Final Project (Part 2): Audio Spectrum Display

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INTRODUCTION

The goal of this project was to build circuit which displays the spectrum of a given audio signal in real time. The circuit will have three frequency sections (bass, mid, treble), and four LEDs per section. These LEDs will indicate how strong the given frequency range is in real time. For example, if there's little bass but a lot of mid frequencies at a given time, then only the bottom-most bass LED may be on, while three of the mid LEDs may be on. In other words, each LED will have a different cutoff. Since there are four LEDs, these cutoffs were chosen to be multiples of fifths of power. Specifically, the bottom LED will turn on if the frequency is at or above 1/5 of max power, the next LED will turn on if the frequency is at or above 2/5 of max power, the third at 3/5 max, and the fourth at 4/5 max.

Furthermore, the intended behavior of this circuit is that the LEDs will also get brighter as there is more of the given frequency. So not only will an LED turn on at a given power level, but its brightness will also depend on how high the power is above that level.

The circuit will also be able to be tuned using a potentiometer; if this feature wasn't available, then a quiet song would always only turn on one or two LEDs, so it would not show the relative strengths of the frequencies very well. In other words, the user will be able to essentially set the "max power" value, and in turn will change the cutoffs of the LEDs.

An example visual of the final product is shown in Fig. 1 below.

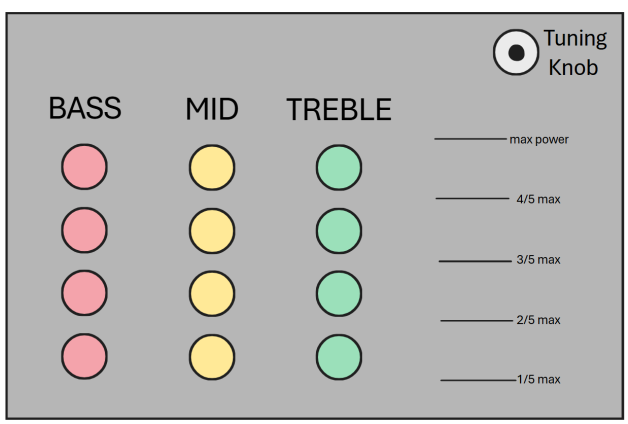

FIG. 1. Example of final product

CIRCUIT DESIGN

Input Amplification

Firstly, the audio input was received from a computer using a pair of wired earbuds. The wires were cut and stripped, so they could be accessed and plugged into the breadboard. The first step, before sending the song to all the modules, was building a variable amplifier. The reason for this was described in the introduction; if a song is quiet, then the circuit display will not be sensitive enough to give a reasonable output. We will design the circuit using the maximum possible earbud output amplitude as our reference. This way, if the user is playing a song that reaches this amplitude, then they don't need any gain, but if they are playing a song that doesn't reach this amplitude, then they will need some gain. Therefore, this amplifier will have a minimum gain of 1, and a large maximum gain (reasonably, at least 10). Therefore, we used a non-inverting amplifier, since this has a minimum gain of 1. For the feedback resistor, we used a potentiometer, which could range from 0Ω to $100\text{k}\Omega$. For the other resistor in the amplifier, we used $10\text{k}\Omega$ (see Fig. 2). This gives a maximum gain of 11, which should be sufficient for the vast majority of cases. If we made this resistor too small (like 1k), then the sensitivity of the tuning knob would be so large that it would be very hard to tune precisely (since a smaller change in the potentiometer resistance would cause a larger change in gain). The 10k value acts as a reasonable balance between allowing high enough gain and having reasonable tuning sensitivity.

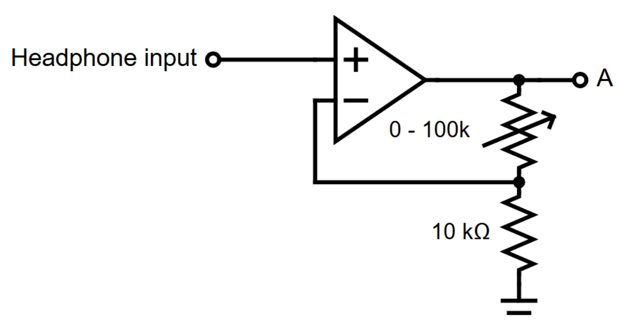

FIG. 2. Variable amplifier. Gain range is 1 to 11.

Filtering

The output of the amplifier (point A in Fig. 2) goes into the filtering stage of all three modules. The type of filters used in this project were second-order Sallen-Key style filters, with a Butterworth style response curve. This type of filters have several benefits compared to a standard passive filter:

- Sharp Roll-Off: The two poles (second order) lead to a -40 dB/decade roll-off outside of the pass-band, as opposed to the standard -20 dB/decade roll-off of a standard single pole filter.

- Active: This type of filter uses an op-amp (active), which allows for better output impedance, as well as more controlled (and higher) gain.

- Butterworth Response: The Butterworth response shape can be thought of as the "standard" filter response shape. The ideal Butterworth response has a fixed gain in the pass-band, exhibits half power at the cutoff frequency, and has a consistent roll-off slope per decade.

In order to achieve the Butterworth response with the two-pole Sallen-Key filter design, the gain must be set to about 1.586. Therefore, we used resistor values of 5.6k and 10k, which leads to a gain of 1.56 (which is within 10% of the desired value). The filter for the bass module is shown in Fig. 3 below. This filter is of low-pass type (since it must pass the low bass frequencies), with a cutoff frequency of about 234 Hz (the bass range cutoff is about 250 Hz).

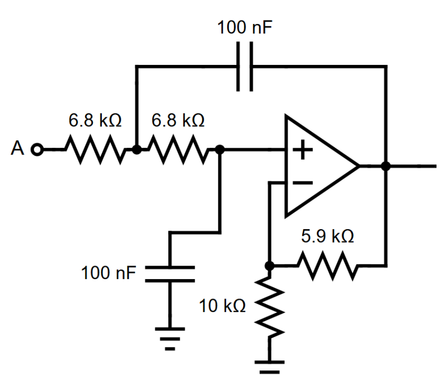

FIG. 3. Low-pass second-order Sallen-key filter for bass module. Cutoff frequency $f_c = 234$ Hz.

The filter for the treble module is high-pass filter of the same type (two-pole Sallen-Key), with a cutoff frequency of about 5.88 kHz (the treble range starts at about 6 kHz). The mid-range module is a bit more complicated, since it needs to pass a band of frequencies, and attenuate frequencies on both sides. This was achieved by cascading a high-pass filter and a low-pass filter. The high-pass filter has a cutoff frequency of about 284 Hz and the low-pass filter has a cutoff frequency of about 4.82 kHz. Realistically, the mid range upper cutoff should have been a bit lower (anywhere from about 2 kHz to 4 kHz), however the values used in this project were sufficient to demonstrate the concept.

Amplification & Rectification

After the signal is filtered, it is then amplified slightly (and then buffered), in order to generate sufficient voltage for the upcoming diode drop. For the bass and treble modules, this amplifier had a gain of 2.2 (see Fig. 4), while for the mid-range module, the amplifier had a gain of about 1.4. The reason this is the case is because the signal was double-amplified by the two filters in this mid-range module, so it already had a large amplitude.

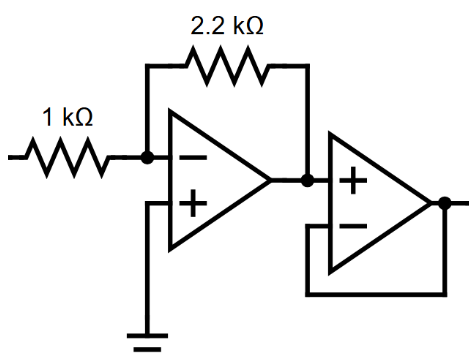

FIG. 4. Post-filtering amplifier (and buffer) for bass and treble modules. Gain = 2.2

At this point, we have a signal whose amplitude depends on the strength of the given frequencies in the original input signal. If we want to drive LEDs based on this amplitude, then we need to rectify it. This way, a higher amplitude will lead to more DC to drive the LEDs. This was simply achieved using a diode as a half-wave rectifier, followed by a smoothing capacitor and resistor in parallel to GND (see Fig. 5).

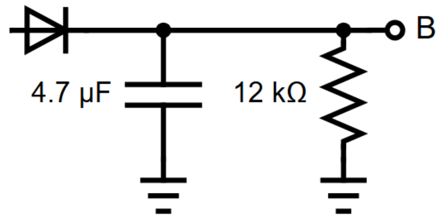

FIG. 5. Half-wave rectifier + smoothing. The R and C values here are for the bass module.

Regarding the RC time constant, this value depended on the frequency range in question. For the bass module, we used a time constant of roughly 56 ms. This relatively large time constant is due to the fact that bass frequencies can get quite low, so they can have large periods. For example, a frequency of 20 Hz has a period of 50 ms, and the time constant must be large enough to reasonably smooth any frequency in the given range. For the mid and treble modules, we were able to use a much smaller time constant of about 3.3 ms, since any frequency above about 300 Hz will have a period smaller than this. The reason we were able to use the same time constant for both the mid and treble modules is because there is no downside to using this time constant in the treble range, since it still leads to a very fast response to changes in amplitude. In other words, the only downside of using an extremely large time constant is that it will cause a lagging response to a quick change in signal amplitude. Since 3.3 ms is small enough to lead to a reasonably quick response time, it can be used for both modules.

LED Drivers

At this point, we have a DC signal with some ripple, with the DC value set based on the strength of the target frequencies in the input signal. We can drive LEDs with this signal through the use of MOSFETs. We know that a MOSFET only turns on at some minimum gate-source voltage. Therefore, we can precisely set each LED cutoff by setting the source voltage to some constant.

For example, consider a case in which the MOSFET starts to conduct enough current to turn the LED on when the gate-source voltage reaches 2V. We can connect an LED (and a current-limiting resistor) from VCC to the drain. Then we can connect our incoming DC signal to the gate. If we want the LED to turn on only if the signal is at or above 1.5V, then we can set the source voltage to -0.5V using a voltage divider. This causes the gate-source voltage to reach 2V only when the incoming signal reaches 1.5V. However, before putting the -0.5V directly to the source pin, we need to buffer it through an op-amp, so that the op-amp sinks the current. If this op-amp wasn't present, then the current would flow through the voltage-divider resistors, which would affect the voltage set by the divider.

In summary, we now have a setup that causes the gate-source voltage to reach 2V or above when the incoming DC signal reaches 1.5V or above; the MOSFET starts to conduct enough current to turn on the LED only when this is the case. This current flows from VCC, through the LED, through the MOSFET, and into the op-amp (hence the op-amp sinking the current). This setup is shown in Fig. 6. In the figure, R1 can be precisely set to choose the specific cutoff voltage at which the LED will turn on.

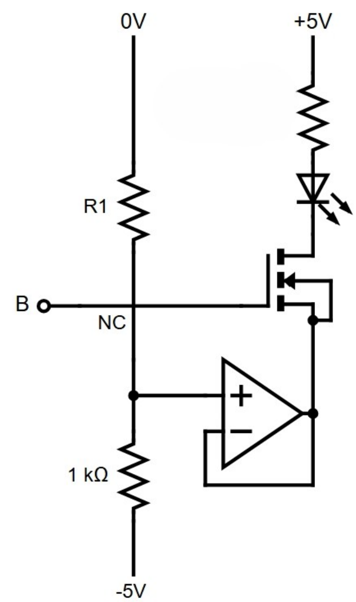

FIG. 6. LED driver circuit stage. The value of R1 determines the minimum voltage at which the LED turns on. The unlabeled resistor is a current-limiting resistor.

In this project, these R1 resistor values were chosen experimentally (i.e. by setting a frequency to the desired fraction of max power, and testing various resistance values until the given LED started to turn on). However, in the Results section, we will perform some calculations to determine the theoretically optimal resistor values for each LED, and compare these to the experimentally determined values.

Summary

Shown in Fig. 7 below is a schematic of the circuit stages. However, it should be noted that these stages are present more than once. For example, the middle section of the schematic (the filter, amplifier, and rectifier) is present in each of the three modules. Additionally, there are four LED drivers in each module, each with a different threshold value (leading to 12 total LED driver modules).

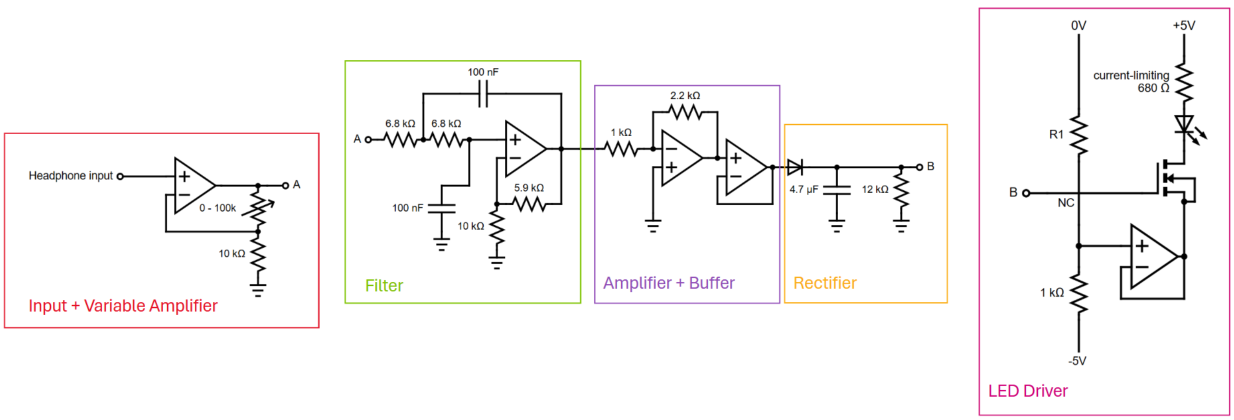

FIG. 7. Summary of all circuit stages. Note: Many of these stages are present in multiple places.

Shown in Fig. 8 is the full breadboarded circuit, with each stage indicated using the color scheme of Fig. 7. As has been neglected up to this point, we also included a speaker on the breadboard to be able to hear the music along with watching the LEDs. This speaker was simply implemented by using a non-inverting amplifier to drive it. The input to the amplifier was taken from point A in Fig. 7.

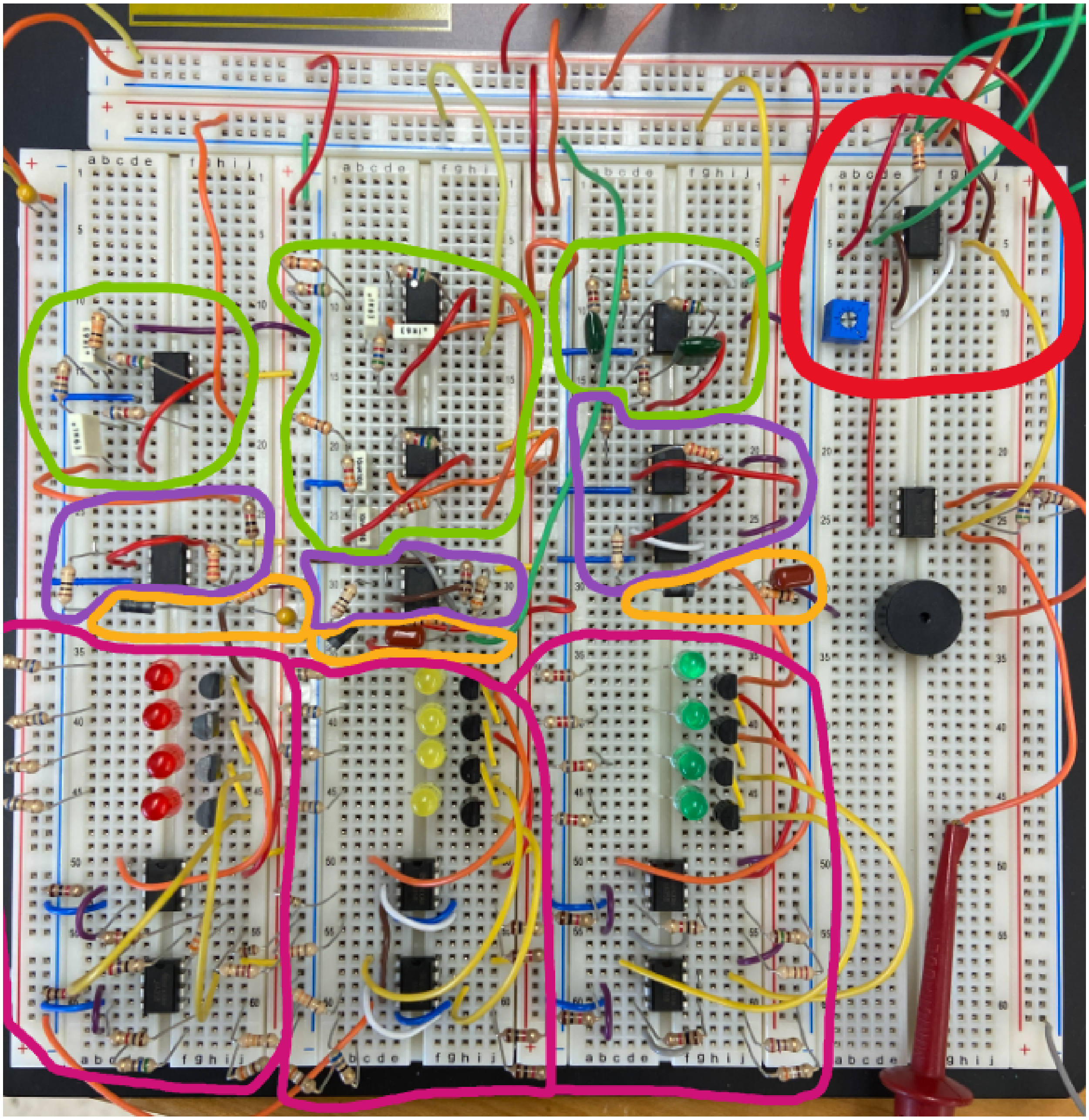

FIG. 8. Full breadboard. Stages are labeled with same color scheme as Fig. 7.

RESULTS

Filter Measurements

To quantitatively determine how the filter behavior compares to theoretical predictions, we measured several data points (gain vs. frequency) for each module. Pure sine waves were fed into the filters to gather these measurements. The results are shown in Table I, Table II, and Table III below. The uncertainties can be said to be implied by the number of significant figures.

TABLE I. Mid module filter data; post-filter amplitude at various frequencies. Gains were calculated relative to the pre-filter amplitude of 1.71.

Frequency (Hz)	**Post-Filter Amplitude (V)**	**Gain**
250	2.37	1.39
285	2.75	1.61
300	2.89	1.69
350	3.28	1.92
400	3.51	2.05
500	3.78	2.21
700	3.94	2.30
900	4.00	2.34
1500	4.00	2.34
2000	3.95	2.31
2500	3.85	2.25
3000	3.70	2.16
3500	3.50	2.05
4500	2.98	1.74
4820	2.81	1.64
5500	2.45	1.43

TABLE II. Bass module filter data; post-filter amplitude at various frequencies. Gains were calculated relative to the pre-filter amplitude of 1.71.

Frequency (Hz)	**Post-Filter Amplitude (V)**	**Gain**
50	2.59	1.51
100	2.57	1.50
200	2.17	1.27
235	1.93	1.13
245	1.83	1.07
300	1.46	0.85
350	1.17	0.68

TABLE III. Treble module filter data; post-filter amplitude at various frequencies. Gains were calculated relative to the pre-filter amplitude of 1.71.

Frequency (Hz)	**Post-Filter Amplitude (V)**	**Gain**
5000	1.53	0.89
5400	1.65	0.96
5820	1.80	1.05
6000	1.85	1.08
7000	2.08	1.22
8000	2.24	1.31
9000	2.34	1.37
10000	2.41	1.41
15000	2.53	1.48
20000	2.52	1.47

Using this data, we can find the experimentally determined values of gain and cutoff frequency for each filter, and compare them to the theoretically predicted values. This can be achieved using the scipy.optimize.curve_fit function in python. We know that second-order high-pass or low-pass Butterworth filters will obey the following response functions:

$$G_{\text{low\_pass}}(f) = \frac{G_0}{\sqrt{1 + (\frac{f}{f_c})^4}}$$

$$G_{\text{high\_pass}}(f) = \frac{G_0}{\sqrt{1 + (\frac{f_c}{f})^4}}$$

Where $G(f)$ is gain as a function of frequency, G_0 is the gain in the pass-band (which should be 1.56 in our filters, as mentioned in the Circuit Design section), and f_c is the cutoff frequency for the filter at question. If we feed the gain vs. frequency data into the curve_fit function, it will return the optimized values of G_0 and f_c for the function to best fit the measured data. Once these values are found (which will be deemed as the experimentally determined values), we can compare them to the theoretically predicted values for each filter. It's important to note that since the mid-range module cascaded two filters, then its response function will be a product of high-pass and low-pass response functions with different cutoff frequencies. Therefore, it will look like

$$G_{\text{bandpass}}(f) = \frac{G_{0,LP}}{\sqrt{1 + (\frac{f}{f_{c,LP}})^4}} \cdot \frac{G_{0,HP}}{\sqrt{1 + (\frac{f_{c,HP}}{f})^4}} = \frac{G_1}{\sqrt{(1 + (\frac{f}{f_{c,LP}})^4)(1 + (\frac{f_{c,HP}}{f})^4)}}$$

where G_1 is the total gain of the bandpass filter, which will be the product of the gains of the low-pass and high-pass filters.

Performing the curve fitting, we get the following results shown in Fig. 9, Fig. 10, and Fig. 11.

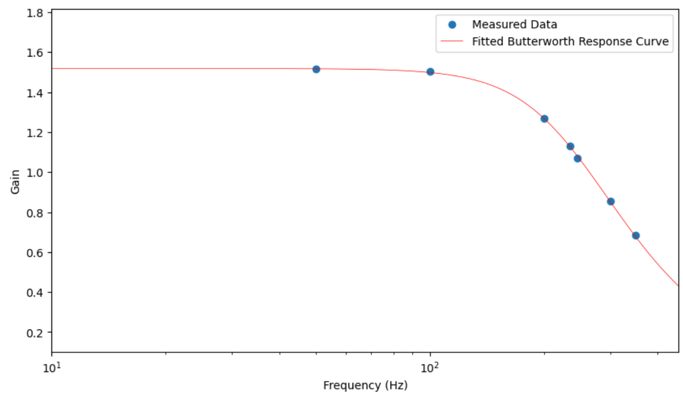

FIG. 9. Bass module filter; frequency vs. gain data and fitted curve. The parameters determined by the curve_fit function are $G_0 = 1.52$, $f_c = 247$ Hz.

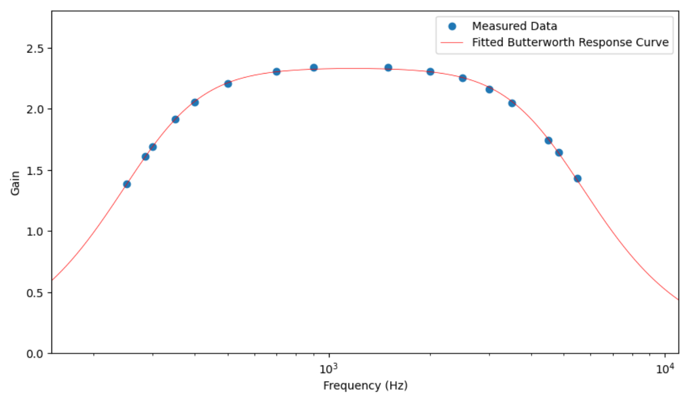

FIG. 10. Mid-range module filter; frequency vs. gain data and fitted curve. The parameters determined by the curve_fit function are $G_1 = 2.34$, $f_{c,\mathrm{HP}} = 293$ Hz, $f_{c,\mathrm{LP}} = 4.79$ kHz.

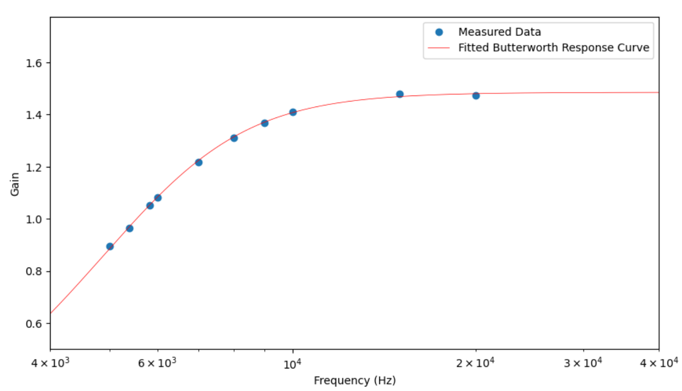

FIG. 11. Treble module filter; frequency vs. gain data and fitted curve. The parameters determined by the curve_fit function are $G_0 = 1.48$, $f_c = 5.81$ kHz.

The expected values for all of the fitted parameters are discussed in the Circuit Design section. As a recap, we had a pass-band gain of 1.56 for the bass and treble filters, and a pass-band gain of $1.56^2 = 2.43$ for the mid-range module. For the bass module, we had $f_c = 234$ Hz. For the treble module, we had $f_c = 5.88$ kHz. For the mid-range module, we had $f_{c,\mathrm{HP}} = 284$ Hz and $f_{c,\mathrm{LP}} = 4.82$ kHz.

For the bass module, we find that the fitted G_0 is 1.52, which is within 10% of the expected 1.56. We also find that the fitted f_c is 247 Hz, which is within 10% of the expected 234 Hz. For the treble module, we find that the fitted G_0 is 1.48, which is within 10% of the expected 1.56. We also find that the fitted f_c is 5.81 kHz, which is within 10% of the expected 5.88 kHz. For the mid-range module, we find that the fitted G_1 is 2.34, which is within 10% of the expected 2.43. We also find that the fitted $f_{c,\mathrm{HP}}$ is 293 Hz and $f_{c,\mathrm{LP}}$ is 4.79 kHz, which are both within 10% of the expected 284 Hz and 4.82 kHz, respectively. Therefore, we can conclude that all three filters roughly obey their theoretically expected response curves.

LED Driver Measurements

Regarding the LED Drivers, recall that we had an unknown resistor value in each driver module (which was denoted as R1), and recall that the values for this resistor were determined experimentally. We can perform some calculations to determine the theoretically optimal R1 values, and compare these to the experimentally determined values. The steps to determine the theoretically optimal R1 value are shown below:

1. Let A_0 be the peak voltage of the signal before the diode.

2. Let $A = A_0 - 0.7$. This should roughly be the peak voltage after the diode drop.

3. Calculate $B = Ae^{-T/\tau}$, where T is the period of the signal frequency, and τ is the time constant for the smoothing resistor and capacitor after the diode. This will be the minimum voltage after the diode drop.

4. We can estimate the curve between A and B to be linear. Therefore, the DC value of the post-diode signal (denoted as X) will be the average of A and B, i.e. $X = \frac{1}{2}(A+B)$

5. Recall that some minimum gate-source voltage is required to turn on each LED. Denote this voltage as V_{min}. Note: V_{min} will differ for the various LED colors.

6. Using the voltage divider equation, we can determine that $V_{\text{source}} = -5\text{V} \cdot \frac{R1}{R1+1000}$

7. Since the circuit is designed so that X will always be less than V_{min}, then we can write $V_{\text{min}} = X - V_{\text{source}}$

8. Since we know the values of X and V_{min}, then we can solve for V_{source}, and thereby solve for R1.

As an example, we will go through the calculation of the bottom-most LED in the bass module. As discussed in the Introduction section, we want this LED to turn on at 1/5 of max power. We measured that the maximum possible earbud output amplitude is 1.71 V peak-to-peak, so at 1/5 of max power, the peak-to-peak voltage will be $\sqrt{1/5} \cdot 1.71 = 0.765$, which means the maximum voltage will be 0.765/2, or about 0.38 V.

This 0.38 V will experience gain passing through the filter and amplifier. Since we want this calculation to be relevant for frequencies in the pass-band, we use the expected pass-band filter gain of 1.56. The amplifier in the bass module has a gain of 2.2. Therefore, this 0.38 V will get bumped up to $0.38 \cdot 1.56 \cdot 2.2 = 1.31$ V after the filter and amplifier. This is our A_0 value.

Following the next step of the calculation, we have $A = 0.61$ V. In order to calculate B, we must choose a frequency. Since we want this calculation to be relevant to an any frequency in the pass-band, then we should be conservative and use a lower frequency rather than a higher frequency. This is because a lower frequency will have less DC after its signal is rectified, due to its longer period leading to a lower value of B. Therefore, reasonable frequencies to use are about 50 Hz for bass, 400 Hz for mid-range, and 7 kHz for treble. In this case, we are doing a bass calculation, so we have $T = \frac{1}{50\text{ Hz}} = 20$ ms. Also, for the bass module, we have $\tau = 56$ ms (as discussed in the Circuit Design section). Now that these values have been established, we can calculate B. We have $B = 0.61e^{-20/56} = 0.43$ V.

Continuing the calculation, we have $X = \frac{1}{2}(A+B) = 0.52$ V. Before continuing from here, we must establish the value of V_{min}. With our LED driver design, this value only depends on the color of the LED. Through experimental testing, we found that V_{min} is about 2.05 V for red and yellow LEDs, and about 1.97 V for green. Note: These are not forward voltages.

They are the minimum necessary gate-source voltages to draw enough current through the LED to power it on at a small but noticeable brightness.

For the bass module, we used red LEDs, so we have $V_{\text{min}} = 2.05$ V. Therefore from step 7, we have 2.05 = 0.52 - V_{source}, which means $V_{\text{source}} = -1.53$ V. Finally, from step 6, we have $V_{\text{source}} = -5\text{V} \cdot \frac{R1}{R1+1000}$, so we can use this to solve for R1. Doing so, we find that R1 = 441 Ω. In our circuit, for this specific LED, we used the experimentally determined value of 446 Ω for this resistor. Therefore, we see that the result of this calculation agrees with the experimentally determined value.

Performing this same calculation for the 11 other LED driver modules, we can obtain theoretical R1 values for each. The results are summarized in Table IV, which shows each theoretically predicted value, along with the corresponding experimentally determined value, for each LED driver module. Note: In these calculations, we used gains of 1.56 for the bass amplifier, 2.17 for the mid-range amplifier, and 1.27 for the treble amplifier, since these are the theoretical gains of each amplifier at 50 Hz, 400 Hz. and 7 kHz, respectively. We also used diode drops of 0.7 V, 0.6 V, and 0.6 V, respectively, since this is approximately what we observed.

TABLE IV. Theoretically predicted vs. experimentally determined resistor values for the LED driver modules.

LED Module / Number	**Predicted R1 Value (Ω)**	**Empirical R1 Value (Ω)**
Bass / #1	441	446
Bass / #2	272	270
Bass / #3	166	180
Bass / #4	90	100
Mid / #1	490	458
Mid / #2	349	330
Mid / #3	257	220
Mid / #4	190	150
Treble / #1	433	390
Treble / #2	275	270
Treble / #3	175	220
Treble / #4	103	150

We see that the majority of the predicted values are within 10% of their corresponding theoretical values, so we can conclude that these calculations generally match our observations. However we also see that some of the value pairs are further apart from each other. This could potentially be explained by the fact that the experimentally determined values were found via trial and error, so they are also not necessarily the optimal resistor values. Therefore, a potential extension to this project could be testing each predicted resistor value, and measuring the current through the LED for each. This could give a clearer idea of how accurate the predictions really are, and could help determine an improved design for the LED modules that allow for greater precision and control over the LED currents and cutoffs.

Final Functionality Test

As a realistic functionality test of our circuit, we tested it with a few different songs, one of which is shown in Video 1. We found that it was quite easy to tune the initial amplifier to get a reasonable LED sensitivity. We also found that the LED display appeared to match up with what we were hearing in the song at any given time. Therefore, we can conclude that the circuit successfully functions as intended.

CONCLUSION

The goal of this project was to build a real-time audio spectrum display capable of dividing an input signal into bass, mid, and treble bands, and indicating their relative amplitudes through LED arrays. The final circuit met the intended goals. Each module (bass, mid, treble) was comprised of a frequency-specific filter, post-filter amplifier, rectifier, and LED driver stage. The measured frequency response parameters (cutoff frequency and gain) agreed with theoretical predictions. Additionally, the calculated LED driver resistor values generally agreed with the experimentally determined values, although there was some slight deviation between some of the values, which could warrant future experiments and lead to improvements in the circuit design. The display responded reliably to music, with LED patterns correlating well with the perceived frequency content. Overall, the project demonstrated successful integration of analog signal processing techniques, and offered a nice visualization of audio spectral content.