Basic Audio Equalizer

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Abstract

The following lab report was to design a basic analog audio equalizer and to integrate it into a circuit in order to match certain criteria. More specifically, a circuit blueprint had to be properly determined for each subsystem involved within the circuit that would theoretically line up with each requirement a completed circuit would reach. This lab report utilized the findings of a variety of labs performed previously in order to determine this topology, such as looking at the findings of Resistor-Inductor-Capacitor filters and op amps from experiment 10 to find specific resistor values. This was achieved through a three-step process. The first step mainly focused on computational measurements to achieve the desired effects of each filter as well as going through multiple kits to find more of a certain type of component than one kit had, as well as agreeing over which components would yield closer results to the desired values due to imprecision. The second step was simply the process of testing each subsystem on its own to ensure that its output aligned with our calculations. The final step revolved around integrating every independent subsystem and making any modifications to the circuit as necessary if any complications arose.

Overview

Three Frequency Bands

The circuit was built with only three separate frequency bands, a significantly lower number of bands than the average audio equalizer for the sake of simplicity. The task was primarily focused on giving each of the frequency bands its own role. More specifically, one band was to behave as a bass filter to handle frequencies below 320 Hz, another was to act as the treble filter to deal with frequencies above 3.2 kHz, and the last filter was built to handle frequencies between the two ranges. These filters were very similar to the low and high pass filters first brought up in experiment 10, which could be used to help with splitting apart the input. A combination of both filters could be used in order to mimic a band path filter, which creates an ideal overlap range between two frequencies. Lastly, since every frequency band would be a parallel branch when implemented into the circuit, all three frequency ranges would have to be added up. Put more simply, the circuit's design would involve taking the input signal, branching into three separate paths each with a frequency band and rejoining back into one path.

Controllable Independent Gains

This aspect mainly exists to establish that each of these frequency bands had to have their own independent gains, all of which had to be controllable. This meant that the op amps first introduced in experiment 8 had to be utilized in order to ensure that each band had its own gain. Additionally, each band would need its gain to be manually controllable by an observer, referring to the usage of several potentiometers which would each behave as a variable gain. This task essentially meant each frequency band would need additional elements to make each gain independent and controllable while the audio signal was still separated.

Adding Signals and Controlling Volume

After the frequency bands had their own independent gain applied, they needed to be recombined into one audio signal. Another op amp configuration had to be utilized here in order to combine the inputs by using a summing configuration, which would reconstruct the signal completely. Afterwards, yet another op amp subsystem was needed in order to manually control the volume of the audio signal itself. In summary, this last task was focused on the recombining and manual controlling of the signal itself.

Requirements

Specification	Requirement
Speaker resistance	8Ω
Bass filter -3 dB cutoff	320 Hz $\pm$10%
Mid filter -3 dB bandwidth	0.32 kHz to 3.20 kHz $\pm$10%
Treble filter -3 dB cutoff	3.20 kHz $\pm$10%
v_{amp} with all knobs at minimum settings	15 mV_{RMS} @ 200 Hz, 2kHz, and 10 kHz
v_{amp} with all knobs at maximum settings	$100mV_{RMS} \pm 10\%$ @ 200 Hz, 2kHz, and 10 kHz
$v_{amp,max} - v_{amp,min}$ max ripple with equalizer max	15 mV_{RMS} from 200 Hz to 10 kHz
Amplifier output power	400 mW from 200 Hz to 10 kHz

Table 1: All conditions to meet.

Theory

Concepts, formulas, and equations used within the report

$$V_{RMS} = \frac{V_{PP}}{2\sqrt{2}} \tag{1}$$

$$P = \frac{V_{RMS}^2}{R} \tag{2}$$

Equation 1 is a formula to quickly calculate the root-means-squared voltage when given a peak-to-peak voltage from the input. Equation 2 takes the rms voltage and uses it to determine the power through a resistor.

$$P = \frac{V_{PP}^2}{8R} \tag{3}$$

Combining equations 1 and 2 creates equation 3, which is a convenient formula for quickly determining the power dissipation through a resistor by using the peak to peak voltage of the input.

$$V_{out} = -\frac{R_2}{R_1} * V_{in} \tag{4}$$

Equation 4 was used within the integrated circuit (op amp) to control the volume of the output by inverting the input wave and controlling the resistance via a potentiometer.

$$V_{amp} = V_{in} * F_F(f) * G_F(f) * S_F(f) \tag{5}$$

Equation 5 is the general calculation of the output through any one of the frequency bands. The "f" refers to the frequency of the input and the "F" refers to the corresponding frequency band to consider depending on the value of f. The three function at the end refers to the filter, the equivalent game, and the summing respectively.

$$f = 200Hz, F_{LP} = 1 \tag{6}$$

$$f = 200Hz, G_{LP} = \frac{10K}{R_{in,LP}} \tag{7}$$

$$f = 200Hz, S_{LP} = \frac{10K}{R_{S,LP}} \tag{8}$$

Equations 6, 7, and 8 are the results of a 200 Hz frequency input wave. The denominators of the last two equations must be solved for (there is no single answer for both, but their product must match the value on the other side of the equation from equation 5).

$$|H(w)| = |\frac{Z_c}{Z_c + Z_R}| = \frac{1}{1 + j * 2 * \pi * f * R * C} = \frac{1}{\sqrt{2}} \tag{9}$$

$$|H(w)| = |\frac{Z_R}{Z_c + Z_R}| = \frac{R * j * 2 * \pi * f * R * C}{1 + R * j2\pi * f * R * C} = \frac{1}{\sqrt{2}} \tag{10}$$

Equations 9 and 10 were utilized in order to find the measurements of the components to use for each of the three frequency bands. The bass band used only equation 9, the treble band used only equation 10, and the middle band used both.

Methods

Computing Voltage for Amplifier

Firstly, the values for each subsystem had to be chosen prior to assembling the circuit proper. For the bass filter, equation 9 was used to determine capacitance and resistance values. A capacitance of 2.2 microFarads was chosen simply out of convenience, leaving a resistance of 220 ohms. Calculations for the low end of the middle band used the same formula, using the same capacitance and 2.2 ohms to compensate for the op amp. The high end of the middle band used equation 10, where 4.7 microFarads was chosen again out of convenience and 110 ohms was calculated to fit. The treble filter used the same equation, wherin the same capacitance was used and 11 ohms was used.

Constructing and Measuring Subsystems

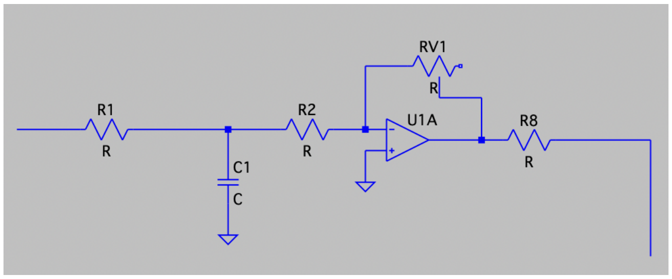

Figure 1: The low pass frequency band (bass).

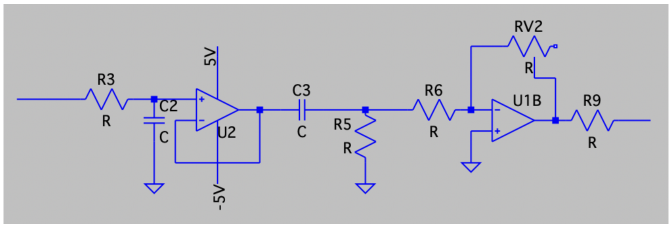

Figure 2: The mid pass frequency band.

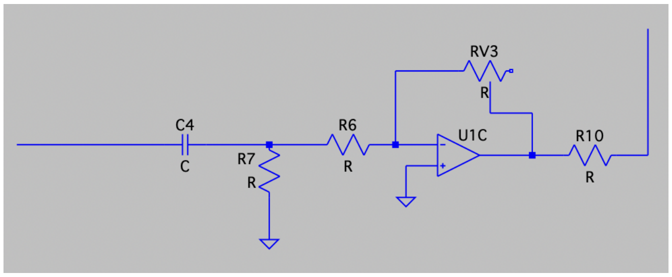

Figure 3: The high pass frequency band (treble).

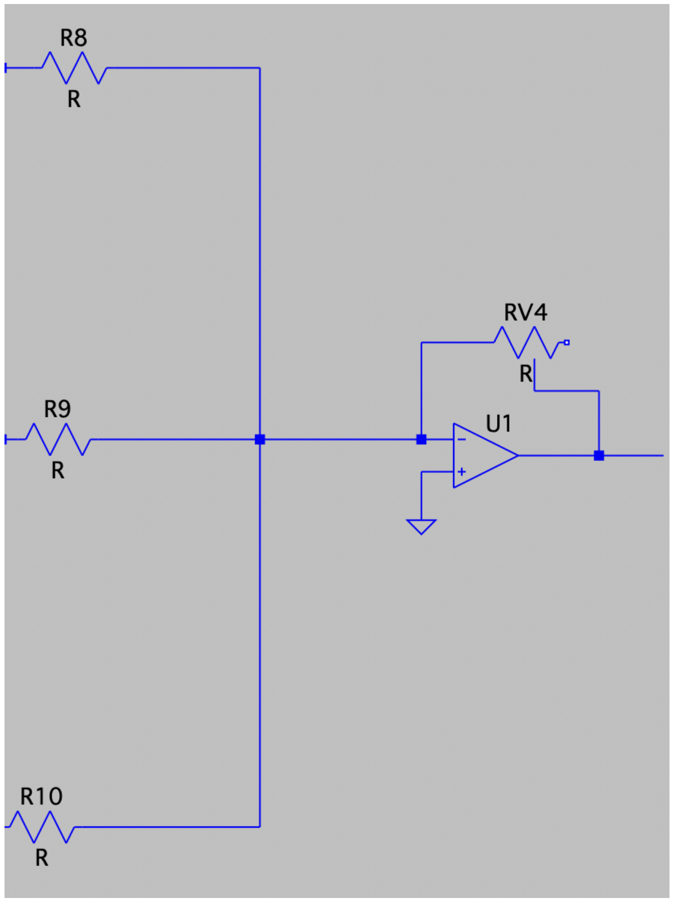

Figure 4: The summing amplifier reconnecting every band.

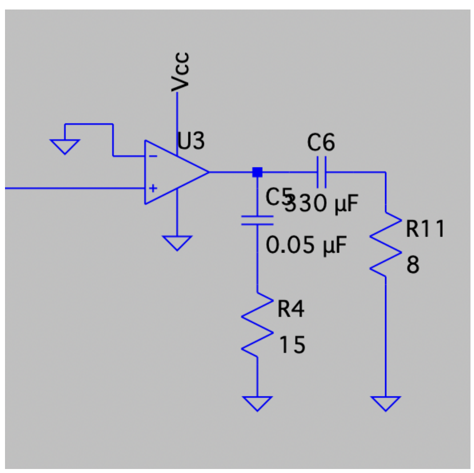

Figure 5: The audio amplifier to provide sufficient voltage and current.

Using the calculated values from before with the topology of the circuit agreed upon, we were able begin with assembling the various sections for individual testing. We had begun testing the circuit while it was still in parts to ensure desirable results, which allowed for quicker adjustments. The mid-pass frequency band in particular required many changes before it began to behave as expected. The summing amplifier, on the other hand, was much similar due to involving minimal calculations to operate correctly.

Integration of Subsystems

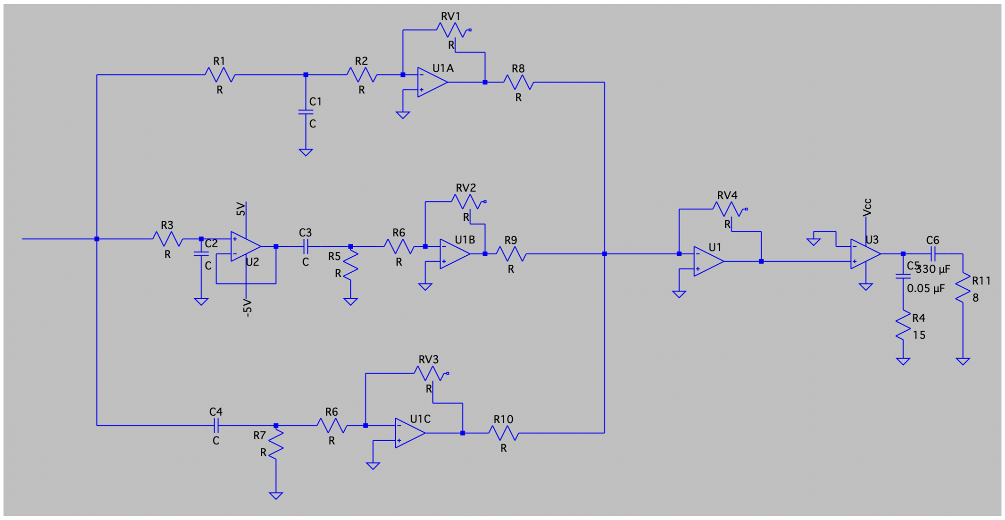

Figure 6: The entire circuit, with all components combined.

With every individual component passing each test, we proceeded to construct the entire system to observe how each subsystem would interact with each other and whether or not it would continue to meet the criteria. After reviewing the resistances over and exchanging several components to ensure they were not faulty, plots were made to showcase the circuit's various aspects.

Results and Discussion

Three Frequency Bands

Each of the bands was tested independently, with the middle band pass receiving the greatest number of adjustments overall. All three filters satisfied the cutoff/bandwidth requirements after several revisions, with their graphs displayed below.

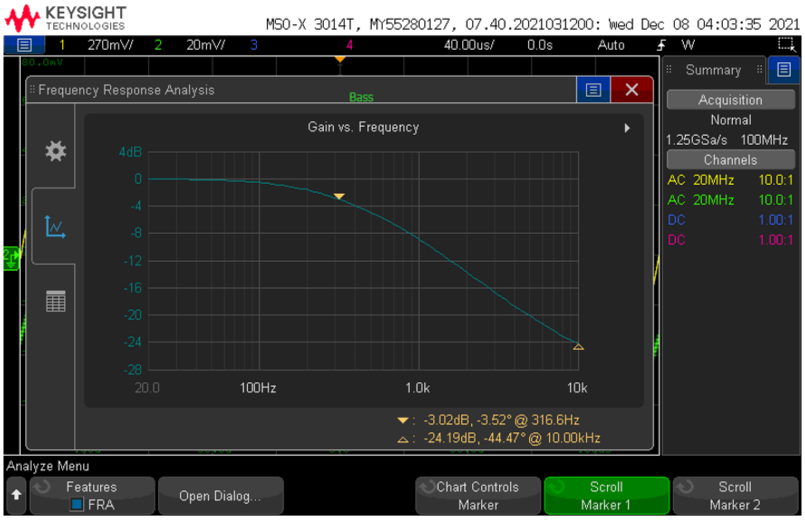

Figure 7: Bass filter.

Figure 8: Mid filter.

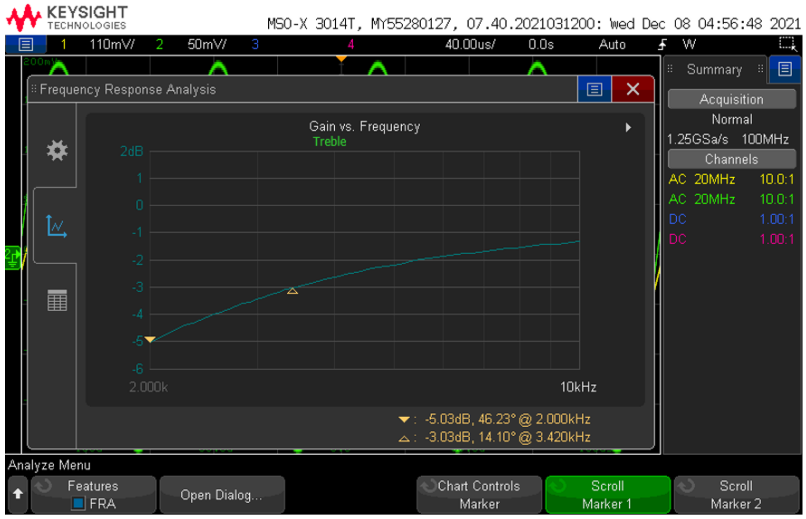

Figure 9: Treble filter.

Controllable Independent Gains

Potentiometers were utilized in each frequency branch as well as the with two of the pins shorted to enable observers to adjust the resistance values anywhere from 0 to 10 kiloohms. These equalizer knobs allowed for testing at both minimum and maximum settings at a variety of frequencies. The plots for each test is shown below.

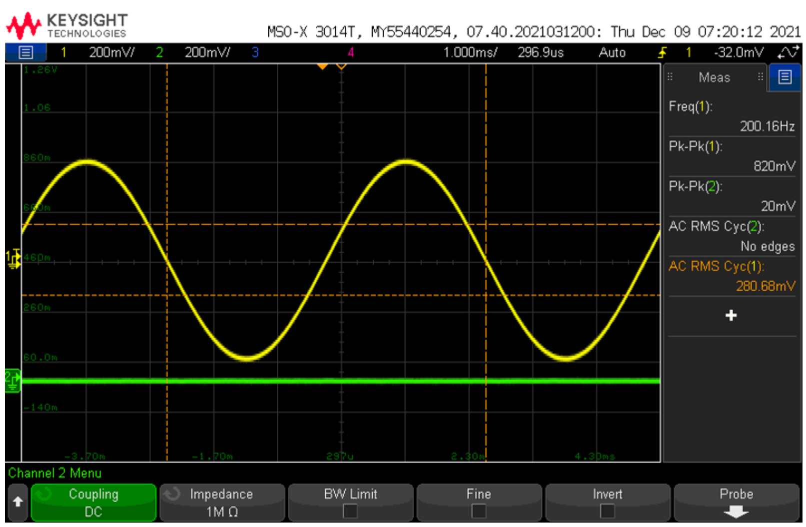

Figure 10: 200 Hz minimum.

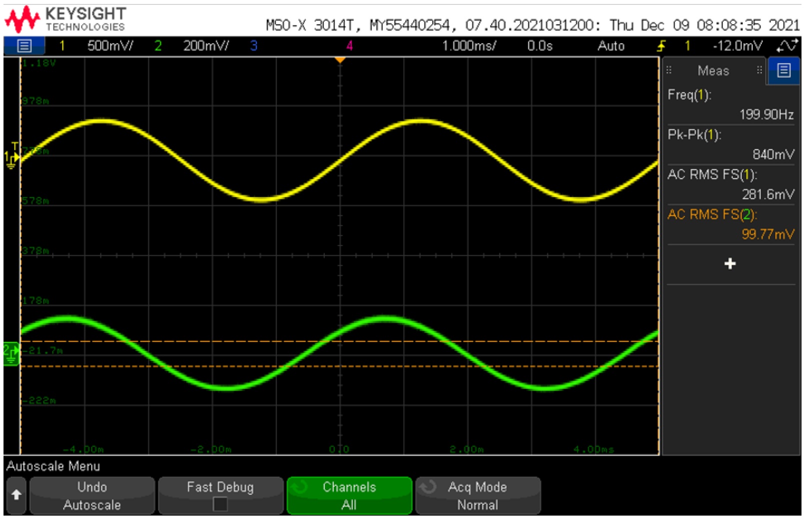

Figure 11: 200 Hz maximum.

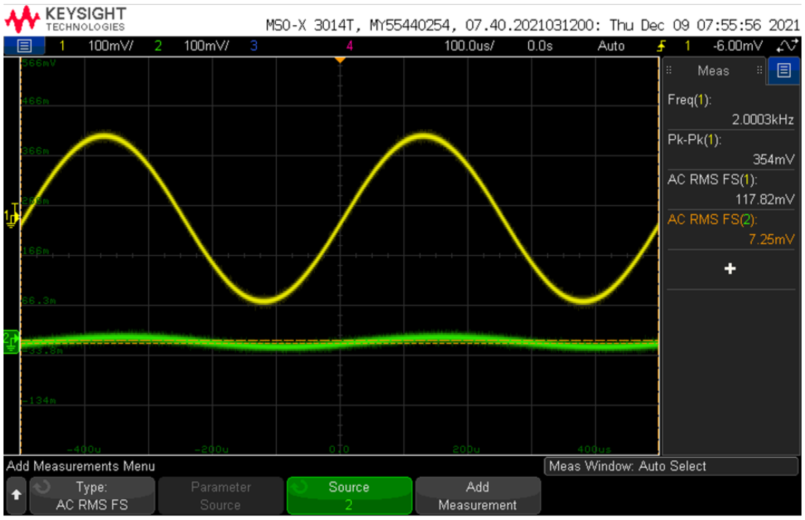

Figure 12: 2 kHz minimum.

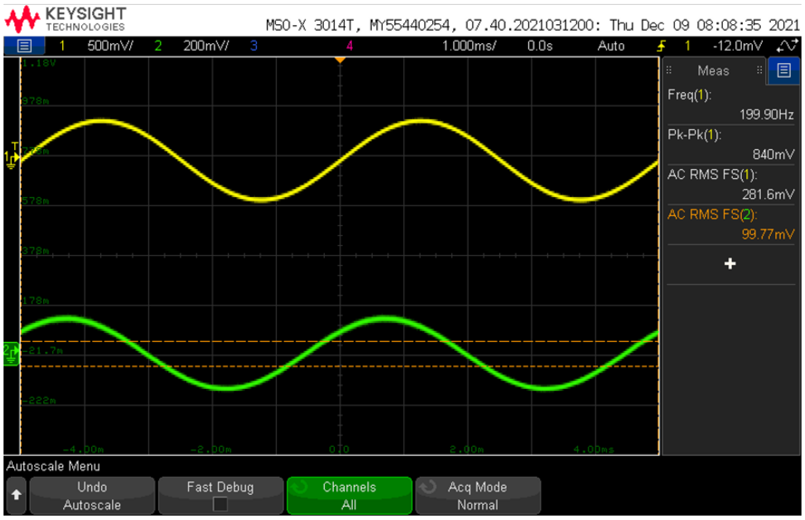

Figure 13: 2 kHz maximum.

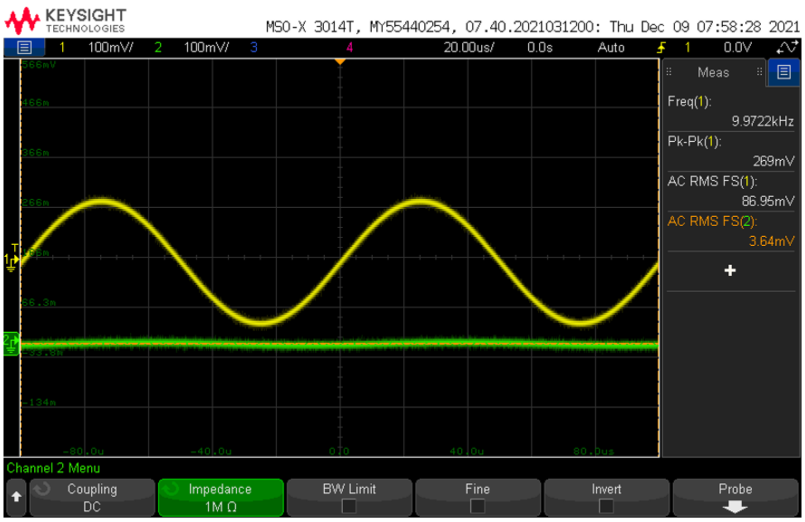

Figure 14: 10 kHz minimum.

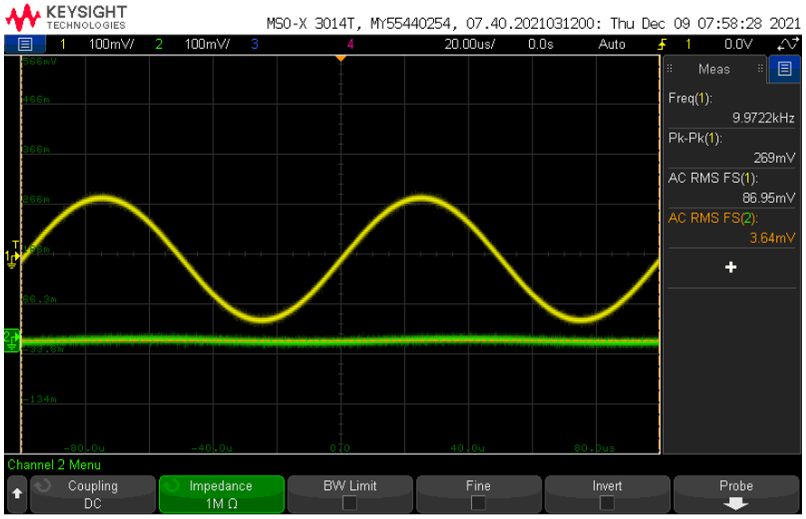

Figure 15: 10 kHz maximum.

Adding Signals and Controlling Volume

After all the frequencies had passed through their respective bands, they were all recombined with an op amp which then led into another op amp and potentiometer subsection which allowed for direct control over the volume of the sound played through the speaker. Equation 3 was used in order to calculate the amplifier output power, where our power amplifier sat at 20. The max ripple was calculated by finding the difference between the maximum and minimum values of the v_{amp}, which fell under $15mV_{RMS}$. Below are the graphs that demonstrate the adder and the power amplifier, as well as a demonstration of volume adjusting (the computer in-lab would not work and photos were taken of the screen instead).

Figure 16: Adder Demonstration.

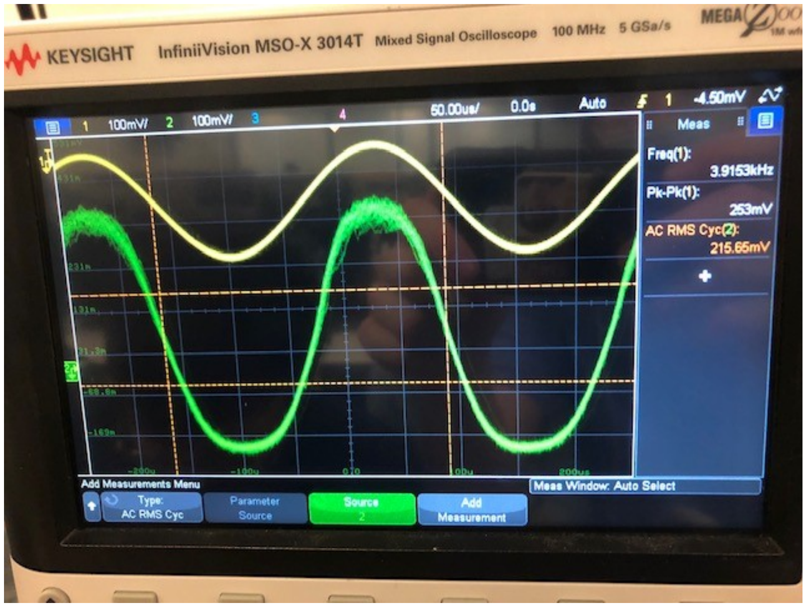

Figure 17: Power Amplification Demonstration.

Figure 18: Volume Demonstration.

Conclusion

Our goals were to create a simple audio equalizer that matched a list of specifications. Additionally, measurements and plots were made in order to ensure that the circuit worked on a theoretical level. However, there were complications with creating each subsystem, especially when it came to the value of resistors used within the circuit. Additionally, we had made many unneeded adjustments to the circuit during its integration by replacing components that we believed were broken due to not acquiring desired results but likely weren't. The implementation of the volume control, on the other hand, went very smoothly. The objectives were met because all three tasks were accomplished and all the criteria was met during review.