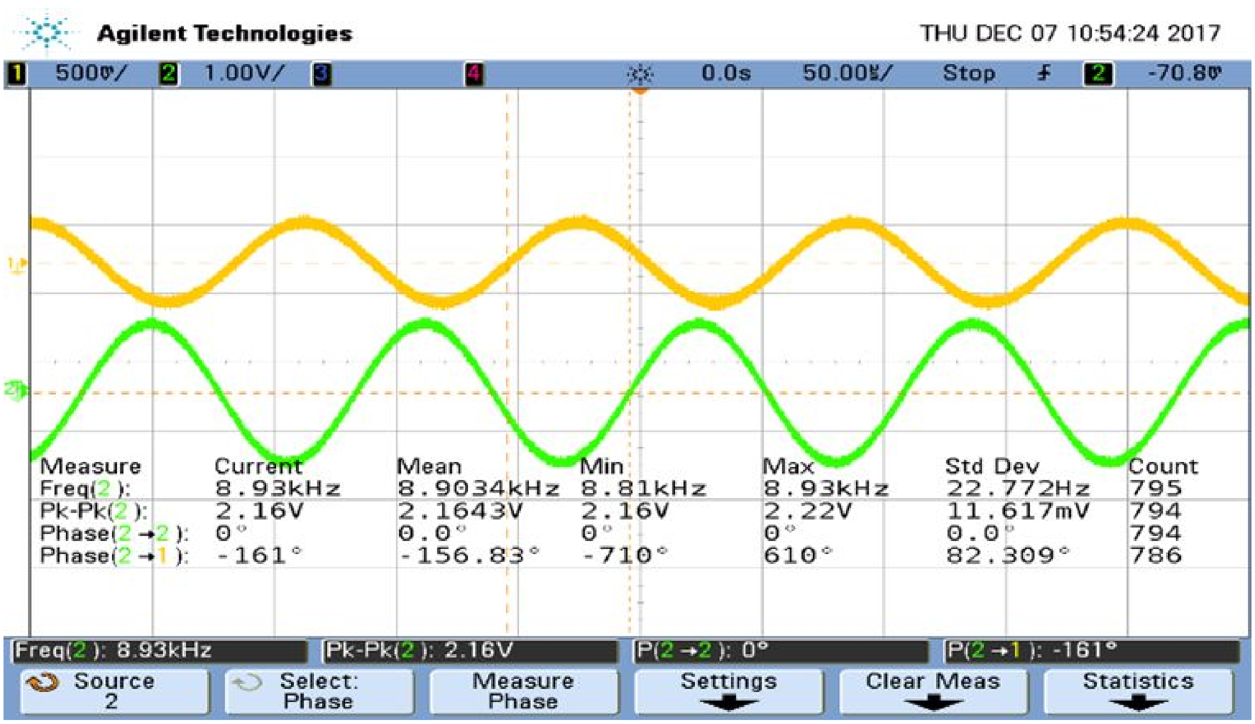


Figure 40: Phase difference between Treble input and output waveforms.

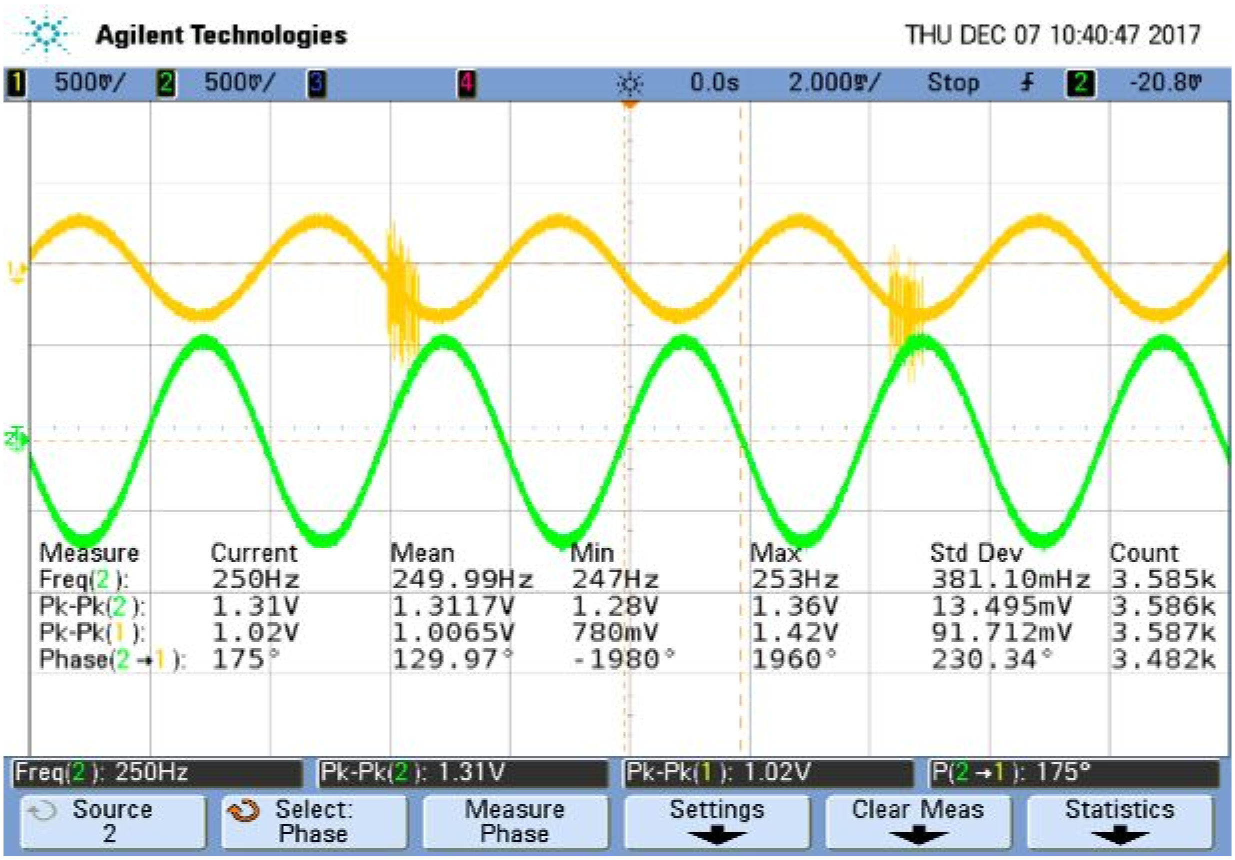


Figure 40: Phase difference between Bass input and output waveforms.

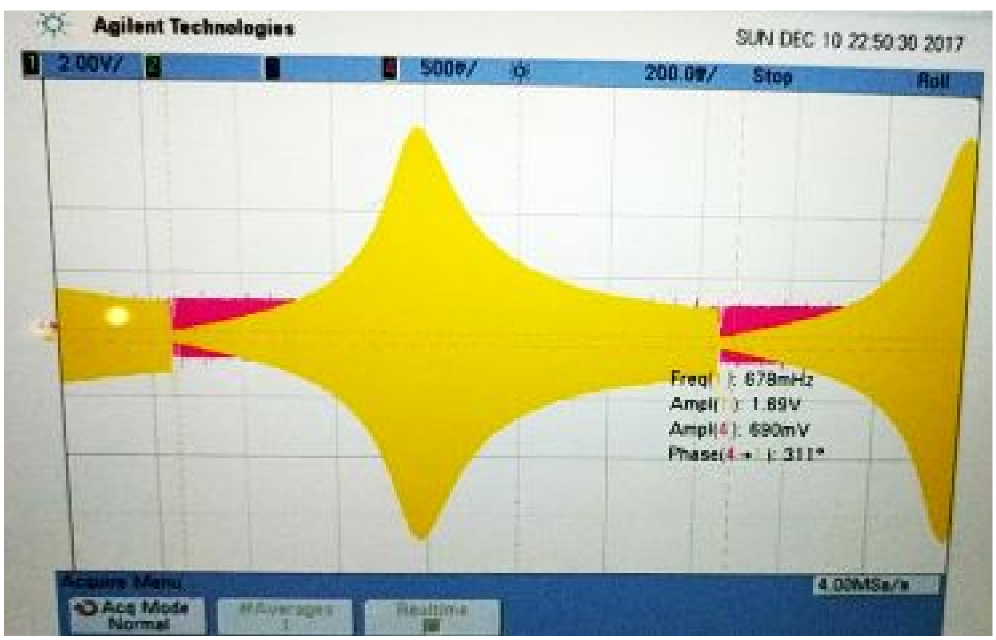


Figure 41a: Bode plot of magnitude on oscilloscope: Treble.

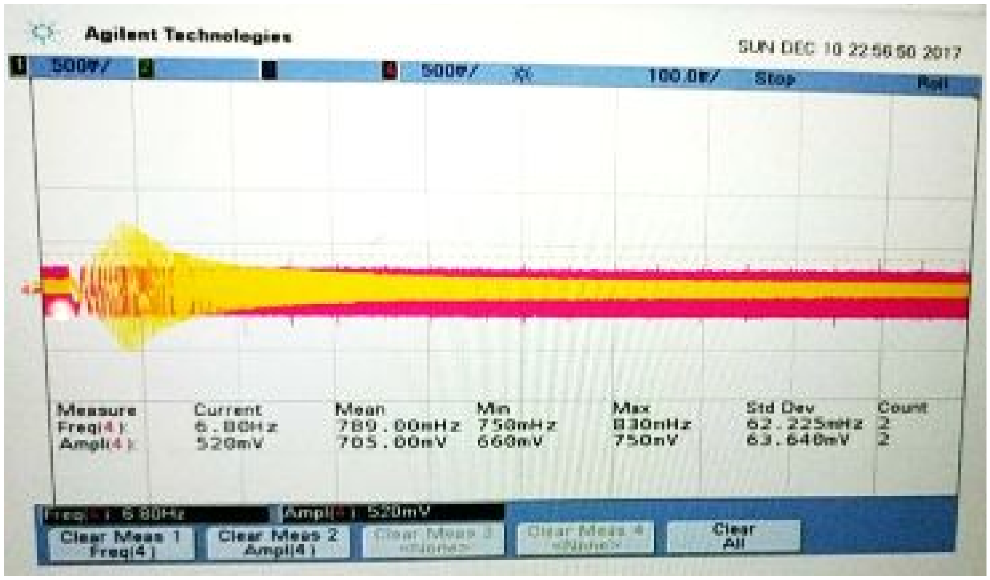


Figure 41b: Bode plot of magnitude on oscilloscope: Bass.

<u>Error analysis</u>

Table 1: Treble Filter Error Analysis

	Hand Calc	Simulation	Measurement	% diff btw hand and meas.	% diff btw sim. and meas.
Low cut off	7.9kHz	*7.58kHz*	7.91kHz	0.13%	4.17%
High cut off	9.9kHz	9.89kHz	10.99kHz	9.92%	10.01%
Center freq	8.9kHz	8.8kHz	8.89kHz	0.11%	1.01%
Bandwidth	2kHz	2.31kHz	3.08kHz	35.06%	25%
Q	4.45	3.81	2.88	54.51%	32.29%

The largest error was the percent difference in bandwidth of the Treble filter which affected the Quality factor. Since our filter was first-order, the slightest change in resistor or capacitor values had a great effect on the bandpass range. There was also an unaccounted-for resistance of the audio jack block that may have altered the bandpass. Although the low cutoff was spot-on, and the high cutoff is only 1kHz off, which is not too bad since treble takes in the higher-frequency audio signals anyway, this practically small deviation resulted in a large percent difference.

Table 2: Bass Filter Error Analysis

	Hand Calc	Simulation	Measurement	% diff btw hand and meas.	% diff btw sim. and meas.
Low cut off	150Hz	160.3Hz	149.7Hz	0.20%	7.08%
High cut off	350Hz	338.8Hz	350Hz	0%	3.20%
Center freq	250Hz	237.1Hz	249Hz	0.40%	4.78%
Bandwidth	200Hz	178.5Hz	200.3Hz	0.15%	10.88%
Q	1.25	1.33	1.24Hz	0.81%	7.26%

As for the Bass filter, the actual measured values were a lot closer to expected compared to the simulated values, which caused the 10.88% difference between them. This can be attributed to element natural variance.

Section conclusion

Though percent difference between simulation and measured were fairly significant, our resulting filters are reliable because they do filter the proper general range of frequencies. Especially for bass, which was more selective, our filter exceeded expectations. For treble, the lower cutoff frequency was spot-on, which is excellent. The higher cutoff frequency was 1kHz higher than expected, which is unfortunate, but at least it is the higher cutoff for a range that generally encompasses high frequencies anyway. While designing the filters, we actually started with a cascading low-pass, high-pass filter, but since we were using first-order filters, the resulting bandpass was much too wide. As a result, we instead switched to an infinite gain multiple feedback filter, which used just one op-amp, which fixed the issue of bandpass widening. Ultimately, we succeeded in our goal of designing a power supply, treble filter, and bass filter.

Part 3: Final Integration

Hand calculations

Our power supply can deliver up to We require a DC supply of 15V to rail our op-amps. We also need about 25mA current per op-amp. Power is current times voltage, so for two op-amps, we need to budget 15V * 50mA = 0.75W of power. The power supply can deliver up to 50mA, and each filter requires up to 25mA. 50mA = 25mA + 25mA. KCL is maintained.

Pre-lab simulation:

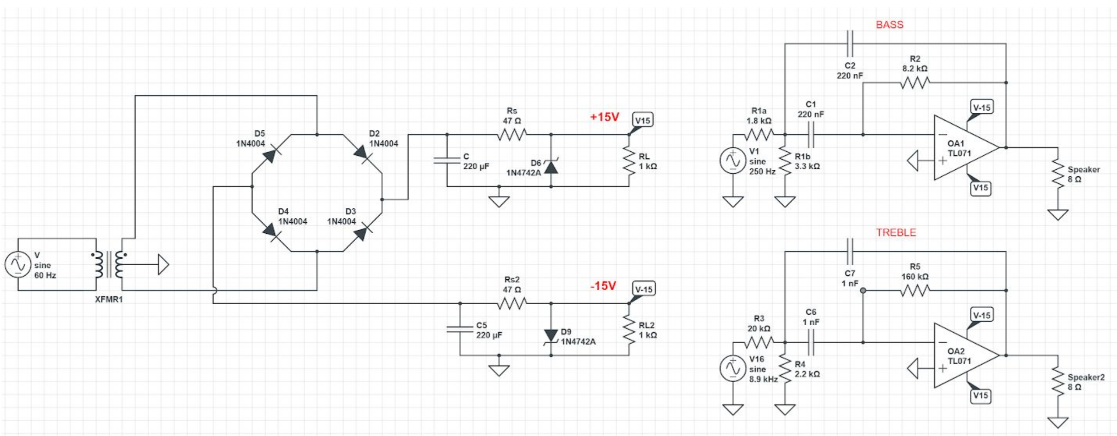


Figure 42: Simulation showing all components properly connected

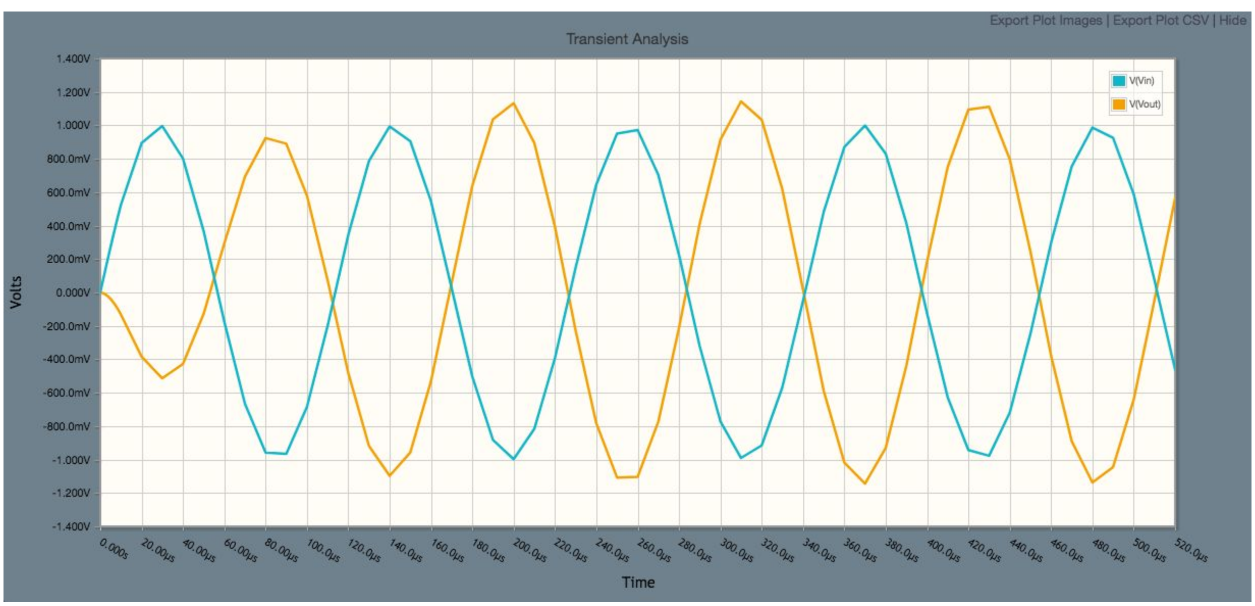


Figure 43: Time Domain Simulation of Treble at center frequency (8.8kHz)

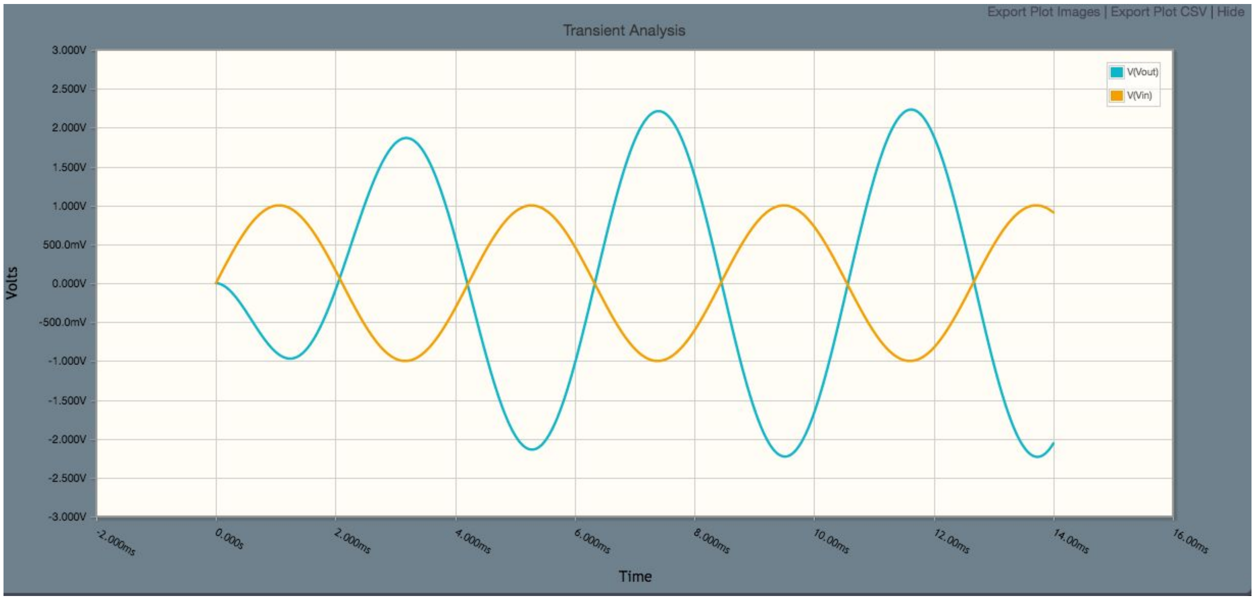


Figure 44: Time domain simulation of bass at center frequency (250Hz)

<u>Experimental setup:</u>

To test the complete speaker filter with power supply, we connected all components as specified in figure 45. Input is from the waveform generator and output is the oscilloscope.

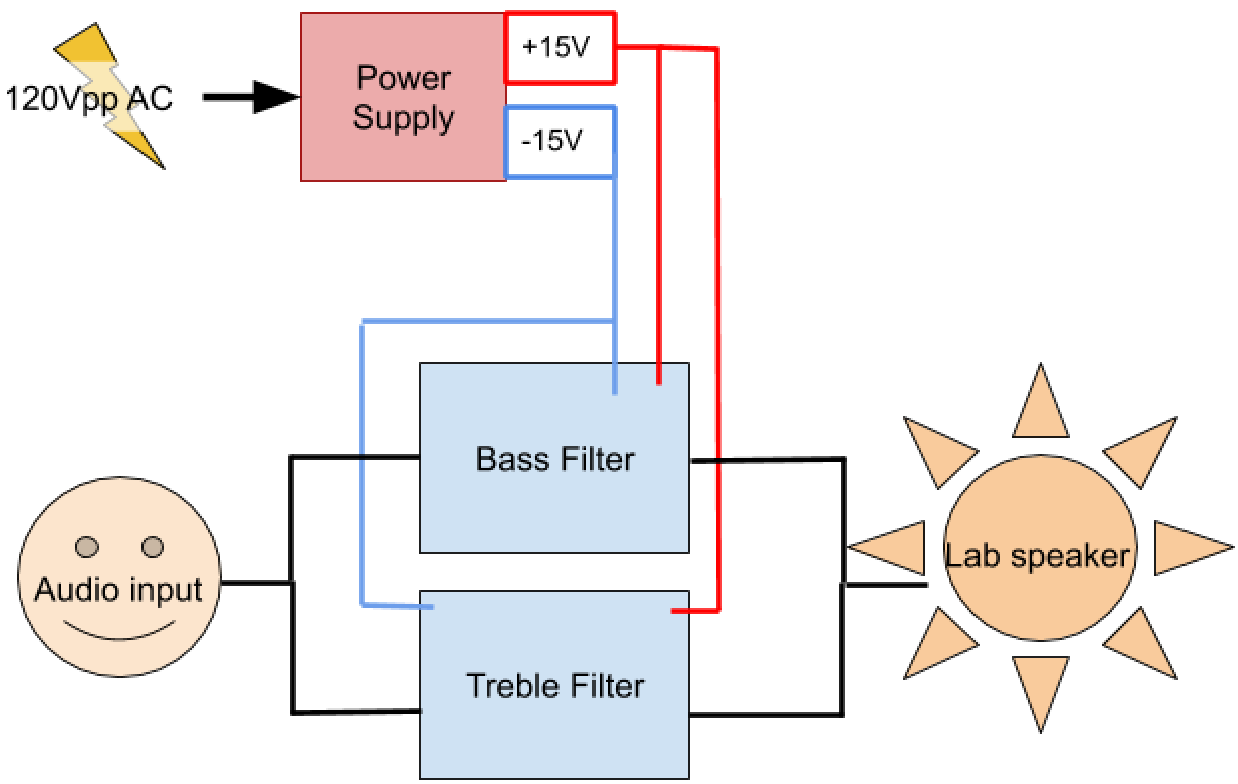


Figure 45: Complete connected project, with power supply powering the op-amps.

We used the oscilloscope and waveform generator to test our filters. To test the treble filter, we set the frequency on the waveform generator to 6kHz (below the cutoff), 7.9kHz (lower cutoff), 8.9kHz (middle of range), 9.9kHz (upper cutoff), and 11kHz (above the cutoff). To test the bass filter, we set the frequency to 100Hz (below the cutoff), 150Hz (lower cutoff), 250Hz (middle of range), 350Hz (upper cutoff), and 400Hz (above the cutoff). At or between the cutoffs, we should get a voltage within max voltage and max voltage divided by $\sqrt{2}$. Below and above the cutoffs, we should get an output less than max voltage divided by $\sqrt{2}$.

We verified the output responses by using the measure function on the oscilloscope, moving frequencies around until we achieved the desired voltage value.

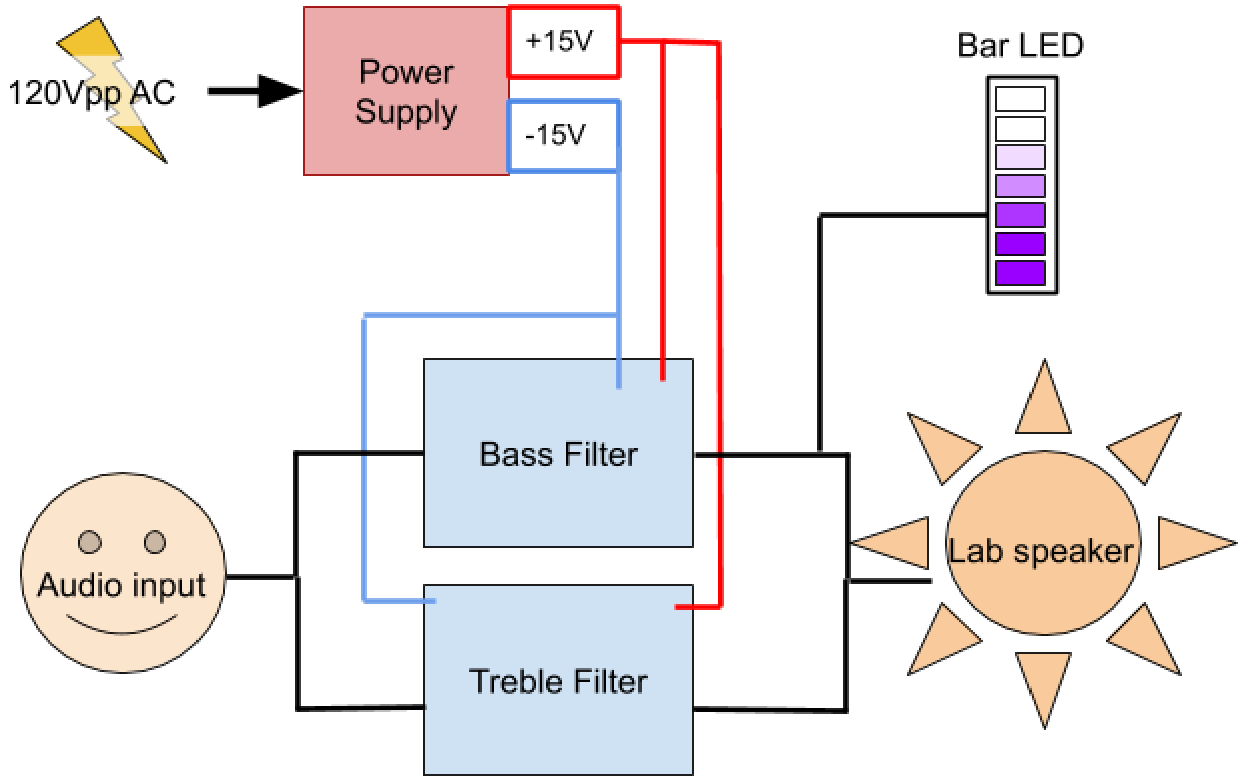
<u>Experimental Procedure</u>

Before attaching the entire project together, we first tested the power supply under various load resistances, to make sure that it would function properly, before connecting it to our filters. Then we used the digital multimeter to ensure that amount of voltage being delivered by the power supply to the filters is 15V. To test the filter itself, we used a triple output power supply to power the rails, and examined output in response to various inputs on the oscilloscope (as detailed in section 2).

Once we knew that both of these worked independently, we put the pieces together and repeated the same testing procedure: make sure that the voltage being delivered from the power supply to the rails were indeed +15V and -15V, and test filter response to various input waveforms.

<u>Experimental results</u>

The final system composed of a power supply, two filters, and an extraneous LED volume bar. See below:



The power supply delivered 15V DC to the filter by scaling down and regulating voltage from a wall outlet. The filters used the 15V DC to rail op-amps, which used in conjunction with our particular arrangement of capacitors and resistors, created a bandpass filter. An audio jack connects from an audio source to the filters, and another audio jack connects from the filter output to the lab speakers.

Our simulated bandwidth for treble were [7.9kHz, 10.99kHz], and for bass were [150Hz, 350Hz]. Our measured bandwidths, however, were [7.9kHz, 9.9kHz] for treble and [160.3Hz, 338.8Hz] for bass. This is a 34% difference for treble, and 13% difference for bass.

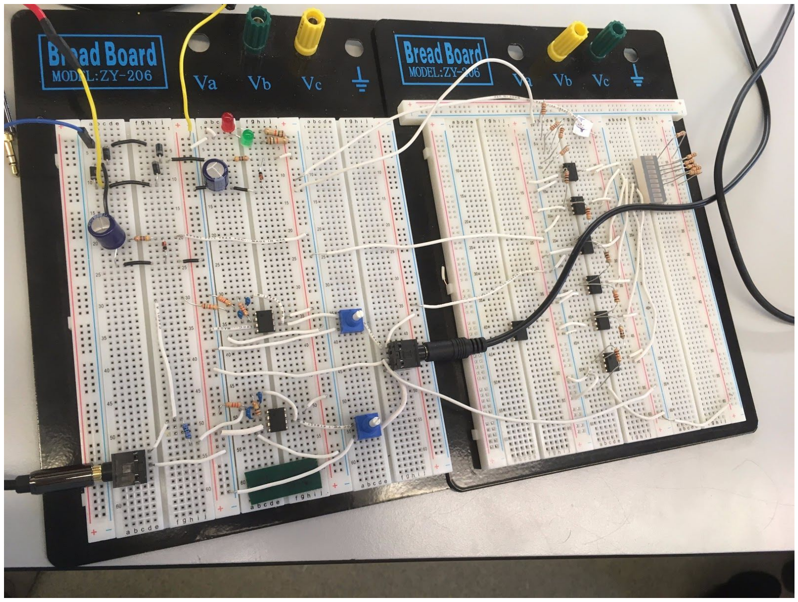


Figure 46: Complete build of final system

<u>Error analysis</u>

The 34% and 13% differences are fairly significant error percentages, and can be attributed to a couple factors. For starters, we built first-order filters, so bandpass range was more sensitive to minor differences in such values as resistance and capacitance. Another reason is that we used an audio jack block to take input and give output, so there is unaccounted-for resistance in that component that may affect the measured bandpass. Lastly, though the percent differences are fairly large, we can point out that for bass, the measured bandpass is actually more selective than the simulated, so this is not too bad of a difference. For the treble, the low cutoff is spot-on, and the high cutoff is only 1kHz off, which is not too bad since treble takes in the higher-frequency audio signals anyway.

<u>Final conclusion:</u>

Our final project worked beautifully. The filters successfully isolated treble and bass frequencies from audio input and gave it a gain for the output. The power supply successfully delivered 15V DC power with minute ripple voltage. Even the extra LED bar successfully showed volume fluctuation.

The biggest difference between measurement and simulation for filters was in our treble filter, at the high cutoff, with a 34% percent difference, as discussed in the error analysis of this section. The biggest difference overall, however, is ripple voltage between our measured and simulated power supply, because CircuitLab did not have the zener diode we wanted. To account for these differences, next time we could build a second-order filter to make the filter more selective, thus narrowing the bandwidth.

From this project, we have learned that the best way to approach electrical engineering projects is to start out with hand calculations and understanding simulations. Then, once we have a clear picture of how to approach the task, start building the project component-by-component. Since some relatively-unpredictable factors will change our output anyway, factor them in while building and make modifications as necessary. Another lesson learned is that when playing with potentially large amp values, always double-check calculations and simulations so that you do not blow too many fuses or capacitors.