FUN II Final Project Report

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***Abstract*—The audio signal from a song is commonly represented as a complex sinusoid that comprises many different frequencies. In this inquiry, LEDs will be used to visually represent the magnitude of different frequency ranges at any time throughout the song. The process of converting the audio signal to an LED flash is done in four steps. First, the left and right audio channels are summed and amplified. This is done with an op amp configuration. Next, the frequencies present in the song are divided into a high partition (treble) and a low partition (bass). This separation is accomplished with filters that utilize op amps and reactive components. Once there is a distinct sinusoid representation for each frequency range, these signals pass through a common source amplifier that drives an LED. A peak detector is included to convert the AC signal into a DC impulse that can flash the LED on and off quickly. Finally, the amplifier is completed with a MOSFET that is biased to flash whenever the audio signal pushes it from cutoff into saturation. For each part of this process, there is an analytical design that uses mathematics and circuit analysis to support the chosen design. Numerical simulations in NI Mulitisim are also included to support conclusions and design choices before they are actually tested. Ultimately, each circuit component is verfifed experimentally on a breadboard, and then is soldered into a PCB design.**

I. Overleaf Read Only Link

https://www.overleaf.com/read/cgfxrghsfbdv

II. Background Information and Rationale

An audio signal coming from a headphone jack contains two signals: left channel and right channel. In order to design an analog signal processing system with these two inputs, first we must consider what we want the output to be. In this project, our goal was to take in two input signals from a headphone jack and vary the intensity of two LEDs depending on the amplitude of the magnitude of each channel. In order to achieve the desired outputs, we started by drawing a block level schematic:

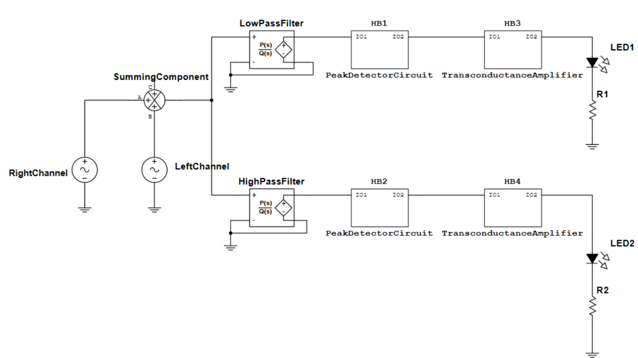

Fig. 1. Block Level Schematic

This is the final project report for ECE 2660 Fundamentals 2, Spring 2021 at the University of Virginia

The first step in the schematic is to add the two input signals using a summing, we can use a summer op-amp configuration to achieve this. Then, split the output from the summing circuit into two parallel paths and feed them into a low pass filter and high pass filter. The low pass filter will let low frequencies through and block high frequencies. The high pass filter will let high frequencies through and block low frequencies. To make the filters work properly, we must first choose a song we want to analyze and look at its frequency spectrum to determine its cutoff frequency. Looking at figure 2, we can see that most of the frequencies in Sultans of Swing by Dire Straits are below 800Hz. So, we choose 800Hz to be the cutoff frequency. After, a peak detector circuit is needed to create an envelope of the incoming audio signal since the the original audio signal are changing many times faster than the human eye can process. Finally, by using a trans conductance amplifier we can set the LEDs brightness with a current rather than voltage because LEDs are sensitive to voltage.

III. Design and Analysis

A. System Level Design and Analysis

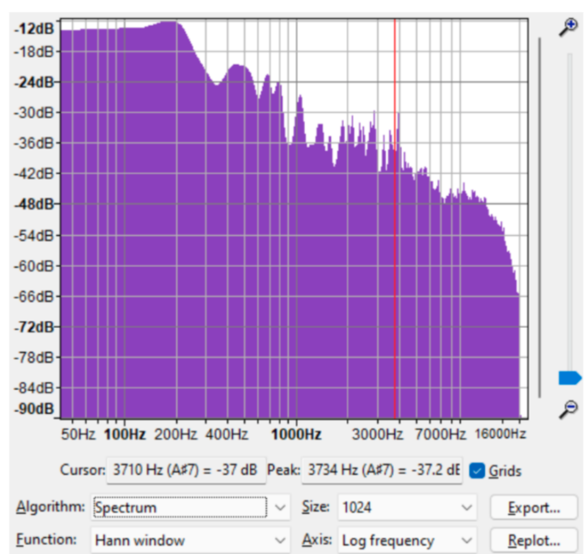

Fig. 2. Dire Straits-Sultans of Swing Frequency Spectrum

What song did your group pick for the final project? Why did you choose that song?

The song we selected for our project is Dire Straits by Sultans of Swing. This choice was based on the song's genre, rock, which provides a distinct diversity of frequencies throughout its duration. There is a consistent bass line that keeps the beat throughout the song, and is sometimes the only source of sound. Now and again, a higher frequency guitar sound

overlaps with high intensity.
What cutoff values did you determine for your low pass and high pass filters? Why? We chose 800Hz to be the cutoff frequency for both filters. This is because the majority of the songs frequency lies below 800Hz.

B. Summing Amplifier

The summing amplifier was developed using an inverting amplifier model, featuring a T resistor network in the feedback loop. Each input also included a high pass filter to eliminate DC and very low frequency signals. The input resistors were chosen first to keep the cutoff frequency near 20Hz. Next, through KCL analysis, an equation for a gain of 3 was determined. Note that this gain was defined for both outputs, and can be changed to just one later if the music voltage signal is too low. Finally, a MATLAB iterative analysis yielded resistor values for the T network that satisfied the gain (from the kit).

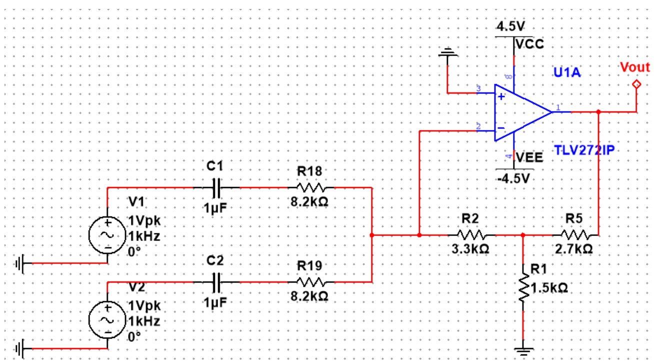

Fig. 3. Summing Amplifier.

$$-\frac{V_i}{Z_{C3}+R_{18}}-\frac{V_1}{R_2}=0$$

$$\frac{V_1}{R_2}+\frac{V_1}{R_1}+\frac{V_1-V_o}{R_5}=0$$

Now that KCL equations are determined, the first is solved for V_1:

$$V_1=-V_i\frac{R_2}{Z_{C3}+R_{18}}$$

Next, this expression is substituted into the second equation to keep it strictly in terms of V_i and V_o:

$$-V_i\left(\frac{1}{Z_{C3}+R_{18}}-\frac{R_2}{R_1(Z_{C3}+R_{18})}-\frac{R_2}{R_5(Z_{C3}+R_{18})}\right)=\frac{V_o}{R_5}$$

Finally, the equation is rearranged to produce the transfer function:

$$\frac{V_o}{V_i}=-\frac{1}{Z_{C3}+R_{18}}\left(R_5+\frac{R_2R_5}{R_1}+R_2\right)$$

Similarly, the right input will produce the following transfer function:

$$\frac{V_o}{V_i}=-\frac{1}{Z_{C14}+R_{19}}\left(R_5+\frac{R_2R_5}{R_1}+R_2\right)$$

The left and right channels are now combined. Assuming that the gain is only relevant for signals with a frequency that exceeds the cutoff frequency, the capacitors can be treated as short circuits, eliminating their impedances:

$$V_o=-V_i\left(\frac{R_5}{R_{18}}+\frac{R_2R_5}{R_1R_{18}}+\frac{R_2}{R_{18}}+\frac{R_5}{R_{19}}+\frac{R_2R_5}{R_1R_{19}}+\frac{R_2}{R_{19}}\right)$$

Now, there are two high pass RC circuits involved in this setup, the corner frequencies of which will help determine the values of R_18 and R_19:

$$f_{cL}=\frac{1}{2\pi R_{18}C_3}$$

$$f_{cR}=\frac{1}{2\pi R_{19}C_14}$$

Given that the corner frequencies must equal 20Hz, resistance and capacitor values are determined based on kit availability. Given that C_3 = C_14 = 1μF, we have:

$$R_{18}=\frac{1}{2\pi C_3f_{cL}}=7958\Omega$$

$$R_{19}=\frac{1}{2\pi C_3f_{cR}}=7958\Omega$$

The equation for V_o is updated with values below, and is simplified thereafter:

$$V_o=-V_i\left(\frac{R_5}{7958\Omega}+\frac{R_2R_5}{R_1(7958\Omega)}+\frac{R_2}{7958\Omega}+\frac{R_5}{7958\Omega}+\frac{R_2R_5}{R_1(7958\Omega)}+\frac{R_2}{7958\Omega}\right.$$

$$V_o=-V_i\left(\frac{R_2R_5}{R_1(3979\Omega)}+\frac{R_2+R_5}{3979\Omega}\right)$$

Next, a gain of 3 is desired for this circuit. The resistor values in the kit are submitted to a MATLAB algorithm to produce appropriate resistor values, shown below:

$$R_1=1500\Omega, R_2=3300\Omega, R_5=2700\Omega$$

C. High Pass Filter

Write about design and analysis of the HPF here. First,

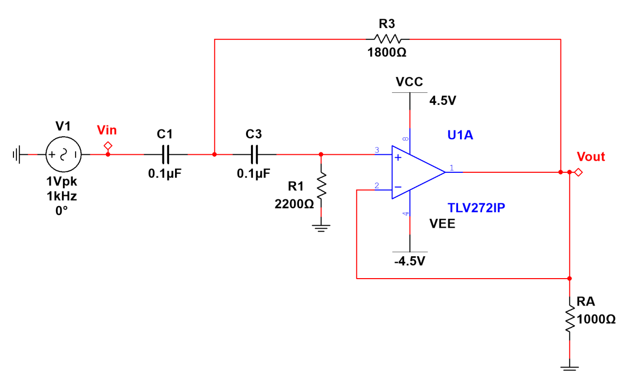

Fig. 4. Sallen Key High-Pass Filter Schematic

the transfer function for a low pass Sallen-Key Filter has the following form:

$$H(s)=\frac{Gs^2}{s^2+s\frac{\omega_c}{Q}+\omega_c^2}$$

Through KCL analysis and lots of very tedious algebra, a similar form can be created from the system components:

$$H(s) = \frac{Gs^2}{s^2 + s(\frac{1-G}{R_1C_1} + \frac{1}{R_2C_1} + \frac{1}{R_2C_2}) + \frac{1}{R_1R_2C_1C_2}}$$

Next, the fundamental quantities specified in the design constraints are derived from these equations, in terms of system components:

$$\omega_c = \sqrt{\frac{1}{R_1R_2C_1C_2}}$$

$$G = \frac{R_A + R_B}{R_A}$$

$$Q = \frac{\omega_c}{\omega_c^2(R_2C_2(1-G) + R_1C_1 + R_1C_2)}$$

$$Q = \frac{1}{\omega_c(R_2C_2(1-G) + R_1C_1 + R_1C_2)}$$

Now, noting the design constraints of the gain K = 0, the Q value 0.5 < Q < 0.707, and the corner frequency f_c = 800Hz, the component values become more clear. First, the gain:

$$K = 20\log(G) = 0 \rightarrow G = 1$$

If the linear gain is unity, the means one of the resistors in the feedback loop must be a short:

$$G = \frac{R_A + R_B}{R_A} = 1 \rightarrow R_A = (1000\Omega)\ R_B = 0\Omega \text{ (short)}$$

Next, through optimization analysis in MATLAB, the following resistor and capacitor values were chosen:

$$R_1 = 1800\Omega,\ R_2 = 2200\Omega,\ C_1 = C_2 = 0.1\mu\text{F}$$

These quantities yield the following values:

$$f_c = \frac{1}{2\pi}\sqrt{\frac{1}{R_1R_2C_1C_2}}$$

$$f_c = \frac{1}{2\pi}\sqrt{\frac{1}{(1800\Omega)(2200\Omega)(0.1\cdot10^{-6}\text{F})^2}} = 799.8\text{Hz}$$

$$Q = \frac{1}{\omega_c(R_2C_2(1-G) + R_1C_1 + R_1C_2)}$$

$$Q = \frac{1}{(2\pi)(799.8\text{Hz})(1800\Omega)(0.1\cdot10^{-6}F)(2)} = 0.55$$

The next step in the design process is to verify the values of the system components numerically. To do this, the transfer function must be converted into a plottable function with real numbers (Note here that G = 1):

$$H(j\omega) = \frac{(j\omega)^2}{(j\omega)^2 + (j\omega)\frac{\omega_c}{Q} + \omega_c^2}$$

$$H(j\omega) = \frac{-\omega^2}{-\omega^2 + (j\omega)\frac{\omega_c}{Q} + \omega_c^2}$$

Next, the numerator and denominator are reduced:

$$H(j\omega) = \frac{-\omega^2}{-\omega^2(1 - j\frac{\omega_c}{\omega Q} - \frac{\omega_c^2}{\omega^2})}$$

$$H(j\omega) = \frac{1}{1 - j\frac{\omega_c}{\omega Q} - \frac{\omega_c^2}{\omega^2}}$$

Now, preparing to find the magnitude, the denominator is converted into the form a + bj:

$$H(j\omega) = \frac{1}{(1 - \frac{\omega_c^2}{\omega^2}) - j\frac{\omega_c}{\omega Q}}$$

Finally, the theoretical magnitude is determined:

$$|H(j\omega)| = H(\omega) = \sqrt{\frac{1}{(1 - \frac{\omega_c^2}{\omega^2}) - j\frac{\omega_c}{\omega Q}} \cdot \frac{1}{(1 - \frac{\omega_c^2}{\omega^2}) + j\frac{\omega_c}{\omega Q}}}$$

$$|H(j\omega)| = H(\omega) = \frac{1}{\sqrt{(1 - \frac{\omega_c^2}{\omega^2})^2 + (\frac{\omega_c}{\omega Q})^2}}$$

Using actual values from the design, the following plottable function is determined:

$$H(f) = \frac{1}{\sqrt{(1 - \frac{639653}{f^2})^2 + (\frac{800}{0.55f})^2}}$$

D. Low Pass Filter

Write about design and analysis of the LPF here. First,

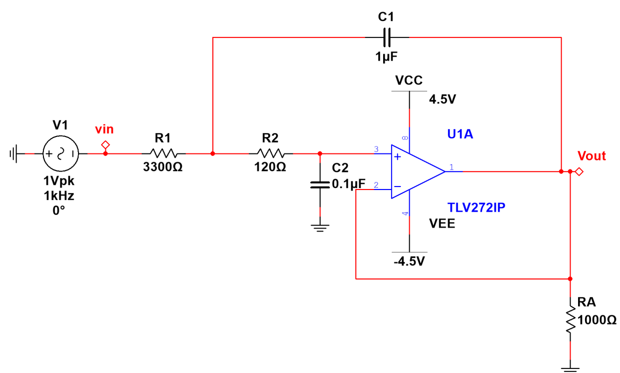

Fig. 5. Sallen Key Low-Pass Filter Schematic

the transfer function for a low pass Sallen-Key Filter has the following form:

$$H(s) = \frac{G\omega_c^2}{s^2 + s\frac{\omega_c}{Q} + \omega_c^2}$$

Through KCL analysis and lots of very tedious algebra, a similar form can be created from the system components:

$$H(s) = \frac{\frac{G}{R_1R_2C_1C_2}}{s^2 + s(\frac{1-G}{R_2C_2} + \frac{1}{R_1C_1} + \frac{1}{R_2C_1}) + \frac{1}{R_1R_2C_1C_2}}$$

Next, the fundamental quantities specified in the design constraints are derived from these equations, in terms of system components:

$$\omega_c = \sqrt{\frac{1}{R_1R_2C_1C_2}}$$

$$G = \frac{R_A + R_B}{R_A}$$

$$Q = \frac{\omega_c}{\omega_c^2(R_1C_1(1-G)+R_2C_2+R_1C_2)}$$

$$Q = \frac{1}{\omega_c(R_1C_1(1-G)+R_2C_2+R_1C_2)}$$

Now, noting the design constraints of the gain K = 0, the Q value 0.5 $<$ Q $<$ 0.707, and the corner frequency f_c = 800Hz, the component values become more clear. First, the gain:

$$K = 20\log(G) = 0 \rightarrow G = 1$$

If the linear gain is unity, this means that one of the resistors in the feedback loop must be a short:

$$G = \frac{R_A + R_B}{R_A} = 1 \rightarrow R_A = (1000\Omega)\ R_B = 0\Omega\ \text{(short)}$$

Next, through optimization analysis in MATLAB, the following resistor and capacitor values were chosen:

$$R_1 = 3300\Omega,\ R_2 = 120\Omega,\ C_1 = 1\mu\text{F},\ C_2 = 0.1\mu\text{F}$$

These quantities yield the following values:

$$f_c = \frac{1}{2\pi}\sqrt{\frac{1}{R_1R_2C_1C_2}}$$

$$f_c = \frac{1}{2\pi}\sqrt{\frac{1}{(3300\Omega)(120\Omega)(1\cdot10^{-6}\text{F})(0.1\cdot10^{-6}\text{F})}}$$

$$f_c = 799.8\text{Hz}$$

$$Q = \frac{1}{\omega_c(R_1C_1(1-G)+R_2C_2+R_1C_2)}$$

$$Q = \frac{1}{(2\pi)(799.8\text{Hz})(3300\Omega+120\Omega)(0.1\cdot10^{-6}F)} = 0.58$$

The next step in the design process is to verify the values of the system components numerically. To do this, the transfer function must be converted into a plottable function with real numbers (Note here that G = 1):

$$H(j\omega) = \frac{\omega_c^2}{(j\omega)^2 + (j\omega)\frac{\omega_c}{Q} + \omega_c^2}$$

$$H(j\omega) = \frac{\omega_c^2}{-\omega^2 + (j\omega)\frac{\omega_c}{Q} + \omega_c^2}$$

Now, factoring the square of the corner angular frequency out of the denominator, we have:

$$H(j\omega) = \frac{\omega_c^2}{\omega_c^2(1 - \frac{\omega^2}{\omega_c^2} + \frac{j\omega}{\omega_c Q})}$$

$$H(j\omega) = \frac{1}{1 - \frac{\omega^2}{\omega_c^2} + \frac{j\omega}{\omega_c Q}}$$

Now, preparing to find the magnitude, the denominator is converted into the form a + bj:

$$H(j\omega) = \frac{1}{(1 - \frac{\omega^2}{\omega_c^2}) + j\frac{\omega}{\omega_c Q}}$$

Finally, the theoretical magnitude is determined:

$$|H(j\omega)| = H(\omega) = \sqrt{\frac{1}{(1-\frac{\omega^2}{\omega_c^2}) + (j\frac{\omega}{\omega_c Q})} \cdot \frac{1}{(1-\frac{\omega^2}{\omega_c^2}) - (j\frac{\omega}{\omega_c Q})}}$$

$$|H(j\omega)| = H(\omega) = \frac{1}{\sqrt{(1-\frac{\omega^2}{\omega_c^2})^2 + (\frac{\omega}{\omega_c Q})^2}}$$

Using actual values from the design, the following plottable function is determined:

$$H(f) = \frac{1}{\sqrt{(1-\frac{f^2}{639653})^2 + (\frac{f}{(799)(0.58)})^2}}$$

E. Peak Detector

The peak detector will generate DC-like signals that will be used to drive the LED at different intensities depending on the amplitude of the audio signal. In order to implement this we use a precision half wave rectifier using an op-amp and a diode. That will get rid of the bottom half of a signal, so we now add a resistor and capacitor at output so that the peak voltage is preserved for a certain amount of time, thus producing a somewhat constant signal. Bias resistors are then added at the input to offset the magnitude of the peaks. The offset will depend on the DC bias that each distinct transistor needs to be just under threshold to have the LED off when there is no audio signal. Those will be designed in the next section along with the LED Driver. As for the rest of the detector, the input capacitor simply needs to be a big friendly capacitor to block any DC signals, so we choose $C_{12} = 10\mu F$. For the output capacitor and resistor we must choose values such that the duration of the peaks are long enough to be detected by the human eye. We choose $R_6 = 6.8k\Omega$ and $C_{12} = 100\mu F$.

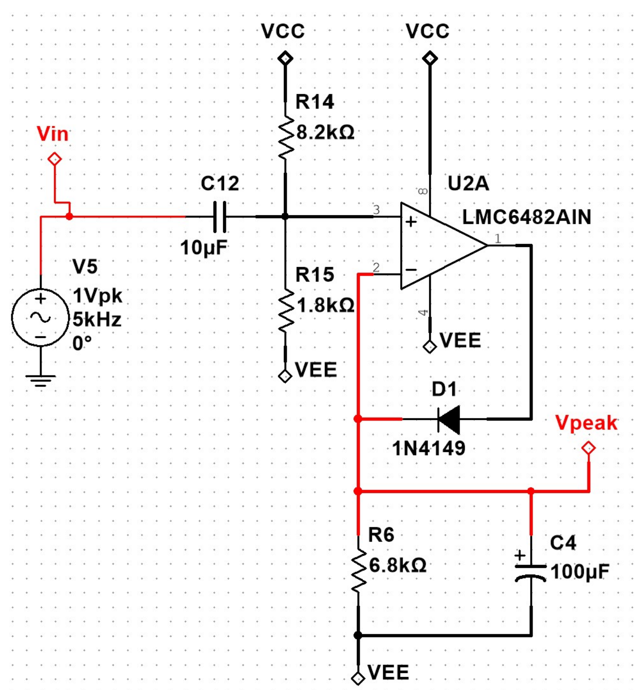

Fig. 6. Low Pass Filter Peak Detector Circuit Schematic

F. LED Driver

The design here will vary for the low-pass input and high-pass input. First we take a look at the design for the low-pass

filter LED driver:
The MOSFET used for the analysis and design in this section requires a gate voltage of about -2.6 V to be just below the threshold of turning on the LED. Therefore, we will bias the gate so that it's at about -2.9 V, a little lower than the threshold so that the LED only turns on when the audio signal is loud enough. To select the bias resistors we perform KCL at the input of the peak detector:

$$\frac{V_{in} - V_{EE}}{R_{15}} + \frac{V_{in} - V_{CC}}{R_14} = 0$$

$$\frac{-2.9 + 4.5}{R_{15}} + \frac{-2.9 - 4.5}{R_{14}} = 0$$

$$\frac{1.6}{R_{15}} = \frac{7.4}{R_{14}}$$

$$\frac{R_{14}}{R_{15}} = 4.63$$

So we choose the closest values available in our component kits:

$$R_{14} = 8.2k\Omega, R_{15} = 1.8k\Omega$$

To be in the safe operating range of the LED, we assume that the maximum current will be 10 mA. With that we can solve for the V_{GS} needed to obtain that current.

$$K_N = \frac{2i_D}{(v_{GS} - V_t)^2}$$

Rearranging the variables to get VGS by itself:

$$V_{GS} = \sqrt{\frac{2i_D}{K_N}} + V_t$$

Plugging in 10mA for i_D, 0.107 for KN and 1.98 for Vt:

$$V_{GS} = \sqrt{\frac{2(0.01)}{0.107}} + 1.98 = 2.41V$$

By analyzing the song with the oscilloscope, one will find that the loudest parts of the song will peak at 2.5 V, which means that the maximum voltage a the gate will be $V_G = -0.4V$. With that information we can solve for V_S and R_7.

$$V_S = V_G - V_{GS} = -0.4 - 2.41 = -2.81V$$

Now we can solve for the source resistance:

$$V_s - V_{EE} = I_D R_7$$

$$R_7 = \frac{V_s - V_{EE}}{I_D} = \frac{1.69}{0.01} = 169\Omega$$

Therefore we choose $R_7 = 180\Omega$ from our kit.
Now we repeat the same process for the LED driver for the high-pass filter. This time the MOSFET requires a gate voltage of about -2.9 V to keep the LED off, so we choose $V_G = -3.2V$. Performing KCL in the same manner we get the following resistor values:

$$R_{16} = 6.8\Omega K$$

$$R_{17} = 1.2k\Omega K$$

Using the expression to solve for V_{GS} at 10 mA using $K_N = 0.032$ and $V_T = 1.2$:

$$V_{GS} = \sqrt{\frac{2(0.01)}{0.032}} + 1.2 = 1.93V$$

For the high frequency portion, the peak voltage is at about 0.7 V, so the maximum voltage at the gate will be $V_G = -2.1V$. Now solving for source voltage and resistance:

$$V_S = V_G - V_{GS} = -2.1 - 1.93 = -4.03V$$

$$V_s - V_{EE} = I_D R_{13}$$

$$R_{13} = \frac{V_s - V_{EE}}{I_D} = \frac{0.47}{0.01} = 47\Omega$$

Therefore we choose $R_{13} = 47\Omega$ from our kit. (Designs for this section will vary from member to member).

IV. Simulations

This section should show your simulations. Break it into 3 parts: 1) DC bias point simulations that show the DC performance with no time varying signals at the input. 2) Frequency dependent AC simulations of each block and the entire project (these will be a bunch of bode plots) and 3) Response to your songs throughout the circuit (these will be a bunch of signals in the time domain as it progresses from the input voltage to the output current going through each LED.

A. Summing Amplifier

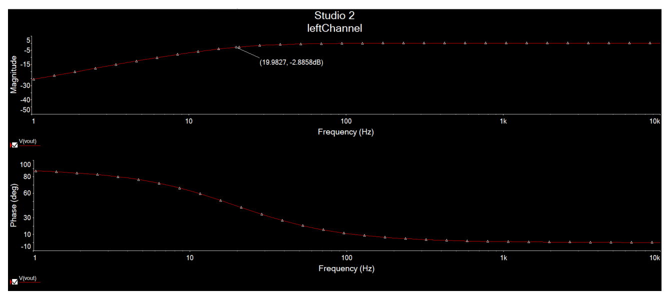

Fig. 7. AC Sweep with only the left channel and it verifies that is has the correct corner frequency and gain.

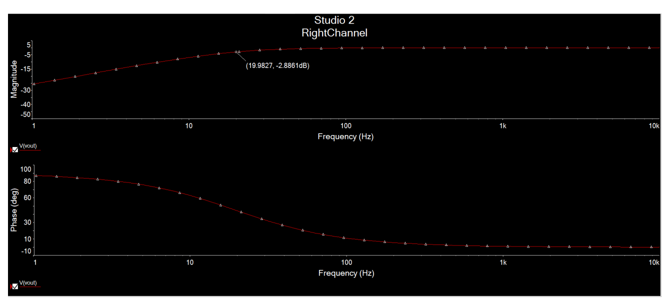

Fig. 8. AC sweep with only the right channel and it verifies that it has the correct corner frequency and gain.

Figure 8 and figure 7 show the response of the summing amplifier at various frequencies. As indicated by the label, the cutoff frequency is at 19.8 Hz, which means that signals below that frequency will not pass through the amplifier just as desired.

B. High Pass Filter

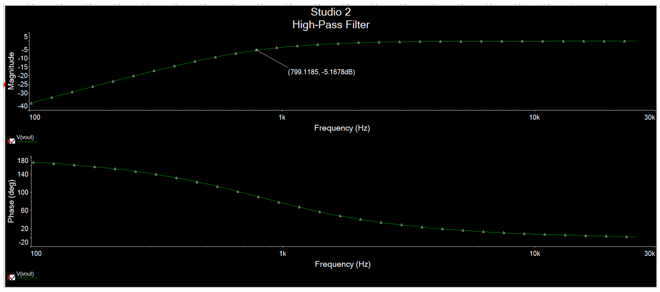

Fig. 9. Sallen Key Low-Pass Filter Multisim Simulation

C. Low Pass Filter

Show your simulations of the LPF here.

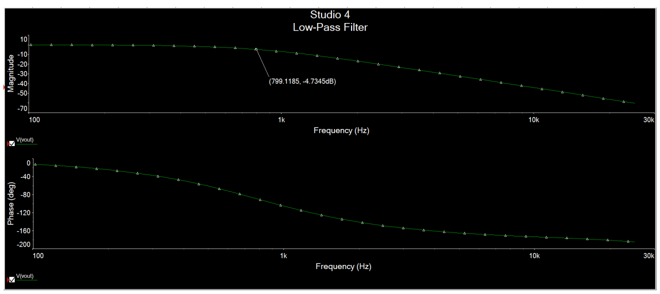

Fig. 10. Sallen Key Low-Pass Filter Multisim Simulation

D. Peak Detector

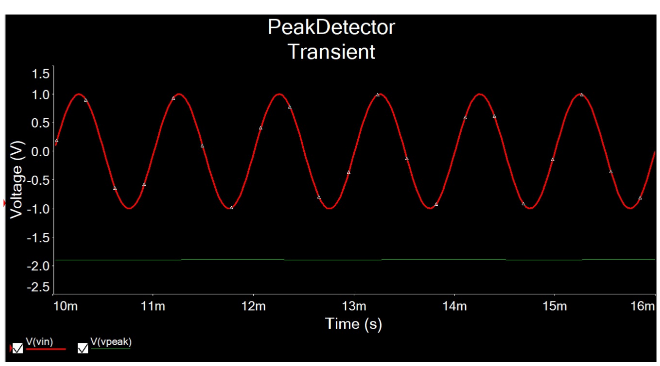

Fig. 11. Low Pass Filter Peak Detector Simulation with 1 V Sinusoidal Input at 1 kHz

As expected, the output signal is almost perfectly constant and offset by -2.9 V as designed.

E. LED Driver

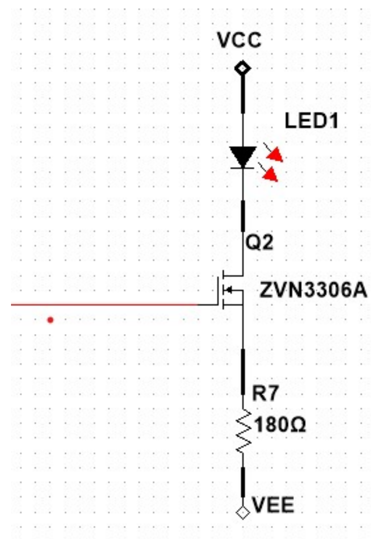

Fig. 12. LED Blinking with Input signal

F. System Simulations

The components designed in the previous sections are put together to test their behavior. A sinusoidal signal with 0.5 V amplitude at 500 Hz is used as the left input while a signal with 0.2 V at 1.5 kHz is used for the right input. Here is the output of the summed signals:

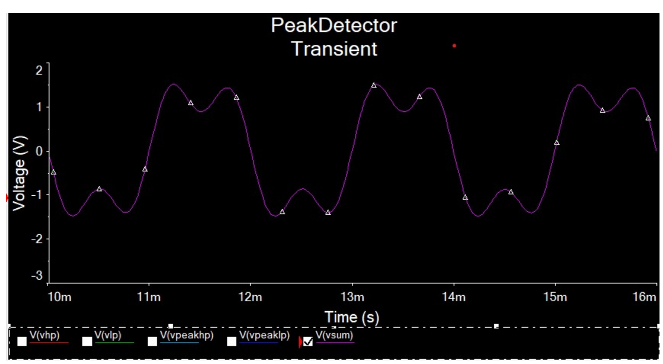

Fig. 13. Summer Amplifier Output

After that the signal is passed through a high-pass and low-pass filter. As seen in Figure 12, the summed input is split into a low and high frequency signal. Those are then fed into their respective peak detectors:

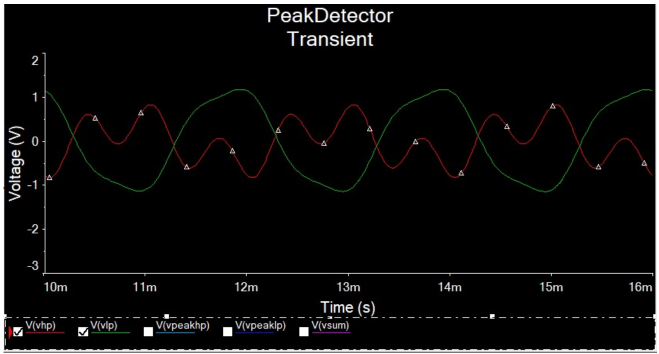

Fig. 14. Low-pass(green) and High-Pass(red) Filter Output

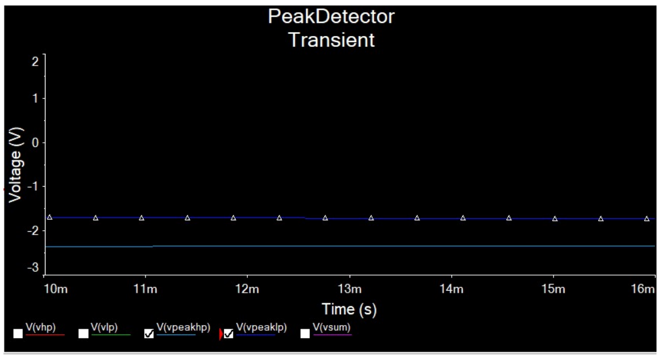

Fig. 15. Peak Detector Output

V. Experimental Results

A. Completed Circuit on PCB

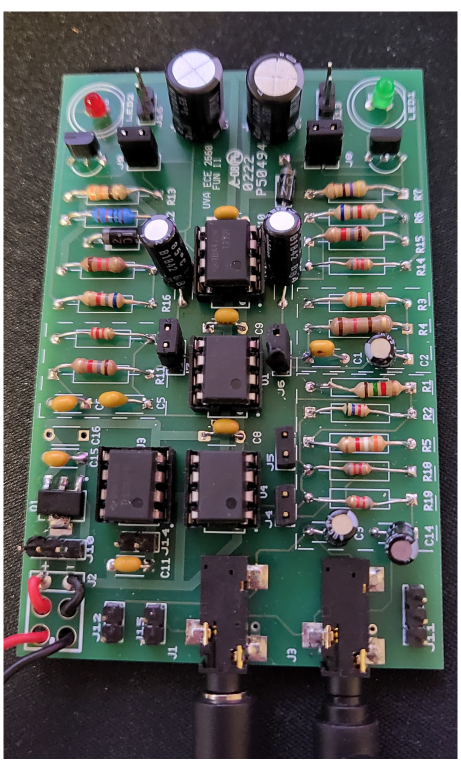

Fig. 16. Front of the PCB with all components soldered in

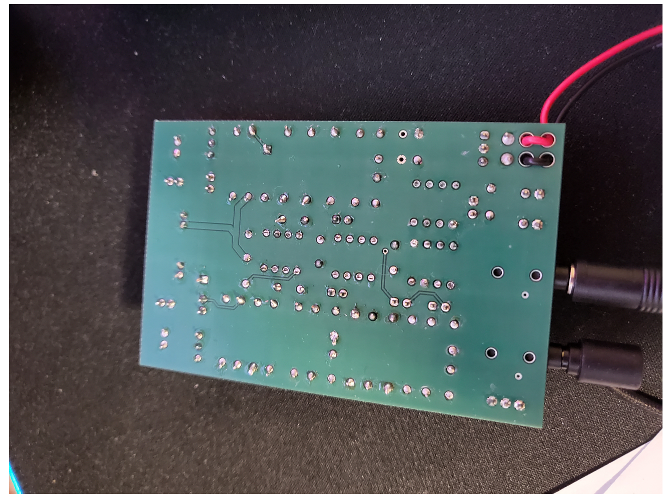

Fig. 17. Back of the PCB with all components soldered in

B. Summing Amplifier

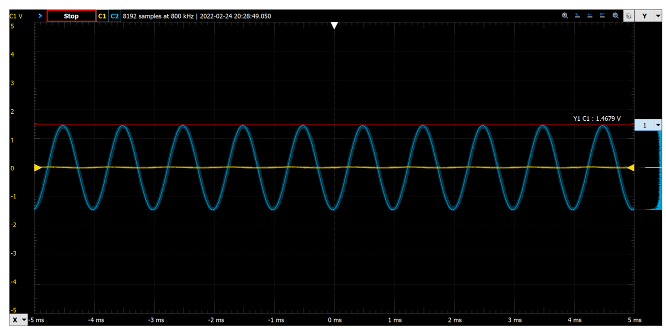

Fig. 18. A wavegen signal with amplitude 1V and frequency 1kHz, inputted into the left channel, produced this response.

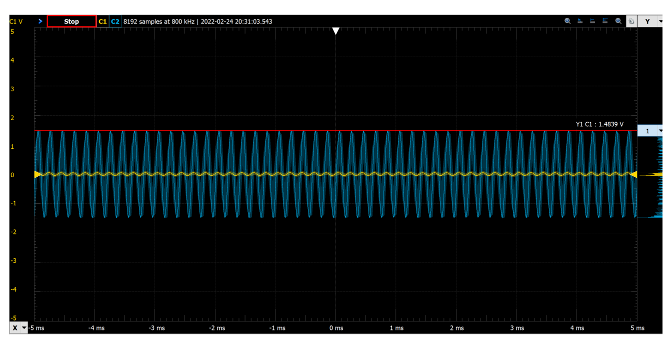

Fig. 19. A wavegen signal with amplitude 1V and frequency 5kHz, inputted into the left channel, produced this response.

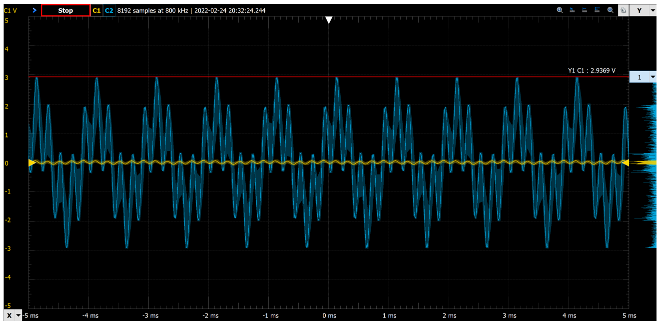

Fig. 20. Signals from the previous two figures were combined to produce this response. The gain is very close to 3 as specified in the design.

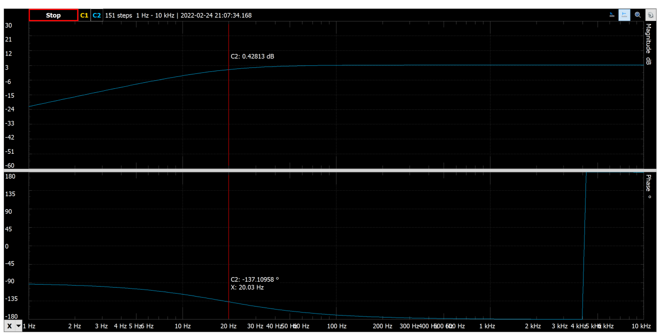

Fig. 21. Signals of various frequencies were tested with the left channel input to verify its DC blocking capability. Note that the passband starts around the design-specified corner frequency of 20Hz. This effectively blocks DC.

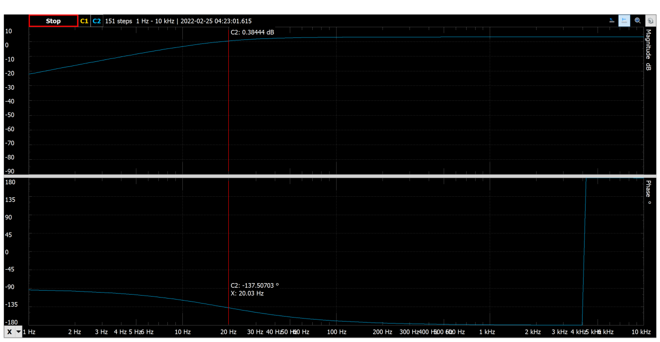

Fig. 22. Signals of various frequencies were tested with the right channel input to verify its DC blocking capability. Note that the passband starts around the design-specified corner frequency of 20Hz. This effectively blocks DC.

C. High Pass Filter Experimental Results

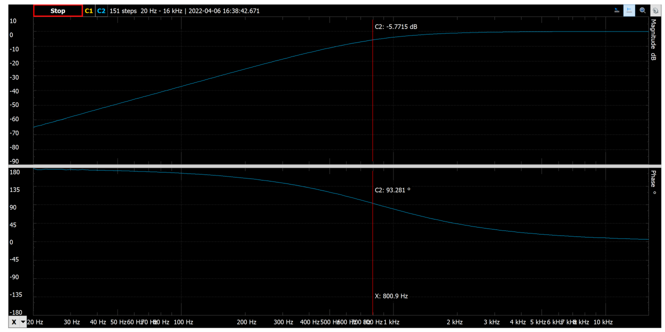

Fig. 23. Once soldered into the pcb, the bode plot of the Sallen-Key High Pass Filter was tested experimentally. The results show a cutoff frequency of about 800Hz for a Q value of 0.58, as specified in the design

D. Low Pass Filter Experimental Results

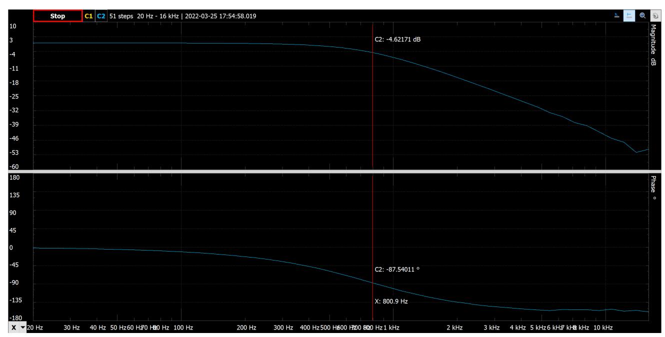

Fig. 24. Once soldered into the pcb, the bode plot of the Sallen-Key High Pass Filter was tested experimentally. The results show a cutoff frequency of about 800Hz for a Q value of 0.55, as specified in the design

E. Peak Detector Experimental Results

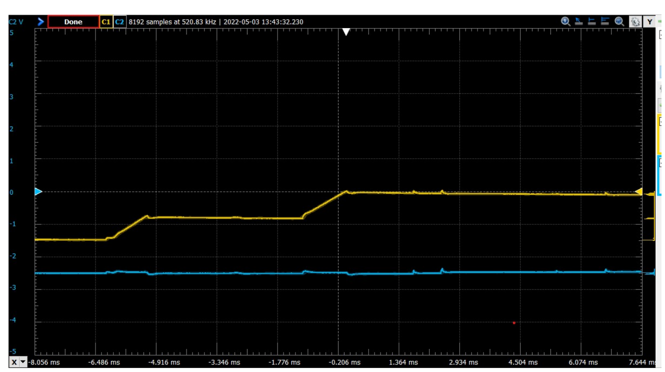

Fig. 25. Peak Detector Output With Song Input

F. LED Driver Experimental Results

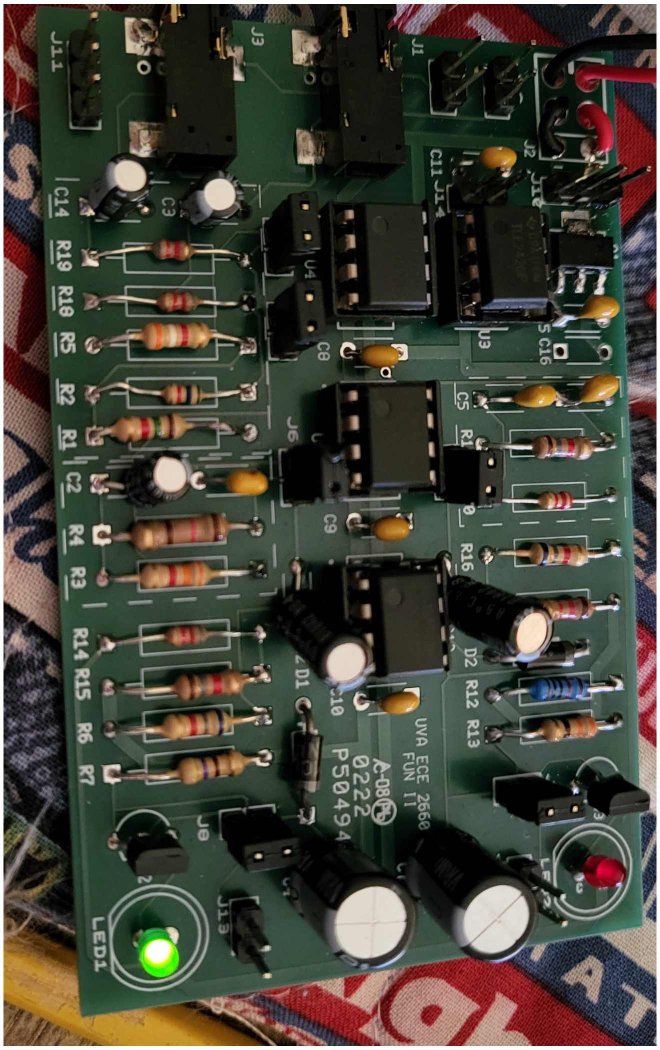

Fig. 26. LED Lighting up with Peak Detector Input

G. System Level Experimental Results

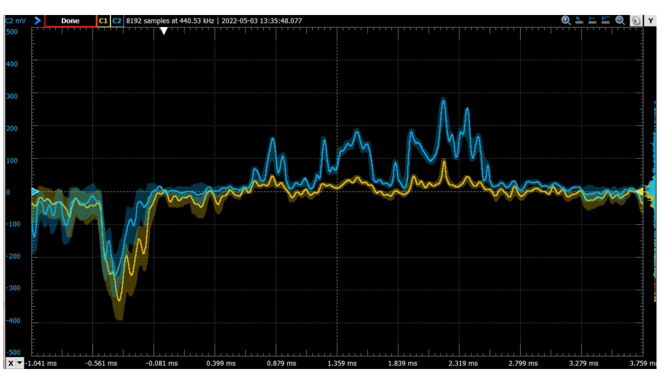

Fig. 27. Song Left and Right Input

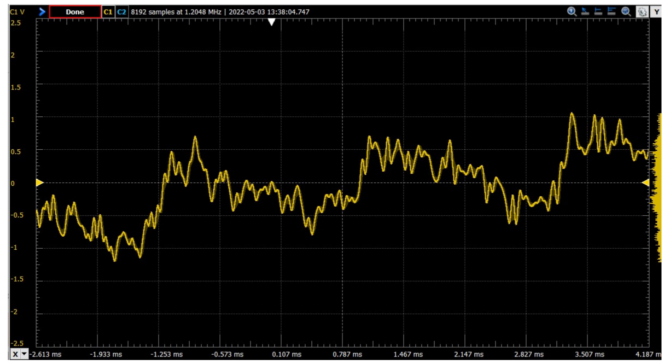

Fig. 28. Left and Right Input Summed

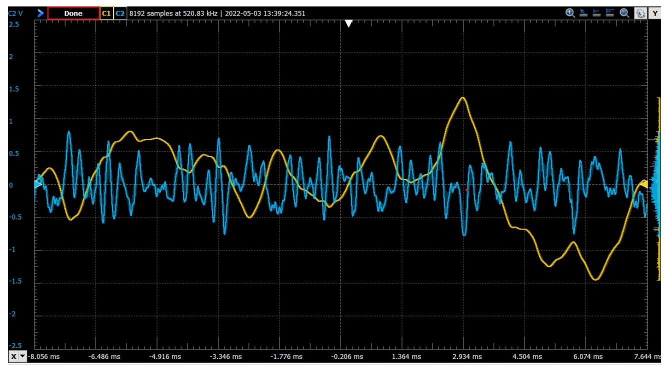

Fig. 29. Summed Song Signal After Low-pass and High-pass Filters

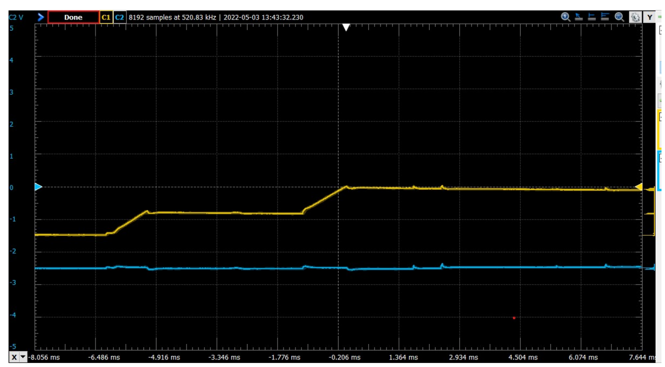

Fig. 30. Peak Detector Output of High and Low Pass Filter

VI. Challenges and Workarounds

- There was no output from the summer amplifier when using the headphone jack. First I checked all the components of the amplifier and made sure it was getting powered but found nothing. It turns out that one of the leads of my headphone jack was not soldered in completely. After re-soldering, the output worked as desired.
- It was really hard to derive the peak detector and LED driver analytically as the MOSFETS had different characteristics, so their behaviors weren't really matching up with what was predicted analytically. The solution to this was to base the design of the peak detector and LED driver by looking at the behavior of the MOSFET on the breadboard.

- Initially our Sallen-Key filters had a Q value that was too low, so we had to resolder some resistors to correct that
- We had trouble with images not appearing on the right subsections. We fixed this by importing 3 additional packages to fix the image formatting.

VII. Conclusions and Thoughts for Future Classes

Although difficult and time consuming, this project was very enjoyable and extremely rewarding as a learning activity. The first few sections of the project were clearly defined and it was hard to get lost; however, the last sections like the LED driver had many parameters that varied from person to person and group to group. This degree of freedom left a lot of room for us to really think about the design and how to implement it. Some of the many things that we learned from this project include how audio signals work and how we can manipulate them with different components to achieve a certain behavior. The most critical concept in this project was how to properly configure op-amps, as we learned to use them as amplifiers, filters, and peak detectors. We also gained a better understanding with regards to the varying behavior of a MOSFET to use it as an electronic switch for our LED. A piece of advice that I would give to future students is to be patient when something is not working right. One of the most important skills for this project is definitely troubleshooting. Beep checking and output testing were our most useful methods of discovering soldering and design issues. Taking a step back to think about what could be the problem and then testing out your circuit individually by sections to diagnose the problem is essential.

VIII. Collaboration Statement

Ezemet Burkut's Contributions: Worked on background information and rationale, summing amplifier, analysis of the frequency spectrum of the song, high pass filter, low pass filter, simulations, experimental results, conclusion.

Kousuke Tapia's Contributions: Worked on summing amplifier, analysis of the frequency spectrum of the song, high pass filter, low pass filter, simulations, experimental results, conclusion, peak detector, led driver

James O'Coneel's Contributions: Worked on the abstract, summing amplifier, analysis of the frequency spectrum of the song, high pass filter, low pass filter, simulations, experimental results, conclusion.