Final Project: Audio Equalizer

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Abstract

This report details the design, construction, and testing of an adjustable audio equalizer circuit as part of ECE 20007's final project. The audio equalizer independently amplifies low ($0 - 320Hz$), mid-range ($320 - 3200Hz$), and high ($3200 + Hz$) frequencies. It must meet ripple, output power, and cutoff specifications. In addition to measuring outputs with lab testing equipment, the circuit's performance is demonstrated with a live adjustment of input audio. This project integrates many topics from previous labs, notably signal filtering and operational amplification. The report details the theoretical foundation of audio equalization and its component processes, outlines the procedure of design and construction, and presents the circuit valdiation results. These results are then discussed and the project's success in meeting specifications analyzed.

Introduction

This project is the culmination of a semester's work, synthesizing passive/active filters and op-amps into a tangible demonstration: an adjustable audio equalizer capable of selectively modifying the frequencies of an input signal. The final circuit must accept a $320mV$ RMS signal and selectively amplify frequencies in the ranges of 0-320, 320-3200, and $3200+Hz$. These seperated signals must be re-integrated and their aggregate independently amplified so that the output signal is suitable for driving a speaker. Additionally, the circuit must meet performance specifications for cutoff, ripple, and extremum amplification settings, as tabulated below.

1. **Cutoff:** each frequency must have a cutoff frequency (or in the case of the middle frequency filter, cutoff frequencies) within 10% of the listed cutoff value.

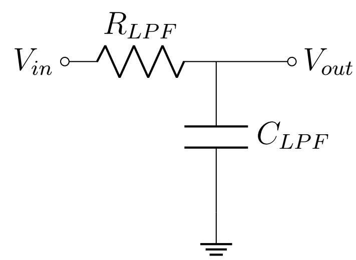
2. **Ripple:** when each volume knob is turned to its maximum, the difference between the highest peak and lowest trough must be no greater than $15mV$.

3. **Extremum:** at each input frequency indicated, the output RMS for each of the low, middle, and high frequency filters set to minimum volume must be at most $15mV$, while the output RMS must be within $100mV$ when set to maximum volume. After summation, the output power must be greater than $400mW$ when the input signal is between 100 and $10000Hz$.

Specification	Value
Bass filter $-3dB$ cutoff	$320Hz \pm 10\%$
Middle filter $-3dB$ cutoff	$320 - 3200Hz \pm 10\%$
Treble filter $-3dB$ cutoff	$3200Hz \pm 10\%$
Maximum ripple	$15mVRMS$
Min output RMS	$15mV$
Max output RMS	$100mV \pm 10\%$
Min output power	$400mW$ for $100 - 10000Hz$ input

The significance of this project extends beyond its technical complexity. It serves as a bridge between theory and practice, drawing on concepts discussed in ECE 20001 as well as empirical observations in ECE 20007. As engineering endeavors progressively demand adeptness in translating theory to functional products, we must not merely meet technical specifications but foster passion for the underlying science of what we design, so that we may always strive to learn and illuminate the dark corners of the universe. Thus, this project is the last in a series of stepping stones into a dark pond. From now on we must pick up the path to move forward and take what we have seen with us as we progress ever farther into that vast expanse. Let us step bravely and lovingly into those unknown waters and move beyond, with the hope that others may walk securely on the stones we lay behind.

Figure 1: RC low-pass filter



Theory

As this project draws from several labs that have come before, an overview of the audio equalizer's subcircuits is in order. Although many concepts and components from past work are present (capacitors, inductors, RMS voltage), we may identify two crucial constituents: **passive filters** and **operational amplifiers**.

Filters

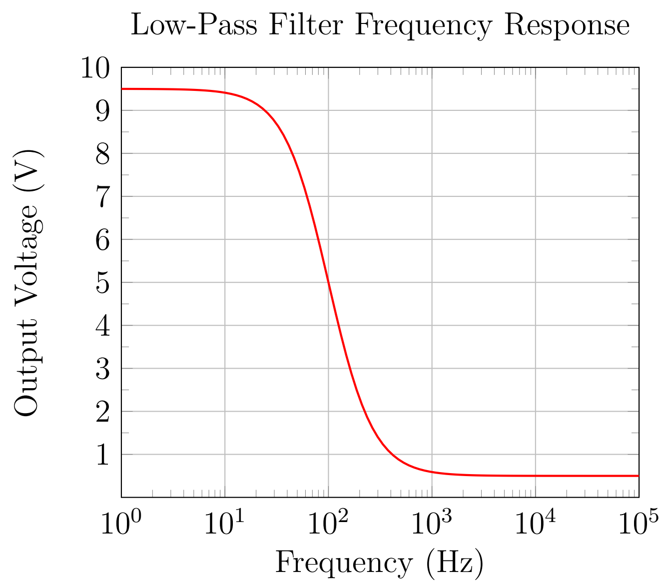
By combining resistors, inductors, and capcitors, we may create circuits that permit signals within a range of frequencies to pass unimpeded while attenuating frequencies outside this range. By carefully selecting values for the inductors and capacitors, we can adjust the permitted frequencies to any range desired. Let us examine a type of passive filter which permits all frequencies below a specified cutoff, the RC low-pass filter shown in Figure 1. If we recall the behavior of a capacitor, as frequency increases, the impedence decreases according to the formula

$$Z_C = \frac{1}{j\omega C} \tag{1}$$

Thus, the higher the frequency of the input signal, the more of that signal the capacitor diverts to ground. If we plot the output signal of the circuit in Figure 1 as a function of frequency, we obtain a graph similar to Figure 2. Typically, the signal cutoff frequency is defined as the frequency at which the gain $\frac{V_{out}}{V_{in}}$ is equal to $\frac{1}{\sqrt{2}}$, or on the decibel scale, the $-3dB$ point. We can analyze the circuit to determine precisely at what frequency this occurs. Notice that the circuit is set up in a voltage divider configuration, so the output voltage will be given by

$$\begin{aligned} V_{out} &= V_{in} \times \frac{-jX_C}{R_{LPF} - jX_C} && (2) \\ &= V_{in} \times \frac{-jX_C}{R_{LPF} - jX_C}. && (3) \end{aligned}$$

Figure 2: Frequency Response of an RC Low-Pass Filter



Ergo, we have that

$$\frac{V_{out}}{V_{in}} = \frac{-jX_C}{R_{LPF} - jX_C} \tag{4}$$

$$\frac{1}{\sqrt{2}} = \frac{-jX_C}{R_{LPF} - jX_C} \tag{5}$$

$$\frac{1}{\sqrt{2}} = \left|\frac{-jX_C}{R_{LPF} - jX_C}\right| \tag{6}$$

$$= \frac{X_C}{\sqrt{R_{LPF}^2 + X_C^2}} \tag{7}$$

$$= \frac{1}{\sqrt{\frac{R_{LPF}^2}{X_C^2} + 1}} \tag{8}$$

Therefore, we finally have

$$\frac{R_{LPF}^2}{X_C^2} = 1 \tag{9}$$

$$R_{LPF} = X_C \tag{10}$$

$$= \frac{1}{\omega C_{LPF}} \tag{11}$$

$$\rightarrow \omega = \frac{1}{R_{LPF}C_{LPF}} \tag{12}$$

$$\rightarrow f_c = \frac{1}{2\pi R_{LPF}C_{LPF}} \tag{13}$$

With this process, we have found the specific frequency at which the gain is $\frac{1}{\sqrt{2}}$. We can repeat this process for any kind of filter, three different kinds of which this project uses. The other two

Figure 3: RC high-pass filter

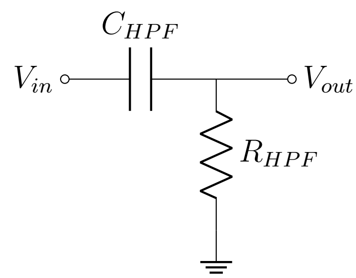
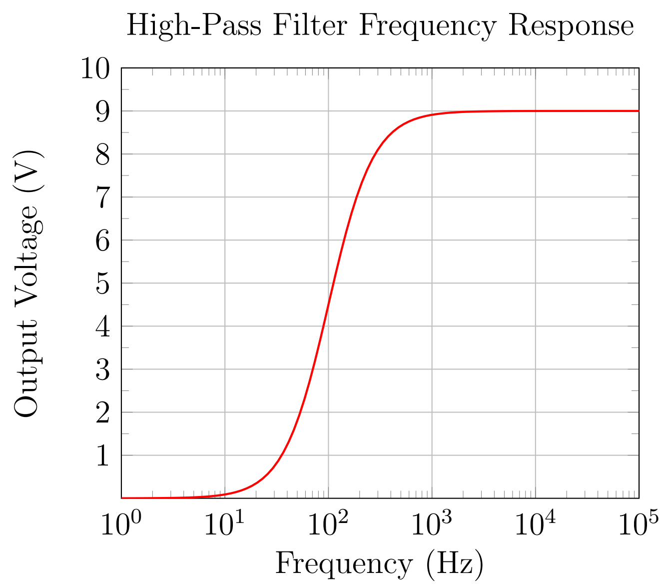


Figure 4: Frequency Response of an RC high-Pass Filter



are a RC high-pass filter (Figure 3, whose frequency response analysis graph is given in Figure 4) and a band-pass filter (Figure 5). The cutoff frequency of the RC high-pass filter, below which all frequencies are attenuated, is given by the same expression as the RC low-pass filter,

$$f_c = \frac{1}{2\pi R_{HPF} C_{HPF}}. \tag{14}$$

The band-pass filter is different from the other two in that a range of frequencies are permitted, while outside frequencies are not. Thus its characteristic equations take a different form. The frequency around which the pass band is centered (creatively called the *center frequency*) is given by

$$f_C = \frac{1}{2\pi\sqrt{L_{BPF} C_{BPF}}}. \tag{15}$$

The width of the pass band is given by the (equally creatively titled) *bandwidth*,

$$\beta = \frac{R}{L}. \tag{16}$$

Figure 5: RLC band-pass filter

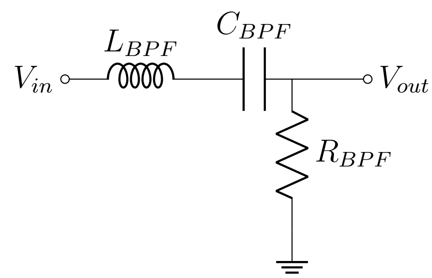
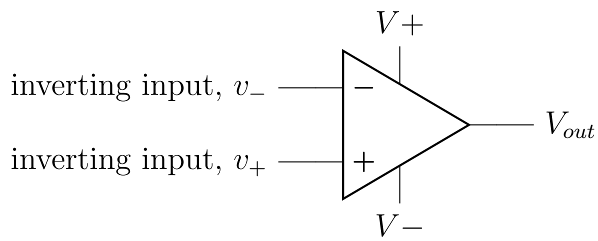


Figure 6: Operational amplifer schematic

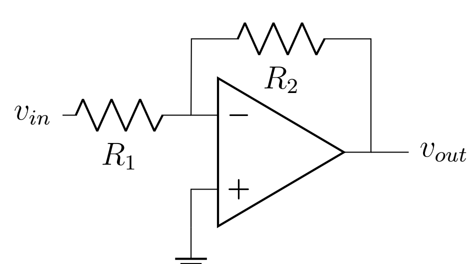


Thus, by tuning the values of R_{HPF}, L_{HPF}, and C_{HPF} we may center and adjust the width of the pass band to whatever we desire. This project requires the use of all three filter types to seperate the frequencies into the $0-320$, $320-3200$, and $3200+Hz$ bands. Once seperated, these signals may be seperately amplified or attenuated before recombination. Let us now turn to the component used to amplify signals, the **operational amplifer**.

Amplifiers

Operational amplifers, or *op-amps*, accept two signals and output one. Different combinations of op-amps and other components can achieve high gains (10^5 or more), isolate subcircuits to reduce loading, or sum many signals. The schematic for an op-amp is displayed in Figure 6. The audio equalizer circuit in this project uses op-amps configured as inverting amplifers, whose layout is shown in Figure. 7. Application of Ohm's law at the inverting input terminal reveals

Figure 7: Inverting amplifier schematic



$$I_{R_1} = \frac{v_{in} - v_-}{R_1} \tag{17}$$

$$I_{R_2} = \frac{v_{out} - v_-}{R_2} \tag{18}$$

Now, since I_{R_1} and I_{R_2} are the only currents delivered to the inverting input, Kirchhoff's current rule informs us that

$$I_{R_1} + I_{R_2} = 0 \tag{19}$$

$$\frac{v_{in} - v_-}{R_1} + \frac{v_{out} - v_-}{R_2} = 0 \tag{20}$$

$$\frac{v_{in} - v_-}{R_1} = -\frac{v_{out} - v_-}{R_2} \tag{21}$$

$$(v_{in} - v_-)R_2 = -(v_{out} - v_-)R_1 \tag{22}$$

$$v_{in}R_2 - v_-R_2 = -v_{out}R_1 + v_-R_1 \tag{23}$$

Since the non-inverting input is grounded ($v_+ = 0$) and the op-amp is in a buffer configuration, we have that $v_- = 0$ and thus

$$v_{in}R_2 - v_-R_2 = -v_{out}R_1 + v_-R_1 \tag{24}$$

$$v_{in}R_2 - 0 \times R_2 = -v_{out}R_1 + 0 \times -R_1 \tag{25}$$

$$v_{in}R_2 = -v_{out}R_1 \tag{26}$$

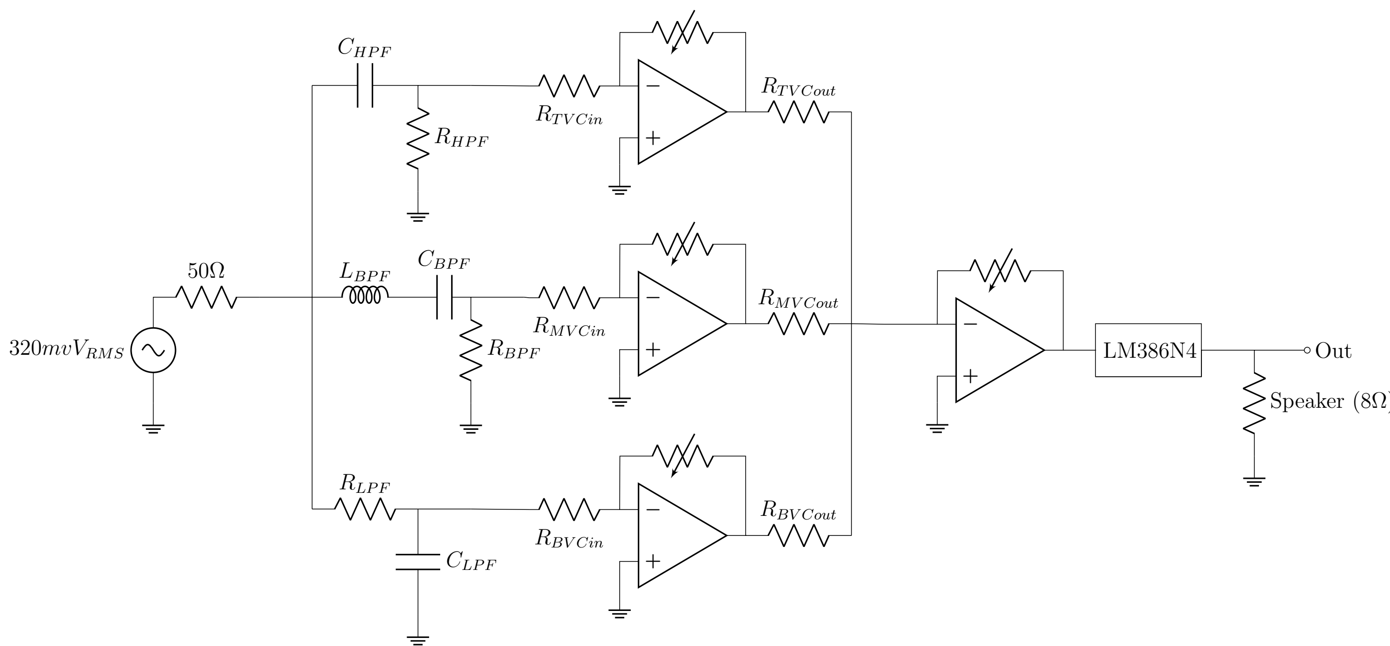
$$\frac{v_{out}}{v_{in}} = -\frac{R_2}{R_1} \tag{27}$$

So by altering the ratio of R_2 to R_1, we may adjust the gain of the amplifier. This useful property finds application as a volume adjuster in the audio equalizer. By substituting a variable resistor in place of R_2, the gain of each frequency may be independently controlled. The output of each is then fed into the input of another inverting amplifier, whose gain may be adjusted to control the volume of the entire output signal.

Audio Equalizer

Now that we are equipped with an understanding of the audio equalizer's fundemental components, let us integrate them and see how they adjust the different frequencies of an input signal. Once a singal of $320mv$ RMS is connected, each filter will isolate its respective range of frequencies. The output of each filter then enters its amplifier, which allows for selective tuning. The amplified output is then recombined and fed into the master amplifier, whose amplification may be adjusted to control the volume of the entire signal. The signal then passes through the power amplifier and terminates in a speaker, where the effects of the equalization may be auditorily verified. The entire audio equalizer circuit schematic is displayed in Figure 8.

Figure 8: Audio equalizer schematic



Procedure

With a firm basis of the underlying logic of the circuit, let's now step through how it was constructed.

1. First, values for the resistors, capacitors, and inductors were calculated as follows.

   Low-pass filter specifications: as seen in section Theory, the cutoff frequency for a low-pass filter is given by

   $$f_c = \frac{1}{2\pi R_{HPF} C_{HPF}}.$$

   Since we require that $f_c = 320Hz$, a capacitor value of $0.1\mu C$ was chosen for its availability in the minikit and a resistance of

   $$\begin{aligned} R_{HPF} &= \frac{1}{2\pi f_c C_{HPF}} && (28) \\ &= \frac{1}{2\pi(320Hz)(0.1\mu C)} && (29) \\ &= 4973.59197162\Omega && (30) \\ &\approx 5.1k\Omega && (31) \end{aligned}$$

   was calculated. The value was rounded to align with resistances provided in the minikit.

   Band-pass filter specifications: the center frequency for the band-pass filter is

   $$f_C = \frac{1}{2\pi\sqrt{L_{BPF} C_{BPF}}},$$

while the bandwidth is

$$\beta = \frac{R}{L}.$$

Our purposes require $f_c = 1760Hz$ and $\beta = 2880Hz$. Again, we select a capacitor value before calculating resistance (and inductance). Let $C_{BPF} = 47nF$, then

$$\begin{aligned}
L_{BPF} &= (\frac{1}{2\pi f_c})^2 \frac{1}{C_{BPF}} && (32)\\
&= (\frac{1}{2\pi(1760Hz)})^2 \frac{1}{47nF} && (33)\\
&= 0.173987108143H && (34)\\
&\approx 160mH && (35)\\
R &= \beta L_{BPF} && (36)\\
&= 2880Hz \times 0.160H && (37)\\
&= 460.8\Omega && (38)\\
&\approx 500\Omega && (39)
\end{aligned}$$

High-pass filter specifications: The formula for the cutoff frequency of a high-pass filter is the same as that of the low-pass filter,

$$f_c = \frac{1}{2\pi R_{HPF} C_{HPF}}.$$

However, now $f_c = 3200$. We therefore again select the convenient $0.1\mu C$ for the capacitor and find that

$$\begin{aligned}
R_{HPF} &= \frac{1}{2\pi f_c C_{HPF}} && (40)\\
&= \frac{1}{2\pi(3200Hz)(0.1\mu C)} && (41)\\
&= 497.359197162\Omega && (42)\\
&\approx 500\Omega && (43)
\end{aligned}$$

Amplifier specifications: we know that the gain of an inverting amplifier is given by

$$\frac{v_{out}}{v_{in}} = -\frac{R_2}{R_1}.$$

Since the amplifier here needs to function as a volume control, the gain ought to be between 0 and 1. We would also like to dynamically and conveniently adjust the volume, for which we should use a potentiometer. Since the only potentiometer in the minikit is $10k\Omega$, we have that

$$\begin{aligned}
R_1 &= -R_2 \frac{v_{in}}{v_{out}} && (44)\\
&= 10k\Omega && (45)
\end{aligned}$$

R_2, the potentiometer, may be tweaked to adjust the gain lower.

Output power: We know to avoid loading we must make R_{TVCout}, R_{MVCout}, R_{BVCout} large in proportion to R_{TVCin}, R_{MVCin}, R_{BVCin}, which are all $10k\Omega$. Let us choose $R_{TVCout} = R_{MVCout} = R_{BVCout} = 100k\Omega$. Simulation of the circuit then reveals that the output voltage, without power amplification, is approximately $150mV$ RMS at $3200Hz$. We require the final output power to be over $400mW$ with an 8Ω speaker. Thus, the output RMS voltage must be greater than

$$\begin{aligned} V &= \sqrt{PR} \\ &= \sqrt{400mW 8\Omega} \\ &= \sqrt{3.2}V \\ &\approx 1.789V \end{aligned}$$

The gain of the LM386 is 20, which brings the pre-amplification $150mV$ RMS to $150mV \times 20 = 3V$, well above the threshold of $1.789V$.

2. Next, the circuit outlined in Figure 8 was constructed in LTspice and simulated with a transient analysis to ensure functionality. Since LTspice lacks a native LM386 component, one was created and inserted using code provided by user Jonk on StackExchange [1].

3. Then, each subcircuit was created and independently tested in the following order:

   (a) High-pass filter. Tested with frequency response analysis (FRA) on oscilloscope.

   (b) Band-pass filter. Tested with FRA.

   (c) Low-pass filter. Tested with FRA.

   (d) High-pass volume control. Tested with digital multimeter (DMM) and power supply unit (PSU).

   (e) Band-pass volume control. Tested with DMM and PSU.

   (f) Low-pass volume control. Tested with DMM and PSU.

   (g) Total volume control. Tested with DMM and PSU.

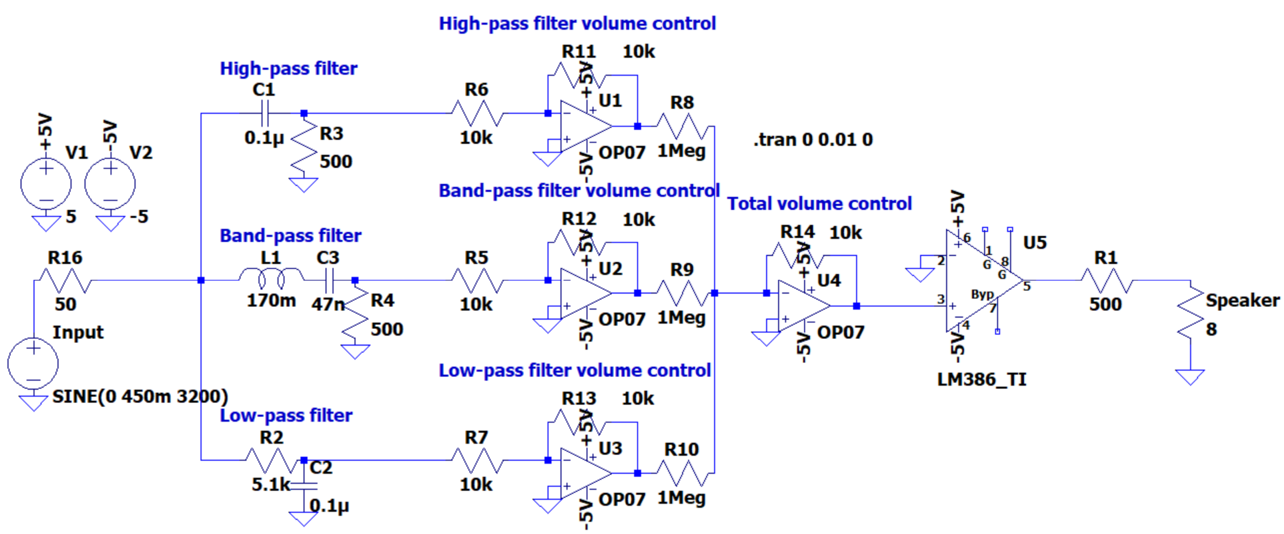
   (h) Power amplifier. Tested with DMM and PSU.

   After each subcircuit was operational, it was integrated into the larger audio equalizer.

4. Next, the circuit was judged on its ability to meet the benchmarkes outlined in the Introduction. Measurements were gathered and compiled under Results.

5. Finally, the circuit was connected to an audio input and output. Sound was played and adjusted to confirm functionality.

Figure 9: Spice schematic of audio equalizer



Results

Spice Simulation

The Spice schematic created for simulation is shown in Figure 9. To demonstrate proper functionality of the circuit, it was shown that the filters worked as expected, the output power was sufficient for each, and that the volume may be adjusted as needed.

1. *Filters*: Figure 10 shows the simulated output of each filter at frequencies of 320, 3200, and $32000Hz$. As can be seen from the figure, each filter attenuates the signal below the $-3dB$ point at its respective cutoff frequency.

2. *Output voltage*: Figure 11 shows the magnitude of the output before and after amplification at the lowest frequency, $320Hz$. It can be easily seen that amplification brings the voltage above the value required to drive a speaker of 8Ω, which was previously calculated as $\approx 1.789V$ RMS.

3. *Volume control*: Figure 12 shows the output signals at each of the three volume controls, as well as the complete output volume. As the figure shows, when all three volume controls are turned down, the output voltage is on the order of nanovolts. When one is turned down, its corresponding output is so small in relation to the others as to seem flat on the graph.

Empirical Analysis

The following aspects of the circuit were tested: filters, output voltage at minimum volume, output voltage at maximum volume, total volume control, signal recombination, ripple, and output power.

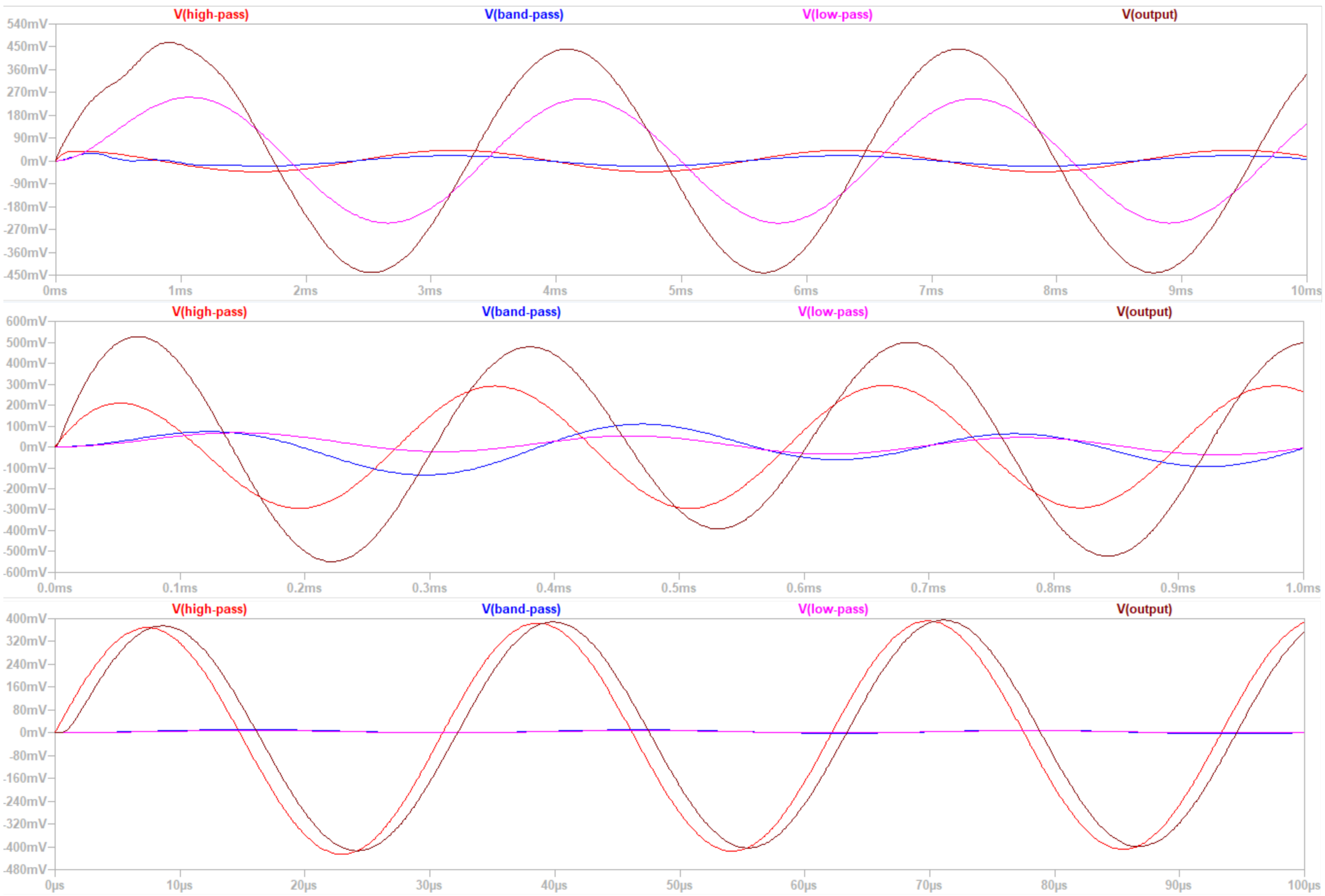


Figure 10: Spice simulation results at 320 (top), 3200 (middle), and 32000 (bottom) Hz input

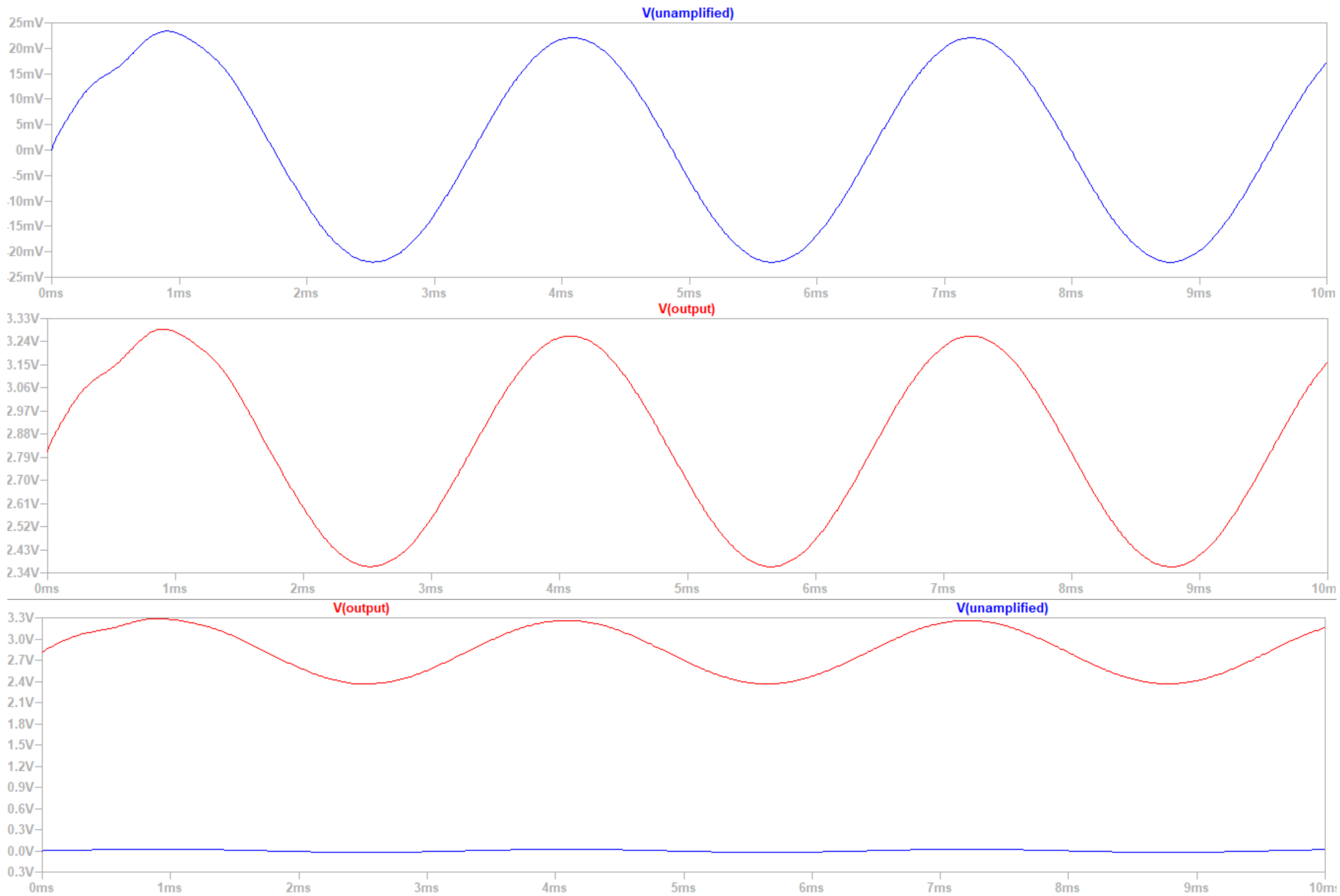


Figure 11: Output at $320Hz$ for unamplified (top), amplified (middle), and both (bottom)

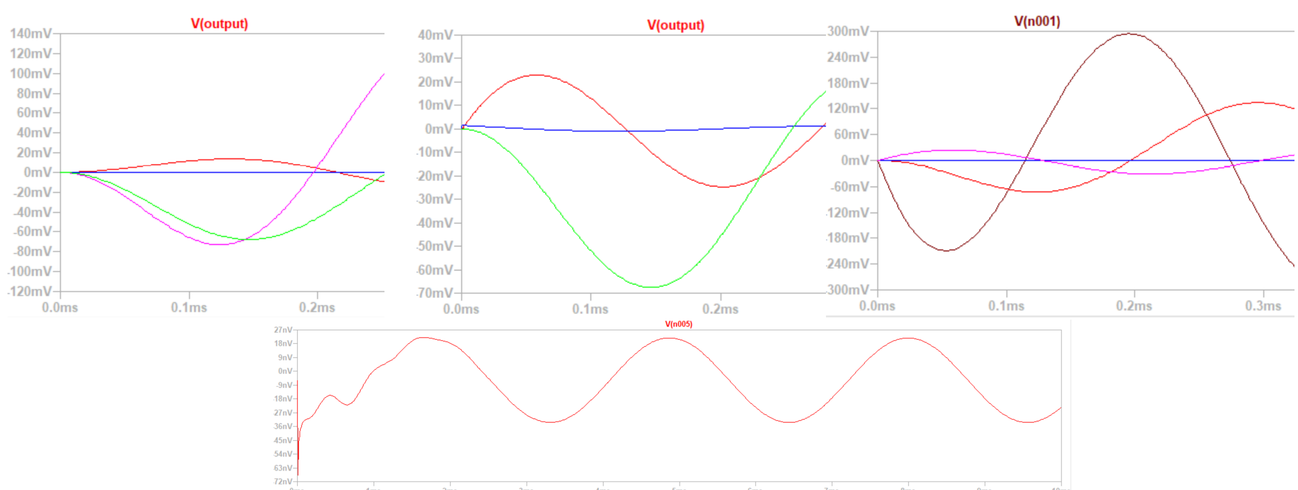


Figure 12: 0 volume for HPF (left), BPF (middle), LPF (right), and all (bottom)

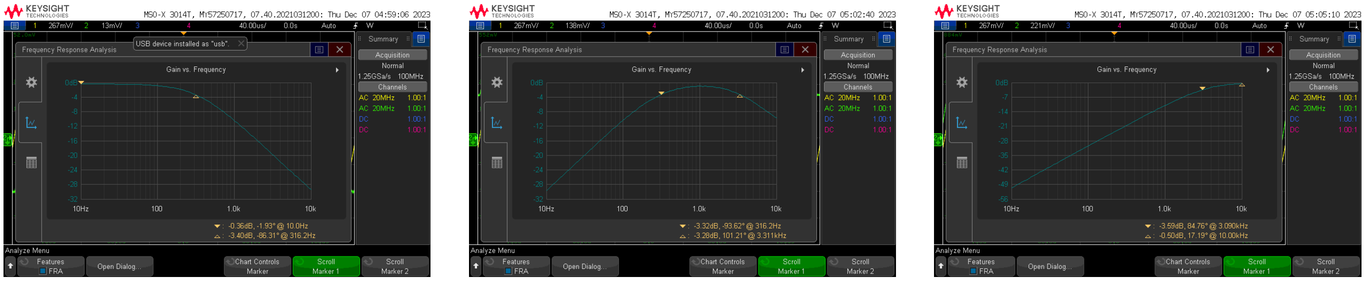


Figure 13: FRA results for low-pass (left), band-pass (middle), and high-pass (right) filters

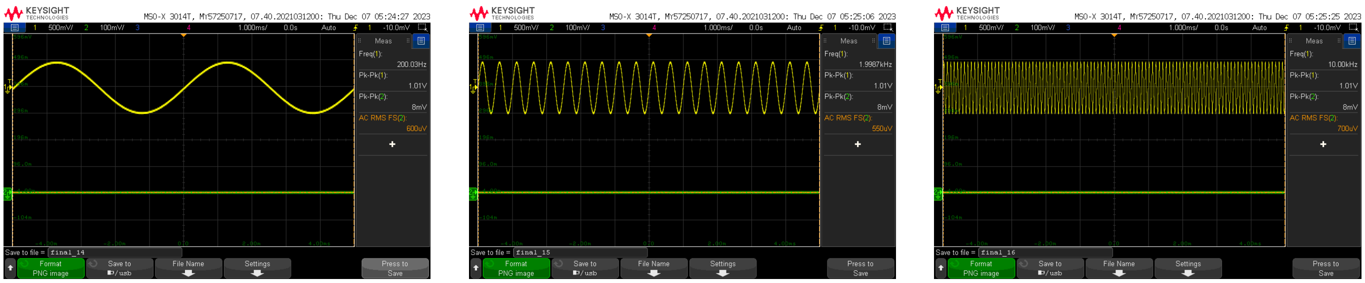


Figure 14: Output signal with minimum volume for 200, 2000, and 10000Hz

1. *Filters*: Figure 13 shows the measured output of each filter at frequencies of 200, 2000, and 10000Hz (the upper limit of human hearing).

2. *Output RMS voltage at minimum volume*: Figure 14 shows the magnitude of the output signal at a range of frequencies when the master volume dial is turned to a minimum.

3. *Output RMS voltage at maximum volume*: Figure 15 shows the RMS values of the voltage feeding into the power amplifer when the volume control is set to max.

4. *Volume control*: Figure 16 displays the output signal with the volume control at zero, turned slightly, and turned to the max.

5. *Signal recombination*: Figure 17 shows three input signals to each of the three filters: a since wave, triangle wave, and square wave.

6. *Ripple*: Figure 18 displays the gain at maximum volume during FRA.

7. *Output power*: Figure 19 displays the RMS voltage of the circuit.

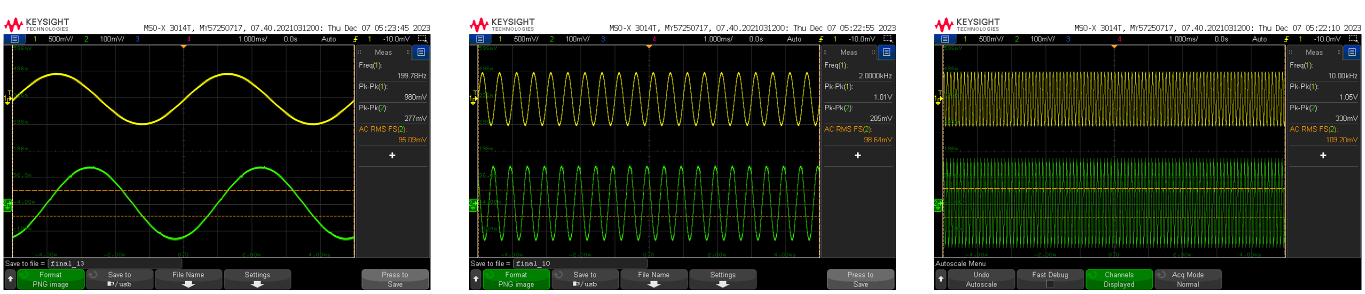


Figure 15: Output signal with maximum volume for 200, 2000, and 10000Hz

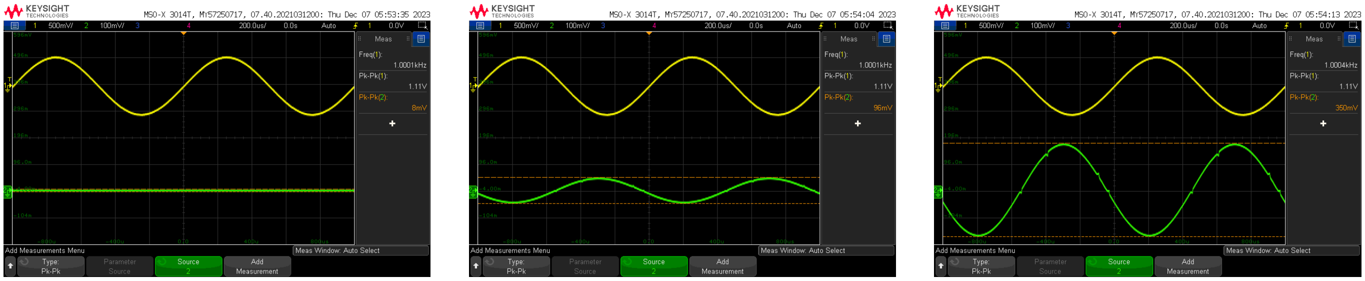


Figure 16: Input and output signal for low (left), medium (middle), and high (right) volume

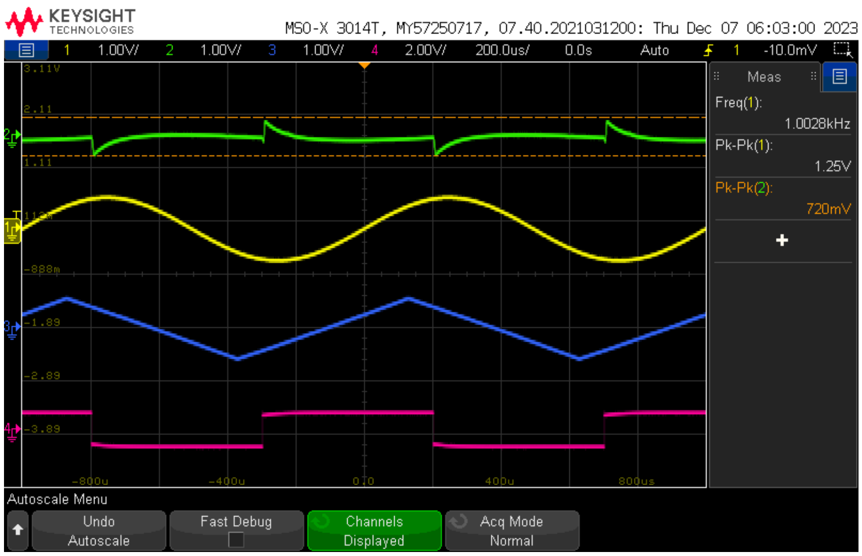


Figure 17: Input signals to each filter and recombined output

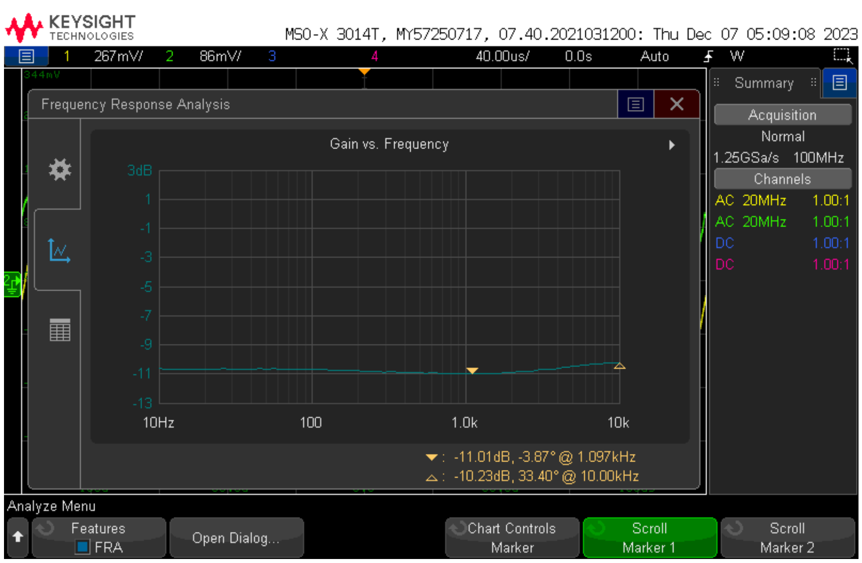


Figure 18: Gain at maximum volume across frequency range

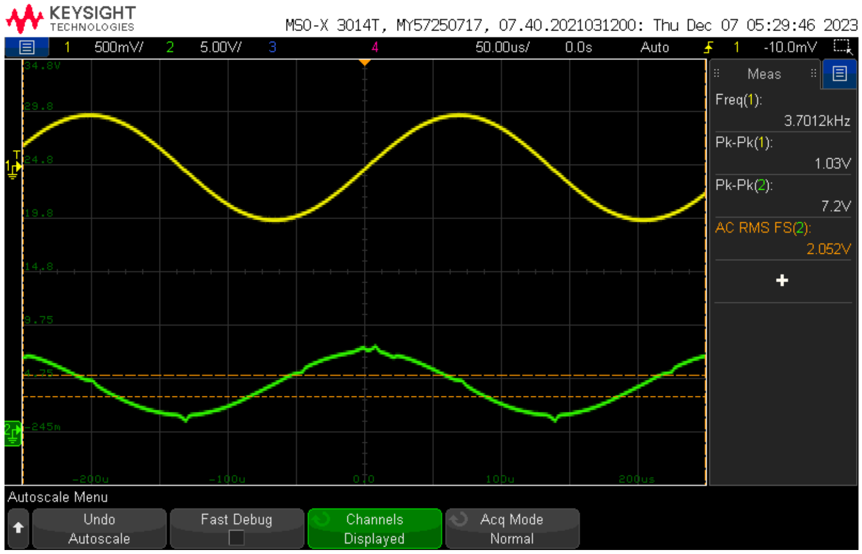


Figure 19: Output RMS voltage

Discussion

The project is a resounding success, meeting and in many cases significantly exceeding benchmarks on all tests: filtering, output voltages at minimum and maximum volumes, volume control functionality, signal recombination, ripple control, and output power.

1. *Filters*: As can be seen from Figure 13, each filter attenuates the signal below the $-3dB$ point at its respective cutoff frequency.

2. *Output RMS voltage at minimum volume*: As Figure 14 shows, the output at minimum volume is on the order of μV for all frequencies, which would be inaudible for most humans.

3. *Output RMS voltage at maximum volume*: The RMS values must be $100mVRMS \pm 10\%$, and our observed values are $95.09mV$, $98.64mV$, and $109.20mV$. Although the latter value approaches the 10 % threshold, all values are within and therefore satisfy the requirement.

4. *Volume control*: As can be qualitatively seen in Figure 16, the volume varies according to the degree at which the master volume knob is turned.

5. *Signal recombination*: As Figure 17 shows, the combined graph displays the output of the summing inverting amplifier and represents a superposition of these waves with combined characteristics of the three input waves, showing proper signal summation.

6. *Ripple*: Figure 18 shows the gain at maximum volume during is mostly linear with a maximum difference of $0.78dB$. That means that the human ear will percieve a roughly 10% difference in loudness between the frequency which produces the lowest gain and the frequency which produces the highest. This is absolutely acceptable and will barely be perceptible when listening to music from the audio equalizer.

7. *Output power*: Figure 19 shows that the RMS voltage of the circuit is $2.052V$. With an 8Ω speaker, this produces a power of

$$\begin{aligned} P &= \frac{V_{RMS^2}}{R} \\ &= \frac{(2.052V)^2}{8\Omega} \\ &= 0.526338W \\ &= 526.338mW \\ &> 400mW \end{aligned}$$

   Since we exceed the power needed to drive the speaker, this requirement, too, is met.

Overall, these findings affirm the device's functionality, precision, and compliance with specifications, validating its potential for effective audio signal processing.

Conclusion

Now we are done. This report has exhaustively detailed the theory, simulation, construction, and testing of an audio equalizer. The circuit successfully filtered the input signal into the desired frequency bands, used inverting op-amps to attentuate select frequencies, recombined these signals while allowing for further volume adjustment, and amplified the result so that it could drive a speaker with measured specifications.

This rock has been heaved into place, and the path behind us is a bit longer now. Perhaps in some time we will walk it again. If we do it will be as builders with new eyes and new minds, for returning to where we began is not the same as never leaving.

References

[1] Jonk. URL: https://electronics.stackexchange.com/questions/619723/im-trying-to-simulate-an-audio-amplifier-circuit-but-i-couldnt-find-how-to-add.