

Audio Equalizer

Introduction

Objective

This project focuses on building a Audio Equalizer, which will utilize the key knowledge as we learned during the class. This final implementation will examine how well students handle the course content.

Function

Range of frequency human is able to hear is from 20Hz to 10kHz, so audio equalizer, in this project, define $Frequency < 320\text{ Hz}$ to be Bass filter, $320\text{ Hz} < Frequency < 3200\text{ Hz}$ to be Mid filter, and $Frequency > 3200\text{ Hz}$ to be Treble filter. By implementing this, people can adjust the output ratio for each part, in order to adjust sound to a range people desired.

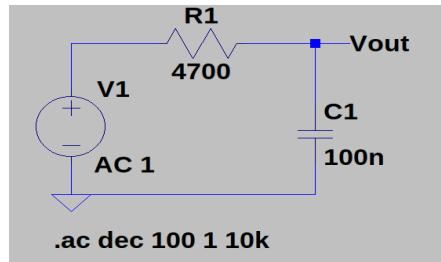
Application

Audio devices like headphones, speakers are able to use Audio Equalizer as a feature that allows customers to adjust and customize sound effect as they want. Noise Reduction can also be accomplished by implementing Audio Equalizer, like engine noise from the road ranges from 50Hz to 200Hz and fan noise ranges from 1kHz to 5kHz. We can adjust Audio equalizer accordingly to decrease the noise from a specific range of frequency.

Theory

Passive Filters

RC Low-pass Filter



(Picture 1)

Low-pass filter is a passive filter that allows frequency lower than cutoff to pass through with less than -3 dB, while frequency higher than cutoff will have gain greater than -3dB.

This is a voltage divider with impedance from both the resistor and capacitor.

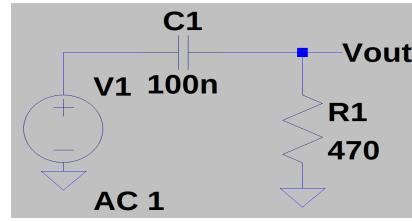
We have:

$$V_{out} = V_{inp} \times \frac{Z_c}{R + Z_c} , \quad Z_c = \frac{1}{j \omega C} , \text{ and} \quad \omega = 2\pi f_c$$

After Calculation, We have:

$$f_c = \frac{1}{2\pi RC}$$

RC High-pass Filter



(Picture 3)

High-pass filter only allows frequency higher than cutoff to pass through with gain higher than -3dB.

Similar to low-pass filter, this is also a voltage divider, but with position of the resistor and the capacitor flipped,

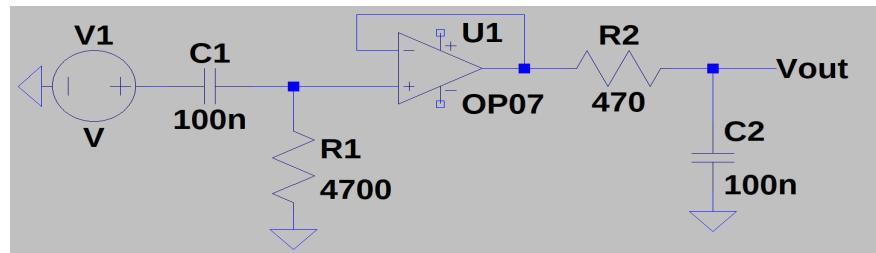
We have:

$$V_{out} = V_{inp} \times \frac{R}{R+Z_c} , \quad Z_c = \frac{1}{j \omega C} , \text{ and} \quad \omega = 2 \pi f_c$$

After Calculation, We have:

$$f_c = \frac{1}{2 \pi R C}$$

Buffered Band-pass Filter



(Picture 2)

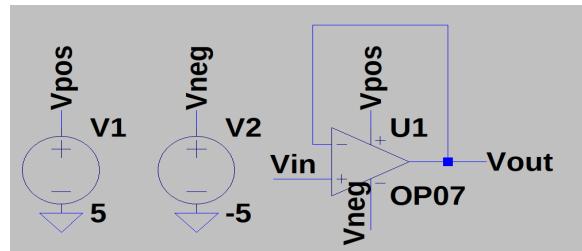
Band-pass filter is a filter only allows frequency within the range of frequency to pass through with gain higher than -3dB.

We will take about the Buffer later, now let's focus on feature of high-pass and low-pass filters combination:

By observing the graph, we can know that AC comes in first meets with high-pass filter, and going through a buffer, and, finally, going through a low-pass filter. We can easily know that having a higher cutoff frequency for low-pass filter and a lower cutoff for the high-pass filter will achieve the effect of band-pass filter/

Operational Amplifiers

Buffer



(Picture 4)

Buffer can be accomplished by using op-amp.

Considering the feature of a op-amp, we have:

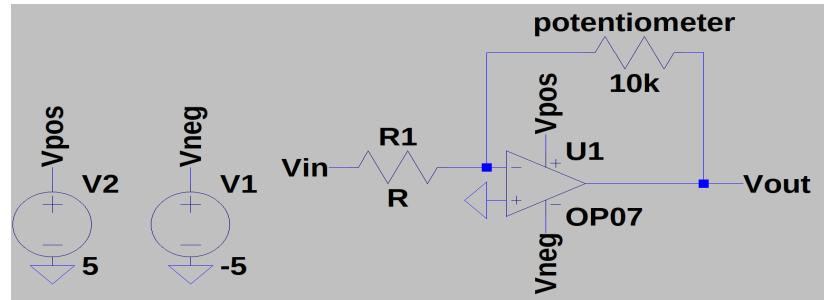
$$V_{input}^+ = V_{input}^- \text{ , and } V_{input}^- = V_{out}$$

Therefore:

$$V_{inp} = V_{out}$$

Since input has very large resistance between them, so it can act like a open circuit, where voltage can still pass on, so we can isolate two parts using this structure.

Inverting Amplifiers



(Picture 5)

Same feature of op-amp can be used here:

$$V_{input}^+ = V_{input}^-$$

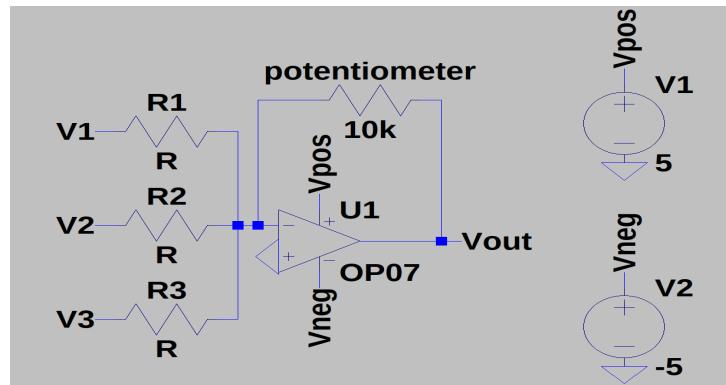
So, we know that node at $V^- = 0V$, and we can do a nodal analysis to it:

$$\frac{V_{inp} - 0}{R_1} = \frac{0 - V_{out}}{R_{potentiometer}},$$

We have:

$$\frac{V_{out}}{V_{inp}} = -\frac{R_2}{R_1}$$

Summing Amplifier



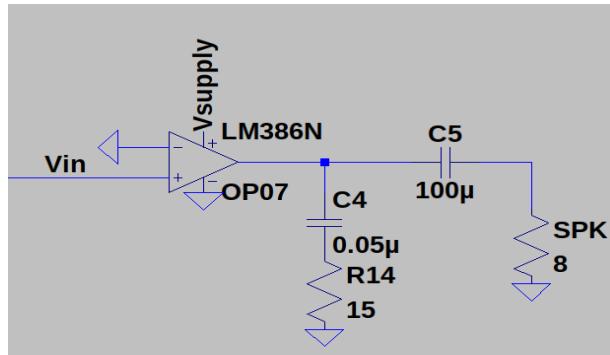
(Picture 6)

Summing Amplifier is the same logic as the inverting amplifier, where we do the same nodal analysis, and use the same feature of the op-amp:

$$V_{input}^+ = V_{input}^- \text{ , and } V^- = 0V \text{ , and } \frac{V_1 - 0}{R_1} + \frac{V_2 - 0}{R_2} + \frac{V_3 - 0}{R_3} = \frac{0 - V_{out}}{R_{potentiometer}} \text{ , so we can end up with:}$$

$$V_{out} = -R_{potentiometer} \times \left(\frac{V_1}{R_1} + \frac{V_2}{R_2} + \frac{V_3}{R_3} \right)$$

Power Amplifiers



(Picture 7)

The Power Amplifier is different from operational amplifier, where it can magnify the voltage that directly goes to the V_{input}^+ pin for many times, and the amplifier itself can endure more current than the op-amp. Therefore, we use the LM386N power amplifier here, which can make voltage 20 to 200 times higher, to magnify the power delivering to the speaker.

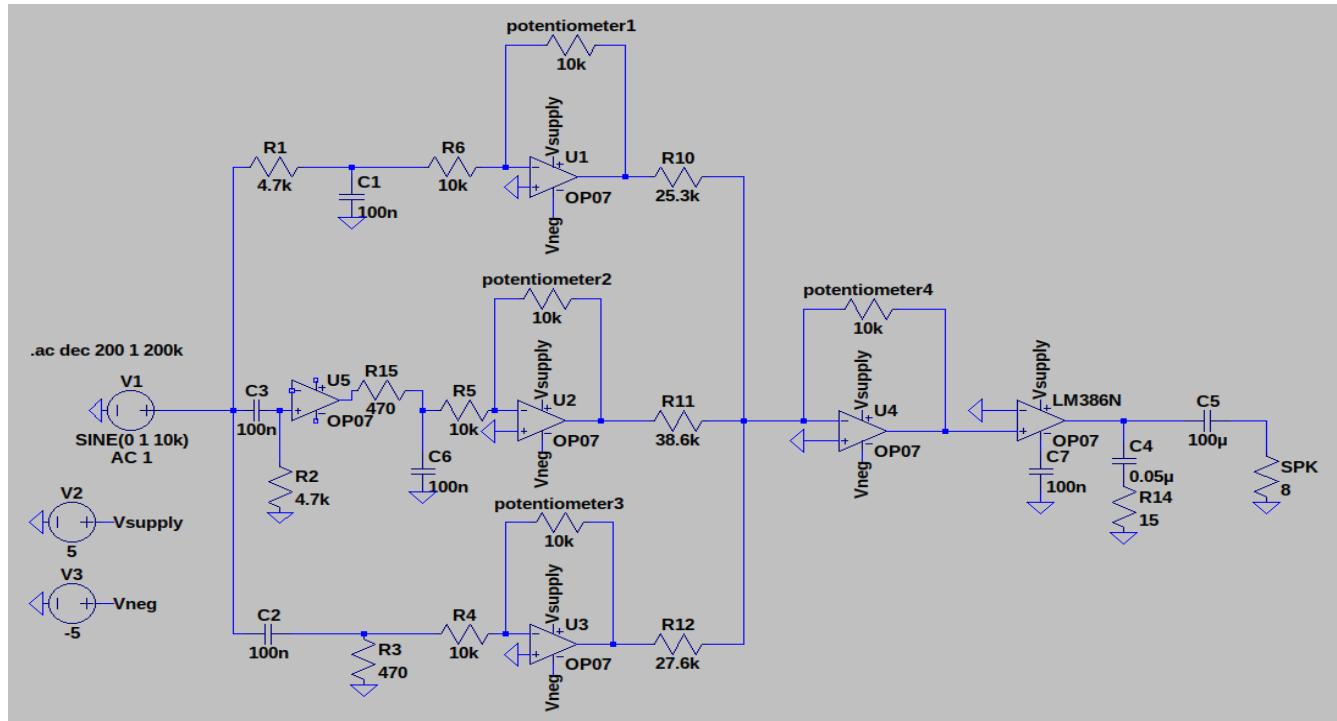
Procedure

Specification	Requirement
Speaker Impedance	8 Ohm
Bass filter -3dB cutoff	320Hz \pm 10%
Mid filter -3dB cutoffs	320-3200Hz \pm 10%
Treble filter -3dB cutoff	3200Hz \pm 10%
V_{amp} with all volumes turned to minimum settings	<15mV _{RMS} @ 100 Hz, 1000 Hz, 10000 Hz
V_{amp} with all volumes turned to maximum settings	100 mV _{RMS} \pm 10% @ 100 Hz, 1000 Hz, 10000 Hz

(Picture 8)

These are the values we are calculating below.

Schematics



(Picture 9)

This is the schematics that experimentally correct and meets with requirement. The calculations of parameters below are theoretical. The formulas are from the Theory section above.

Parameter

Filters

Low-pass Filter

We know that:

$$f_c = \frac{1}{2\pi RC} \quad \text{, and } f_c = 320 \text{ Hz}, C = 100 \text{ nF} \quad \text{, as we designed, meaning frequency lower than 320Hz}$$

can pass with gain higher than -3dB.

Therefore, the calculated resistance is:

$$R=4973\Omega .$$

High-pass Filter

We know that:

$f_c=\frac{1}{2\pi RC}$, and $f_c=3200\text{ Hz}$, $C=100\text{ nF}$, as we designed, meaning frequency higher than 3200Hz can pass with gain higher than -3dB.

Therefore, the calculated resistance is:

$$R=497\Omega .$$

Band-pass Filter

We know that band-pass filter in the design using high-pass and low-pass combination, and we need frequency from 320Hz to 3200Hz to pass through with gain higher than -3dB, so we need 3200Hz cutoff frequency for the low-pass filter with 320Hz cutoff frequency for the high-pass filter.

Since they have the same formula:

$$f_c=\frac{1}{2\pi RC} ,$$

So I just input the value of 320Hz and 3200Hz with selected 100nF capacitor, We have:

$$R_L=497\Omega , R_H=4973\Omega .$$

Volume Control

All three voltage after pass through the filters will each go to a inverting amplifier, and our goal with the input voltage is to invert them, with ratio of voltage input and output to be from 0 to -1, so we can turn it to 0 or the original volume.

We have ratio equation:

$$\frac{V_{out}}{V_{inp}} = -\frac{R_2}{R_1} \quad , \text{ and } R_2 = R_{potentiometer} \in [0, 10k] \Omega \quad ,$$

Therefore, we need $R_1 = 10k \Omega$.

Summing Amplifier

As the requirement specified, we need to have 100mVrms as our output of summing amplifier with 320mVrms input. According to the equation, when all the potentiometer are maxed:

$$V_{out} = -R_{potentiometer} \times \left(\frac{V_1}{R_1} + \frac{V_2}{R_2} + \frac{V_3}{R_3} \right) \quad , \quad V_1 = V_2 = V_3 = -320 \text{ mVrms} , R_{potentiometer} = 10k \Omega$$

We can calculate the value of resistance:

$$R_1 = R_2 = R_3 = 33k \Omega \quad .$$

Description

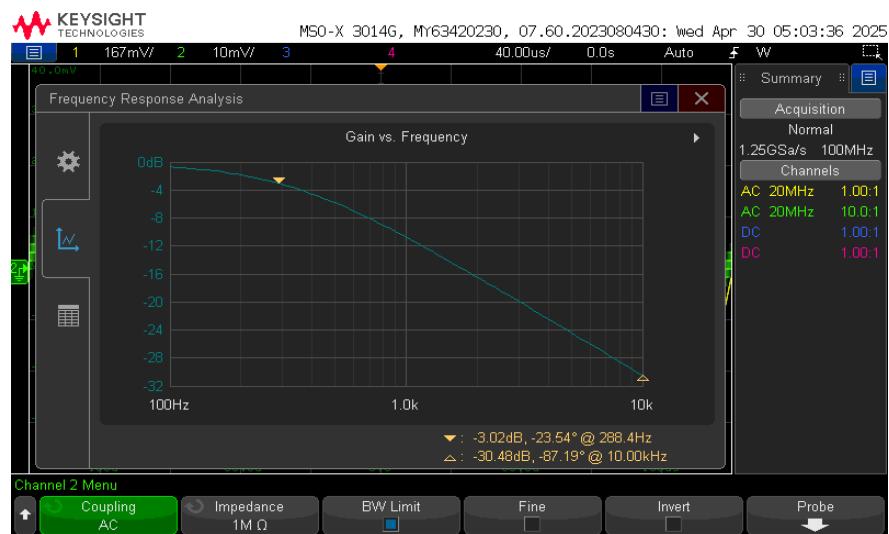
According to the picture 9, we can know that when a AC voltage comes in, the low frequency will go through top part of the circuit, the middle frequency will go through middle part of the circuit, and the high frequency will go through the lower part of the circuit, as how it is designed.

After the AC input is filtered to three parts, we are able to use the potentiometers at the inverting amplifiers to adjust the volume of each range of frequency.

Finally, they are combined together with a summing amplifier, with controlled voltage output that goes to the power amplifier.

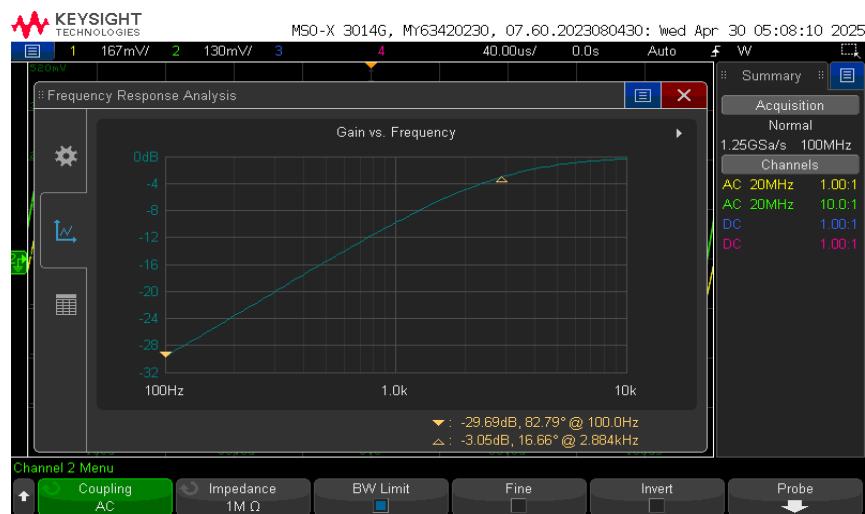
Result

Low-pass Filter



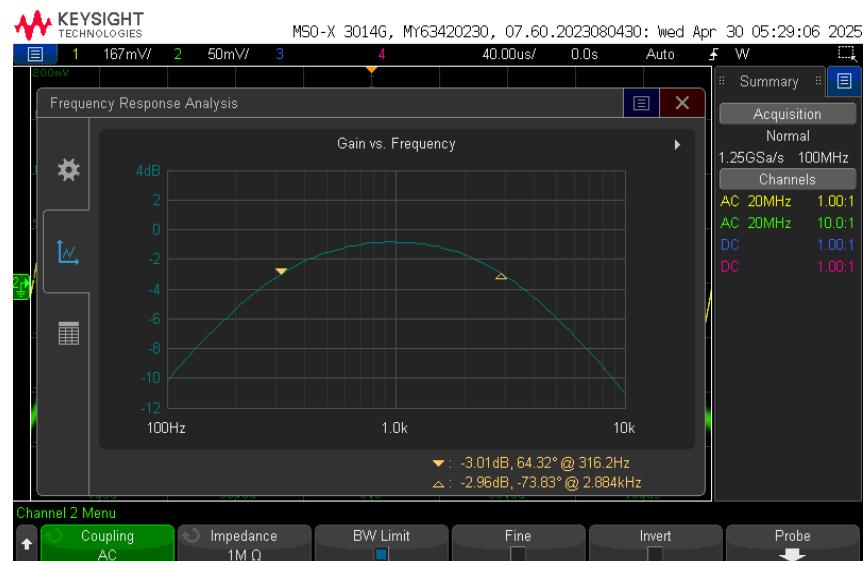
(Picture 10)

High-pass Filter



(Picture 11)

Band-pass Filter



(Picture 12)

Explain

We use FRA to sweep through each frequency at the output of each filter, and observe the frequency at -3dB gain position, in order to see if the filters met with requirement.

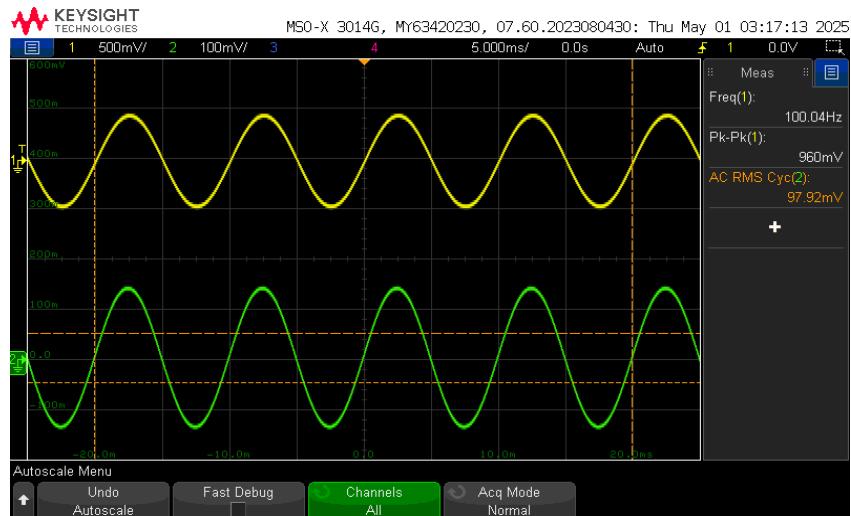
Filter Type	Lower Cutoff (Hz)	Higher Cutoff (Hz)	Lower %Error (Hz)	Higher %Error (Hz)
low-pass	288.4	N/A	9.8%	N/A
high-pass	N/A	2884	N/A	9.8%
band-pass	316.2	2884	1.2%	9.8%

(Table 1)

The result of three filters matches the requirement of tolerance within 10%, so the function will be accomplished.

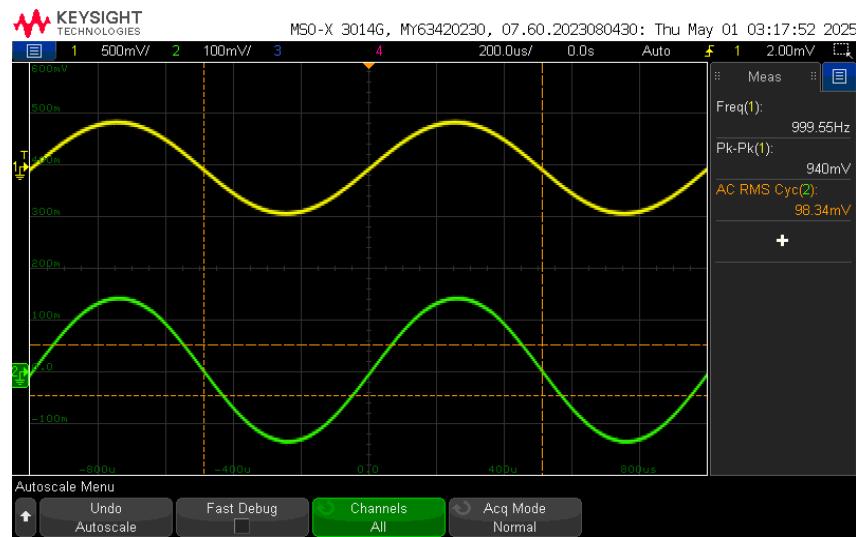
Max Output

100Hz



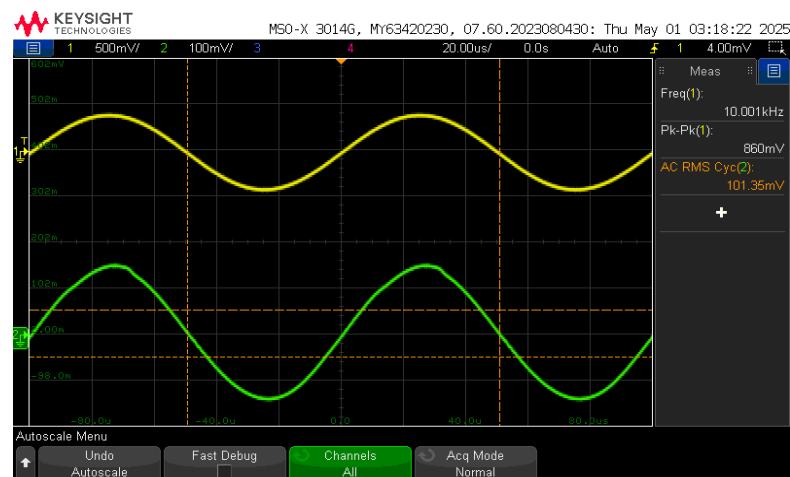
(Picture 13)

1kHz



(Picture 14)

10kHz



(Picture 15)

Explain

We turned all the potentiometers to the max setting, and measure through different frequencies at the output of the summing amplifier, and below is our result:

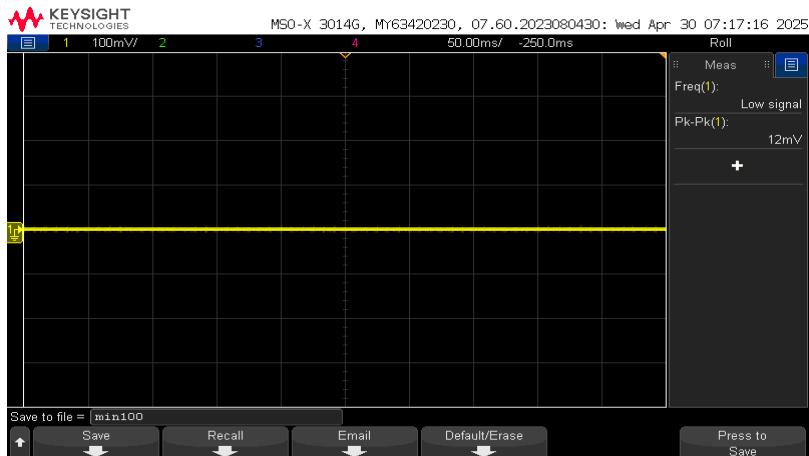
Frequency (Hz)	Vrms (mV)	%Error
100	97.92	2.1%
1k	98.34	1.7%
10k	101.35	1.4%

(Table 2)

The result of three different frequencies matches the requirement of tolerance within 10%, so the function will be accomplished.

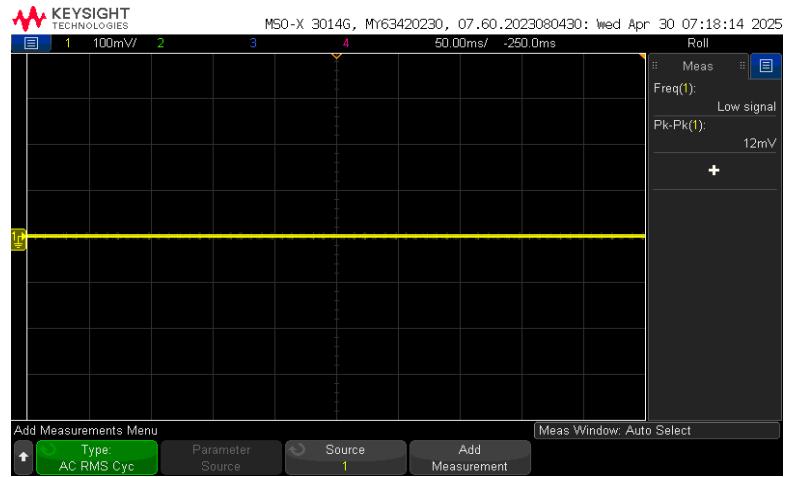
Min Output

100Hz



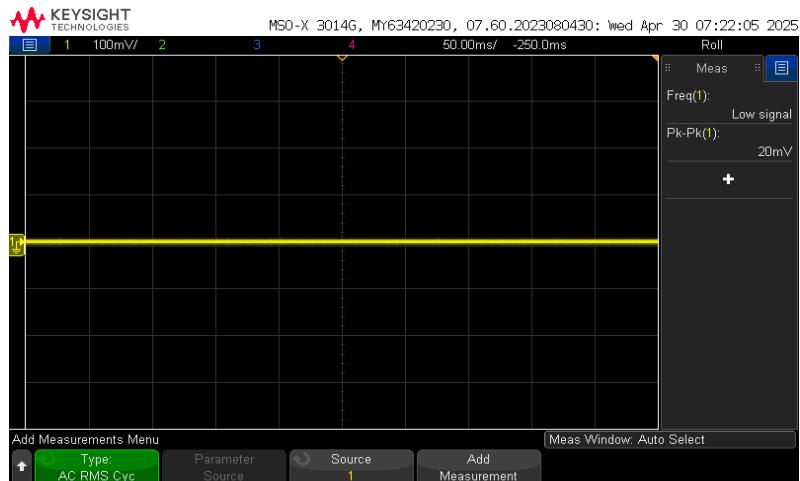
(Picture 16)

1kHz



(Picture 17)

10kHz



(Picture 18)

Explain

The equation for V_{pp} converting to V_{rms} given below:

$$V_{pp} = 2\sqrt{2}V_{rms}$$

Therefore, we can calculate Vpp from the graph into Vrms.

We turned all the potentiometer to the minimum setting, and measure through different frequencies at the output of the summing amplifier, and we get the data below:

Frequency (Hz)	Vpp (mV)	Vrms (mV)	Status
100	12	4.24	Pass
1k	12	4.24	Pass
10k	20	7.07	Pass

(Table 3)

The result of three different frequencies matches the requirement of Vrms lower than 15mV, so the function will be accomplished.

Ripple

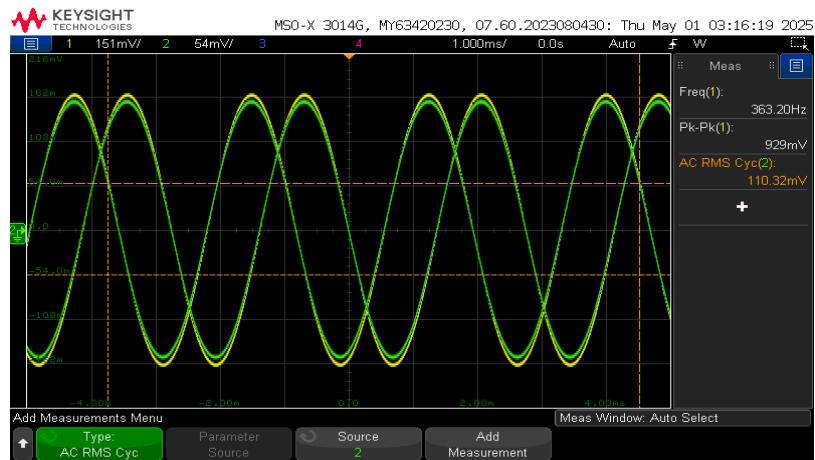
Measurement

FRA



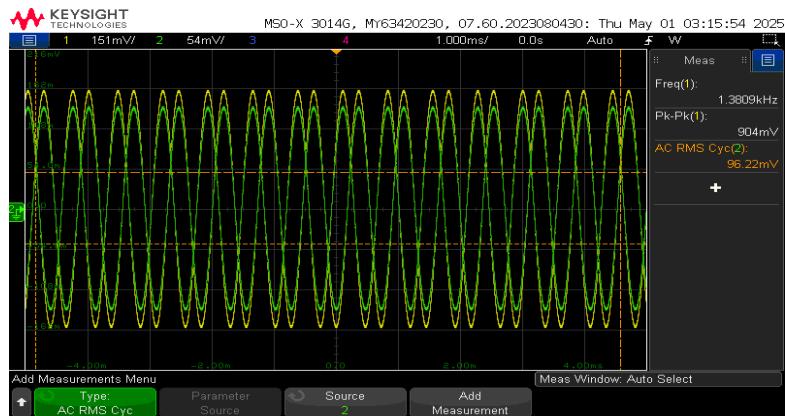
(Picture 19)

Min Gain



(Picture 20)

Max Gain



(Picture 21)

Explain & Calculation

The ripple means for all range of frequencies, the difference of max and min Vrms.

Firstly, we do a FRA at the output of the summing amplifier, so we can find the frequencies with the minimum and maximum gain, which are the positions with min and max Vrms.

We get:

$$f_L = 363 \text{ Hz}, f_H = 1380 \text{ Hz} ,$$

Then, we need to use wave generator to measure the Vrms at these two frequencies, and use:

$$V_{\text{ripple}} = V_{\text{max}} - V_{\text{min}}$$

According to picture 20 and picture 21, we have:

$$V_{\text{min}} = 96.22 \text{ mV}_{\text{rms}}, V_{\text{max}} = 110.32 \text{ mV}_{\text{rms}} ,$$

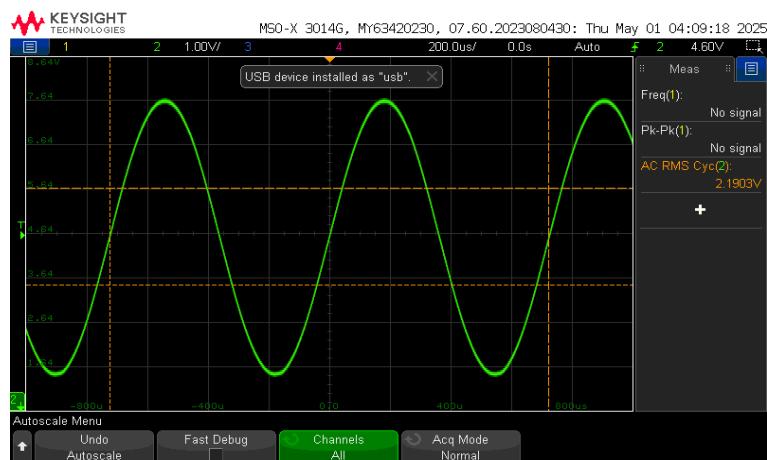
Therefore, we can calculate:

$$V_{\text{ripple}} = 14.1 \text{ mV}_{\text{rms}} ,$$

Which is under 15mVrms, so the ripple meets the requirement.

Power Output

Voltage Measured



(Picture 22)

The voltage measured at the output of the Power Amplifier is $V_{rms}=2.19V$. Assume that we have input voltage of $V_{rms}=100mV$, we can calculate the ratio to show how many times of voltage the power amplifier raised:

$$\frac{V_{inp}}{V_{out}}=21.9 ,$$

Therefore, we know that the Power Amplifier will raise the input power by 21.9 times when the input voltage is at 100mV, which is reasonable according to the Theory section above.

Power Calculation

Now lets calculate for Power output and compare it with the requirement, we have:

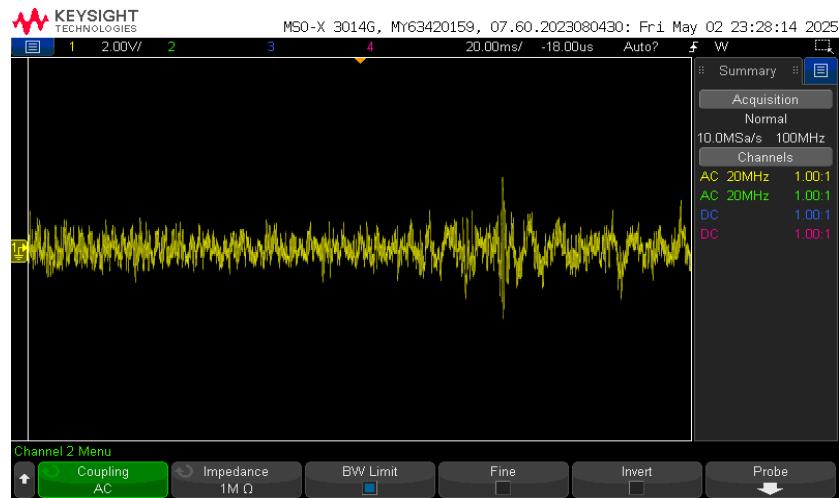
$$P=\frac{V^2}{R}, R=8\Omega, V_{rms}=2.19 , \text{ after calculation, we have:}$$

$$P_{experiment}=0.60W > P_{required} .$$

Therefore, this voltage output meets with the requirement.

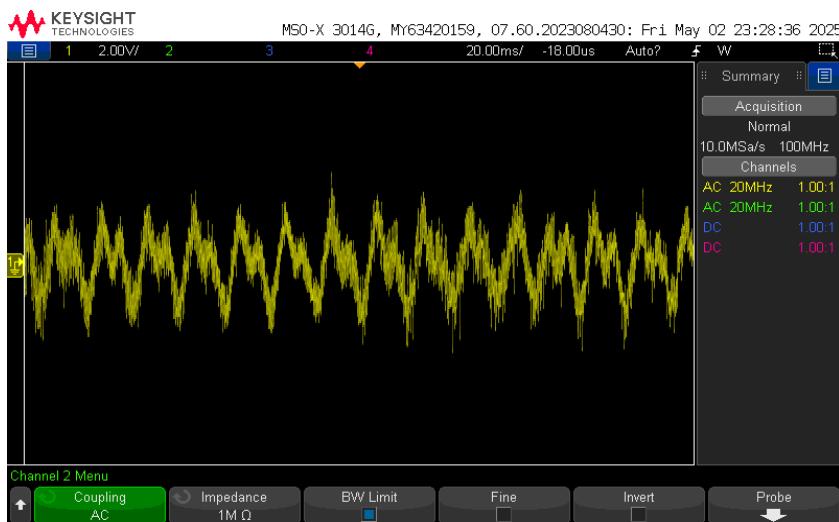
Volume Control

Low



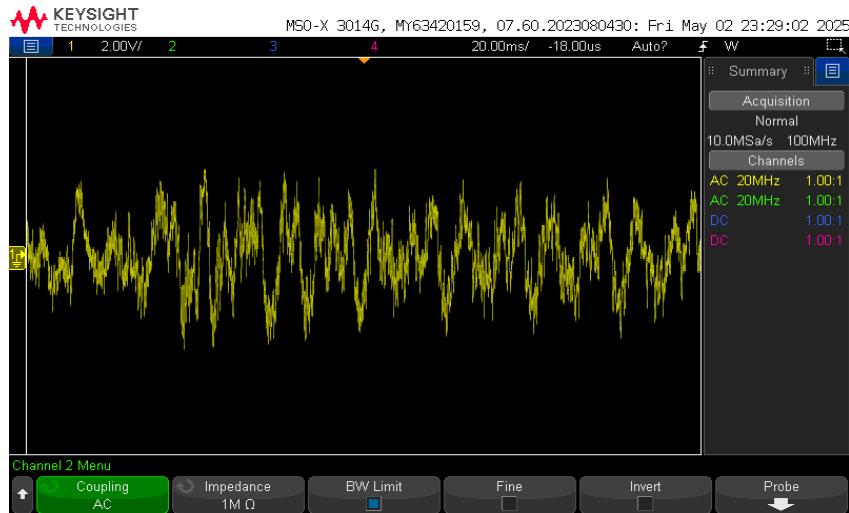
(Picture 23)

Medium



(Picture 24)

High



(Picture 25)

Explain

When we turn the bass volume to 0, we observe on picture 23 that frequencies are very high, and the waves are close together.

When we turn the mid volume to 0, we observe on picture 24 that we have a general structure with low frequencies, but with the noise from high frequencies, which means the mid frequencies are turned down.

When we turn the treble volume to 0, we observe on picture 25 that we have a comparatively lower frequencies than the picture 23, which means high-pass filter works as intended.

Review

According to the pictures and result when playing music, which we can observe on the picture 23, 24, and 25, where they all meet with requirement with corresponding frequency.

Therefore, the project of audio equalizer function as expected. All the parameter used are according to picture 9, since there are some deviation between experimental values and the theoretical values. The reason will be discussed below.

Conclusion

Idea

This project is very straight forward that we use filter to separate AC input to three parts, and, by adjusting each part, we can sum them up to output to speaker with a power amplifier. The key point is to test every function carefully so they meet with the requirements.

Findings

After finishing the project, I understand the importance of making calculations before actually doing the experiment, because I spend a lot of time in useless adjusting parameters, where I can have a general image of what they look like by calculating them in advance.

Experiment & Expectation

Most of the calculated parameters are different experimental values, like filters and the summing amplifier. The filters all have lower resistance in the experiment, and resistances for low and high frequencies are lower than calculated, where the mid frequency is higher for the summing amplifier. I think this is because the accuracy problem of the breadboard and the tolerance for each components may have.

Reference

1. D. Gormley, B. Manning, and C. Hack, *ECE20007 Final Project: Audio Equalizer*, Purdue University, Apr. 28, 2025.
2. R. Beasley and B. Manning, *ECE20007: Experiment 7 – Passive Filters*, C. Hack, Ed., Purdue University, Oct. 16, 2024.
3. B. Manning, *ECE20007: Experiment 8 – Operational Amplifiers*, C. Hack and G. Shaju, Eds., Purdue University, Mar. 10, 2025.
4. Texas Instruments, *LF356: Low drift JFET-input operational amplifier*, Rev. H, Datasheet, 2022. Texas Instruments. Available: <https://www.ti.com/lit/ds/symlink/lf356.pdf>
5. Texas Instruments, *LM324: Low-power quad operational amplifier*, Rev. AF, Datasheet, 2023. Texas Instruments. Available: <https://www.ti.com/lit/ds/symlink/lm324.pdf>
6. Texas Instruments, *LM386: Low voltage audio power amplifier*, Rev. J, Datasheet, 2020. Texas Instruments. Available: <https://www.ti.com/lit/ds/symlink/lm386.pdf>