Chris Fischer, Prasanna Poudyal

Lab TA: Eileen Huang

Instructor: Thomas Farmer

ESE 215 (Electrical Circuits and Systems)

December 17, 2017

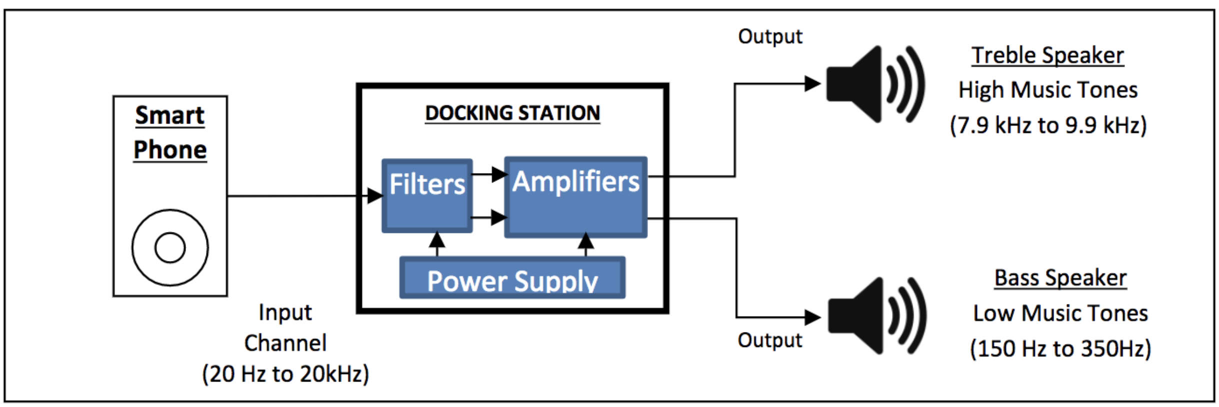
Final Project: Audio Docking Station

Contents

1 Introduction

The goal of this lab is to design, build, and test an inexpensive audio docking station. The docking station will filter and separate coming from a any device with the ability to output and audio signal, such as a smartphone or laptop. The filter aspect of the lab is two fold, take the incoming music from the source and separate its content based on frequency into two channels: high (treble) and low (bass). The output of these filters will then go into an external speaker with a built in amplifier which can play the separated audio assigning each channel to a distinct driver. To make the final result as practical as possible, an AC/DC power supply will also be built to convert the 120 V_{RMS} (AC) power from a standard wall socket to $\sim \pm 15V$ (DC). Figure 1 shows a high level block diagram of the goal of the project.

Figure 1: *High level block diagram of the final product.*

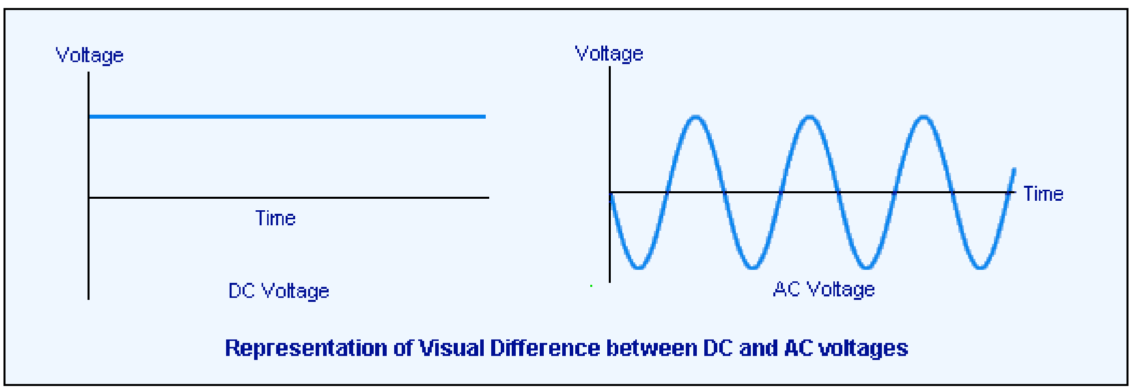


2 Background

There are two main components of this project, the AC/DC power supply and the filter components. Each are critical to the overall function of the final product.

The AC/DC power supply converted the alternating current (AC) from a wall socket to direct current (DC) which can be used by the components of our circuit. The main difference between the two (and the reason we must convert) is that in DC, the electric charge (current) only flows in one direction. Electric charge in AC, on the other hand, changes direction periodically, specifically in a sinusoidal fashion. Because of this behavior, the voltage in AC circuits also periodically reverses because the current changes direction. Figure 2 offers a graph demonstrating this behavior. The

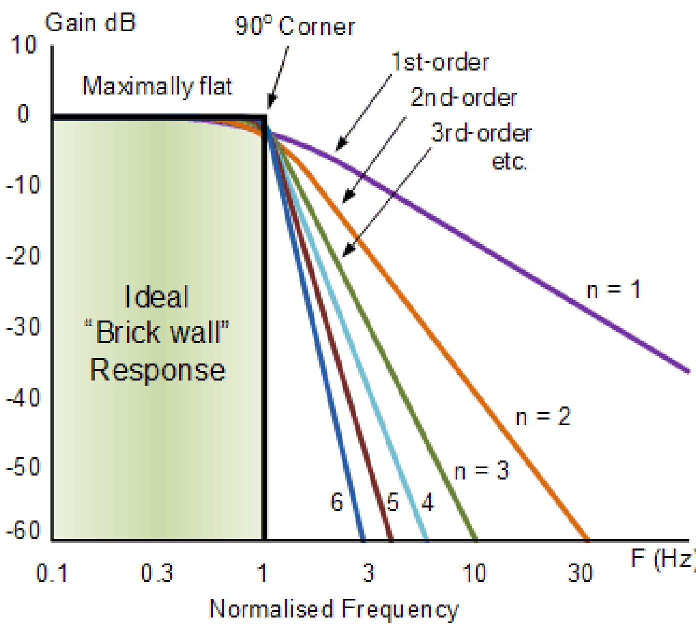
Figure 2: *Alternating current vs direct current.*



components that will make up the filter, which is the second aspect of the project, require DC power, so the converter subsystem must change the sinusoidal voltage to a constant one.

The job of the filter components is to take in a music signal and output a modified signal in which some frequencies have been blocked where others are allowed to pass through. Music typically consists of a wide range of frequencies, and the generally accepted standard range of audible frequencies is 20 to 20,000 Hz. We will be building two separate filters: a bass and a treble. Ideally, the bass filter will block all frequencies except for those in the range 150 to 350 Hz. However, in reality such a hard cut-off line is not achievable. Figure 3 shows this more clearly. Thus, we instead focus on getting the cut-off frequencies (the point where the magnitude decreases

Figure 3: *Generic one-sided filter comparing and ideal model with a realistic first and second order filter.*



past a certain established threshold) to be as accurate as possible, and for the roll-off rate (the rate at which the magnitude decreases from the range of allowed frequencies) to be high. This will give us good isolation of the desired frequencies. The second filter, the treble, will do the exact same thing, just with a different frequency range. The treble should let though anything from 7.9 to 9.9 kHz and block out the rest.

For each part of the project, we will take a systematic approach to design. The first step is hand calculations. We will do all math necessary to ensure that once we build the product in the lab, it behaves as expected. Then, we will simulate our design. This intermediary step helps with testing and debugging, as it offers a simplified view of how the circuit would actually behave if it was built. It allows us to ensure our math is correct before spending time building something that may not work. Lastly, after we have completed these two steps and all results are in agreement with the desired specifications, we can build the circuit in the lab using real parts.

3 Power Supply

The purpose of the power supply aspect of this lab is to output two regulated outputs of approximately +15 V and -15 V DC with a ripple of less than 2% given an input of 120 V_{RMS} AC power supply with a frequency of 60 Hz. This supply then powers up the operational amplifiers (rails) used in the filter component of this lab (discussed later).

3.1 Calculations

3.1.1 Transformer

It is given to us that center-tapped transformer has the following,

$$\frac{N_{Primary}}{N_{Secondary}} = 5.22 = \frac{V_{Primary}}{V_{Secondary}} = \frac{I_{Secondary}}{I_{Primary}}$$

And since we have an AC source of 120 $V_{RMS} \Rightarrow V_{max} = 120 \times \sqrt{2} = 169.706\ V$, we get,

$$V_{Secondary} = \frac{V_{Primary}}{5.22} = \frac{169.706}{5.22} = 32.511V \approx 32.5V$$

And since the center tap is grounded, we expect each terminal to output exactly half of 32.5 V, i.e., 16.25 V

3.1.2 Full wave recifier

Our choice of full wave bridge rectifier is justified by the fact that the positive output terminal of the rectifier has a voltage range of approximately 0V to + 15V and the negative output terminal has the voltage range of approximately 0V to -15V.

While for an ideal diode the forward-bias voltage is 0 V, the forward-bias voltage for the 1N4001 diodes in the lab is approximately 0.75 V. This implies that we need an initial 0.7V across the diode for its resistance to decrease significantly. Furthermore, this forward-bias voltage affects the output voltage as follows:

$$v_{out}(t) = |v_{in}(t)| - 2 \times V_F$$

, where $V_F \approx 0.75V$

The coefficient 2 in the above equation comes from the fact that at any given point in time, exactly 2 diodes are in forward-bias. So now,

$$v_{out}(t) = |32.5| - 2 \times 0.75 = 32.5 - 1.5 = 31V$$

This implies that the positive output terminal now ranges from -0.75V to + 15.5V and the negative output terminal has the voltage range of approximately +0.75V to -15.5V

3.1.3 RC Smoothing Filter

We should now choose smoothing filters such that the voltage across the load never falls less than 98% of the maximum voltage supplied by the rectifier. We know that the upswing time constant, τ_{up}, for the circuit is given by

$$\tau_{up} = (2 \times R_D||R_L)C \approx 2R_DC,$$

if $R_L \gg R_D$ and the downswing time constant, τ_{dn}, for the circuit is given by

$$\tau_{dn} = R_L C$$

And the rectifier's time period is,

$$T_{rectifier} = \frac{1}{120}s = 0.008\bar{3}s$$

To find the downswing time constant, we know,

$$V_c(t) = V_o \times e^{-\frac{t}{\tau_{dn}}}$$

But at time t = 3.65ms after the first peak, the voltage across the capacitor should be at least 98% of V_o, i.e.,

$$V_c(3.65 \times 10^{-3}) = 15.51 \times e^{-\frac{3.65 \times 10^{-3}}{\tau_{dn}}}$$

$$0.98V_o = V_o \times e^{-\frac{3.65 \times 10^{-3}}{\tau_{dn}}}$$

$$\tau_{dn} = \frac{-3.65 \times 10^{-3}}{ln 0.98} = 0.181s$$

Picking C = 800 μF since 800 is available in the lab, we get $R_L \geq 226\Omega$, This makes

$$\tau_{up} = 2R_D C = 2 \times 5 \times 800 \times 10^{-6} = 0.008s$$

Since the time period for the wave itself is 8.3 ms, this value is small enough that the capacitor charges in time for the next cycle.

3.1.4 Voltage regulator

Now even though we have smoothed out the output, we can still do better by regulating the output voltage by using a zener diode.

The expression to obtain peak-to-peak ripple voltage, V_r is given by,

$$V_r = \frac{[(V_{s_1} - 1.5) - V_z]T_{rectifier}}{R_s C} \times \frac{(R_z||R_L)}{R_s + (R_z||R_L)}$$

, where V_z is the voltage zener diodes maintains, $V_{s_1} = 5.22 \times V_{in}$

$$R_z||R_L = \frac{1 \times 1000}{1 + 1000} = 0.999\Omega \approx 1\Omega$$

and,

$$v_{out} = V_z = 15V$$

$$V_r = \frac{[(32.5 - 1.5) - 15]0.008\bar{3}}{50 \times 800 \times 10^{-6}} \times \frac{1}{50 + 1}$$

$$V_r = 0.065V$$

This ripple voltage is well under 2% of the output voltage given by the power supply.

3.1.5 Thevenin Equivalent

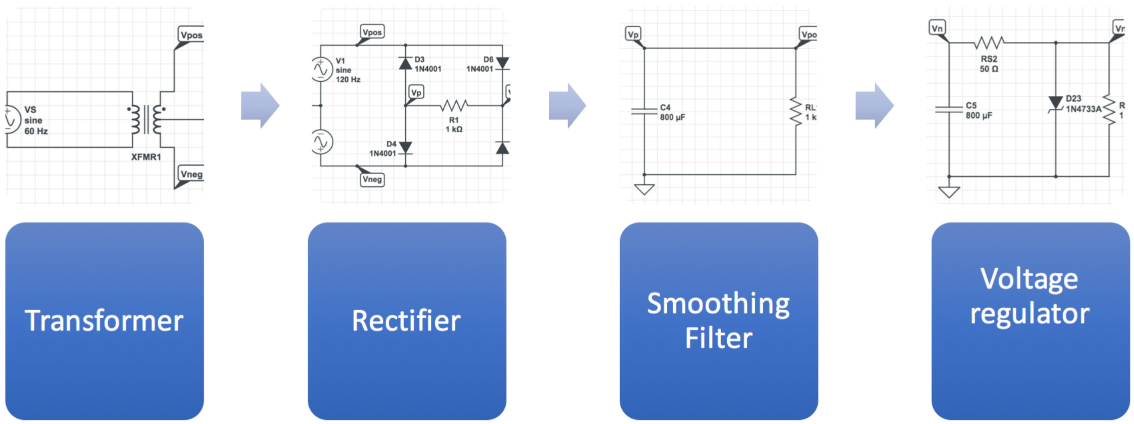
Here, the equivalent resistance of the system is given by,

$$R_{eqiv} = (120\Omega||1k\Omega + 50\Omega) + 5\Omega + 5\Omega = 160.99\Omega$$

Now $V_{Th} = V_{OC} = 15.51V$ so the equivalent circuit is a voltage source with V = 116.25 V and Equivalent resistance of 160.99 $\approx$ 161Ω resistance. Given a load of 1kΩ maximum current the system can supply is $\frac{V_{Th}}{1161} \approx 13.99mA$

3.1.6 Block Diagram

Figure 4: *Block Diagram of the power supply.*



3.2 Power supply simulations

The following section describes the simulations performed for the power source.

3.2.1 Transformer

First we only simulate the transformer to check if the simulations are consistent with hand calculation.

Figure 5: *Transformer schematic with 120 V input at 60 Hz.*

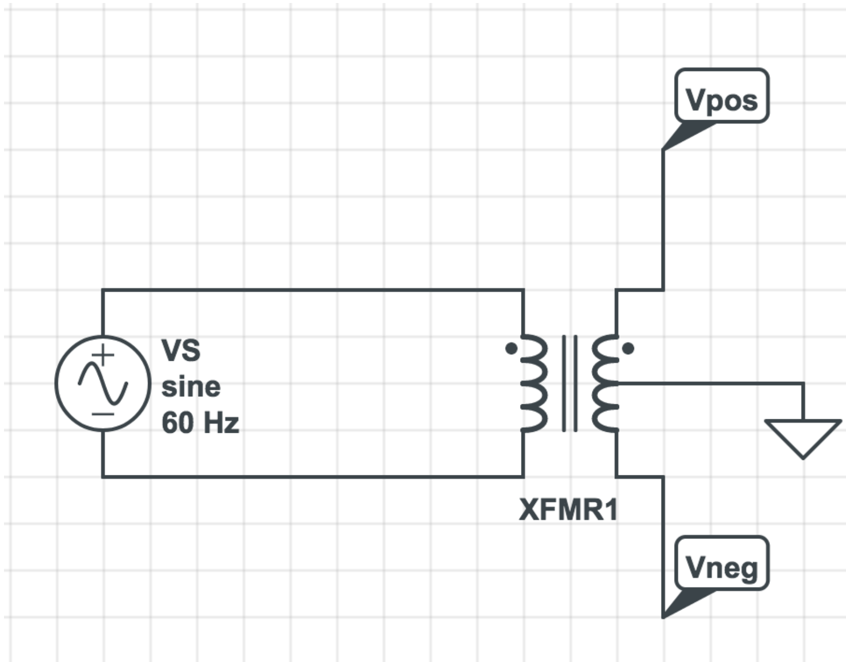
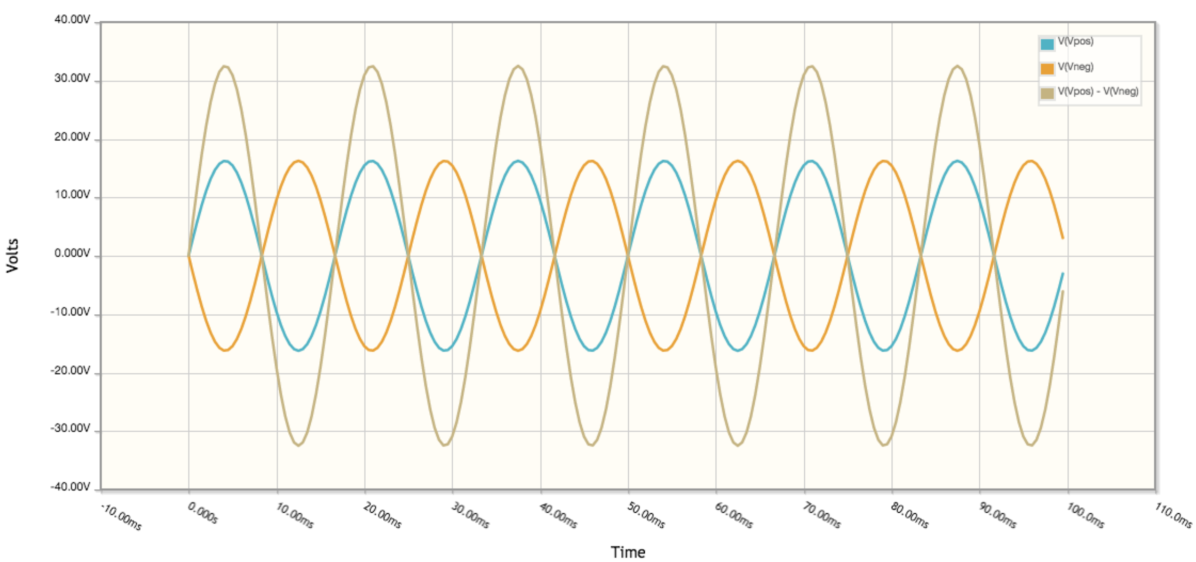


Figure 6: *Graph of Voltages at two ends of the transformer and their differences*



3.2.2 Rectifier

The following schematic shows our rectifier circuit, using 1N4001 diodes outputting a positive and negative lead. We opted to isolate this schematic from the transformer for debugging purposes.

Figure 7: *Rectifier schematic simulated separately*

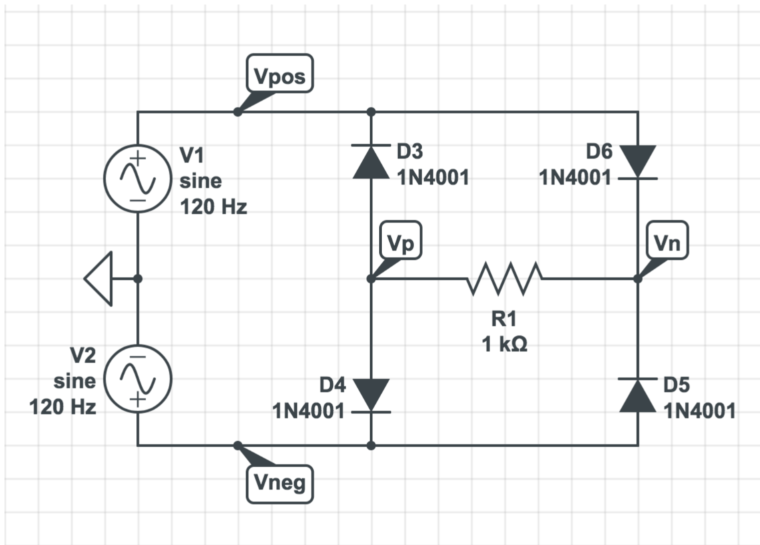


Figure 8: *Graph of V_p and V_{pos}*

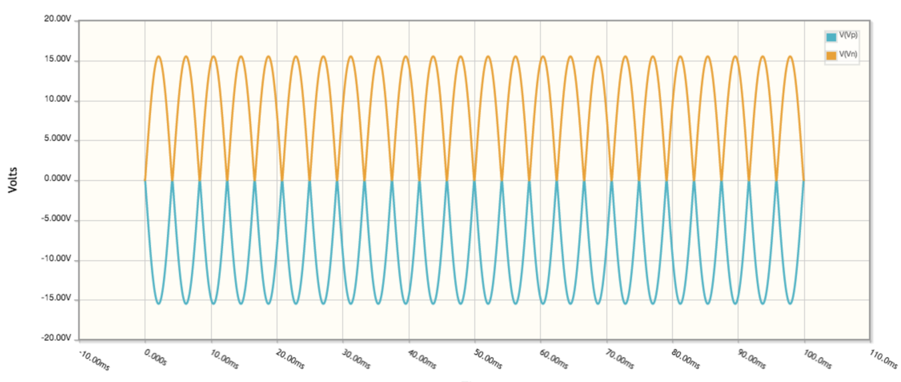
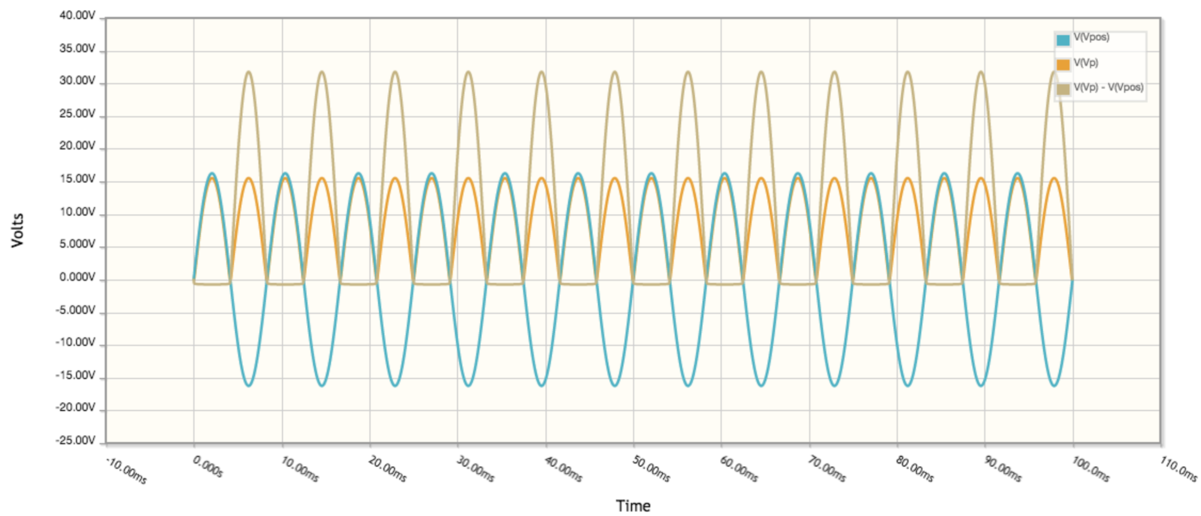


Figure 9: *Graph of* $V_p - V_{pos}$



3.2.3 Smoothing RC Filter

The following schematic shows the transformer and rectifier now combined with the smoothing RC filter described in the above calculations.

Figure 10: *RC smoothing filter schematics*

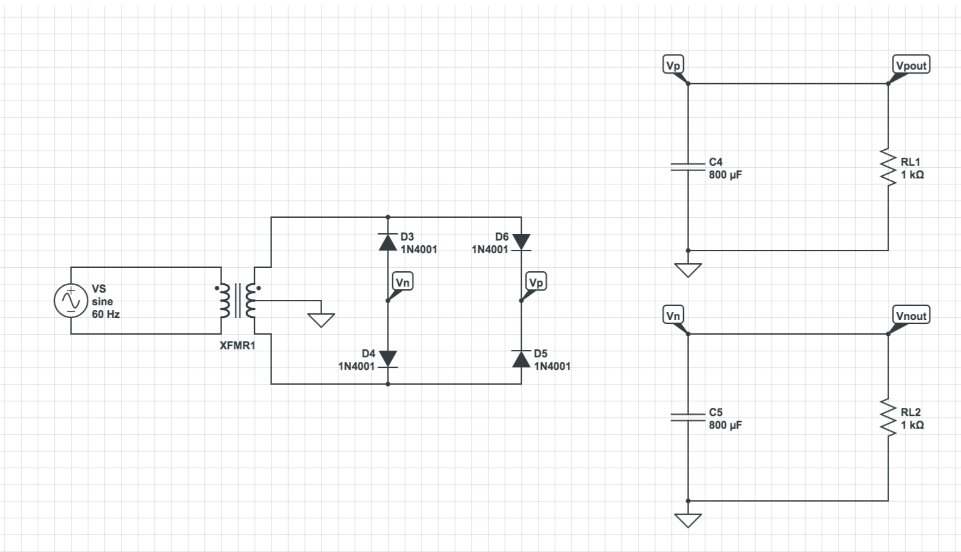
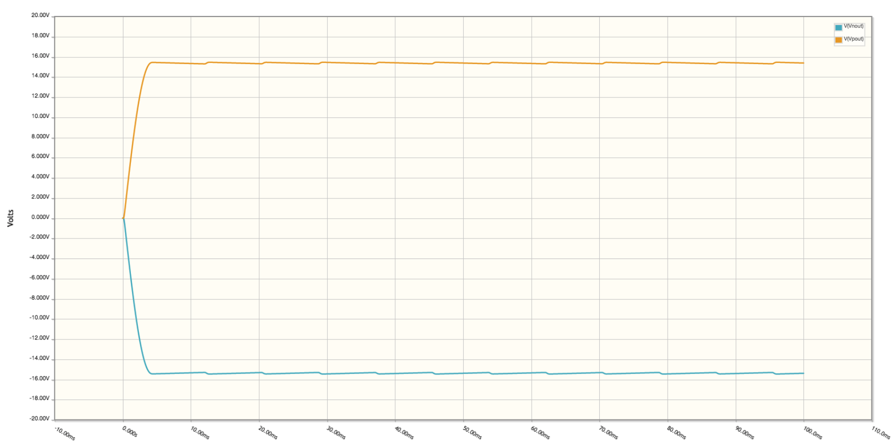


Figure 11: *Voltage vs time graph of the rectifier's positive and negative outputs with smoothing filter*



3.2.4 Voltage regulator

The following schematic shows the previous schematic with an added voltage regulator employing a 15V zener diode and 50 Ohm jumper.

Figure 12: *Voltage regulator schematics*

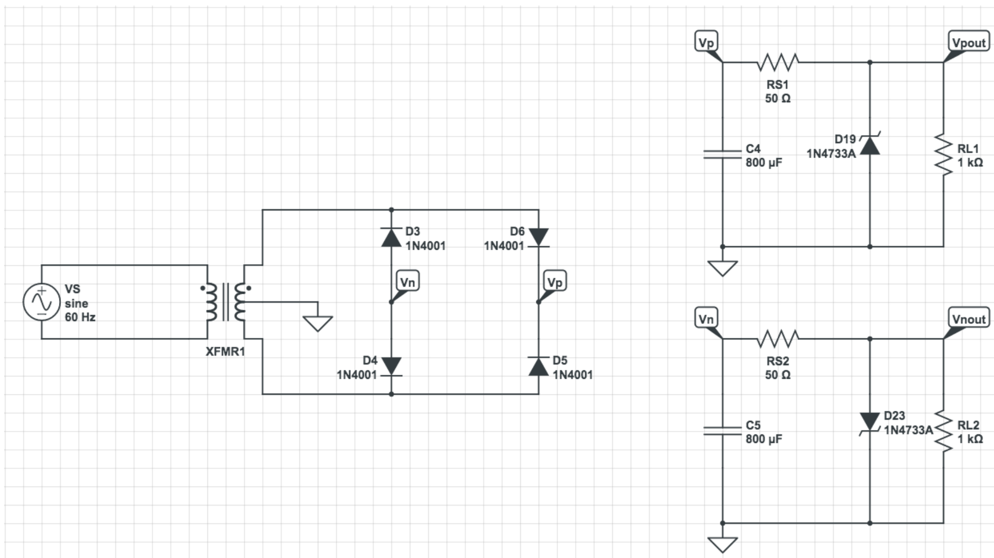
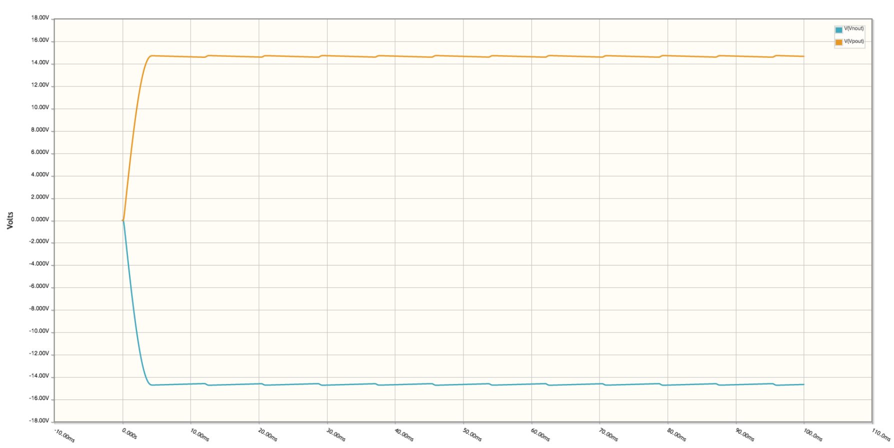


Figure 13: *Graph of the output after using voltage regulator*



3.2.5 Power Supply as a whole system

The following schematic contains the entire power supply system (includes the transformer, rectifier, smoothing filter, and voltage regulator sub components). We have attached a 1 kΩ load.

Figure 14: *Schematics for the entire power supply as a whole with an equivalent load*

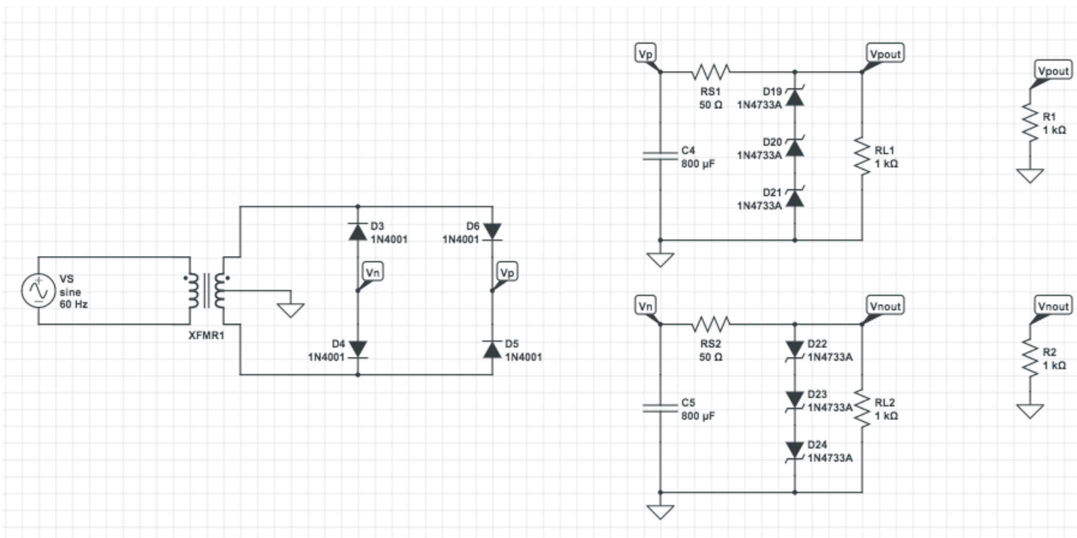


Figure 15: *Graph of the output of the entire power supply as a whole with an equivalent load of 1k Ω*

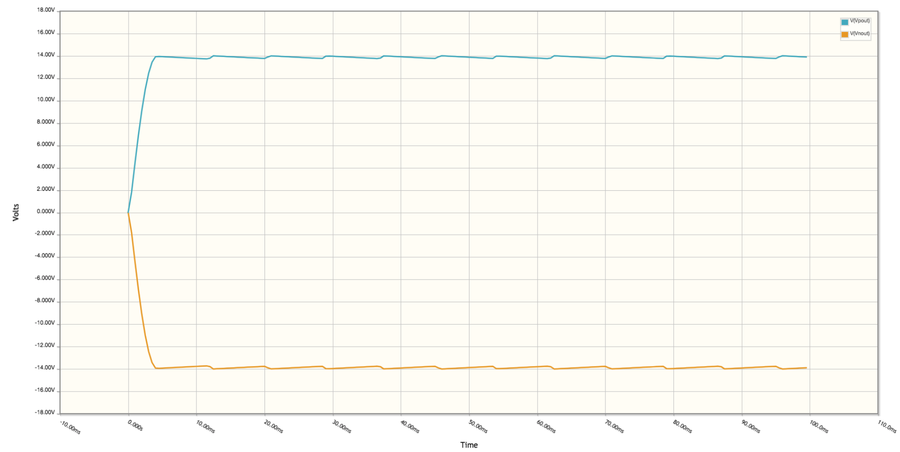


Figure 16: *Graph of the output of the entire power supply as a whole open circuit*

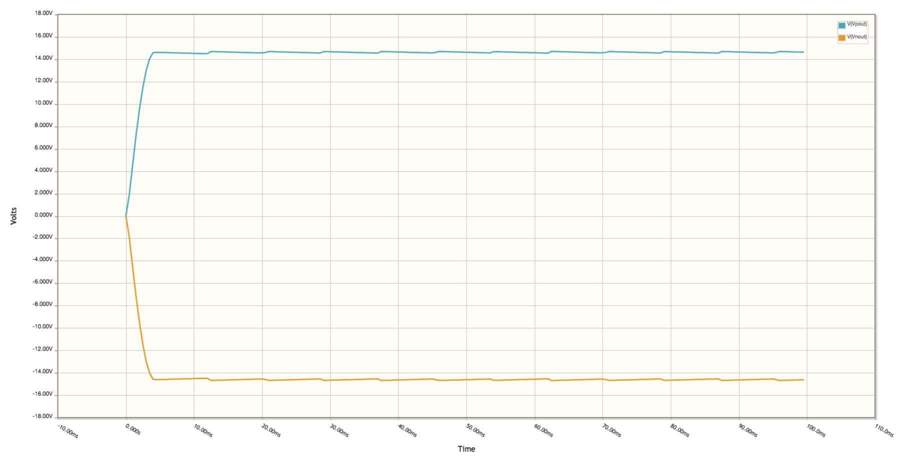
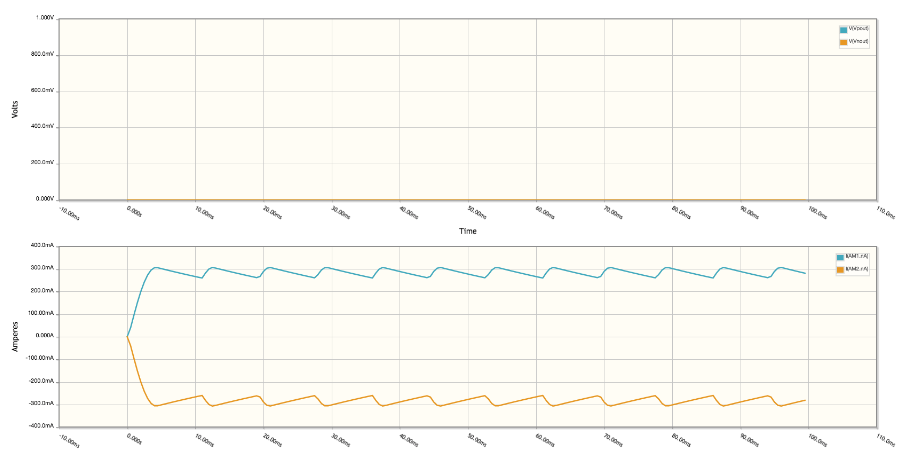


Figure 17: *Graph of the output of the entire power supply as a whole short circuit*



3.3 Experimental

3.3.1 Set-Up

We designed the power supply on a three-column bread board. This allowed us maximum flexibility while debugging and improving the circuits. The equipment used for testing is as follows. A Philips PM6303 RCL Meter was used to measure resistance, capacitance, and parasitic resistance values. To analyze the input and output of the power supply, we used a Agilent Technologies MSO7034B oscillo- scope. An oscilloscope is better suited in this case, as compared to a DMM, for its ability to show periodic waveforms and plot voltage throughout time. This behavior is ideal for our testing, which will involve graphing the input and output of each system over time. When testing, we generally used three channels on the oscilloscope: one for input, one for stage one output, and one for stage two output. The negative end of each cable was connected to a common ground, and the positive end was probed into our bread board.

3.3.2 Building and Testing procedure

We build the power supply as follows. Since all the resistor and capacitor values were already calculated and simulated, all we had left was find a combination of parts which match the simulated values as closely as possible. For resistors, this meant sifting through the resistors, measuring their true resistance on the DMM, and eventually finding a combination of unique resistors that built the target value.

Testing the power supply mainly involves the use of oscilloscope to find whether we met the target at each point of building the circuit. So, first we tested whether the transformer's output was 32.5V and then proceeded with the full wave bridge rectifier. Then again we connect the oscilloscope to the positive and negative output terminals of the rectifier to verify that indeed we got the voltage and the voltage peak that we are supposed to get, i.e., whether we got around +15.51V and -15.51V and whether the shape is that of the absolute value of the sinusoidal graph.

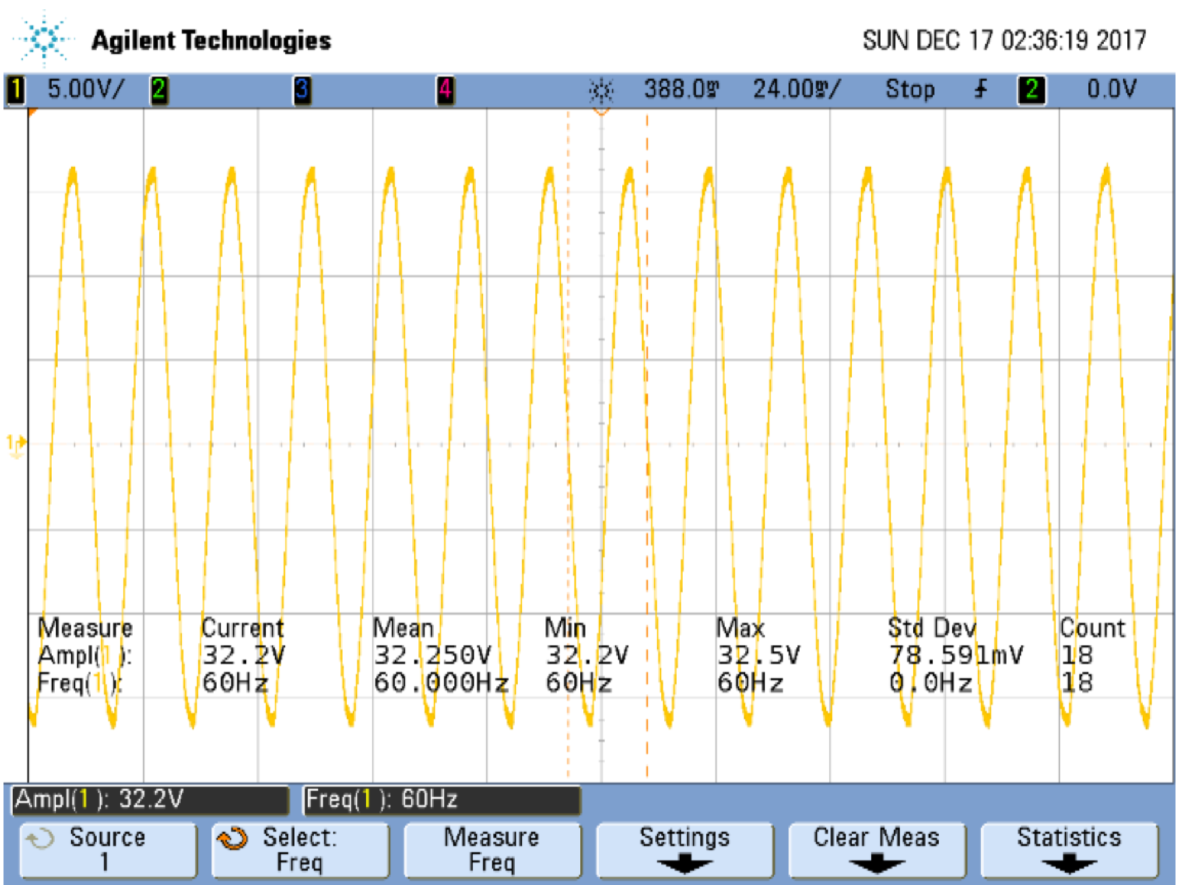
After the rectifiers are verified, we come to the smoothing filter. Here too, we measure the output voltage of the positive and negative terminal and check whether the ripple DC output is within the specified range. After this verification is done, we add the zener diodes and see if the ripple is within the 2% range that was specified.

3.4 Results

3.4.1 Transformer

We found, as shown in the figure, that the transformer output 32.2V at 60Hz. This reveals a % error of 0.9.

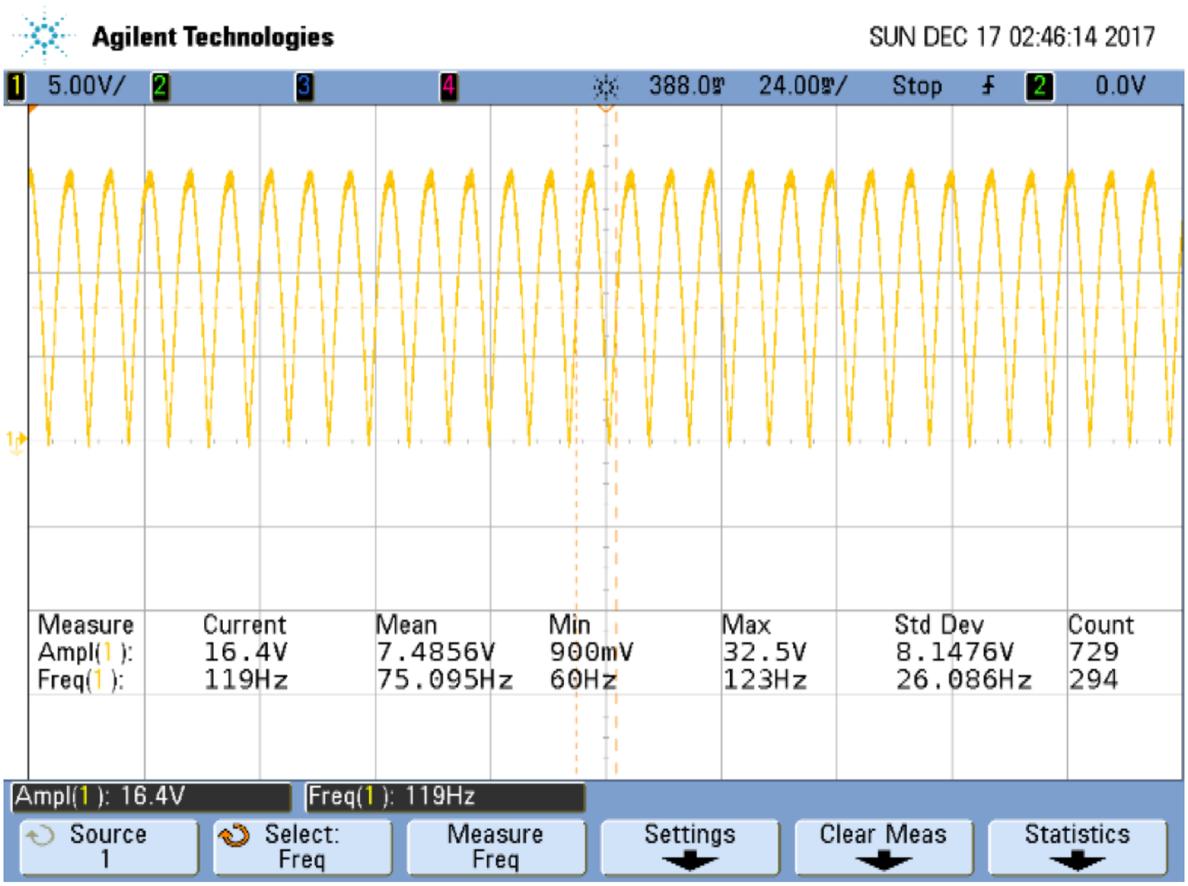
Figure 18: *Oscilloscope output for the transformer*



3.4.2 Rectifier

As shown in the figure, the rectifier outputs 16.4V at 119Hz. And the waveform is also only in the positive half of the plane of the plot as expected.

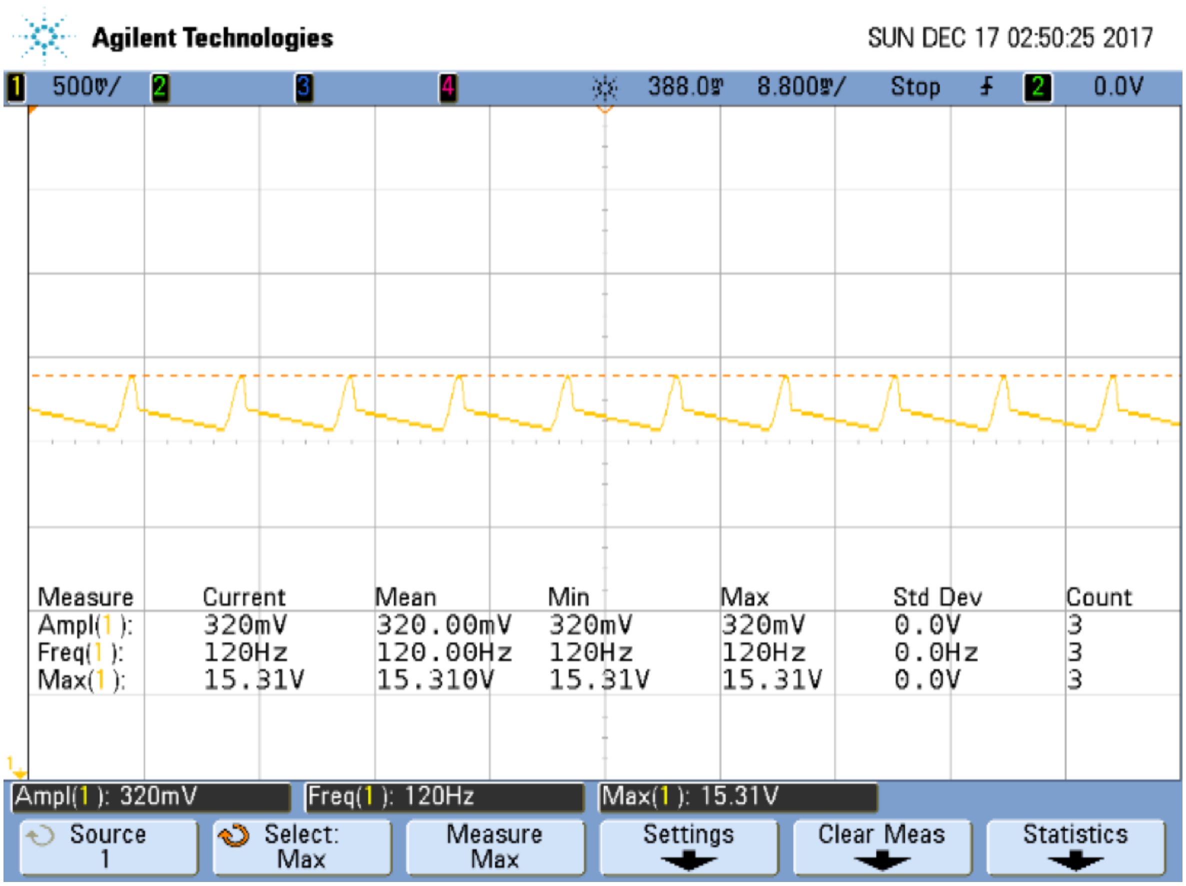
Figure 19: *Oscilloscope output for the rectifier*



3.4.3 Smoothing Filter

The smoothing filter, as shown in the oscilloscope output, has a voltage of 15.31V and the ripple is 320 mV at 120 Hz. This makes up $\frac{0.32}{15.31} = 2.0\%$ of the max-voltage. In simulations, the max was 15.51 V. Thus, the percent error in measurement is $\frac{15.51-15.31}{15.31} = 1.28\%$

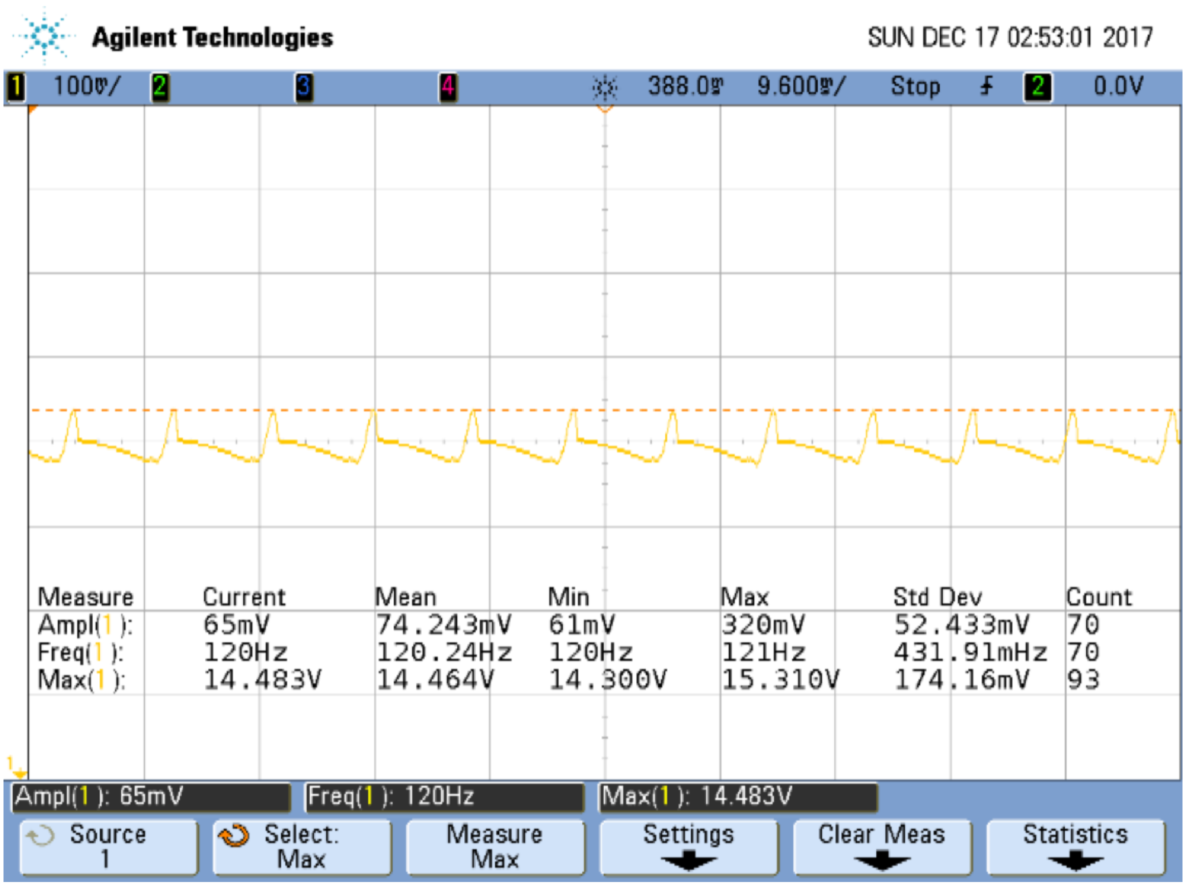
Figure 20: *Oscilloscope output for the smoothing RC filter*



3.4.4 Voltage Regulator equivalent load

The voltage regulator, as shown in the oscilloscope output, has a voltage of 15.31V and the ripple is 0.065 V at 120 Hz = $\frac{0.4}{15.31} \approx 2\%$ ripple voltage.

Figure 21: *Oscilloscope output for the voltage regulator*



3.5 Error Analysis

Overall, the experimental results matched with the hand calculations and simulations. Here is the breakdown of each section:

3.5.1 Transformer

While we expected the output of the transformer to be 32.5V, it turned out to be 32.2 V, a very close match but not an exact one. Calculating the percent error, we get 0.92 %. This can be explained through any possible resistance in the secondary coil of the transformer. To calculate the voltage drop across the coil, the resistance is 0.01 Ω while the oscilloscope we used in series with this small resistance has an internal resistance in the MΩ range. So voltage division would yield less voltage than 32.5V.

3.5.2 Rectifier

While we expected the output to be 16.25V, we got 16.40V. This might be because of the parasitic capacitance of the diodes we used as rectifiers which acted as active devices. The listed parasitic capacitance if the diode is, $2.65 \times 10^{-11}F$. So 2 parasitic capacitors in series yield an overall capacitance of $1.325 \times 10^{-11}F$. This caused the rectifier to output more than 16.25V.

3.5.3 Smoothing Filter

Meanwhile we expected the output to be at 15.51V, experimentally we found it to be at 15.31 V - an error of 1.2%. This error could be best described by the fact that listed capacitance value of the capacitor was not exactly 800 μF. Using Philips PM6303 RCL Meter gave us the value of 785 $\mu F, which suggests a 1.8\%$ margin. Such a difference in capacitance explains the voltage reading we got for the smoothing filter.

3.5.4 Voltage Regulator

While we were expecting 15V, the output was 14.5V. We conjecture here that this could possibly because of the zener diodes heating up too much and chipping away some of the energy of the system.

3.6 Conclusion

Overall the power supply met the specification closely. The calculated/expected values tended to somewhat differ slightly from the measured values, in general, the output were not very far out of the margin of error. While the power supply did deliver around 14.48V but we were expecting around 14.9V, it was due to the resistance values and capacitor values not being precisely what was listed. The rectifier and smoothing filter were highly successful. The voltage regulator was also successful in that the ripple voltage was under 2% but the peak voltage itself was slightly lower. We initially started with 300 Ω as our resistance value for the RC smoothing circuit but quickly realized that this value was unsuitable because it didn't give us the expected peak for V_{out} The aim of this design was to make us apply the principles learned in class and build a custom power supply based on our project's needs. When we tested this system, we were satisfied with the power supply's performance.

4 Filter Components

The purpose of the filter aspect of this lab is to take as input an unamplified audio audio signal (0 to approximately 300 mV_{RMS}) ranging from 20 Hz to 20 kHz (human audio spectrum), and out two channels separated by its frequency components and deliver it to the amplifier component. Specifically, these two channels will be treble and bass, each delivering a very specific range of frequencies. Thus, it is extremely important that we perform the necessary calculations of each filter design's transfer function before simulating or building any circuits. The specifications we are aiming for in this section are as follows:

a) Treble filter

  a) Low cut-off: 7900 Hz, High cut-off: 9900 Hz

  b) -40 dB/decade roll-off rate

  c) 1.5 V/V gain

b) Bass filter

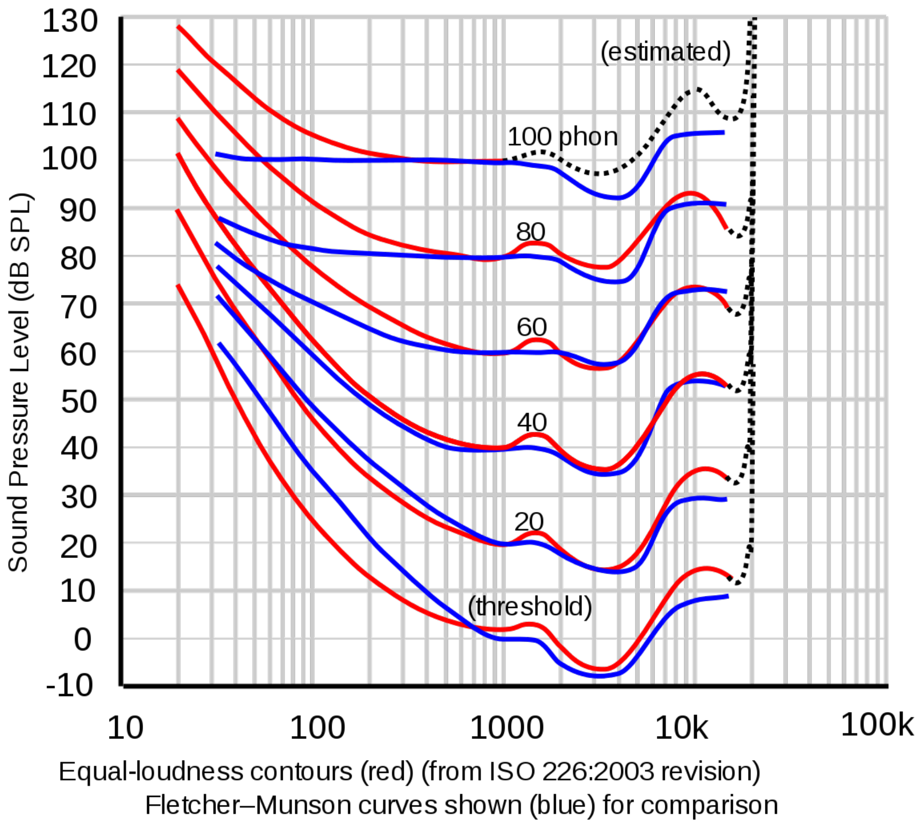
  a) Low cut-off: 150 Hz, High cut-off: 350 Hz

  b) -40 dB/decade roll-off rate

  c) 2.5 V/V gain

The treble will be responsible for letting high frequencies pass while the bass low. Because we are aiming for -40 db/decade roll-off, both filter system will have to be second order. The reason for a higher treble gain, compared to the bass filter, is that high frequencies with the same amplitude as low frequencies are perceived to be quieter, which is an intricacy of the human ear, which is optimized for hearing speech. See Figure 22 for more information. Note that both target gains are greater than one. This is only achievable using active filters, which is what we will employ in Section 4.2.

Figure 22: *Fletcher-Munson curves show that humans perceive frequencies in the range 200-300 Hz to be louder than frequencies in the 10 kHz range. This was the motivation for having different gain levels.*

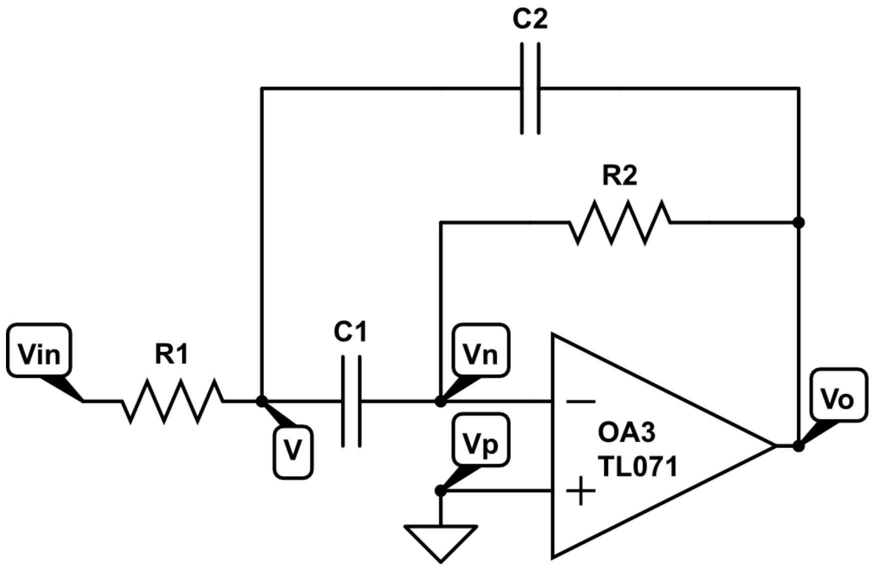


4.1 Calculations

4.1.1 Treble Filter

We now describe the calculations necessary for obtaining the correct specifications for the Treble Filter. We first apply nodal analysis to one of the two multiple feedback, band-pass filters.

Figure 23: *Generic multiple feedback bandpass active filter.*



$$\frac{V - V_{in}}{R_1} + \frac{V - V_n}{\frac{1}{j\omega C_1}} + \frac{V - V_o}{\frac{1}{j\omega C_2}} = 0 \quad (1)$$

$$\frac{V_n - V}{\frac{1}{j\omega C_1}} + \frac{V_n - V_o}{R_2} = 0 \quad (2)$$

$$V_p = V_n = 0 \quad (3)$$

Combining equations (1) and (2) and subbing in 0 for V_n from equation (3) allows use to easily solve for the gain.

$$\frac{\mathbb{V}_{out}}{\mathbb{V}_{in}} = \frac{V_o}{V_{in}} = \frac{-j\omega R_2 C_1}{1 + j\omega(C_1 R_1 + C_2 R_1) + (j\omega)^2 R_1 R_2 C_2 C_2}$$
$$= \frac{-(j\omega)\frac{1}{R_1 C_2}}{(j\omega)^2 + (j\omega)\frac{C_1 + C_2}{C_1 C_2 R_2} + \frac{1}{R_1 R_2 C_1 C_2}}$$

Thus, we have found the transfer function for a single stage multiple feedback, band-pass filter.

$$H_1(\omega) = \frac{-(j\omega)\frac{1}{R_1 C_2}}{(j\omega)^2 + (j\omega)\frac{C_1 + C_2}{C_1 C_2 R_2} + \frac{1}{R_1 R_2 C_1 C_2}}$$

Note that while the transfer function is second order, the roll-off rate will still be -20 dB/decade because this filter is acts as a bandpass. We will now calculate other aspects of the filter: w_o, center frequency; B, bandwidth; w_{c1}, w_{c2} cut-off frequencies; and Q, selectivity. For simplicity's sake, we set $C_1 = C_2 = C$.

$$\omega_o = \frac{1}{C\sqrt{R_1 R_2}}$$
$$f_o = \frac{\omega_o}{2\pi} = \frac{1}{2\pi \cdot C\sqrt{R_1 R_2}}$$
$$Q = \frac{1}{2}\sqrt{\frac{R_2}{R_1}}$$
$$B = \frac{\omega_o}{Q} = \frac{\frac{1}{C\sqrt{R_1 R_2}}}{\frac{1}{2}\sqrt{\frac{R_2}{R_1}}} = 2 \cdot \frac{1}{R_2 C}$$
$$G = -\frac{R_2}{2R_1} = -2Q^2$$

Note that as a consequence of decreasing B, and thus increasing Q, we also increase gain significantly. If we do not take some action to counter this, our op-amps may hit the rails. As we will mention later in more depth, we will precede the filters with an inverting op-amp to scale down the voltage accordingly. Second, as previously noted, the roll-off rate of this filter is -20 dB/decade. Because we want to achieve a -40 db/decade rate, we are going to have to cascade two similar filters. The first of which will have a $B \approx 2000$ Hz, and the second will have $G = 1$. These two in combination with an inverting op-amp will provide us with both the desired cut off frequencies and gain. We can find the total transfer function for the entire cascaded system by multiplying the three together.

$$\begin{aligned} H_{total}(\omega) &= H_1(\omega) \cdot H_2(\omega) \cdot H_3(\omega) \\ &= \frac{-(j\omega)\frac{1}{R_{1_1} C_1}}{(j\omega)^2 + (j\omega)\frac{2}{C_1 R_{1_2}} + \frac{1}{R_{1_1} R_{1_2} C_1^2}} \cdot \frac{-(j\omega)\frac{1}{R_{2_1} C_2}}{(j\omega)^2 + (j\omega)\frac{2}{C_2 R_{2_2}} + \frac{1}{R_{2_1} R_{2_2} C_2^2}} \cdot \frac{-R_f}{R_s} \end{aligned}$$

We can now choose values for the resistors and capacitors. At the advice of the lab TAs, we decided to make our bandwidth slightly narrower than our specifications suggested. The reasoning behind this is as follows: when the desired cut off frequencies are fairly close together, their imperfect roll-offs (when compared to the Bode Straight Line Approximations) can superimpose, lowering the true peak, and this moving the cut-off frequencies farther away from ω_o.

For the first of the two cascaded filters, we settled on $B = 1750$. Thus, an ideal gain of $G = 50$. Because we want a high input resistance for max voltage transfer, we set $R_1 = 9.80$ kΩ. Then, $R_2 = 980.0$ kΩ and $C = 182.5$ pF. Using these values, which we were able to find or combine to create in the lab, give a $w_o = 8.9$ kHz and $B = 1748$ Hz, which is spot on. This should give us cut-off frequencies of 8.026 kHz and 9.774 kHz.

For the second of the two cascaded filters, $G = 1$. Thus, $B = 12580$ rad/s. While this is obviously much greater than the desired cut-off points, we already have achieved those with the first of the cascaded filters. This filter serves only the increase the roll-off rate for frequencies not in the pass band. Because we want a high input resistance for max voltage transfer, we set $R_1 = 9.86$ kΩ. Then, $R_2 = 19.72$ kΩ and $C = 1.282$ nF. Using these values, which we were able to find or combine to create in the lab, give a $w_o = 8.9$kHz and $B = 12586$, which is spot on.

As mentioned, we preface these filters with an inverting op-amp. This is because the total gain of the two filters is $50 \cdot 1 = 50$. The max voltage from a smartphone's audio jack is around 500 mV, which means the resultant voltage would be $0.5 \cdot 50 = 25V$ which is significantly above the $\pm V_{cc}$. The target gain for the treble filter system was 2.0. We need to scale our voltage down by a factor of ~ 15, as there will be some loss of gain when the circuit is built for real. Like always, we want the input resistance to be high for maximum voltage transfer, so we chose a 15 kΩ resistor as

R_s and a 1.0 kΩ as R_f, giving us a local gain of $G = -\frac{1}{15}$.

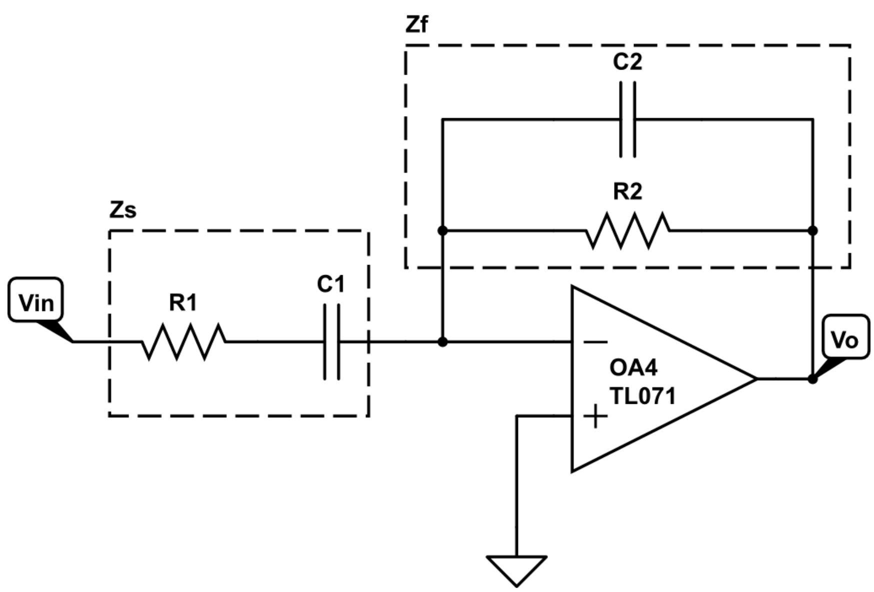
This makes the total gain of the entire filter -3.33 V/V.

The expected input resistance of the treble filter system is just R_s of the inverting op-amp, which is 20 kΩ. The output resistance is approximately equal to the output resistance of the last op-amp, which is 100 Ω. To find the expected DC power requirements, we consider the worst case. For each op-amp, this is 50 mW, according to the TL-071 data sheet. Thus, the total power requirement is 150 mW for both the positive and negative sources. Voltage requirement is $\pm 15V$. Current requirement is thus 10.0 mA.

4.1.2 Bass Filter

We now describe the calculations necessary for obtaining the correct specifications for the Bass Filter. The following is transfer function for the first half of the bass filter. Because the configuration

Figure 24: *Generic bandpass active filter.*



is an inverting-op-amp, we can use the gain equation previously derived.

$$\begin{aligned}
H_1(\omega) = \frac{\mathbb{V}_{out}}{\mathbb{V}_{in}} = -\frac{Z_f}{Z_s} &= -\frac{Z_C||Z_{R_2}}{Z_{R1} + Z_{C1}} = -\frac{\frac{R_2}{1+j\omega R_2 C_2}}{\frac{1+j\omega R_1 C_1}{j\omega C_1}} \\
&= -\frac{j\omega R_2 C_1}{(1 + j\omega R_1 C_1)(1 + j\omega R_2 C_2)} \\
&= -R_2 \frac{j\omega C_1}{(1 + j\omega R_1 C_1)(1 + j\omega R_2 C_2)} \\
&= -\frac{R_2}{R_1} \cdot \frac{j\omega/\omega_{c1}}{(1 + j\omega/\omega_{c1})(1 + j\omega/\omega_{c2})}
\end{aligned}$$

It is clear now that the transfer function consists of one zero and two poles. The zero is at the origin, the first pole is at $\omega_{c1} = 1/(R_1 C_1)$, and the second pole is at $\omega_{c2} = 1/(R_2 C_2)$. We also see that the gain $G = -R_2/R_1$.

Although this filter is in fact of second order, as the degree of the denominator of the transfer function is 2, to achieve the goal of a -40 db/decade roll-off rate we still must build a second, identical filter such that $H_2(\omega) = H_1(\omega)$. Using the same R_1, R_2, C_1, and C_2 values through both filters would give it the same gain, cut-off, and center frequencies. The resulting transfer function would simply be the product of the two. We now see that the order of each pole is 2, which indicates that the roll-off rate of each poll is -40 dB/decade.

$$\begin{aligned}
H_{total}(\omega) &= H_1(\omega) \cdot H_2(\omega) \\
&= \left(-\frac{R_2}{R_1} \cdot \frac{j\omega/\omega_{c1}}{(1 + j\omega/\omega_{c1})(1 + j\omega/\omega_{c2})}\right)\left(-\frac{R_2}{R_1} \cdot \frac{j\omega/\omega_{c1}}{(1 + j\omega/\omega_{c1})(1 + j\omega/\omega_{c2})}\right) \\
&= \left(\frac{R_2}{R_1}\right)^2 \cdot \frac{(j\omega/\omega_{c1})^2}{(1 + j\omega/\omega_{c1})^2(1 + j\omega/\omega_{c2})^2}
\end{aligned}$$

We can now choose values for the resistors and capacitors. First, we chose gain of each individual filter to be equal to 1.8 because we knew that value would be only the ideal case, and furthermore because our ω_c values, the cutoff frequencies, were fairly close together, there would be some interference between the two roll-offs. This would result from the roll-offs not being sharp, as we assume for ideal Bode straight line approximation plots. We also want to set the input impedance high enough to get maximum voltage transfer from the audio source, whether that is the function generator or a cell phone or laptop. We know the function generator output resistance to be 50 Ω so we chose $R_1 = 1.0986$ kΩ because it was convenient from the resistors available in the lab. Thus, $R_2 = 1.0986 \cdot 1.8 = 1.989 \approx 1.978$ kΩ, again using the values available in lab. To calculate C_1 and C_2 we must first convert the cut-off frequency specifications to rad/s.

$$150 \text{ Hz} = 150 \cdot 2\pi \text{ rad/s} = 942.5 \text{ rad/s} = \omega_{c1}$$

$$350 \text{ Hz} = 350 \cdot 2\pi \text{ rad/s} = 2199.1 \text{ rad/s} = \omega_{c2}$$

The center frequency, ω_o, is 1570.8 rad/s, or 250.0 Hz. Bandwidth is 1256.6 rad/s. Q is 1.25. We now calculate C_1 and C_2.

$$C_1 = 1/(R_1 \cdot \omega_{c1}) = 1/(1098.6 \cdot 945.5) = 962 \text{ nF}$$

$$C_2 = 1/(R_2 \cdot \omega_{c2}) = 1/(1978 \cdot 2199.1) = 230 \text{ nF}$$

We chose 965 nF and 229 nF capacitors respectively, as they were available in the lab. To reiterate, the input resistance of the bass filter is just R_1, which equals 1.0986 kΩ. The output resistance is approximately equal to just the output resistance of the last op-amp, which is 100 Ω. To find the expected DC power requirements, we consider the worst case. For each op-amp, this is 50 mW, according to the TL-071 data sheet. Thus, the total power requirement is 100 mW for both the positive and negative sources. Voltage requirement is $\pm 15V$. Current requirement is thus 6.66 mA.

4.2 Simulations

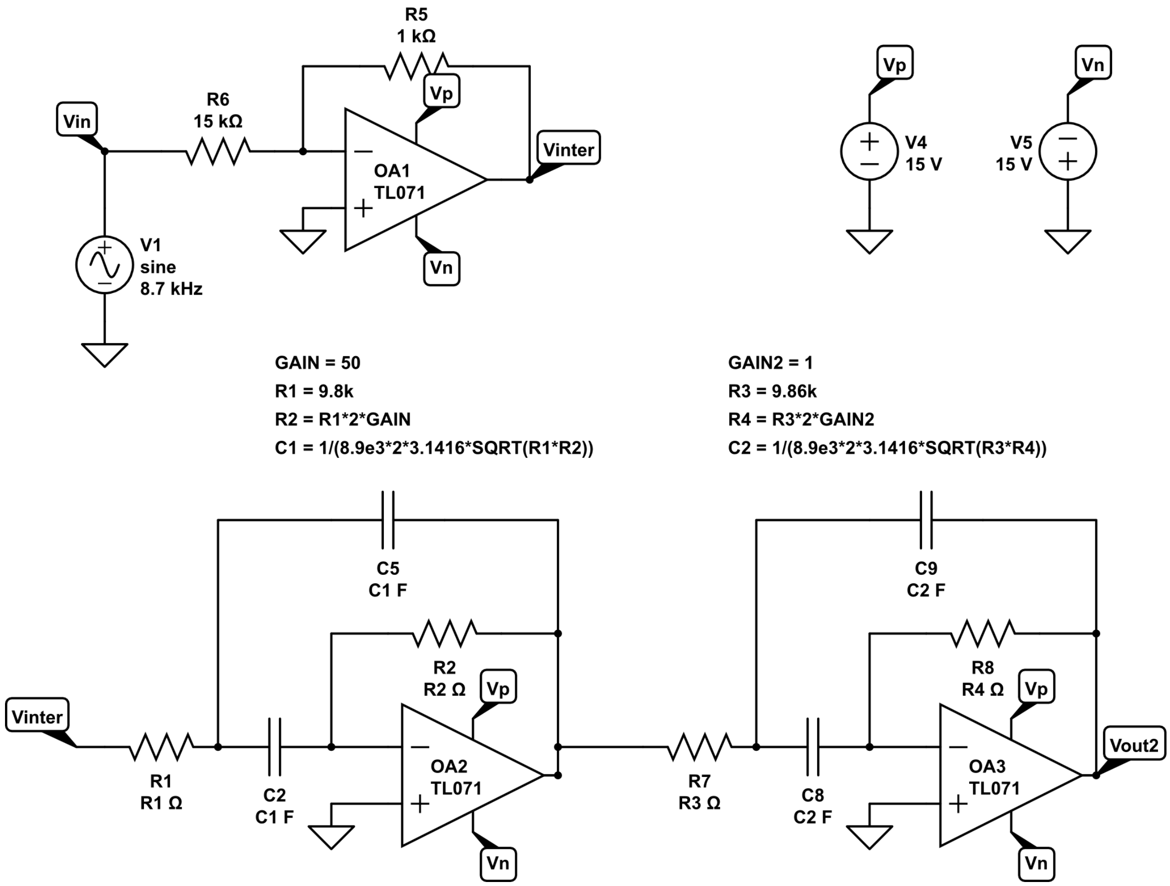
The following section outlines the simulations performed for the lab. We made several design choices when creating the schematics. First was the choice to use TL071 op-amp in the filters for their low noise. This would mean better sound quality when we later built and tested the filter systems with a music input. In the simulations, we powered the op-amps with a DC ± 15 V; however, when we would build the circuit in the lab this power supply was exchanged for the AC/DC converted described in the previous section. We also made a conscious choice to keep our capacitor values low to avoid the having to use the physically larger polarized capacitors.

4.2.1 Treble Filter

Figure 25 shows the schematic representation of our treble filter system. V_{in} represents the audio input, whether that is a simple sine wave or complex signal covering many frequencies. V_{out2} is the output node of our filter. The transfer function $H_{total}(\omega)$ described above is thus $\frac{V_{in}}{V_{out2}}$. V_p and V_n are the positive and negative rail voltages respectively. For the treble filters, because there were more parameters and a more complex transfer function, we decided to use CircuitLab's parameter feature, which allows one to avoid hardcoding values into the resistors and capacitors. This allows us to quickly change gain and center frequency as needed.

Figure 26 shows the semi-log and Bode diagrams for the treble system. There are a few things to note. Because we have an odd number of op-amps with negative gain, the output is 180 degrees out of phase with respect to the input. This can also be seen in Figure 27. The maximum gain of the system, as seen in the semi-log, is approximately 3.25 V/V in the ideal case. On the Bode magnitude plot, we can see that the maximum is equal to $20 \log 3.25 = 10.23dB$. This is one of the

Figure 25: *Treble filter schematic with op-amp rails.*



advantages of using active filters, as if passive filters were employed the maximum gain achievable is 0 dB.

Using the Mode magnitude plot, we can calculate the roll-off rate. At 1 kHz, the gain is -38.0 dB. At 100 Hz, exactly one decade below, the gain is -78.1 dB. The roll-off is then $(-38.0-78.1)/1 = 40.1 \approx 40$ dB/decade. We see that on the bode plot that the roll of rate on the high cut-off side is just the negation, so -40 dB/decade. This achieves our specification roll-off goal.

Figure 27 shows time domain plots of output and input of the treble system at its center and cut-off frequencies. These are 8.9 kHz, 7.9 kHz, and 10.4 kHz respectively. We see the phase shift more prominently here, as the input and output are nearly perfectly out of phase. This should not affect anything during real-world use, as music that is inverted will produce identical sound waves as a non-inverted output when played through a speaker. We also see the same gain as on the semi-log plot: approximately 3.25 V/V. At the cut-off frequencies, the gain is by definition $1/\sqrt{2}$ of the center, which is 2.3 V/V.

Lastly, Figure 28 shows the DC current draw at the filters center frequency, 8.9 kHz. This is where the op-amps would be drawing the most current as it is where the gain is highest. The values are significantly lower than we had calculated, which makes sense as the calculations were worst case.

Figure 26: *Treble filter plots showing V_{out2}/V_{in}. Top is semi-log, middle is Bode magnitude, bottom is Bode phase.*

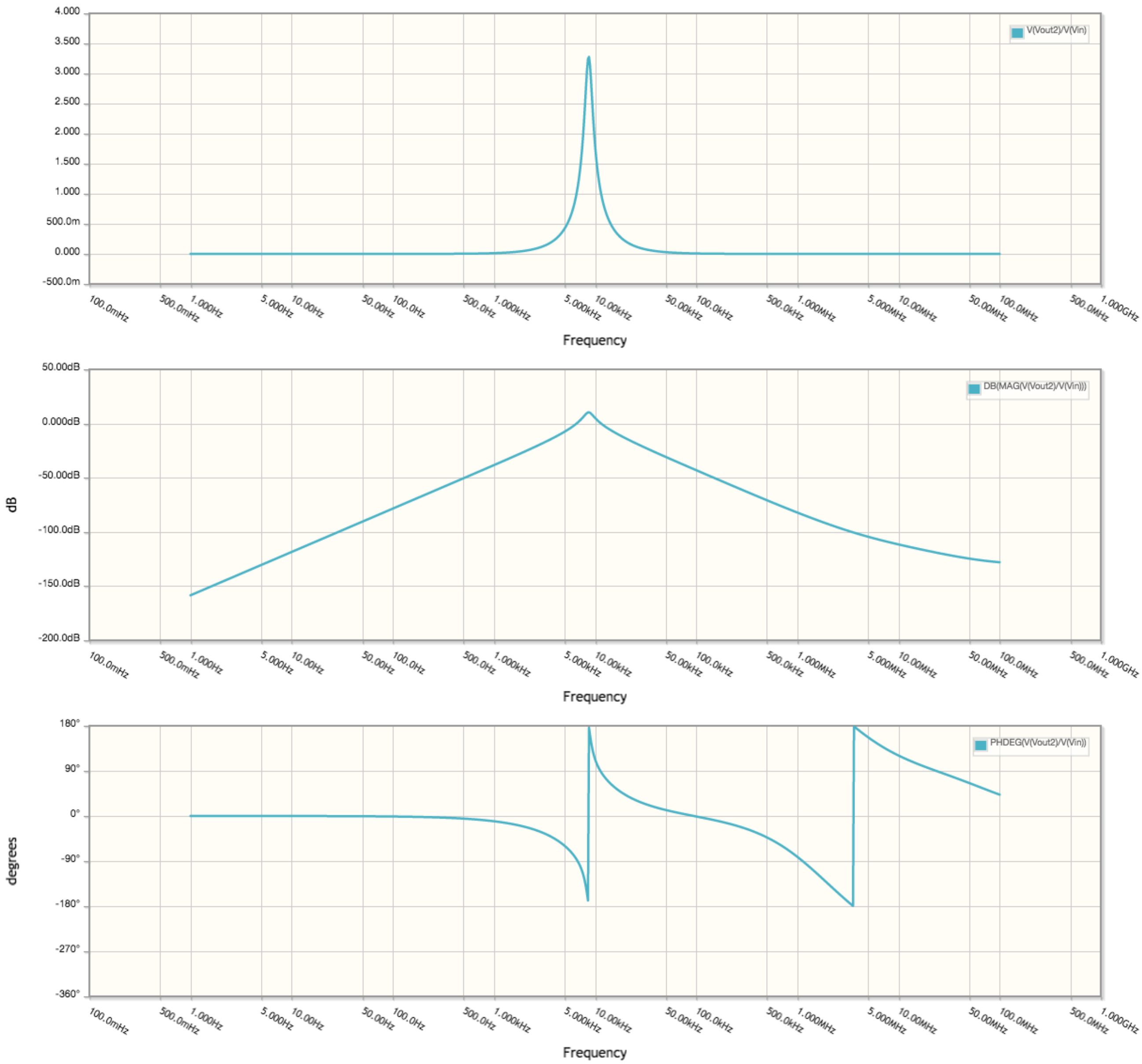
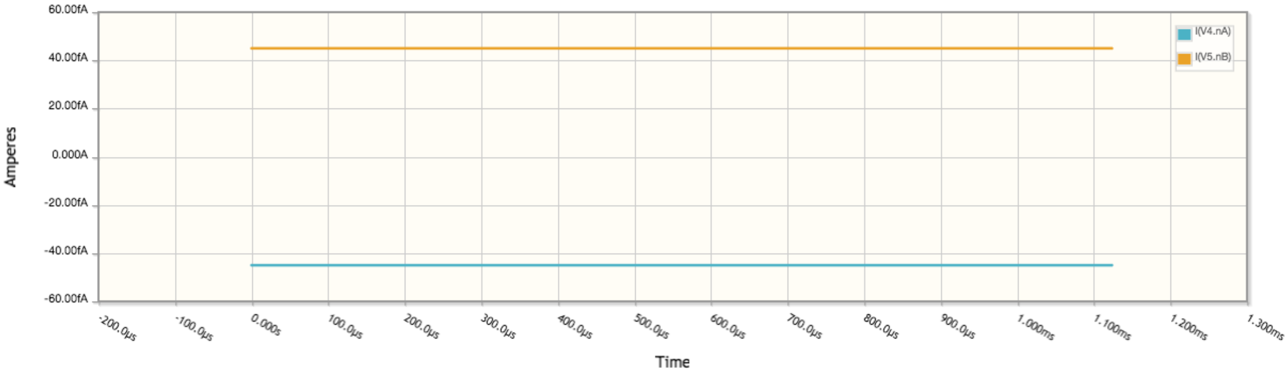


Figure 27: *Treble filter time domain plots showing V_{in} and V_{out2}. Top is low cut-off frequency (7.9 kHz), middle is center frequency (8.9 kHz), bottom is high cut-off frequency (10.4 kHz).*

Figure 28: *Treble filter time domain plots showing DC current draw at center frequency on the positive and negative power supply.*

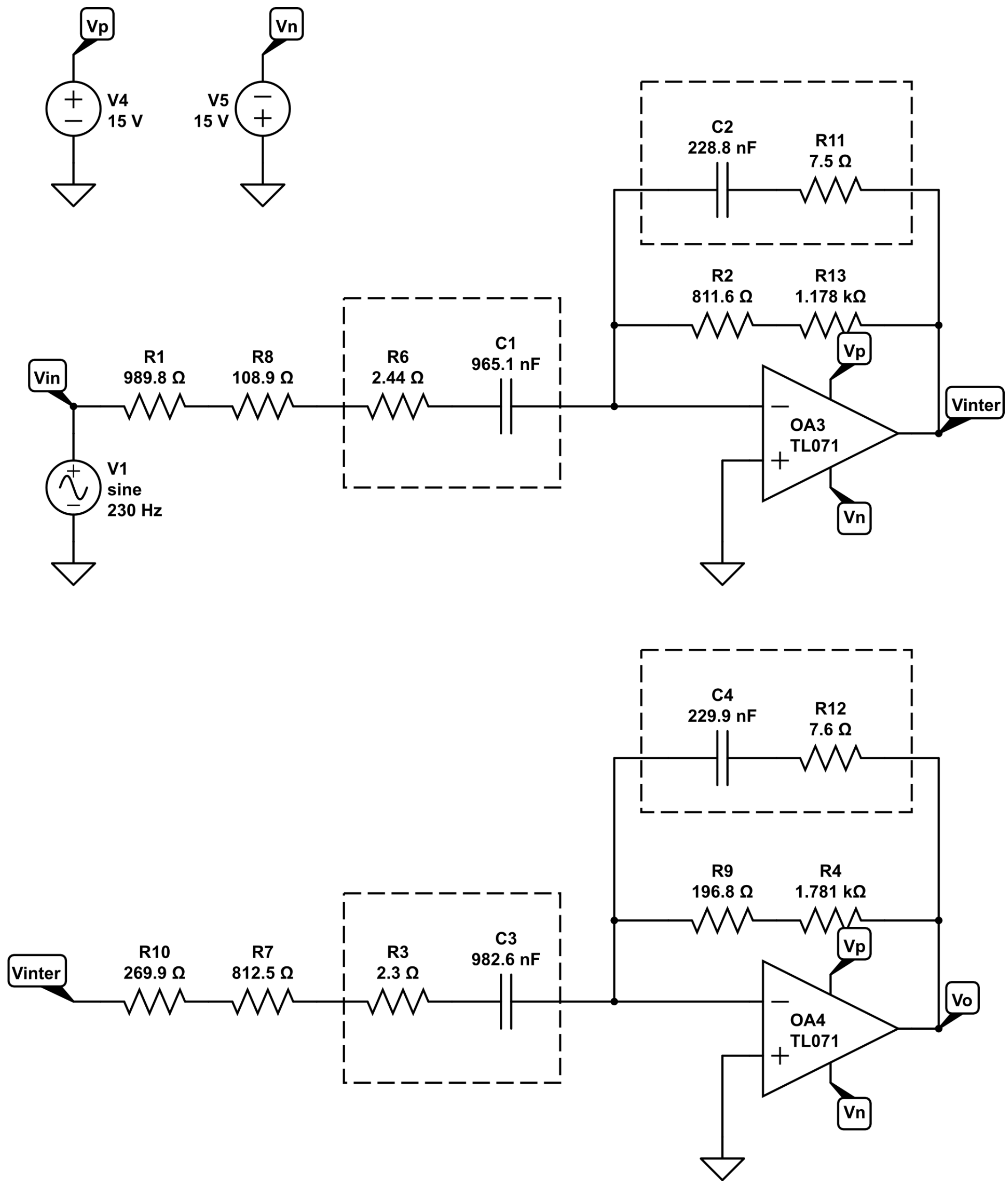


4.2.2 Bass Filter

Figure 29 shows the schematic representation of our bass filter system. V_{in} represents the audio input, whether that is a simple sine wave or complex signal covering many frequencies. V_o is the output node of our filter. The transfer function $H_{total}(\omega)$ described above is thus $\frac{V_{in}}{V_o}$. V_p and V_n are the positive and negative rail voltages respectively. For the bass filters, because we had discussed this type of active inverting bandpass filter design in lecture extensively, we decided to hardcode our actual measured values into our simulation after determining what was available in the lab. This includes the parasitic resistances of the capacitors, which can be seen in the dotted boxes in the schematic.

Figure 30 shows the semi-log and Bode diagrams for the bass system. There are a few things to note. Compared to the treble system, there is a smaller phase shift across all frequencies because we are cascading an even number of inverting op-amps. The maximum gain of the system, as seen in the semi-log, is approximately 1.65 V/V in the ideal case. On the Bode magnitude plot, we can

Figure 29: *Bass filter schematic with op-amp rails.*



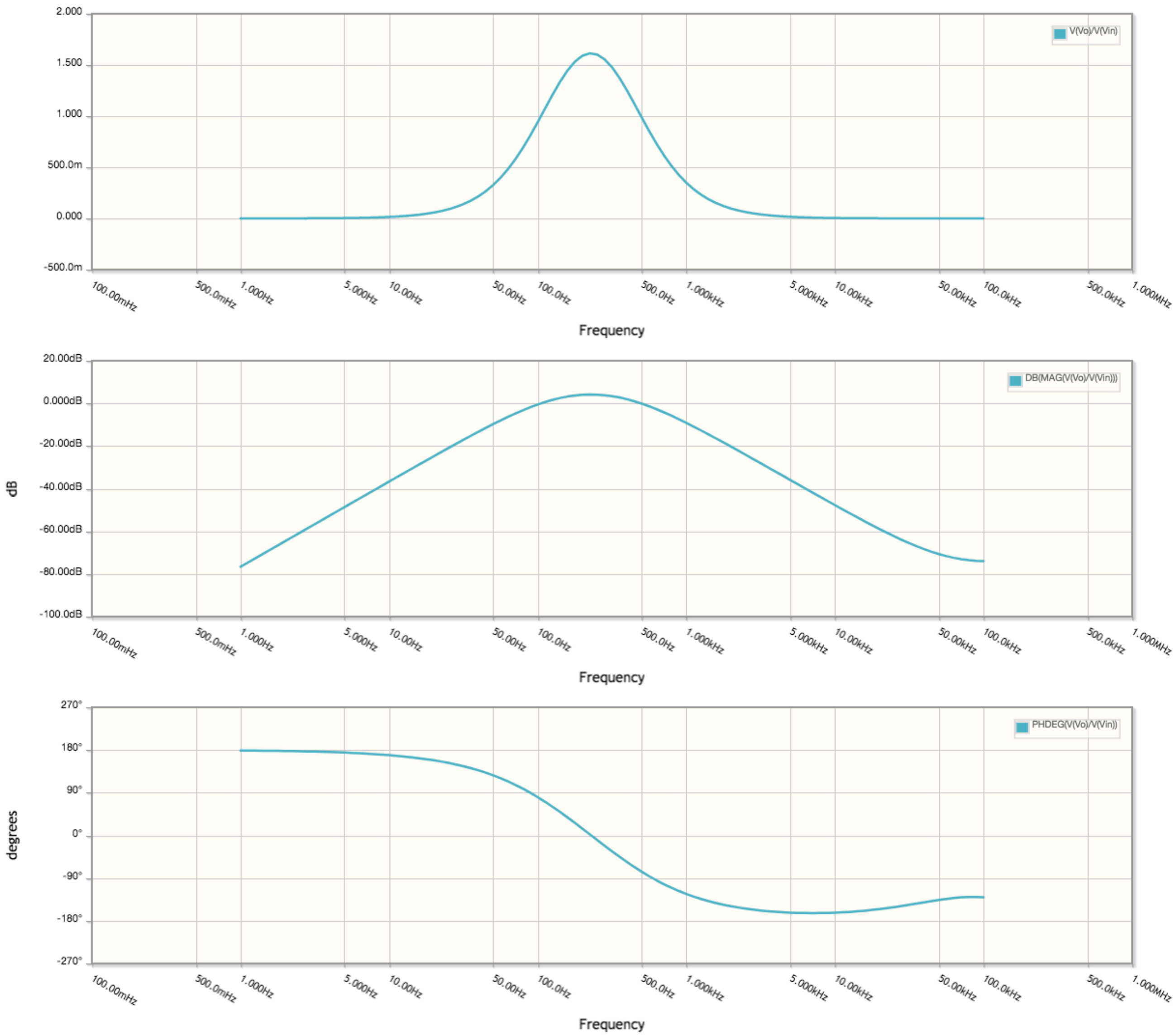
see that the maximum is equal to $20 \log 1.62 = 4.350dB$. Again, such a gain is only achievable using active filters. Using the Mode magnitude plot, we can calculate the roll-off rate.

At 50 Hz, the gain is -9.7 dB. At 5 Hz, exactly one decade below, the gain is -48.6 dB. The roll-off is then $(-9.7 - 48.6)/1 = 38.9 \approx 40$ dB/decade. We can see that the slope on the high cut off side is just the negation, -40 dB/decade. This achieves our specification roll-off goal.

Figure 31 shows time domain plots of output and input of the bass system at its center and cut-off frequencies. These are 230 Hz, 130 Hz, and 420 Hz respectively. Note that the input and output are perfectly in phase at the center, as we have an even number of inverting op-amps. At the low cut-off, the output is leading with respect to the input, and at the high cut-off, the output is lagging with respect to the input. We also see the same gain as on the semi-log plot: approximately 1.65 V/V. At the cut-off frequencies, the gain is by definition $1/\sqrt{2}$ of the center, which is 1.167 V/V.

Lastly, Figure 32 shows the DC current draw at the filters center frequency, 230 Hz. This is where the op-amps would be drawing the most current as it is where the gain is highest. The values are significantly lower than we had calculated, which makes sense as the calculations were worst case. Note that the draw is in fact less than that of the treble system, as we are using one fewer op-amp.

Figure 30: *Base filter plots showing V_o/V_{in}. Top is semi-log, middle is Bode magnitude, bottom is Bode phase.*



4.3 Experimental

The following section outlines the steps take to design, build, test the aforementioned filters in the lab.

Figure 31: *Bass filter time domain plots showing V_{in} and V_o. Top is low cut-off frequency (130 Hz), middle is center frequency (250 Hz), bottom is high cut-off frequency (420 Hz).*

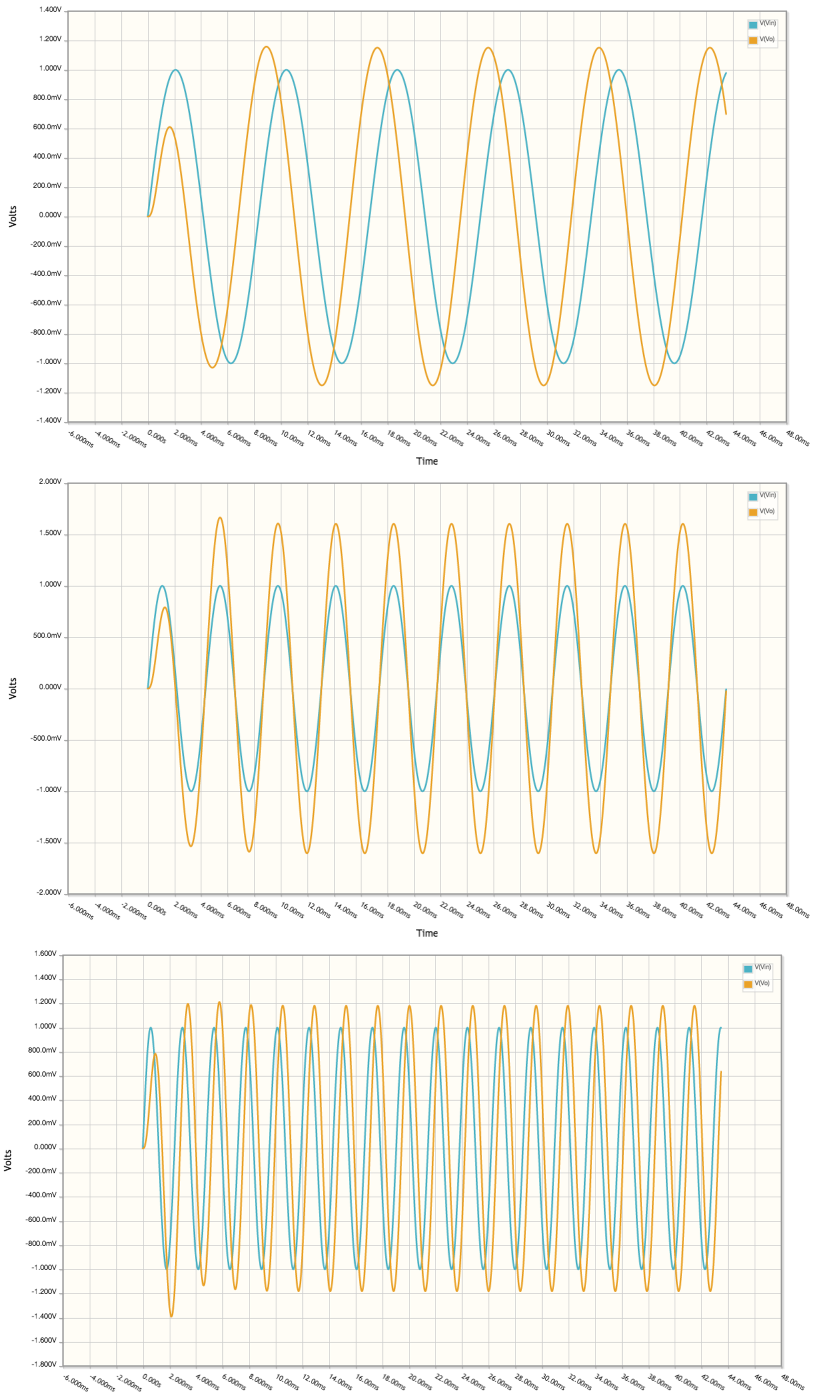
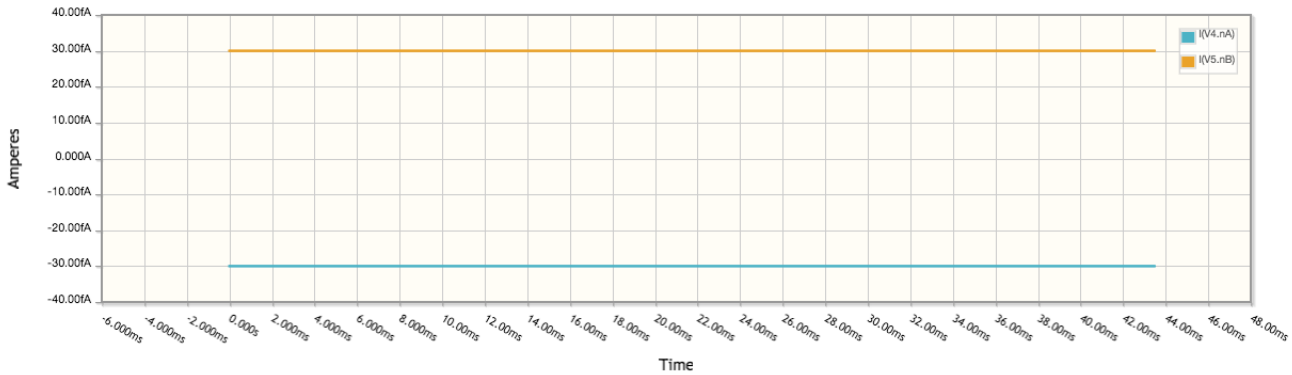


Figure 32: *Treble filter time domain plots showing DC current draw at center frequency on the positive and negative power supply.*



4.3.1 Set-up

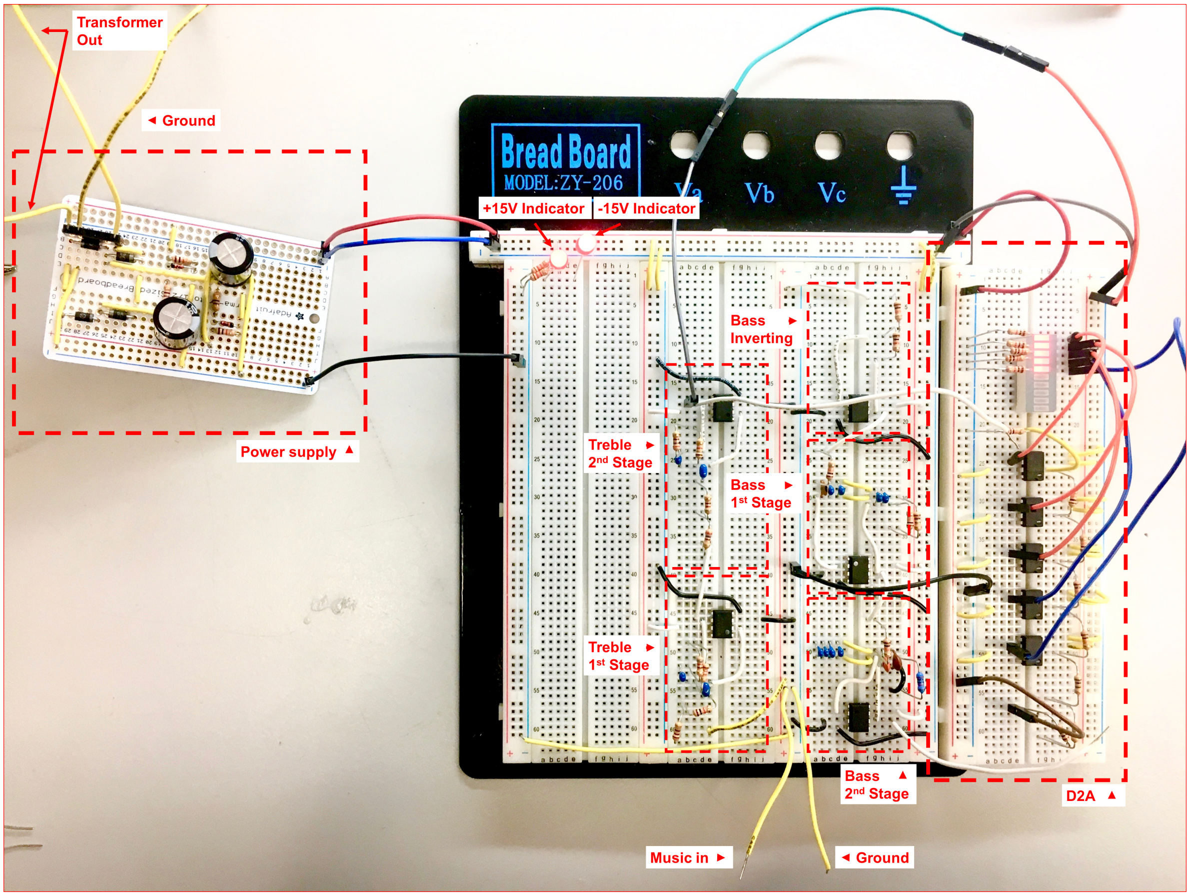
We built the filters on a three-column bread board. This allowed us maximum flexibility while debugging and improving the circuits. The equipment used for testing is as follows. Through the earlier phases of testing, before our AC/DC converted was completed, a Hewlett Packard E3631A Triple Output DC Power Supply, was used to supply a stead ±15 V to the rails of each op-amp. A Philips PM6303 RCL Meter was used to measure resistance, capacitance, and parasitic resistance values. To analyze the input and output of the, we used a Agilent Technologies MSO7034B oscilloscope. An oscilloscope is better suited in this case, as compared to a DMM, for its ability to show periodic waveforms and plot voltage throughout time. This behavior is ideal for our testing, which will involve graphing the input and output of each system over time. When testing, we generally used three channels on the oscilloscope: one for input, one for stage one output, and one for stage two output. The negative end of each cable was connected to a common ground, and the positive end was probed into our bread board. When using the E3631A power supply, we used its ground terminal for this purpose. However, eventually when we powered everything off of the AC/DC converted, we used the ground terminal from the wall socket as the common ground. To test the filters on pure sine waves, we used a Agilent 33521A function generator which was attached to the V_{in} terminal of each circuit as well as ground. A Hewlett Packard 34401A Digital Multimeter (DMM) was used to measure DC current draw.

4.3.2 Building and Testing Procedure

The process of building the filters is as follows. Because all resistor and capacitor values were already calculated and simulated, all we had to do was find a combination of parts which match the simulated values as closely as possible. Because a small percent difference can have a large impact of the characteristics of the filter, we could not rely solely on the labeled values for resistors and capacitors, as the variation in true value was too high for our tolerances. For resistors, this meant sifting through the resistors, measuring their true resistance on the DMM, and eventually finding a combination of unique resistors that built the target value. Typically this was able to be done within 0.5% error (ie. for a 1 kΩ resistor, we would achieve ±5 Ω with only two resistors in series. For capacitors, because there is less of a selection in the lab, it typically took 3 or 4 in parallel to achieve the value needed with a low percent error.

The process of testing the filters mainly involved the use of the PM6303. Using a pure sine wave as input, as opposed to music, allows us to take measurements of the transfer function in a controlled manner. We used two methods to find the center, ω_o, and cut-off frequencies, ω_{c1}, ω_{c2}. The first was based in the time domain and consisted of a manual sweep across a range of frequency values with a fixed amplitude. This allowed us to pin-point the point of maximum gain (ω_o). From

Figure 33: *Annotated photo of the completed circuit.*



there, we manually calculated $1/\sqrt{2}$ of that measured amplitude, and followed the same procedure to find ω_{c1} and ω_{c2}. The second method used the function generator's sweep mode to graph the transfer function in the frequency domain and recreate the Bode plot. The result of this is a semi-log plot which can allow for easy visualization of the frequency response, and calculation of key frequency points.

After ensuring that the cut-off frequencies were correct. we tested the filters with a music signal to gauge their noise level. This step was particularly crucial after switching from the lab power supply to our AC/DC converter. At first the converter's small deviations created noise, which prompted a re-design of the power supply with different capacitor, resistor, and zener diode values.

To measure DC power consumption on first the positive and then negative lead, we inserted a DMM between the AC/DC output and the bread board. The enabled us to get an accurate reading on the current draw for the entire circuit.

4.4 Results

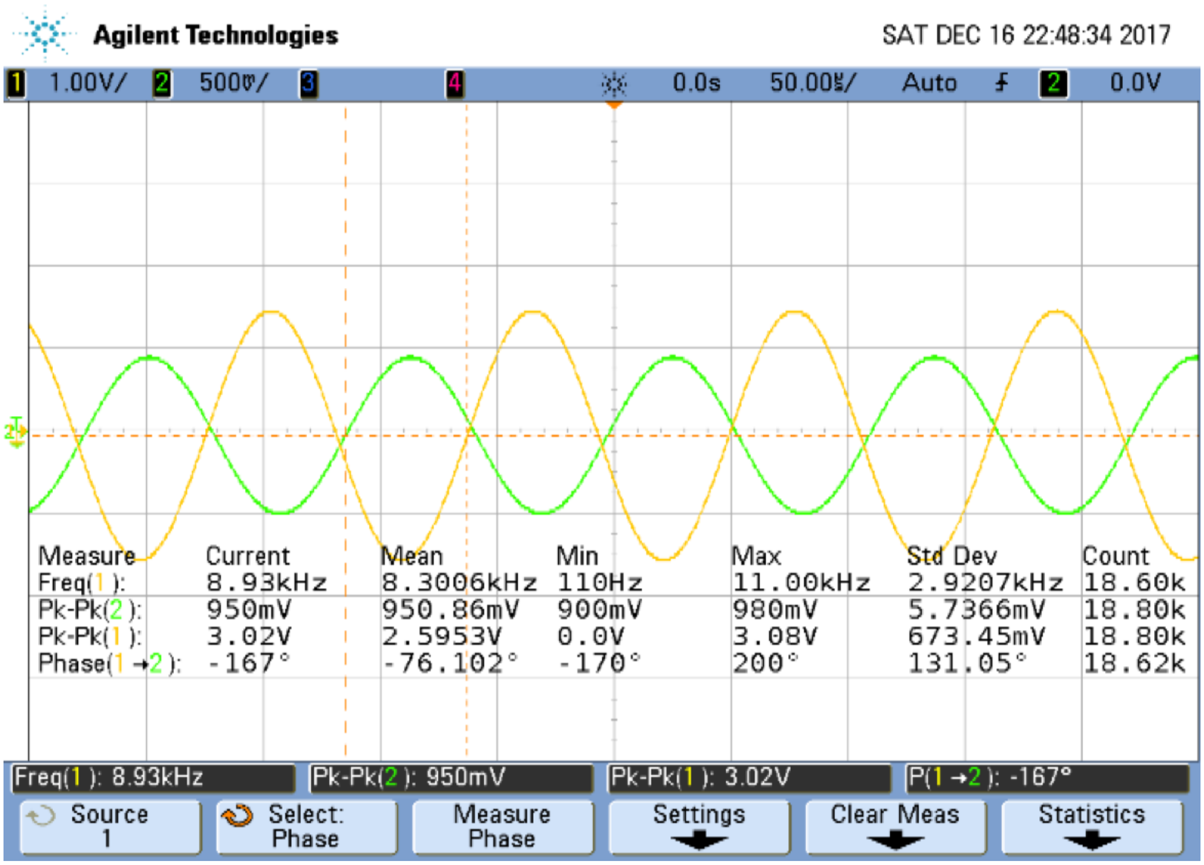
All results were obtained using the AC/DC power supply we built in the previous section.

4.4.1 Time Domain

Figures 34 and 35 show time domain results at center and cut-off frequencies for the treble channel. We see that the max gain equals $3.02/0.95 = 3.178$ V/V. Multiplying this by $1/\sqrt{2}$ gives a cut-off gain of 2.247 V/V. Figure 35 shows that at 7.78 kHz, the low cut-off, we have a gain of $2.15/0.95 = 2.26$ V/V. At 10.7 kHz, we have the same gain. Thus, we have found the cut-off and center frequencies. The phase difference is also present on those same figures. Note that at the center frequency, the phase difference is $-167°$. At the low and high cut-off points, the angle is $-119°$ and $129°$ respectively. These values match what was seen in simulation.

Figures 36 and 37 show time domain results at center and cut-off frequencies for the bass channel (note the difference in scales between the two channels) We see that the max gain equals $1.63/0.96 = 1.69$ V/V. Multiplying this by $1/\sqrt{2}$ gives a cut-off gain of 1.124 V/V. Figure 37 shows that at 117 kHz, the low cut-off, we have a gain of $1.09/0.970 = 1.23$ V/V. At 10.7 kHz, we have a gain of $1.10/0.970 = 1.24$ V/V. Thus, we have found the cut-off and center frequencies. The phase difference is also present on those same figures. Note that at the center frequency, the phase

Figure 34: *Oscilloscope screenshot showing input (green) and treble output (yellow) at center frequency. Note the difference in scales between the two channels.*



difference is exactly 0°. At the low and high cut-off points, the angle is 67° and −60° respectively.

Figure 35: *Oscilloscope screenshot showing input (green) and treble output (yellow) at low cut-off (top) and high cut-off (bottom). Note the difference in scales between the two channels.*

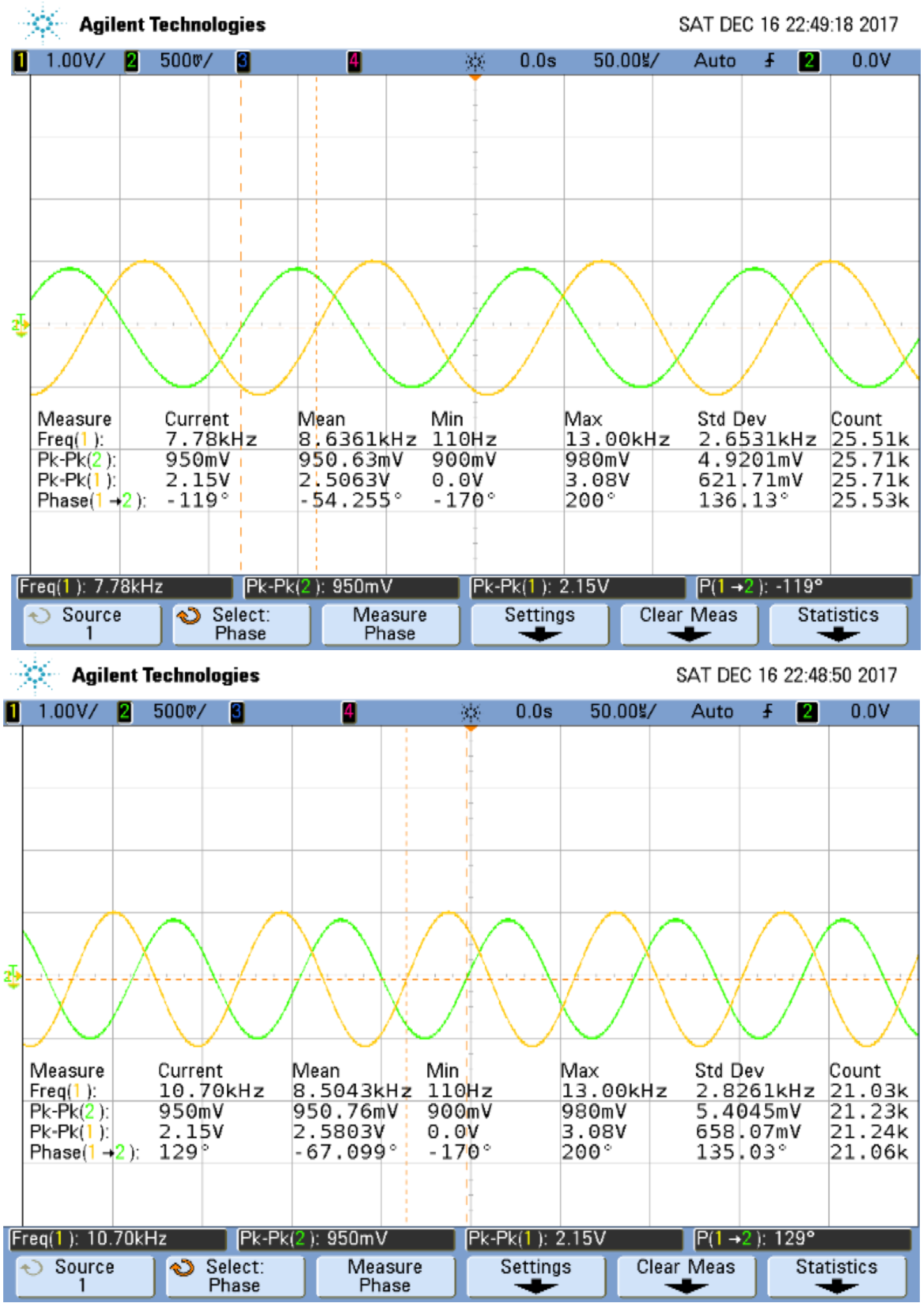
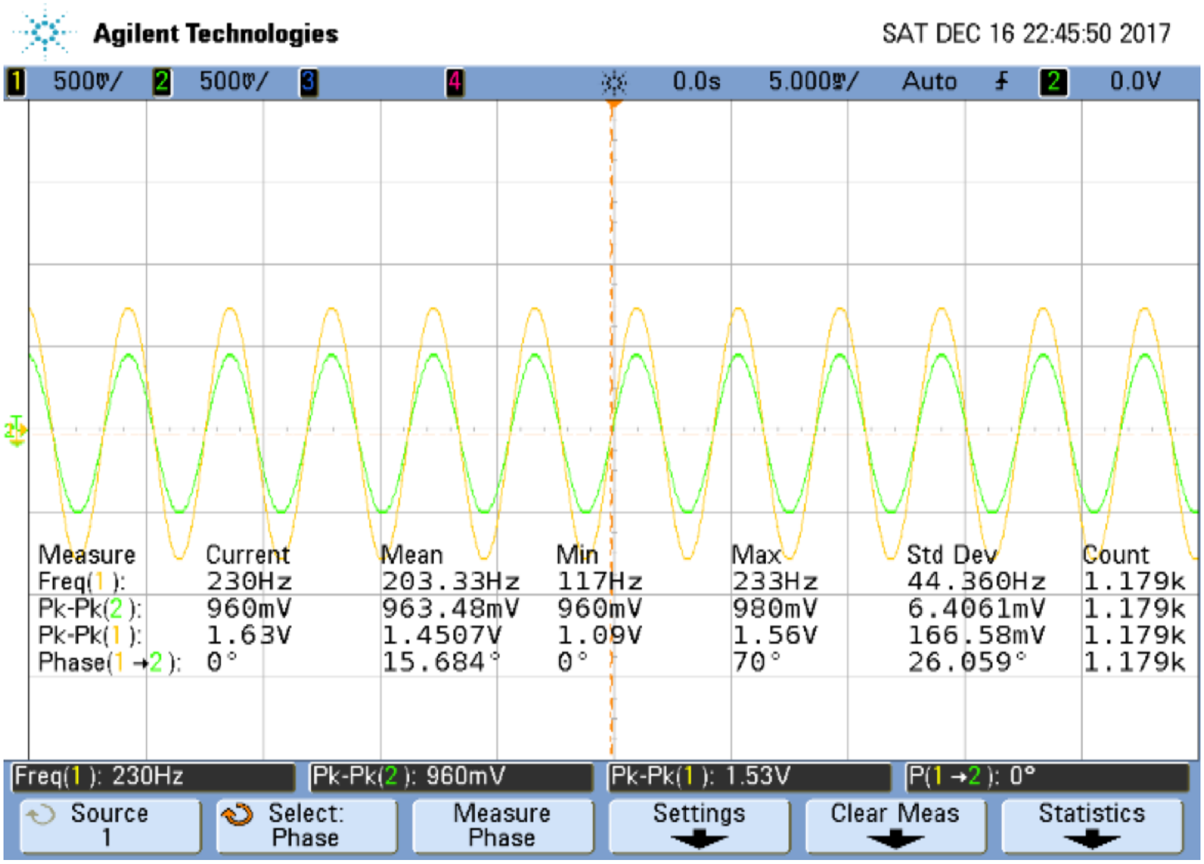


Figure 36: *Oscilloscope screenshot showing input (green) and bass output (yellow) at center frequency.*



4.4.2 Frequency Domain

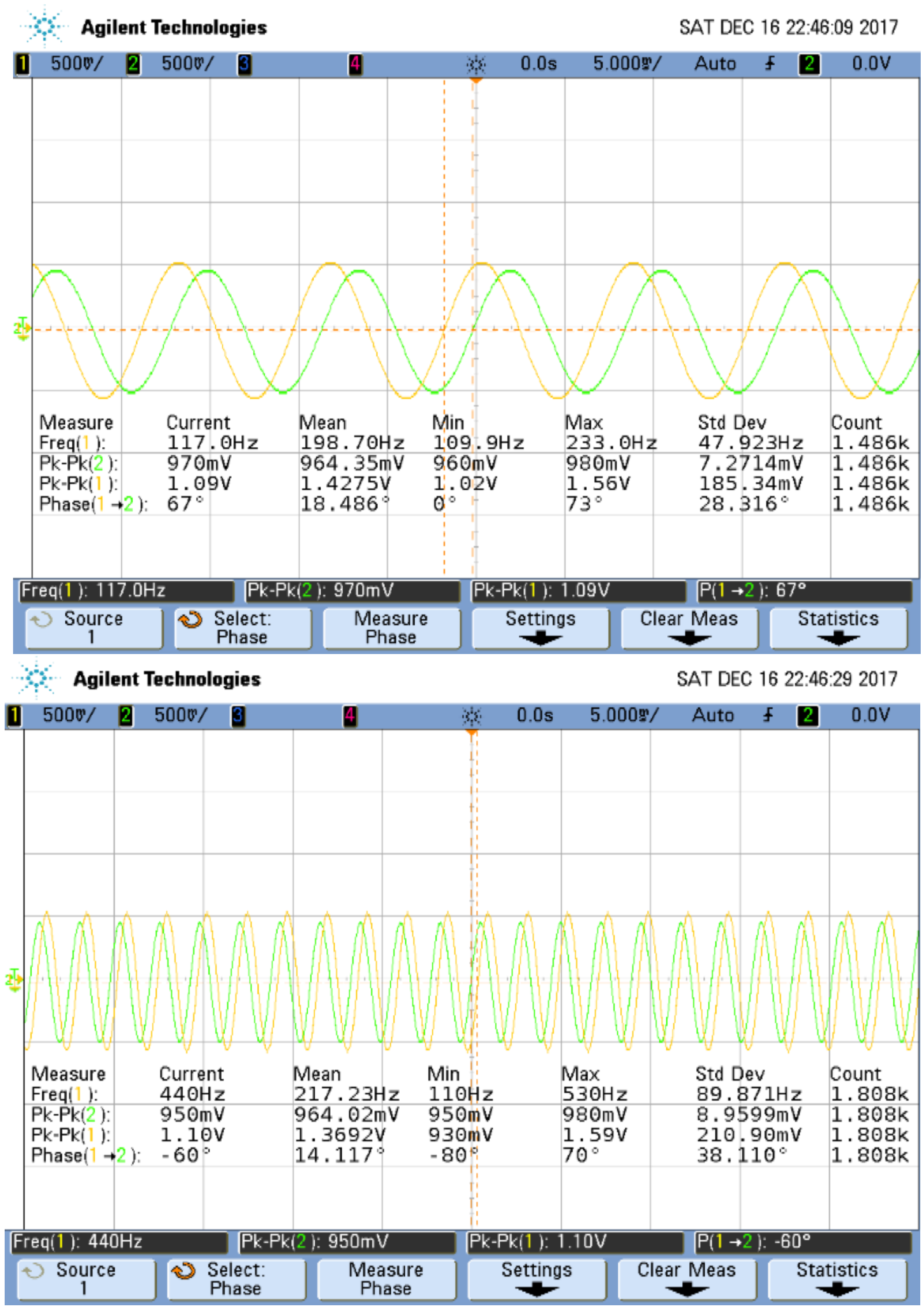
Figures 38 and 39 show screenshots from the oscilloscope after running a logarithmic sweep from 1 Hz to 100 kHz with a period, T, of 1 sec. We can easily convert from a specific time, t, to a specific frequency, f, using this information.

$$\frac{\log f}{\log 10^5} = \frac{t}{T}$$

$$f = 10^{5 \cdot \frac{t}{T}}$$

Using this formula on the values shown in Figure 38, we see that the center frequency equals $10^{5 \cdot 0.473} = 231.7$ Hz, which aligns with our simulated bass center frequency. For treble, the cen-

Figure 37: *Oscilloscope screenshot showing input (green) and bass output (yellow) at low cut-off (top) and high cut-off (bottom).*



ter frequency equals $10^{5 \cdot 0.790} = 8.910$ kHz, which also aligns with the time domain experimental analysis and simulations.

These oscilloscope screenshots also contain magnitude information. The input signal has an identical magnitude as in the time domain analysis, that is 0.95 Vpp. The bass channel sweep has a max magnitude of 1.666 V. The treble channel has a max magnitude of 3.164 V. Both of these values align with our simulation and time domain experimental results.

Figure 38: *Oscilloscope screenshot from logarithmic frequency sweep. Cursors aligned at bass center frequency and amplitude.*

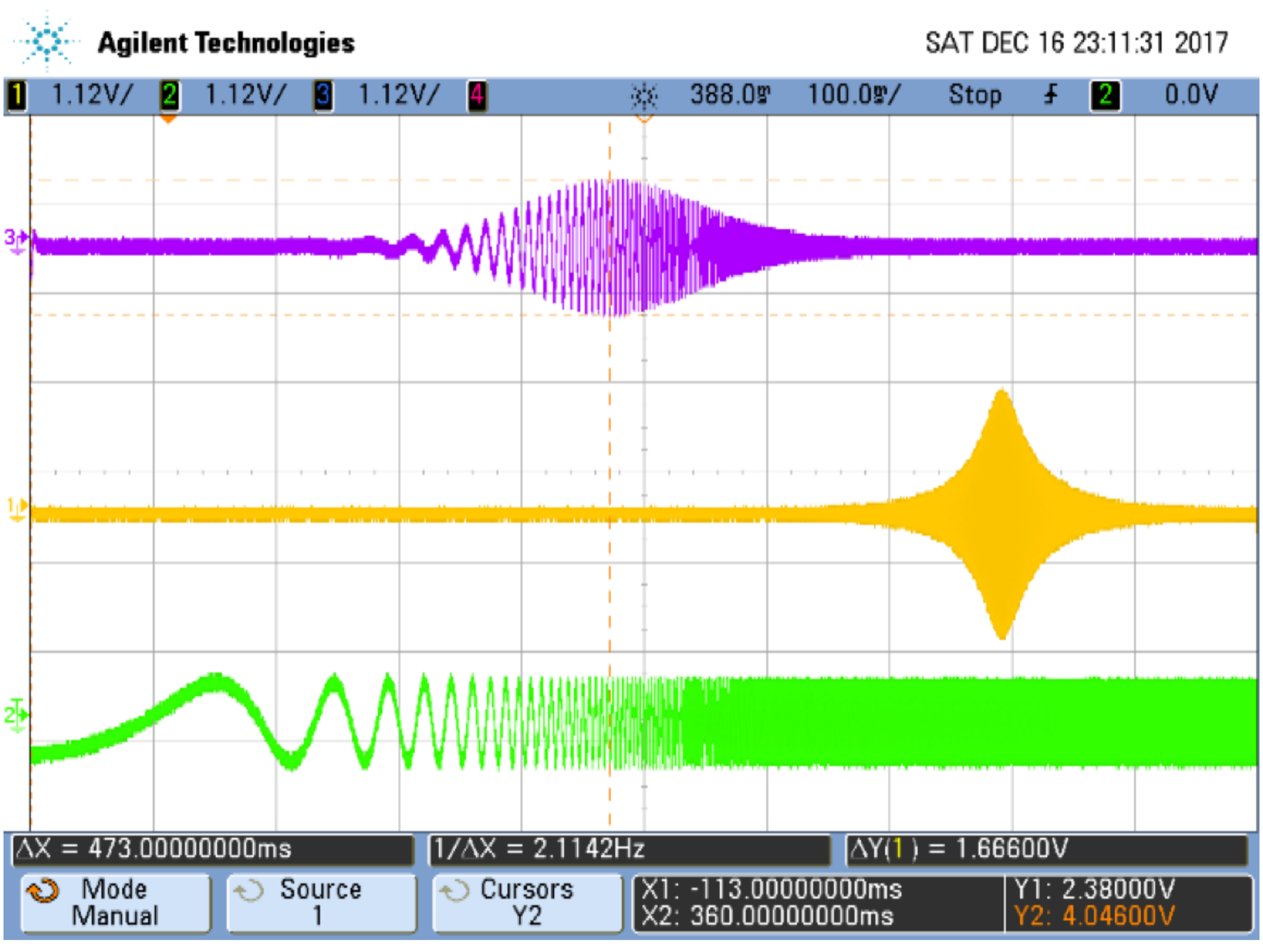
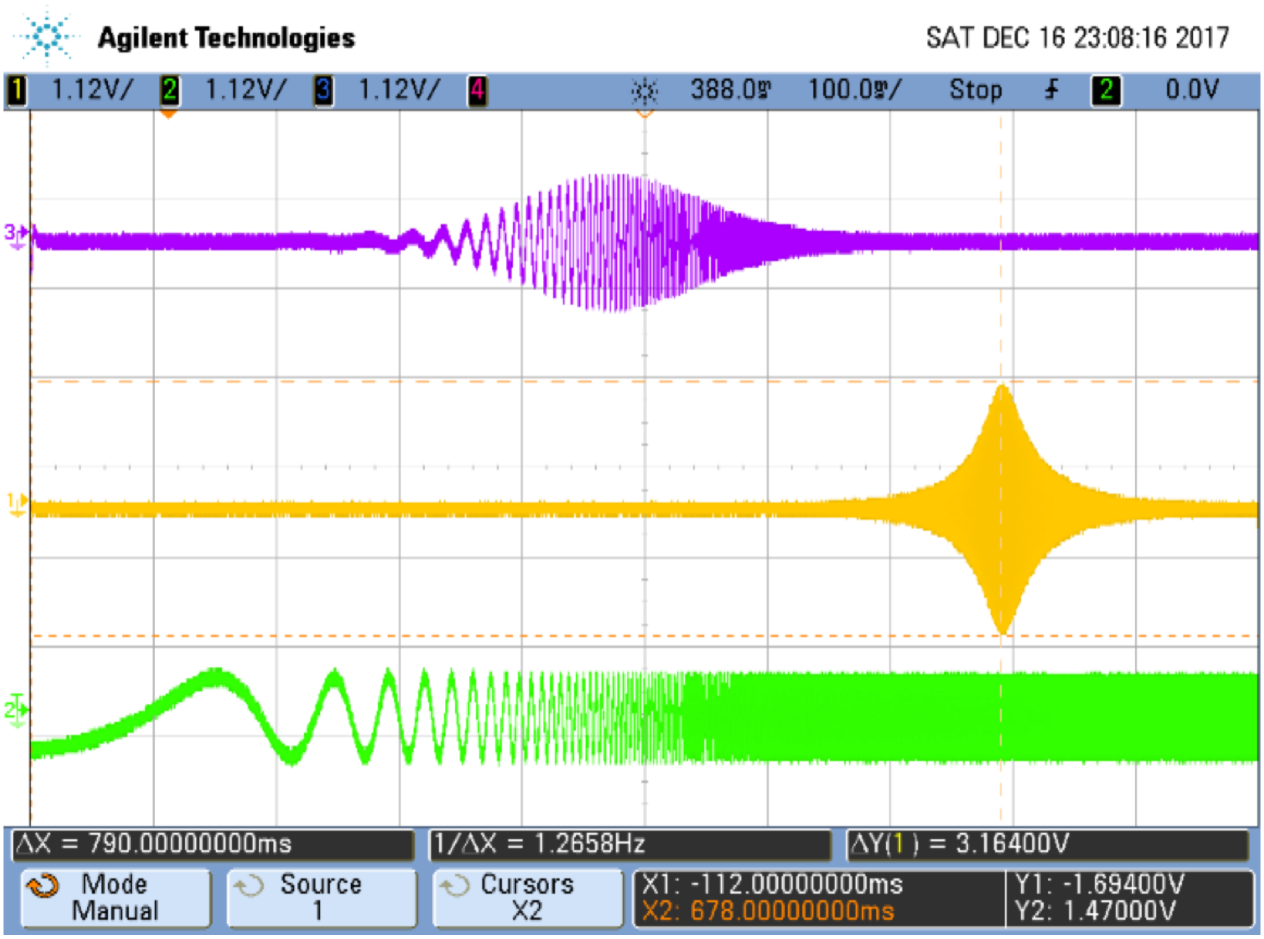


Figure 39: *Oscilloscope screenshot from logarithmic frequency sweep. Cursors aligned at treble center frequency and amplitude.*



4.4.3 DC Power Requirement

Following the procedure outlined above for measuring the DC power requirement, we obtained the following values. For the bass filter, at 230 Hz (center frequency) the filter drew 4.38 mA on the positive lead and -4.38 mA for the negative lead. For the treble filter, at 8.90 kHz (center frequency) the filter drew 6.60 mA on the positive lead and -6.59 mA for the negative lead. These values are reasonable considering that the treble uses three op-amps compared to the bass's two. The current draw for the treble is thus approximately $\frac{3}{2}$ the current draw for bass. This current draw was at a voltage of 13.68 V (from AC/DC converter), so the power requirement for the bass and treble filters is 60.0 mW and 90.0 mW respectively for each of the positive and negative outputs. Thus, the total power draw for bass and treble is 120.0 mW and 180.0 mW respectively.

4.5 Error Analysis

Overall, the results described here match well with our calculations. We break down the error analysis and offer possible justifications for their presence in the following subsections.

4.5.1 Center Frequency

First, let us compare the target center frequencies of our calculations, simulations, and experimental results. For the treble filter, we calculated a center frequency of 8.9 kHz. In the simulation, it was also found to be 8.90 kHz. In our experimental results (time domain and frequency domain) it was 8.93 kHz and 8.91 kHz respectively. Thus, our percent error for the center frequency is 0.11%.

For the bass filter, we calculated a center frequency of 250.0 Hz. In the simulation, it was found to be 230 Hz. In our experimental results (time domain and frequency domain) it was 230 kHz and 231.7 kHz respectively. Thus, our percent error for the center frequency is -7.3% compared to our initial calculations.

Overall, our calculations, simulations, and experimental results showed good agreement for the center frequency. Any small discrepancy, like the error for the bass filter, is most likely due to resistor and capacitor values not equaling their target values exactly. A second reason why the base center frequency may be off is the output resistance of the function generator, which was set to 50

Ω. Because the input resistance of the bass filter is lower than the input resistance of the treble, it may have had a larger, more pronounced, effect. Though, it is worth noting that simulating the bass filter including this output resistance did not have a noticeable impact.

4.5.2 Low Cut-Off Frequency

For the treble filter, we calculated a low cut-off frequency of 8.026 kHz (in a preemptive effort, we decreased the bass band width, so the final result would have 7.9 kHz). In the simulation, it was found to be 7.90 kHz. In our experimental results (time domain) it was 7.78 kHz. Thus, our percent error for the low cut-off frequency is 1.5%.

For the bass filter, we calculated a low cut-off frequency of 150 kHz. In the simulation, it was found to be 120 kHz. In our experimental results (time domain) it was 117.0 kHz. Thus, our percent error for the high cut-off frequency is -22.0% when compared to our calculations.

In the treble filter, our calculations, simulations, and experimental results showed good agreement for the low cut-off frequency. The bass filter error is larger. However, note that our experimental results agree closely with our simulation. Thus, this high error is most likely due to an interference in the low-pass and high-pass halves of the filter. As we have previously mentioned, because the pass band was so narrow, the non-linearity of the true Bode plot it becomes more prominent.

4.5.3 High Cut-Off Frequency

For the treble filter, we calculated a high cut-off frequency of 9.774 kHz. In the simulation, it was found to be 10.4 kHz. In our experimental results (time domain) it was 10.7 kHz. Thus, our percent error for the low cut-off frequency is 8.1%.

For the bass filter, we calculated a high cut-off frequency of 350 Hz. In the simulation, it was found to be 420 Hz. In our experimental results (time domain) it was 440 Hz. Thus, our percent error for the high cut-off frequency is 25.7%.

We see a similar pattern here as with the low cut-off frequency. We believe the errors present for the high cut-off are the same as mentioned in the previous subsection.

4.5.4 Gain

For the treble filter, we calculated a maximum gain of -3.33 V/V. In the simulation, it was also found to be -3.25 V/V. In our experimental results (time domain and frequency domain) it was -3.178 V/V and -3.164 V/V respectively. Thus, our percent error for the low cut-off frequency is -4.6%. Our simulation and experimental results all show good agreement. Such a small error can be expected given the tolerances of the parts we were working with.

For the bass filter, we calculated a maximum gain of 3.24 V/V, as the individual gain of each stage was set to 1.8 V/V. However, it should be noted that when we set those values we knew that the actual gain would be significantly lower, due to interference effects between the high-pass and low-pass halves of the filter. Our target gain from the specifications was 1.5 V/V. In the simulation, it was found to be 1.65 V/V. In our experimental results (time domain and frequency domain) it was 1.69 V/V and 1.67 V/V respectively. Thus, compared to our calculations, our percent error for the low cut-off frequency is -47.8%. This value does not properly represent the filter. Compared to the simulation, our percent error was 1.2%.

4.5.5 Roll-Off Rate

Both of our filters successfully achieved 40 dB/decade roll-off rate, as they were second order.

4.6 Conclusion

Overall, we met the specifications fairly closely. The bass filter was not as successful, though still matched the specifications marginally well. We probably should have recognized the limitations of the design chosen for the bass filter, specifically that the pass band was too narrow, and found a different configuration: possibly even using the configuration employed in the treble section for the bass. The reason we did not was simply because the bass had already been completed before we started work on the treble. The treble filter was highly successful. The experimental results matched very closely with our initial calculations and our specifications were largely met. The multiple-feedback band-pass active filter had a narrow enough pass band for the specifications. Initial, we had built the treble system using the same configuration as the bass, but found that it was impossible to hit the desired cut-off frequencies with such a configuration. Rebuilding it with the different schematic was worth it in that regard.

The aim of the treble and bass filter components was to (1) demonstrate an ability to use the principle learned in lecture on the frequency response of capacitors and inductors, and (2) create a working docking station able to separate music into a bass and treble channel. When we tested our system with music it performed as expected, properly separating the low and high frequencies and removing the mid-range. This section was successful in that regard.

5 Final Integration

This section outlines the process of combining the two subsystems, the AC/DC converted and the filter components, into one working product.

5.1 Calculations

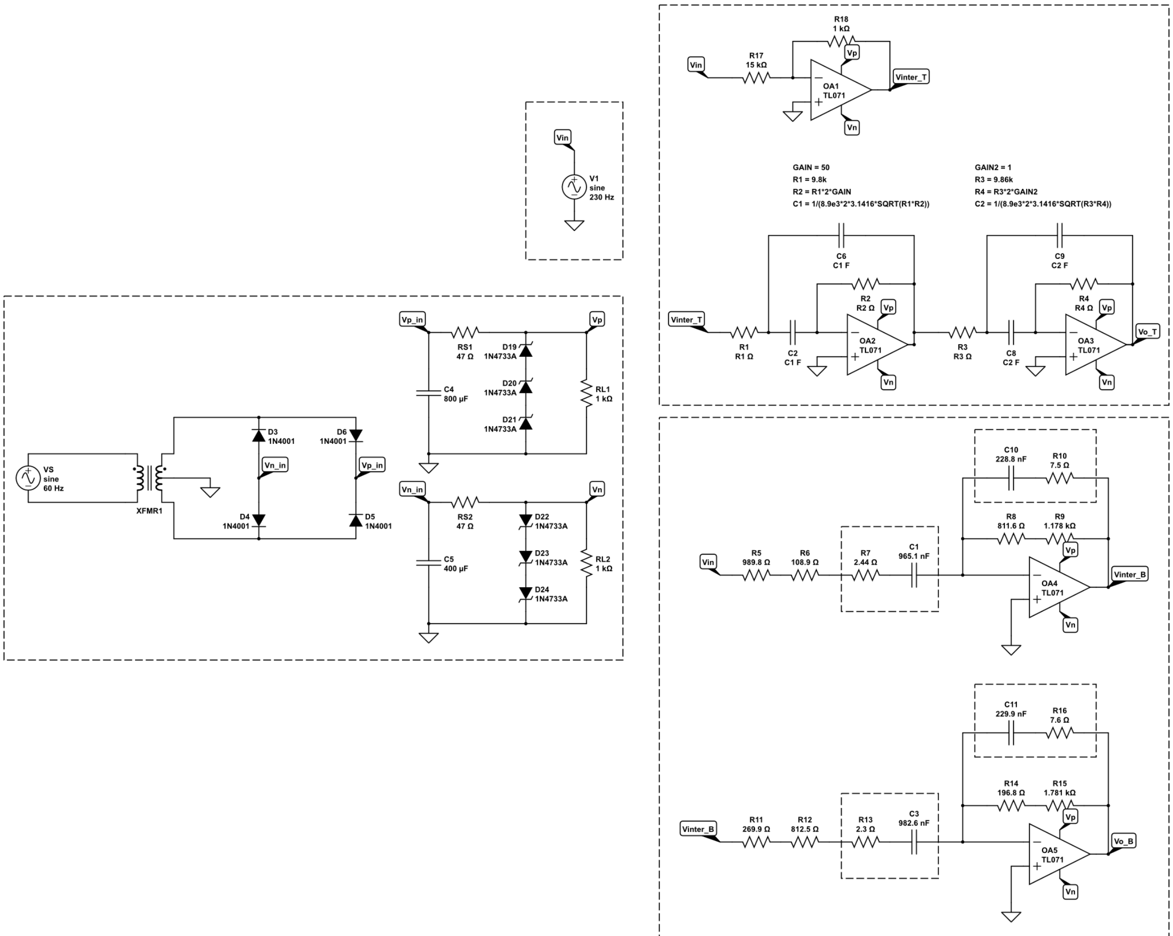
As we measured in the Filter Experimental Results section, the max-current required by the bass filter is 4.38 mA. The treble filter required 6.59 mA at max. Thus, our power supply must be able to supply approximately 10 mA (for both the positive and negative leads). As we will show in the following simulations and experimental results, it is able to do so.

5.2 Prebuild: Simulation

Figure 40 shows a complete schematic of all subsystems. As can be seen, the output of the AC/DC converter is fed into the rails of the op-amps of each filter. The output of the entire system is thus the nodes V_{o_T} and V_{o_B} for treble and bass respectively.

When we attempted to simulate the complete schematic, we ran into an issue where CircuitLab's simulator did not terminate. We believe this is due to there being difference frequency sources, 60 Hz for one, and the center frequency for the other. Thus, we must compromise and simulate each part separately, however with updated loads. For the power supply, we must simulate it with the equivalent load (indicator LEDs and op-amps from both filters). We determined what this load was experimentally with the Hewlett Packard E3631A power supply and a DC ammeter. We determined the current draw was 20.239 mA for the positive lead and -20.239 mA for the negative

Figure 40: *Complete schematic of all subsystems.*

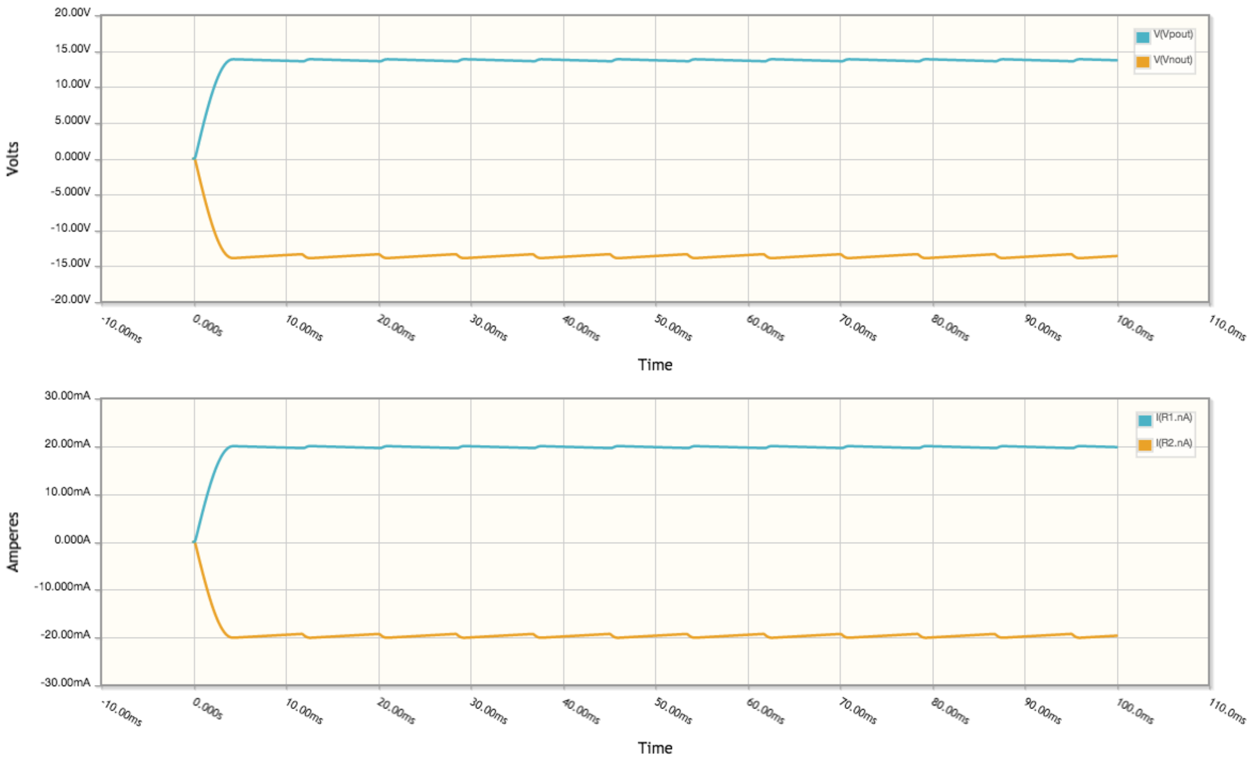


lead at the bass center frequency (230 Hz). We found that the current draw for the treble center frequency was sufficiently similar. Assuming a 14 V supply, the load is $\frac{14}{20.329} = 692\ \Omega$. We thus need to simulate the power supply with a 692 Ω load. This can be seen in Figure 42. Note that the voltage current simulations matched our preliminary experimental results.

Zooming in on the positive lead's ripple, we can see that it varies from 13.57 V to 13.85 V, which is $\frac{0.280}{13.84} \approx 2\%$.

Lastly, we used this information to simulate the treble and bass filters together. Because CircuitLab was unable to simulate our power supply active as the DC source, we used two voltage

Figure 41: *Voltage and current drawn from positive and negative leads using load equivalent to that of the filters and LED indicators (692 Ω).*



sources, set to 13.85 V and -13.85 V for the positive and negative leads respectively. We used a 300 mV_{RMS} as the input. Figure 43 shows the resulting gains of this simulation. The results are as expected, so we can now move onto to experimentally combining subsystems.

Figure 42: *Close up of voltage ripple from positive lead using load equivalent to that of the filters and LED indicators (692 Ω).*

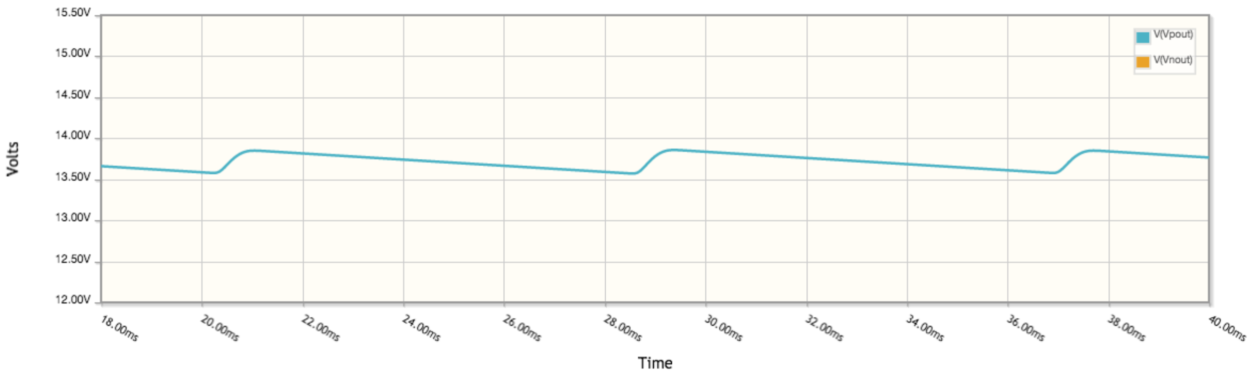
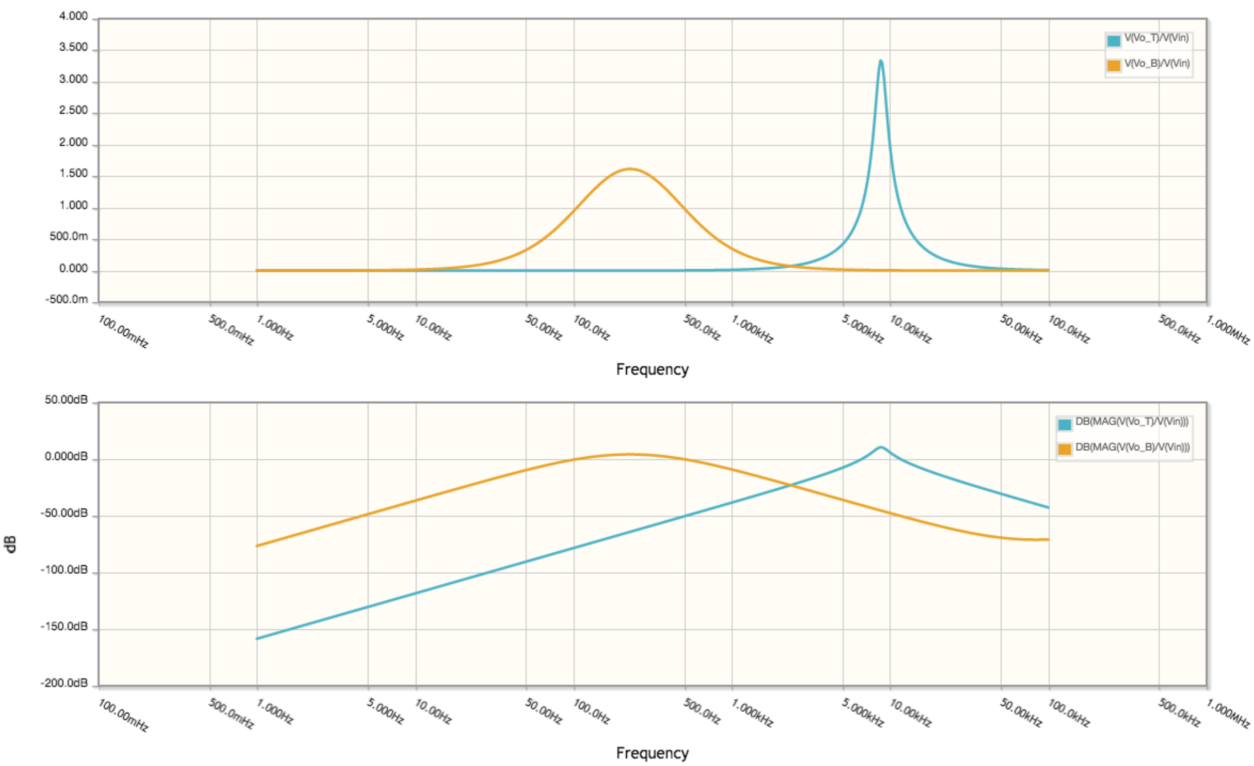


Figure 43: *Simulation of gains of treble and bass filters together, using ±13.85 V DC source.*



5.3 Experimental Setup

Once again, to analyze the input and output of the, we used a Agilent Technologies MSO7034B oscilloscope. An oscilloscope is better suited in this case, as compared to a DMM, for its ability to show periodic waveforms and plot voltage throughout time. This was especially helpful when analyzing the variation in the AC/DC output once it was attached to the filters. An Agilent 33521A function generator was also used as the main input for our testing in this section, although some final tests were run with music output from a MacBook. A Hewlett Packard 34401A Digital Multimeter (DMM) was used to measure DC current draw.

5.4 Experimental Procedure

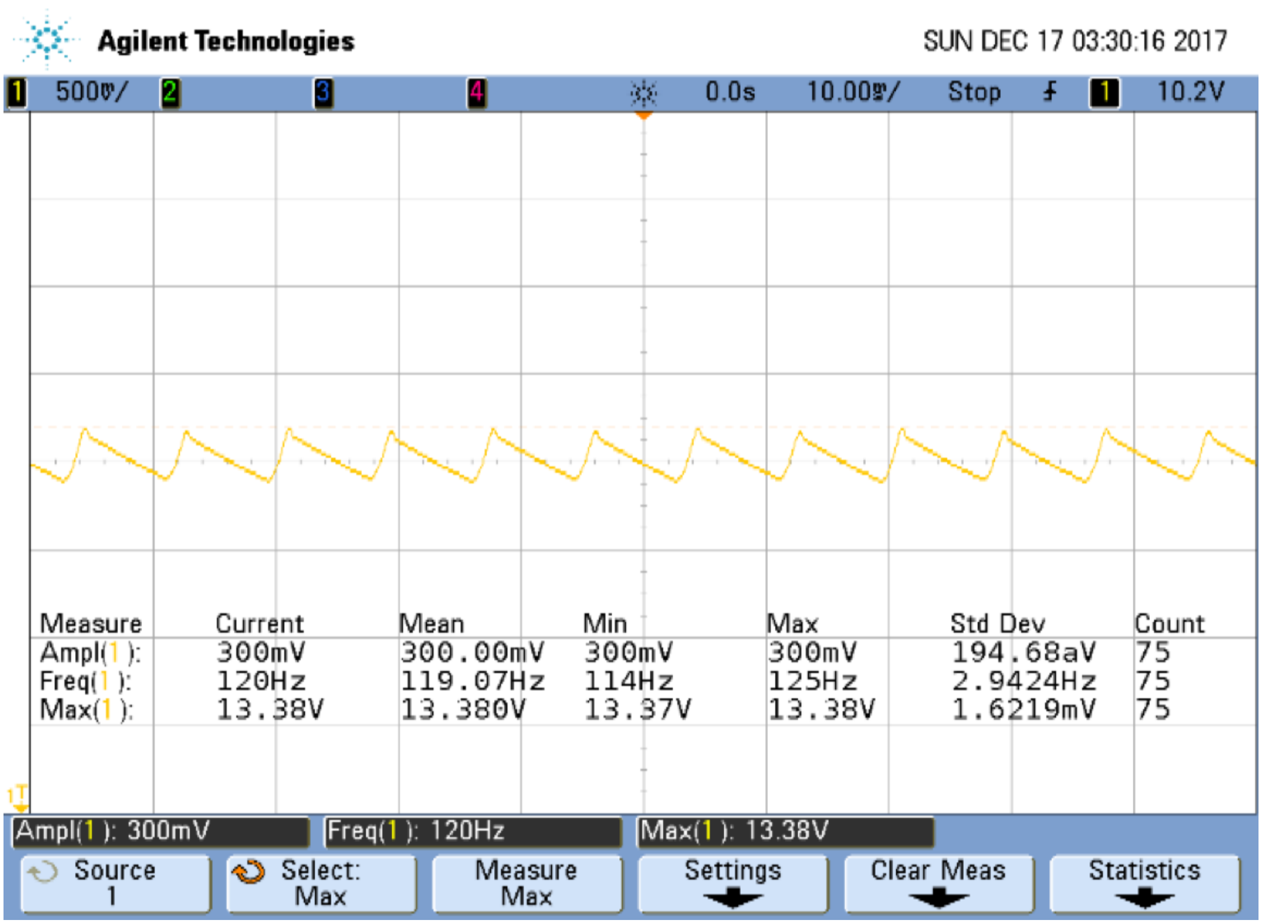
For the experimental procedure of testing the filters running off of the AC/DC converter, please see the Experimental Procedure subsection (Section 4.3.2) of Section 4. All results described there were obtained using the power supply built in Section 3.

Additional steps that we took to test the integration of our subcomponents are as follows. We used the oscilloscope attached to positive lead of the power supply and common ground. We then used the Agilent 33521A function generator attached to the V_{in} terminal of each circuit to supply a sine wave at the proper frequency. The frequencies we chose were the center frequencies of each filter, as this is the point where they should be drawing the most current and putting the smallest load on the AC/DC converter. We then followed the procedure outlined in Section 4.3.2 to verify that the filters were working correctly and with little to no noise.

5.5 Experimental Results

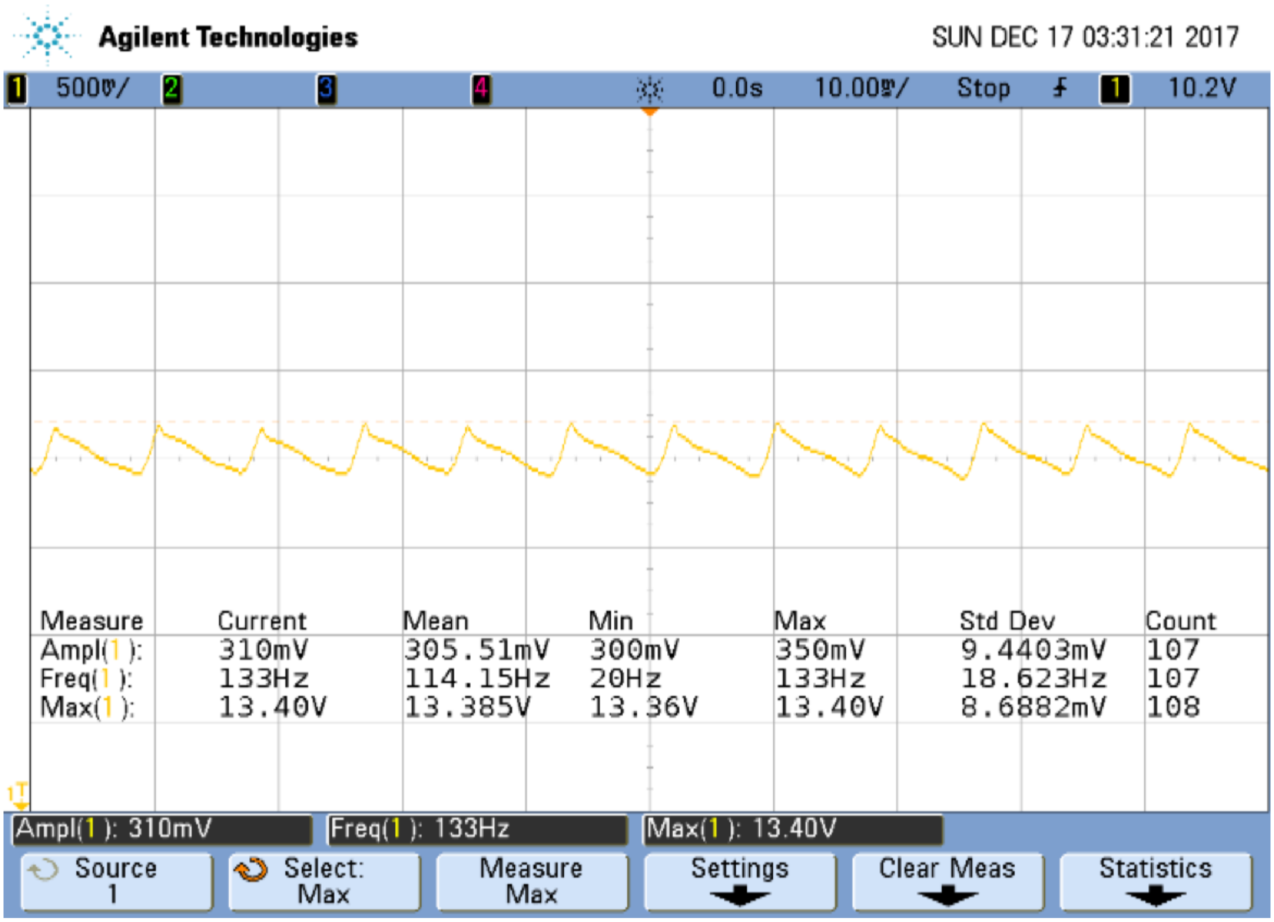
For results regarding the performance of the filters running off of the AC/DC converter, please see the Experimental Results subsection of Section 4. All results described there were obtained using the power supply built in Section 3.

Figure 44: *Oscilloscope screenshot showing the amplitude, frequency, and max-voltage of AC/DC converter voltage (positive lead) at the center frequency of the treble filter (8.90 kHz).*



The following two oscilloscope screenshots show the variation in the output of the AC/DC converter. Figure 44 shows this at the treble filter center frequency (8.90 kHz), and Figure 45 shows

Figure 45: *Oscilloscope screenshot showing the amplitude, frequency, and max-voltage of AC/DC converter voltage (positive lead) at the center frequency of the bass filter (230 Hz).*



this at the bass center frequency (230 Hz). Note that the frequency of both are 120 Hz. This is due to the rectifier which effectively takes the absolute value of the AC wall-socket, doubling the frequency. Also note that the max-voltage at the treble center frequency is slightly lower than the max-voltage at the bass center frequency. This can be attributed to the extra op-amp in the treble filter, which at the treble center frequency will be noticeable whereas at the bass center frequency it would not. The result of this is a higher current draw from the AC/DC converter and thus a slightly lower voltage. These results agree with our DC current draw measurements. The DC current draw was as follows: with the full load (both filters on, LED indicator lights on) at the bass filter center frequency (230 Hz), the current draw on the positive lead was 20.228 mA and -20.228 mA on the negative lead. At the treble filter center frequency (8.9 Hz), the current draw on the positive lead was 20.239 mA and -20.239 mA on the negative lead. The ~ 20 mA draw makes sense as a single LED drew about 10 mA, the bass filter alone (see Filter Experimental Results section for more detail) was approximately 4.4 mA and the treble filter alone drew about 6.60 mA.

Note that with a full load (both filters on, LED indicator lights on) the variation at both center frequencies is still only $\frac{0.3}{13.4} \approx 2\%$.

5.5.1 Error Analysis

We can now compare the full sub-system simulations seen in Section 5.2 to the experimental results seen in Section 5.5. We first compare the max voltage output from the power supply. In simulations it was 13.85 V and experimentally it was 13.40 V. This is a percent error of -3.25%. We now compare the voltage ripple. In simulations it as 0.280 V and experimentally it was 0.300 V. This is a percent error of 6.67%.

These relatively small errors can be attributed to the fact that we were not able to simulate the all sub-systems simultaneously, due to the complexity of the schematics.

The filter simulation using the approximated AC/DC converter produced identical results to those described in more detail in the Filter Components Section.

5.6 Final Conclusion

One lesson learned though failure in this case, is to combine subsystems as early as possible, or at least test them together as soon as one works. There will nearly always be unexpected outcomes in that process, many of which would be easier to fix if known about sooner rather than later. An example from this project was after testing switching the filter's power supply from the E3631A to our AC/DC converter, a large amount of noise was introduced into the output. When music was being played, the result was a barely audible song. We determined that it was the small variation in DC voltage that was causing the larger fluctuations in the music. Although we eventually solved this issue by reworking our power supply, if we had known about this issue soon we could have saved time by building a correct power supply the first time.

Comparing the results of the audio docking station produced to the initial specifications, we were successful in meeting the goals set. Treble cut-off frequencies had a very low error, approximately 1%, and center frequency was also right on target, around 0.1% error. The bass was slightly off specification, but still fairly close with errors around 10%. Overall, it was here that we saw the biggest difference between our hand calculations and our final results. The pass band of the bass filter was slightly too wide; however, this is a product of the filter configuration we chose. It was not capable of producing a 200 Hz pass band from 150 to 350 Hz. If we were to rebuild this product, we would certain try other configurations, such as the multiple feedback type employed for the treble filter.

The gain of each filter was set appropriately high for the external audio amplifier to emit loud music with fairly little noise or distortion.

Overall, the produced docking station was largely successful.