Basic Audio Equalizer Circuit

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Abstract

This experiment aims to design and construct an audio equalizer circuit that is capable of receiving an audio signal as input from an outside source and passing the signal through to a speaker while adjusting the levels of low, medium, and high frequency signal in the output signal, as well as the output volume. This was accomplished through a four total step process that first split the signals by low, medium, and high frequency, adjusted the levels of each through an amplifier, recombined the signals and adjusted the output amplitude, and finally boosted the power of the output signal such that it could drive a speaker.

Each of these steps had a specified success range that is available in Section 1 of this report, and the circuit after construction was measured using an oscilloscope and frequency response analysis tools to determine whether it met the desired specifications. Adjustments needed to be made to the calculated values of components in the circuit, as some resistances were changed by greater than 40% to achieve the desired result, but the overall circuit design did not need to be changed the circuit to succeed in all metrics.

The resulting circuit was able to effectively power a speaker with its output, and while the output audio was noisy, it was still clearly understandable and furthermore could easily be adjusted by the equalizer circuit to vary the sound characteristics. It can be concluded that the equalizer circuit design created in this report is effective at meeting all of the desired metrics and in its functionality as an equalizer.

1. Objectives

1.1. Audio signal filter circuit to split singular input into low, medium, and high frequency signals

Inputting an alternating signal or audio output signal, use passive filtering circuits to isolate the low signals below 320 Hz, mid signals between 320 Hz and 3200 Hz, and high signals above 3200 Hz. All filters must fit the desired cutoff between $\pm$10%

1.2. Signal equalizing amplifier circuit to adjust audio levels

Control the amplitude levels of the low, mid, and high signal ranges to simulate and audio equalizer and control the passthrough voltage for later parts of the circuit.

1.3. Variable amplifier for signal summation and volume control

Combine the isolated low, mid, and high signals together to form an equalized signal that can be output as an audio signal, and control the amplitude of this output to adjust the volume of the total circuit output. The summation of the equalized signals should have a maximum amplitude of V_{RMS} = 100mV and a minimum amplitude below V_{RMS} = 15mV.

1.4. Power amplification circuit

Increase the power of the output signal in order to effectively be used in a physical speaker, P > 400mW in range 200Hz to 10kHz

2. Theory

2.1. Signal filtering circuits work on the fundamental principle that the resistance to flow, or impedance, of an inductor or in this case a capacitor depends on the frequency of the signal that is entering it. The impedance of a capacitor is given by the equation

$$Z_C = \frac{1}{j2\pi fC},$$

which shows that as the frequency of a signal through a capacitor increases, the resistance to flow through that capacitor decreases. This causes the capacitor to act as a block to low frequencies, which enables it to be used to filter which frequencies can pass through a circuit component and which cannot. The primary way to use this feature of capacitors is through voltage division, in which the dividing factor changes as the frequency does.

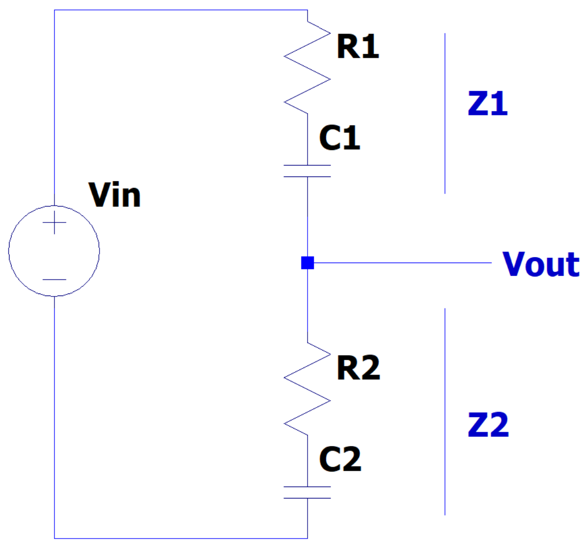

Figure 2.1: Basic voltage division filtering circuit

A basic filter circuit can be seen in figure 2.1, although in the case of this design Z_1 and Z_2 will consist of only a single resistor or a single capacitor. In order to analyze the output of a passive filter, the transfer function of the filter must be analyzed, which is given by the formula:

$$H(f) = \frac{V_{out}}{V_{in}} = \frac{1}{1+j2\pi fRC} \rightarrow |H(f)| = \frac{1}{\sqrt{1+(2\pi fRC)^2}}$$

The cutoff point for a passive filter is the point at which the amplitude of the output has decreased by a specified value measured in decibels(dB), which for this designed circuit is set to be -3.0dB, the point where the output signal is roughly half the power of the input signal. The power gain in decibels is given by the formula:

$$G = 20log(|H(f)|) = 20log(\frac{1}{\sqrt{1+(2\pi fRC)^2}})$$

The order of the resistor and capacitor in the filter circuit determines whether the filter rejects frequencies above or below the cutoff frequency. When the capacitor is at the location Z_1 in Figure 2.1, the filter rejects lower frequencies because the large impedance of the capacitor uses the majority of the input voltage up and what is left mostly goes to ground through the resistor. High frequencies are passed through as the capacitor has a low impedance to the voltage. If the capacitor is instead located at Z_2 in Figure 2.1, high frequencies cause low impedance in the capacitor, shorting the signal to ground instead of allowing it to continue within the circuit.

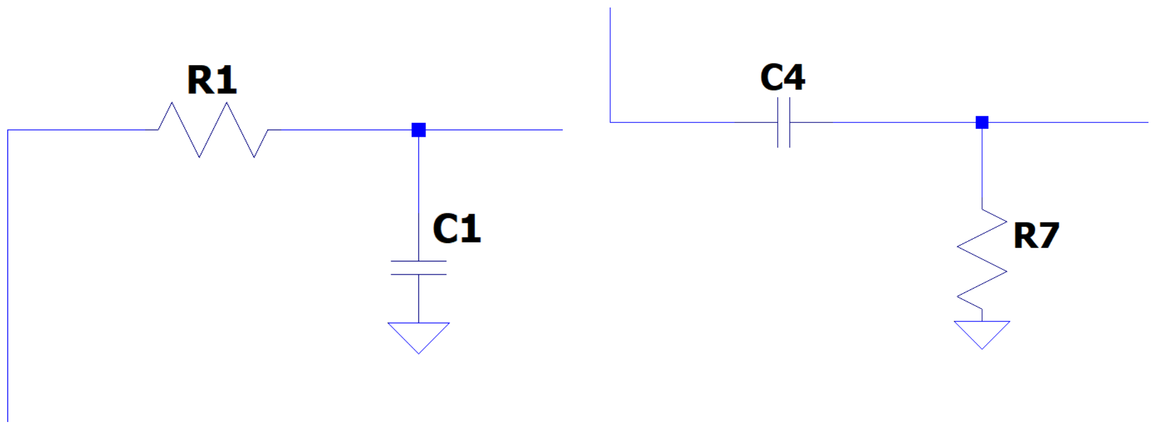

Figure 2.2: Low and High Pass filtering circuits

The final element of the filters needed is for the mid pass function of the circuit, which requires a filter that passes only a specific range of frequencies while rejecting frequencies outside that range, called a band pass. This can be done by combining the low and high pass filters in series, effectively filtering one range and then another, however this may result in additional voltage division occurring in the circuit as the resistors are of comparable magnitude. In order to prevent this, something must prevent the loading effects of the second filter onto the first, and one way to do this is through an operational amplifier. The amplifier is designed to pass the input signal in and output the same signal with a set amount of gain, and prevent loading effects. Gain is not required here, so it is set to zero, creating a band pass circuit with the following schematic:

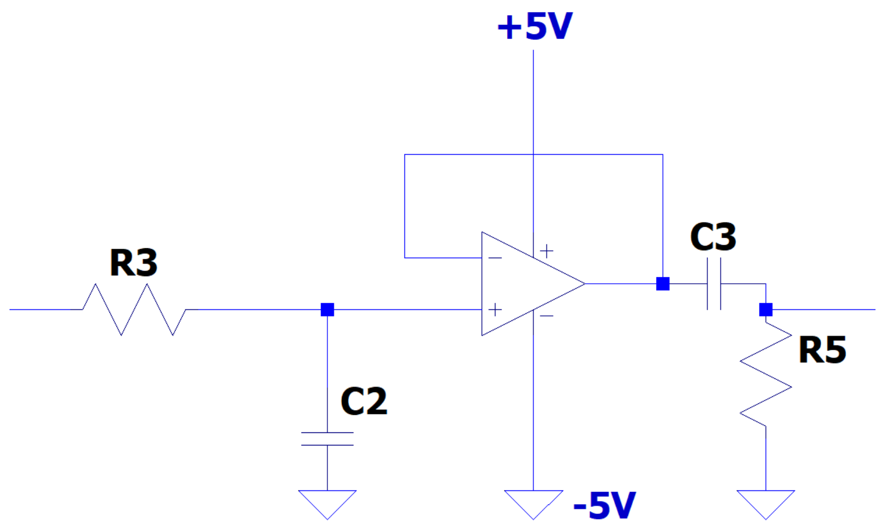

Figure 2.3: Band pass filter with operation amplifier, input voltage on left and output voltage on right

Of note, the operational amplifier requires a separate input voltage of $\pm$5V, and will clip any part of a signal that is outside of this range. All amplifiers of this type require a coupling capacitor at the point of connection between the amplifier and the $\pm$5V signal of 0.1μF to prevent excessive current from flowing. These capacitors will be required for all amplifiers discussed within this report. Additionally, the real values of the resistors and capacitors used may not be the same as the stated value due to error, so adjustments to the values of the resistance and capacitance of a passive filter are required to achieve the desired result.

2.2. The goal of the equalization circuit is to output a percentage of the input signal so that the individual levels of low, mid, and high frequencies can be adjusted in the final output signal. In this circuit an operational amplifier with a variable gain resistance was used so that the levels could be changed easily. Potentiometers are the simplest means to vary the resistance, so they were utilized as the gain resistance. The general amplifier circuit design is shown below:

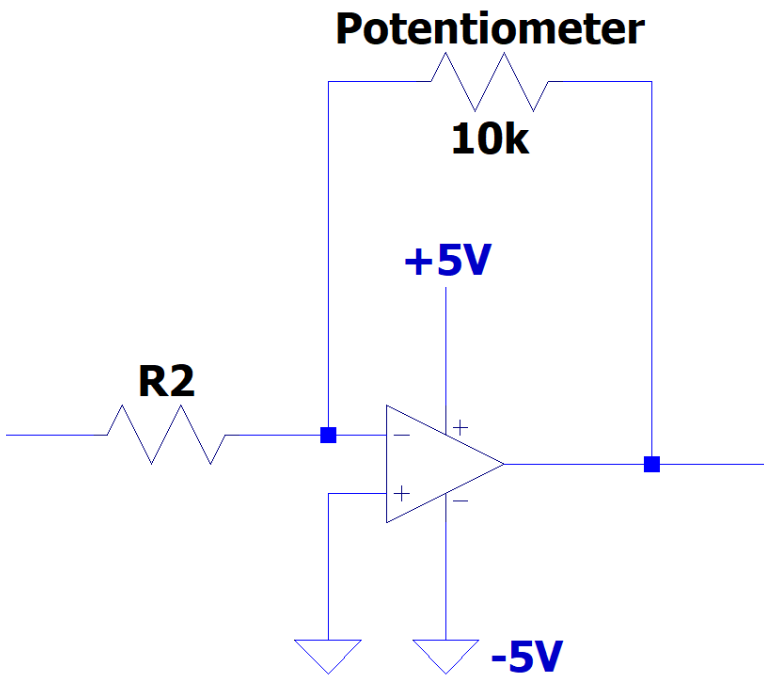

Figure 2.4: Variable gain amplifier circuit

Gain of the operational amplifier is given by $G = \frac{R_G}{R_{in}}$, where R_G is the resistance of the potentiometer and R_{in} is the resistance into the amplifier, shown in Figure 2.4 as R2. The equalizer circuit will be in series with the outputs of the passive filter circuits, so voltage loading may occur. In order to prevent loading effects for this section of the circuit, R_{in} must have a significantly higher value than the resistors of the filters, higher to one or more orders of magnitude. Finally, the operational amplifier requires a separate input voltage of $\pm$5V, and will clip any part of a signal that is outside of this range, just as in the band pass usage.

2.3. Up to the point of the summation and volume control amplifier, the circuit appears as in Figure 2.5. Each of the low, mid, and high signals has an individual signal output that now must be combined together to create a singular output signal with a level that can also be controlled.

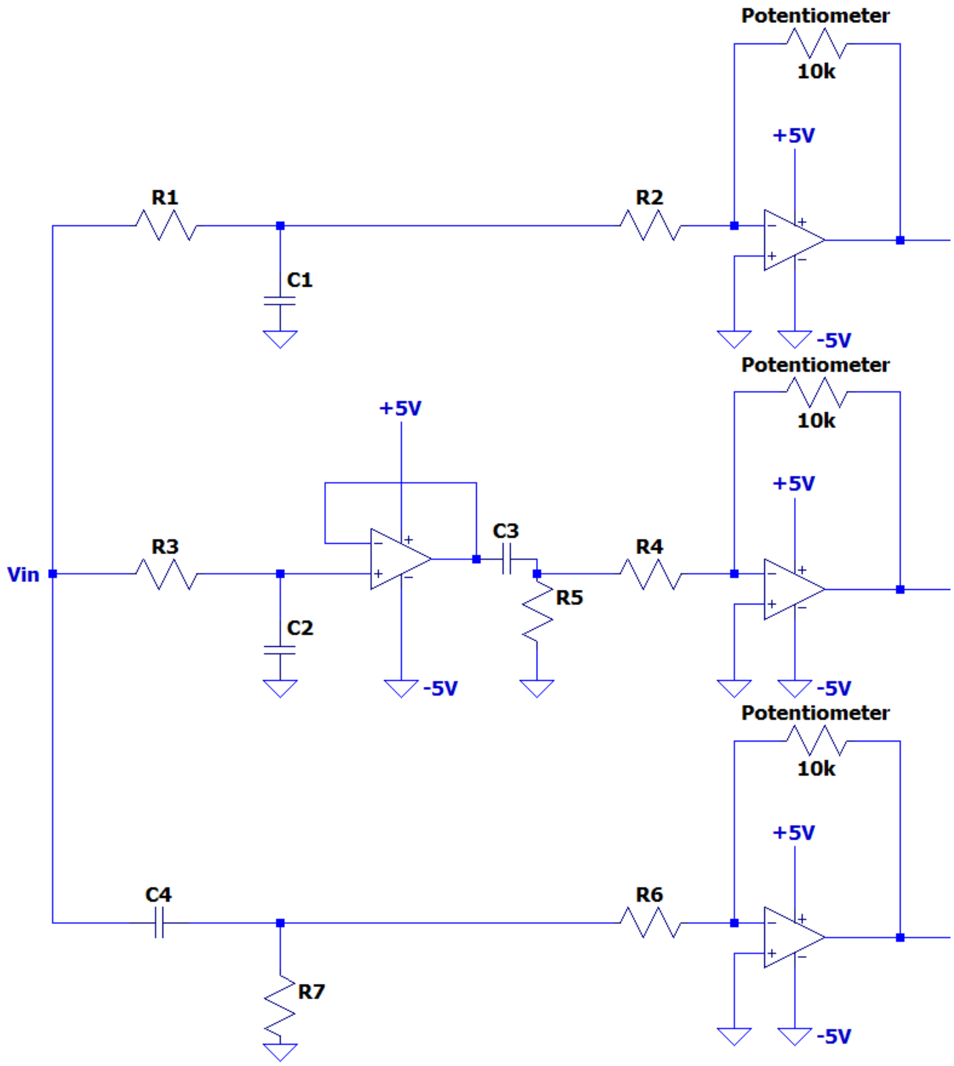

Figure 2.5: Filter and equalization section of the total audio amplifier circuit

One more operational amplifier can be used to do this, inputting all three signals and combining them into an identical amplifier circuit as shown in Figure 2.4. The R_{in} value requires three separate resistors, one for each section of the filter pass, and these each can be treated as an individual pass through the amplifier. The gain is again given by $G = \frac{R_G}{R_{in}}$ where R_G can vary by means of a potentiometer. The total output of the filter, equalizer, and summation and volume circuits must be 100 V_{RMS} with 10% error when a $1V_{PP}$ signal is input, and this output can be modeled by the combination of the gain from the equalizer amplifiers and the volume amplifier circuit. This gives the formula:

$$V_{out} = V_{in} \cdot \frac{R_{G1}}{R_{in,1}} \cdot \frac{R_{G2}}{R_{in,2}},$$

which must be converted to RMS voltage, giving the equation:

$$V_{out,RMS} = V_{in,PP} \cdot \frac{10k\Omega}{R_1} \cdot \frac{10k\Omega}{R_2} \cdot \frac{1}{\sqrt{2}}$$

This equation can be used to set the input resistance values for the individual equalizing circuits and the overall volume control amplifier circuit, where the $\frac{1}{\sqrt{2}}$ converts the output voltage from V_{PP} to V_{RMS}.

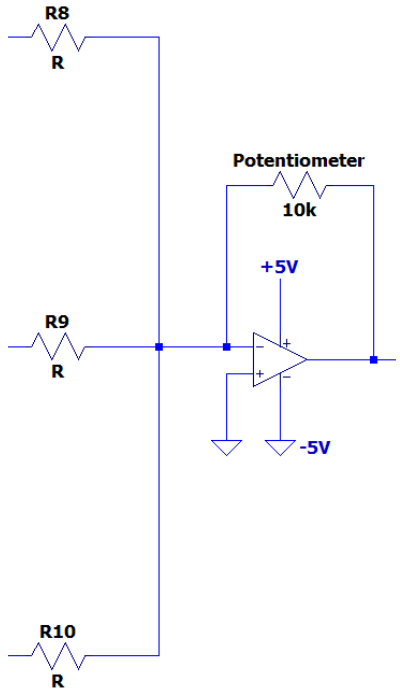

Figure 2.6: Input into summation and volume control amplifier, where R8, R9, and R10 are the R_{in} resistors

The operational amplifier in this circuit also requires a power supply of $\pm$5V, and all waves with a peak outside of this range will be clipped.

2.4. The power amplifier circuit is designed to take the low amplitude input audio signal after it has been equalized through the beginning of the circuit and increase the power it delivers such that it can effectively drive a speaker. The determining factor of the output power of the amplifier is the output voltage, which depends on the impedance characteristics and the supplied voltage. The impedance characteristics have been specified (see Figure 2.8 for values and schematic), so the variable parameter of this amplifier is the supply voltage. Power is given by $P = \frac{V_{out}^{2}}{R}$, and the resistance value of the speaker used is known to be 8Ω. Figure 2.7 shows the voltage output characteristics of the amplifier for different supply voltages and load resistances, which can then be used to calculate the desired supply voltage.

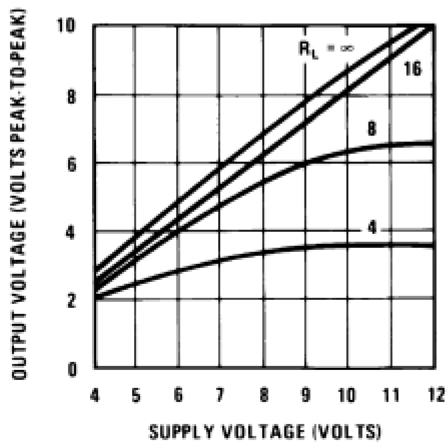

Figure 2.7: Voltage output characteristics of LM386N-4 audio amplifier for given resistance values and supply voltages

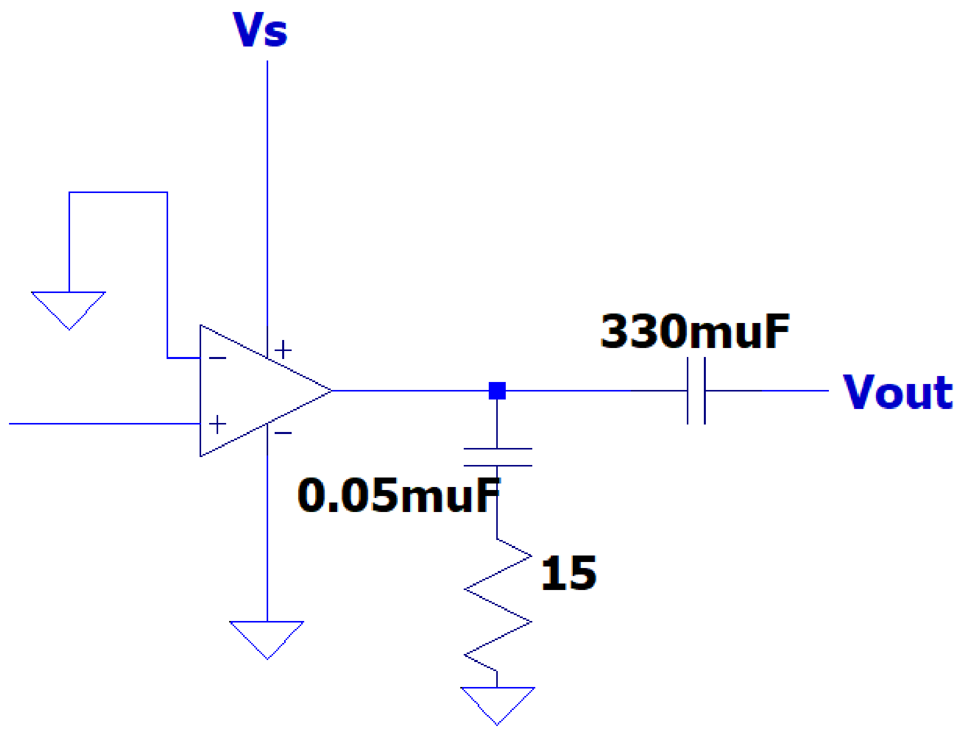

Figure 2.8: Circuit schematic of audio amplifier circuit with specified resistance and capacitance values, where Vs is supply voltage

3. Procedure

1. Values for all components of the circuit were calculated to provide the desired output parameters, listed in Table 3.1

Table 3.1: List of circuit parameters

Component	Value
R1	49.61 Ω
R2	10 kΩ
R3	49.61 Ω
R4	10 kΩ
R5	49.61 Ω
R6	10 kΩ
R7	49.61 Ω
R8	70.7 kΩ
R9	70.7 kΩ
R10	70.7 kΩ
C1	10 μF
C2	1 μF
C3	10 μF
C4	1 μF
Vs	$\approx$5 V

2. The circuit in Figure 3.1 was constructed

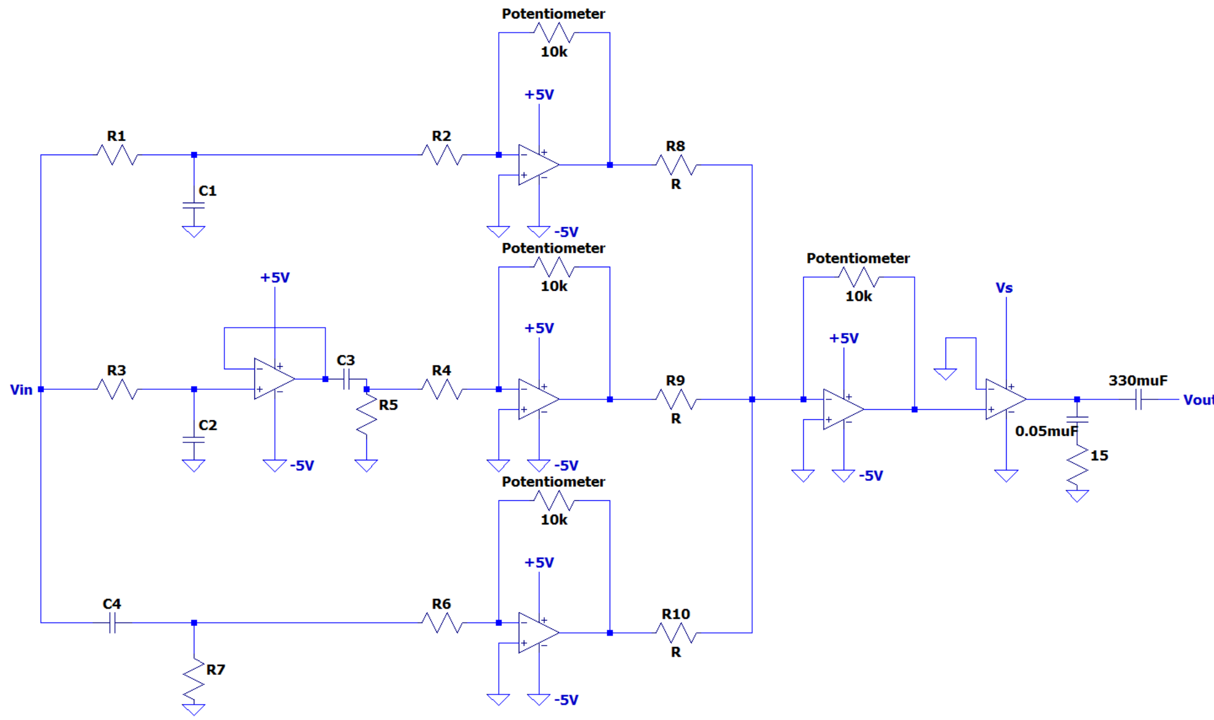

Figure 3.1: Audio equalizer circuit schematic (see Table 3.1 for specific values)

3. The filter characteristics of each filter were tested using the built-in frequency response analysis function of the oscilloscope, with a $1V_{PP}$ input wave ranging from 10Hz to 10kHz with 100 points per decade, and the -3dB cutoff point was recorded
    a. Low pass - The V_{in} probe was measured across at the input and ground, and the V_{out} probe was measured across C1
    b. Mid (Band) pass - The V_{in} probe was measured across at the input and ground, and the V_{out} probe was measured across R5
    c. High pass - The V_{in} probe was measured across at the input and ground, and the V_{out} probe was measured across R7
4. The filter resistance was then adjusted such that the measured value of the cutoff point was closer to the specified value.
5. Steps 3 and 4 were then repeated until the cutoff point is within the error bounds of the specified value
6. The minimum and maximum peak-to-peak voltages across the summation amplifier for 200Hz, 2kHz, and 10kHz were measured by inputting a $1V_{PP}$, setting all potentiometers to the minimum or maximum, and measuring the output waveform across the summing amplifier. The output V_{RMS} value was then calculated from the V_{PP} measurement
7. The ripple was measured by doing a frequency response analysis across the audio amplifier and recording the voltage difference between the frequencies with the highest and lowest decibel levels
8. The peak-to-peak voltage and RMS voltage across the speaker while playing music was recorded in order to calculate the power output of the entire audio equalizer circuit.

4. Results

4.1 Results

Table 4.1: Adjusted values of resistance and capacitance based on actual frequency response measurements

Component	Value
R1	48 Ω
R2	10 kΩ
R3	66 Ω
R4	10 kΩ
R5	60 Ω
R6	10 kΩ
R7	70 Ω
R8	70.2 kΩ
R9	70.2 kΩ
R10	70.2 kΩ
C1	10 μF
C2	1 μF
C3	10 μF
C4	1 μF
Vs	8 V

As seen when comparing Table 4.1 to Table 3.1, the values of resistance in the filtering circuits were changed such that the cutoff point would be within the appropriate bounds. The capacitance and the resistance in all other parts of the circuits remained unchanged, except for R8, R9, and R10 which were approximated by available resistors due to no specific 70.7kΩ resistor.

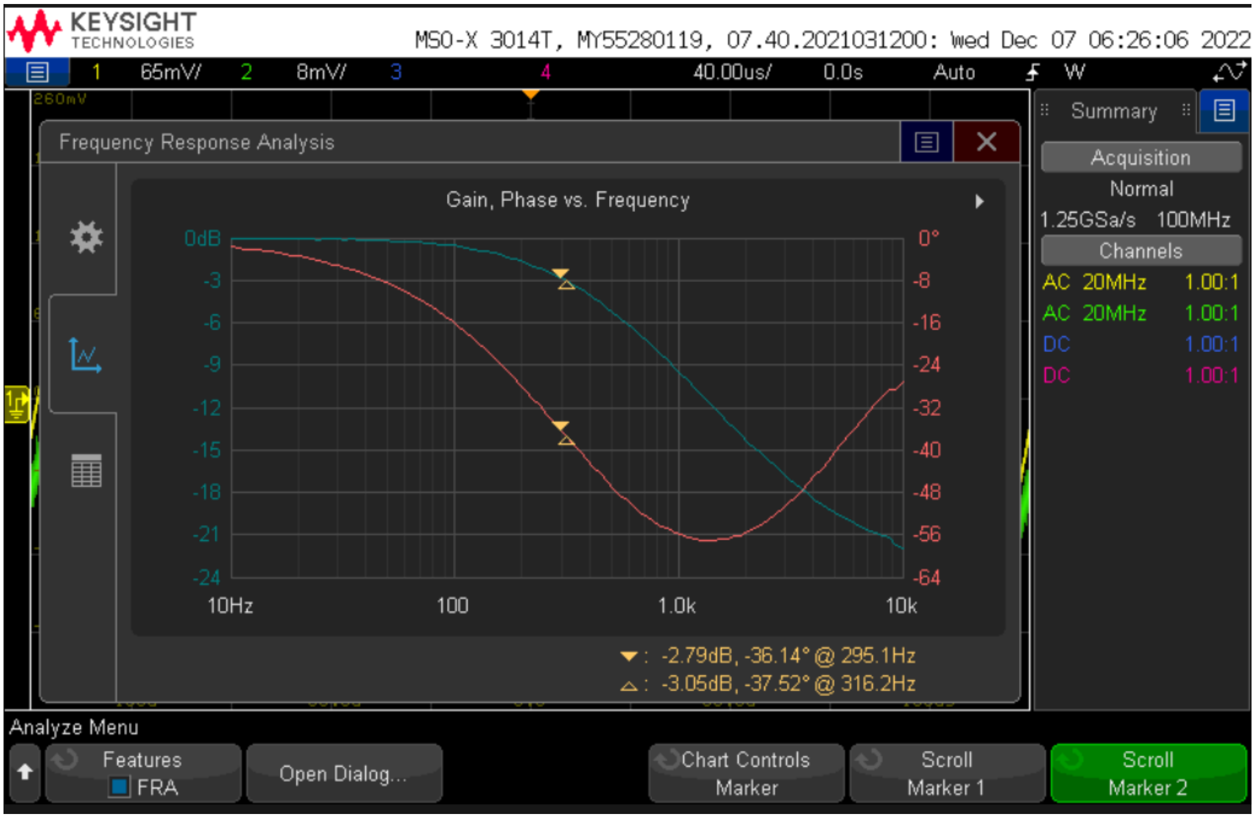

Figure 4.1: Low pass filter frequency response

The frequency response of the low pass filter shows that the -3dB cutoff point is within the range of 295.1 Hz and 316.2 Hz. This is within the accepted range of $320Hz - 10\% = 288Hz$. The filter resistance needed to be decreased slightly to 48Ω to achieve a cutoff point within this range.

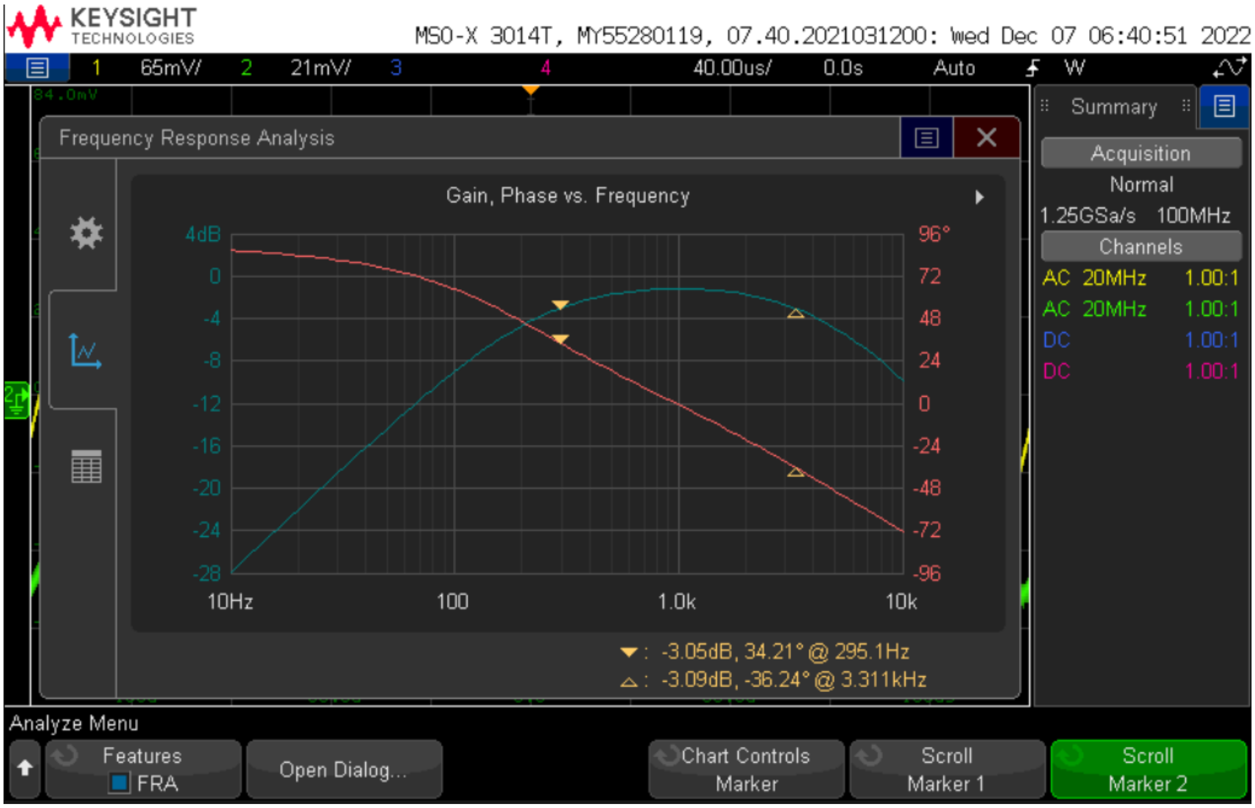

Figure 4.2: Mid (Band) pass filter frequency response

The frequency response analysis of the mid pass filter shows that the low frequency -3dB cutoff point occurred at ≈ 295 Hz which is within the accepted range of $320Hz - 10\% = 288Hz$. The high frequency cutoff occurred at ≈ 3.31 kHz, which is within the accepted range of $3200Hz + 10\% = 3250Hz$. The low pass filter resistance in the mid pass filter needed to be increased to 66Ω and the high pass filter resistance needed to be increased to 60Ω in order to achieve cutoff points within the specified ranges. This increased resistance likely resulted from effects caused by the amplifier circuit separating the two filter sections, possibly loading or some effect on the actual frequency of the signal being passed through.

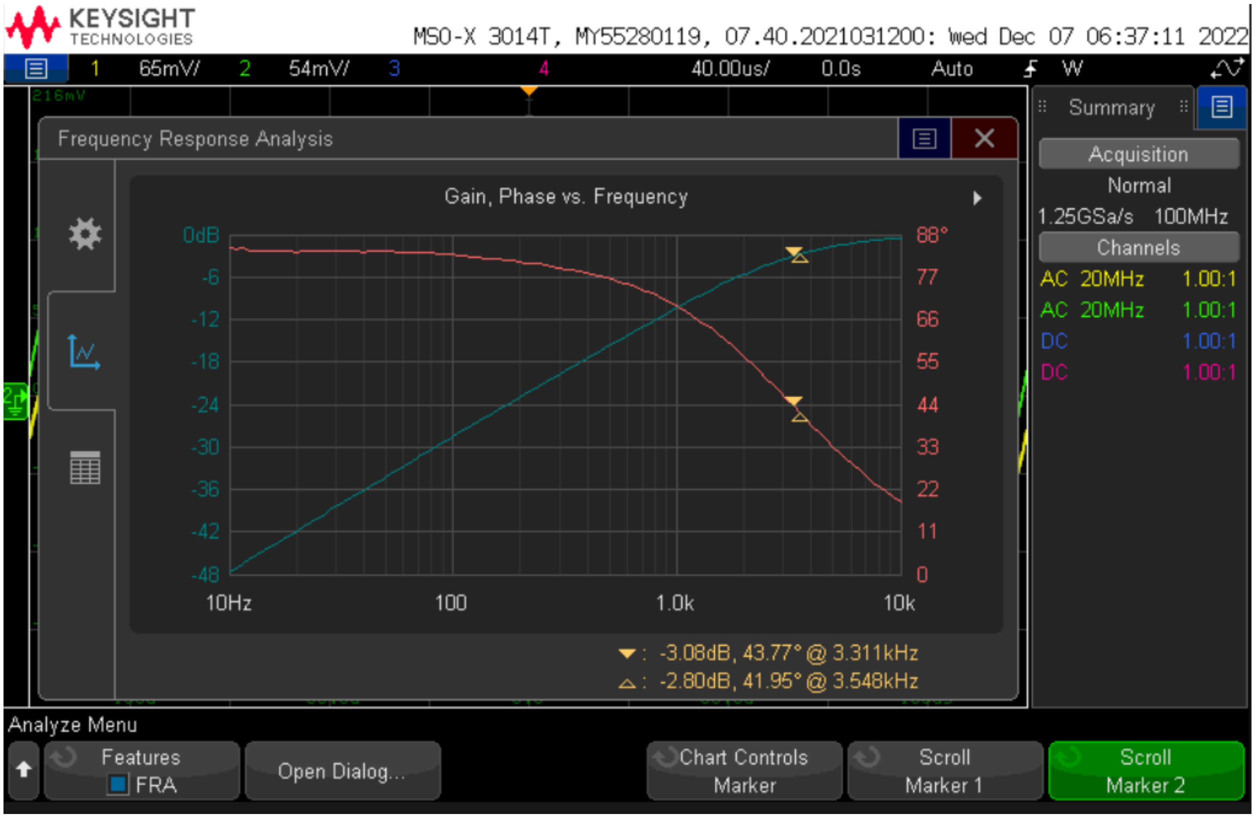

Figure 4.3: High pass filter frequency response

The frequency response of the low pass filter shows that the -3dB cutoff point is within the range of 3.311 kHz and 3.548 kHz, and is much closer to the 3.311kHz point. This is within the accepted range of $3200Hz \pm 10\% = [2.88kHz, 3.520\ kHz]$. The filter resistance needed to be increased to 70Ω to achieve a cutoff point within this range. There does not seem to be any reason for this increase other than potentially a capacitor with a real capacitance much different than the expected capacitance. A more precise and accurate capacitor would likely result in a filter resistance much closer to the calculated value.

4.2 Results

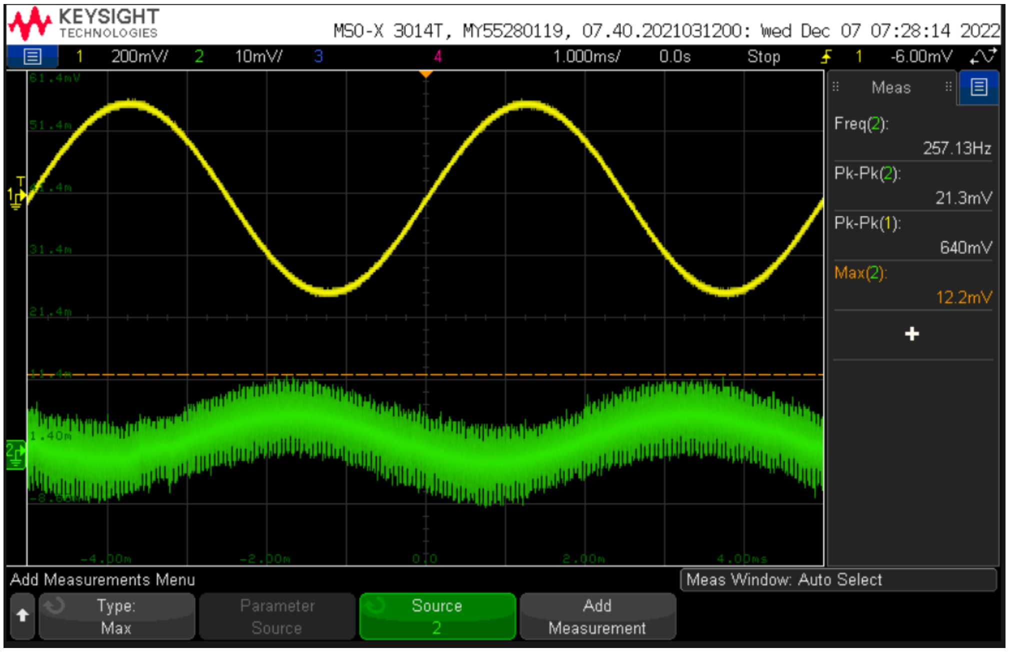

Figure 4.4: Output waveform of combined signals across the summation amplifier at minimum gain, where V_{in} = 1V_{PP} and f = 200Hz

The RMS voltage of the output waveform is given by $V_{RMS,out} = \frac{V_{peak}}{\sqrt{2}} = \frac{12.2mV}{\sqrt{2}} = 8.63\ mV_{RMS}$, which is lower than the required minimum of 15 mV_{RMS}.

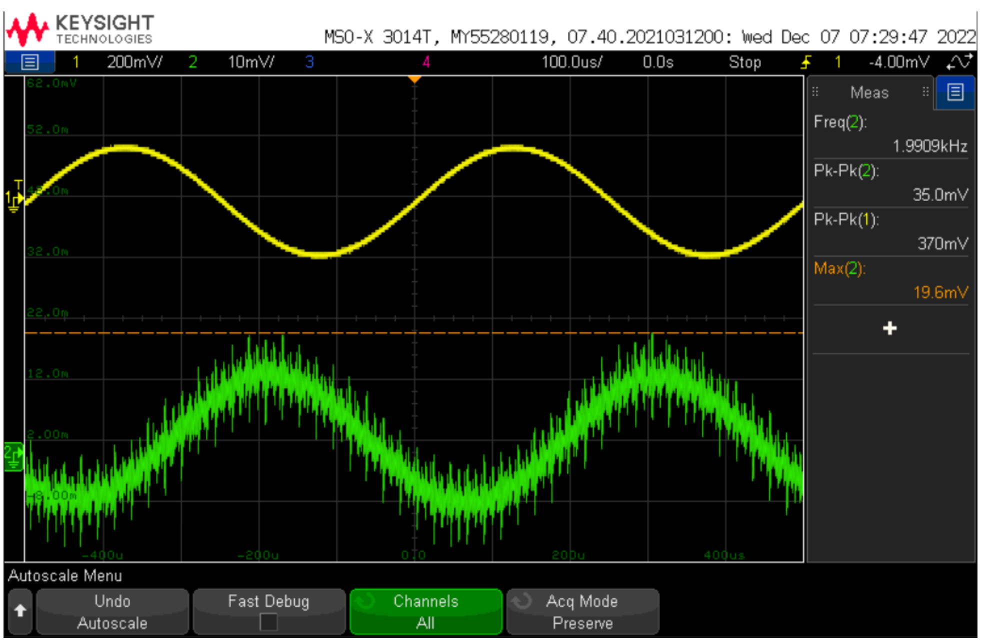

Figure 4.5: Output waveform of combined signals across the summation amplifier at minimum gain, where V_{in} = 1V_{PP} and f = 2000Hz

The RMS voltage of the output waveform is given by $V_{RMS,out} = \frac{V_{peak}}{\sqrt{2}} = \frac{19.6mV}{\sqrt{2}} = 13.86\ mV_{RMS}$, which is lower than the required minimum of 15 mV_{RMS}.

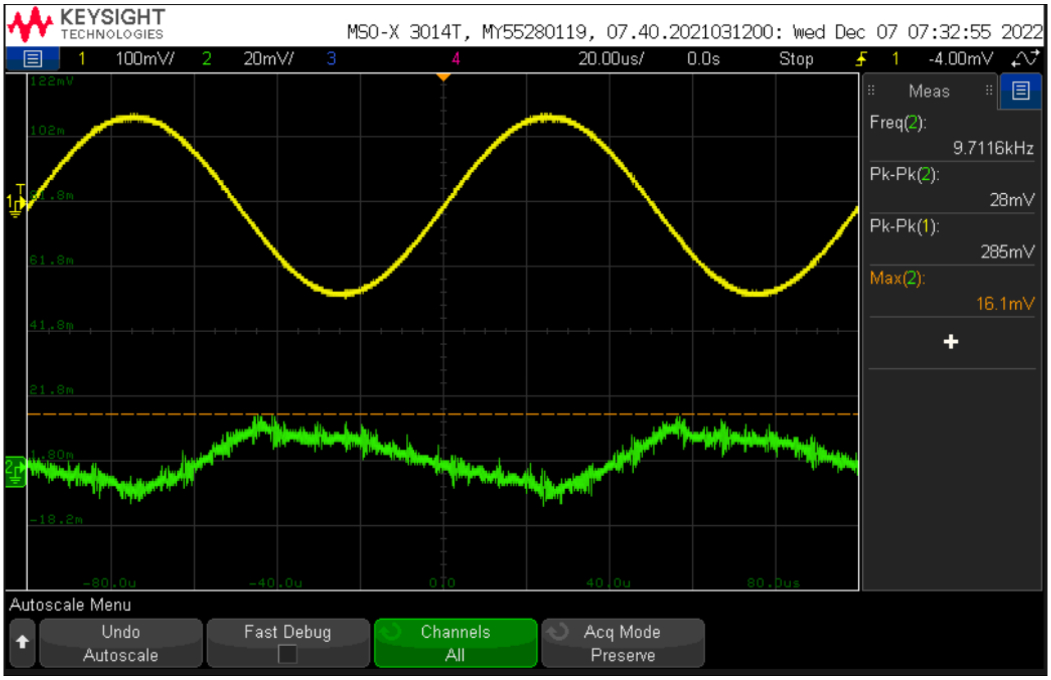

Figure 4.6: Output waveform of combined signals across the summation amplifier at minimum gain, where V_{in} = 1V_{PP} and f = 10kHz

The RMS voltage of the output waveform is given by $V_{RMS,out} = \frac{V_{peak}}{\sqrt{2}} = \frac{16.1mV}{\sqrt{2}} = 11.38\ mV_{RMS}$, which is lower than the required minimum of 15 mV_{RMS}.

While all output signals from the summation amplifier have some noise levels, the overall signal still resembles the input waveform. This noise is likely caused by small amounts of pass through from the undesired filters and general noise that results in a real circuit with this many components situated on a breadboard.

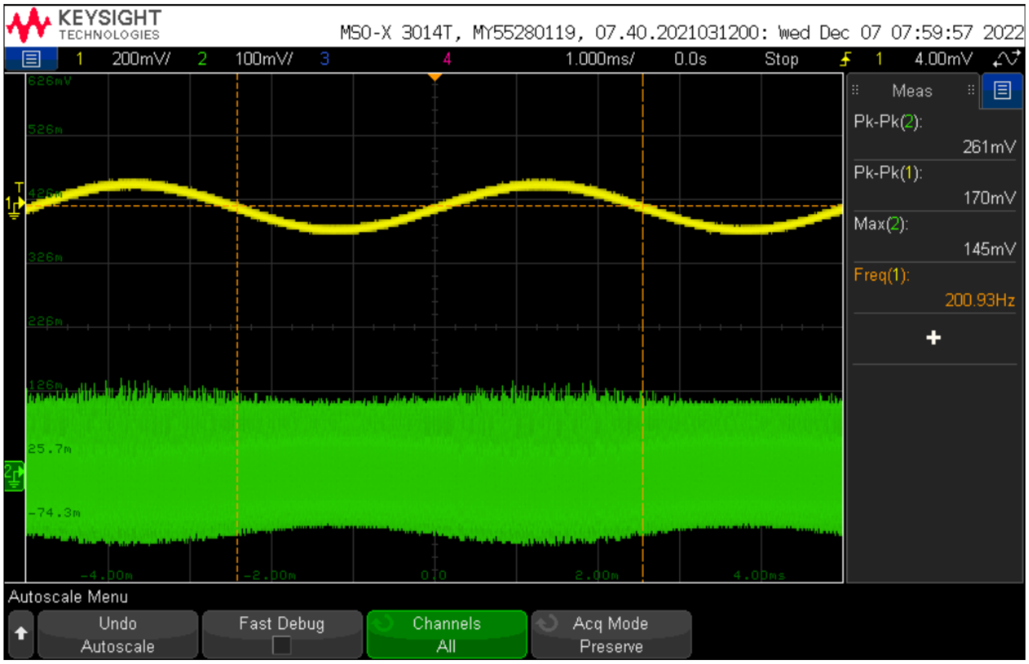

Figure 4.7: Output waveform of combined signals across the summation amplifier at maximum gain, where $V_{in} = 1V_{PP}$ and f = 200Hz

The RMS voltage of the output waveform is given by $V_{RMS,out} = \frac{V_{peak}}{\sqrt{2}} = \frac{145mV}{\sqrt{2}} = 102.53\ mV_{RMS}$, which is within the maximum range of 100mV $\pm$ 10%.

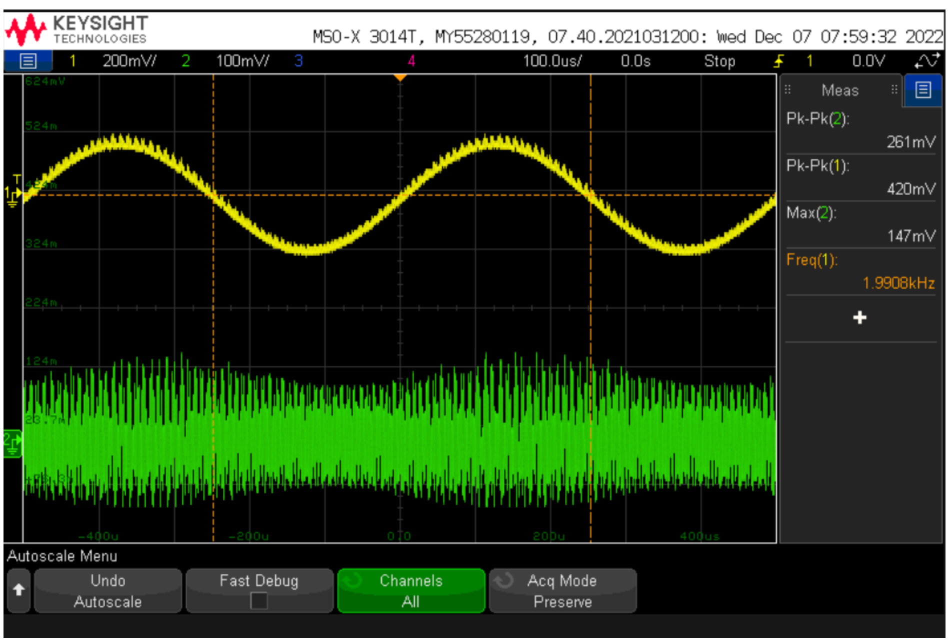

Figure 4.7: Output waveform of combined signals across the summation amplifier at maximum gain, where $V_{in} = 1V_{PP}$ and f = 2000Hz

The RMS voltage of the output waveform is given by $V_{RMS,out} = \frac{V_{peak}}{\sqrt{2}} = \frac{147mV}{\sqrt{2}} = 103.94\ mV_{RMS}$, which is within the maximum range of 100mV $\pm$ 10%.

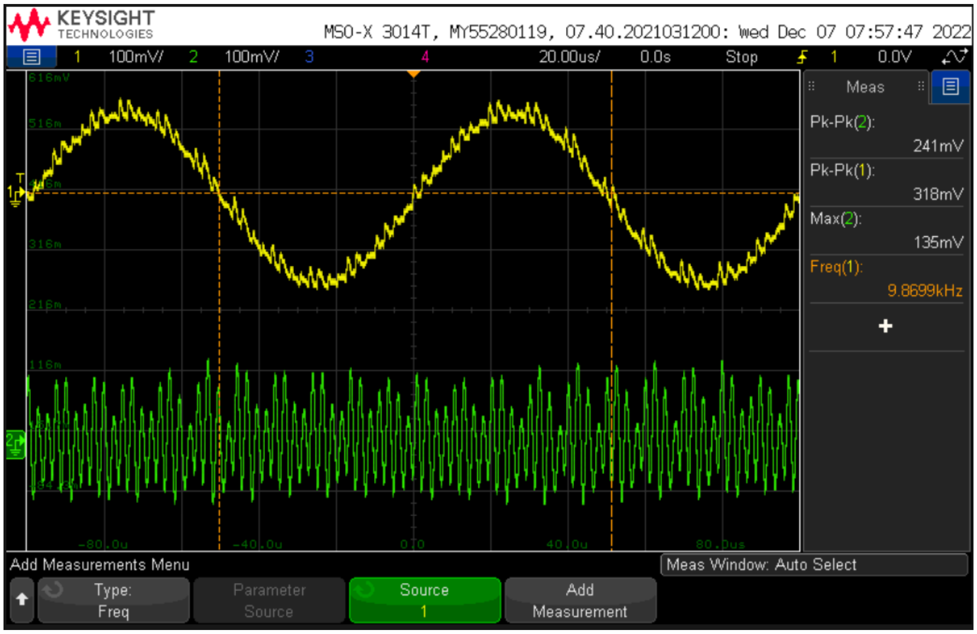

Figure 4.7: Output waveform of combined signals across the summation amplifier at maximum gain, where V_{in} = $1V_{PP}$ and f = 10kHz

The RMS voltage of the output waveform is given by $V_{RMS,out} = \frac{V_{peak}}{\sqrt{2}} = \frac{135mV}{\sqrt{2}} = 95.46\ mV_{RMS}$, which is within the maximum range of 100mV $\pm$ 10%.

The output signals from the summation amplifier have significantly more noise when on maximum gain from all the equalizers and the volume control than when everything is set to minimum. The shape of the input waveform is still visible in each output, but only barely. This is likely caused again by more incidental passthrough from filters that do not completely reject all signals outside of their desired range adding onto the final signal. Additionally, the connections with the pins of the potentiometer were often liable to completely change the output if they were bumped or pressed in the wrong way. Making these connections more stable could also greatly decrease the noise through the summation and volume control amplifier.

4.3 Results

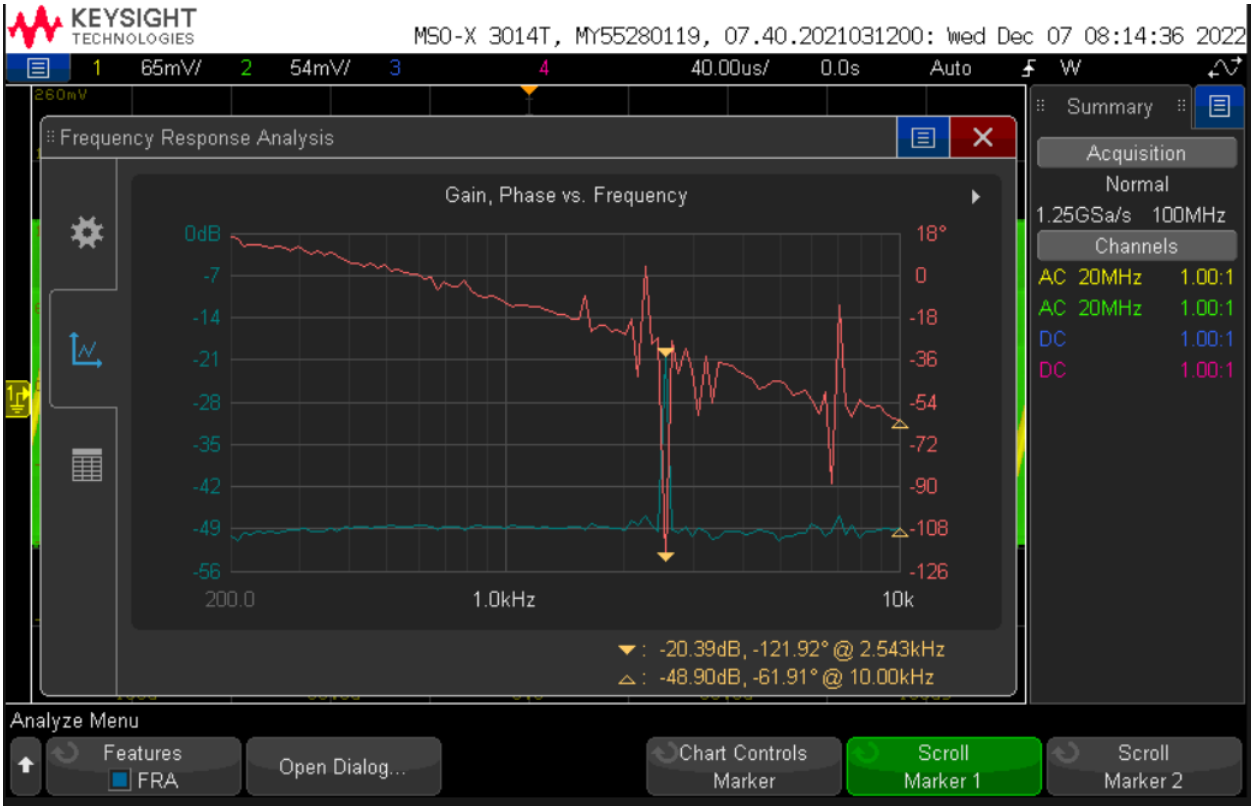

Figure 4.8: Frequency response analysis across the audio amplifier to measure ripple

Figure 4.8 shows the difference in amplitude across the audio power amplifier as the frequency increases when all gain is set to maximum. The amplitude of the output varies no more than roughly 4dB from minimum to maximum, excluding the singular peak. This variation shows that the difference in amplitude between the minimum and maximum power output frequencies is essentially zero, less than the 15mV$_{RMS}$ specification. The peak at 2.543 kHz is an anomaly that recurred each time the frequency response was tested, regardless of the variations to the equalization and volume that were attempted to fix it. This may be caused by oscilloscope error or possibly an unknown resonance property of the circuit, but it will require more testing to diagnose.

4.4 Results

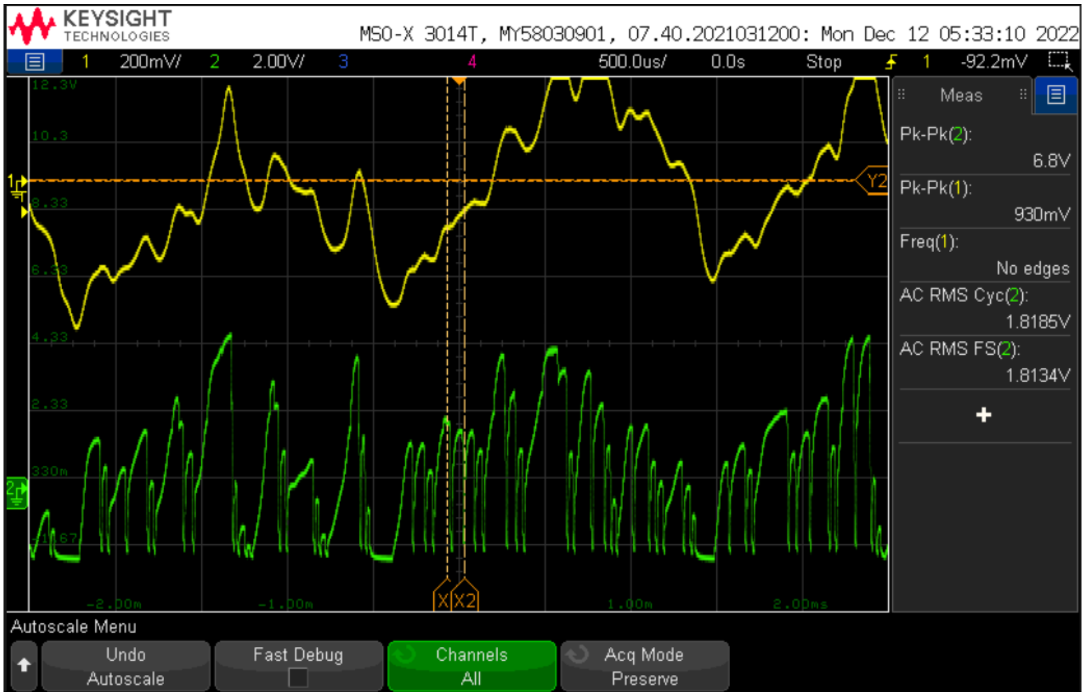

Figure 4.9: Output voltage across the speak when music is playing

As shown in Figure 4.9, the output voltage through the speaker is , which yields a power output of $P = \frac{V_{out}^{2}}{R} = \frac{(1.8185V_{RMS})^{2}}{8\Omega} = 0.413mW > 400mW$, however this only occurred when the input voltage into the audio amplifier was set at 8V, much higher than the calculated 5V value. This may have been the result of some unintentional voltage division or possibly from an input voltage drop across something other than a resistor. More testing is required to determine why the output did not behave as anticipated. Additionally, the output from the speaker was very noisy, especially in the low ranges, and decreased in clarity as the volume of the output grew louder. Noise occurred especially in the low range, so the overall clarity could be increased by using the equalizer functionality to decrease the passthrough of the low frequencies, resulting in an audibly clearer output.

5. Conclusion

Overall the audio equalizer circuit functioned as intended, an audio signal from a computer was passed completely through the circuit to the speaker output, which played audible sounds at reduced quality and clarity from that of the original. The filtering circuits needed to have resistances adjusted due to nonideal behavior from the capacitors and the circuit as a whole, which required constant upkeep as they seemed to change daily and did not seem to remain consistent on different measuring devices. Each circuit's passthrough cutoff fell within the acceptable range, although some were near the extremes of this range, which could have resulted in some of the noise at the output. Additionally, it is likely that frequencies at and around the 320Hz and 3200Hz cutoff points were likely significantly lower in the output signal due to the behavior of the filtering circuits, where the signal strength decreases a fair amount before reaching the actual cutoff point. In order to prevent this, filters with much steeper output drop offs would be needed.

The equalization and volume variable control worked as designed as well, the volume could audibly be adjusted, and the equalization effectively removed frequencies within the set ranges, which could be used to significantly reduce final output noise and increase clarity. These circuits functioned as designed in the ideal space and did not require any adjustments when they were built to meet the required specifications. While as mentioned earlier high quality potentiometers may have increased the quality of output through each of these amplifier circuits, as is the circuit performed to a higher degree than was expected. The final output of the equalization circuit was able to effectively drive the speaker due to the audio amplifier, which was able to both meet the expected power output and exceed it when the supply voltage to it was increased. The speaker output did include much audible noise, as expected from a non-ideal breadboard circuit, but the output still could be clearly understood by any listener, and the effects of the equalization and volume control could be heard in the sound output, which renders the goal of the circuit a success.

This circuit could be improved by lower tolerance parts or a PCB implementation of the circuit. However, the result of circuit testing shows that this audio equalizer circuit design effectively works to meet all specifications and can be implemented to work within the real world.