Electronics Final Project Report

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Abstract—This project involved designing and building an Audio Analyzer that lights up LEDs based on different parts of a song. Using a summing amplifier, filters, peak detectors, and two types of LED drivers, the circuit separates audio into low and high frequencies and visualizes them in real time. The system was tested with the song "Seven Nation Army" and showed clear LED responses to bass and treble. Simulations and lab tests confirmed that each part worked as expected, and the full system successfully combined everything into a functional and engaging audio-visual tool.

I. BACKGROUND INFORMATION AND RATIONALE

The goal of this project is to design, analyze, simulate, and experimentally verify an Audio Analyzer printed-circuit board (PCB) capable of visually representing audio signals through two distinct LEDs. The LEDs flash in response to specific frequency bands and amplitudes of the input audio signal, providing an intuitive visual representation of music. This project integrates multiple electronic subsystems to achieve accurate audio analysis and visually engaging results.

The input stage consists of a Summing Amplifier, which merges and amplifies stereo audio signals from a standard 3.5mm audio jack into a single-channel output for simplified processing. Following this stage, the audio signal is separated into two frequency ranges through high-pass and low-pass filters, which isolate treble and bass frequencies. Each filtered output is then stabilized through dedicated Peak Detectors, ensuring the LEDs respond smoothly to rapid audio changes.

Finally, the processed signals feed into specialized LED driver circuits. The high-frequency path employs an Analog LED Driver, varying LED brightness through continuous changes in current. The low-frequency signal utilizes a PWM (Pulse Width Modulation) LED Driver, modulating brightness by adjusting the duty cycle of a high-frequency pulse signal. Together, these subsystems deliver a dynamic visual display, correlating directly with the musical input and enhancing user engagement through real-time frequency visualization.

For this project, the song selected is "Seven Nation Army" by The White Stripes. This song was chosen due to its strong balance between distinct bass and treble frequencies, making it ideal for demonstrating the effectiveness of both filters clearly. Using the frequency spectrum in Figure 1, the cutoff frequencies determined for the low-pass and highpass filters are 650 Hz and 2 kHz, respectively. These values were selected to effectively separate the prominent bassline,

This is the final project report for ECE 2660 Fundamentals 2, Spring 2023 at the University of Virginia

which predominantly lies below 650 Hz, from the clear and sharp treble elements, such as guitar solos and vocal clarity, which typically occur above 2 kHz. By using these cutoff frequencies, the system achieves clear differentiation and optimal visual representation of the audio spectrum components of the selected song.



Fig. 1. Frequency Spectrum of "Seven Nation Army" by The White Stripes

II. INPUT STAGE: SUMMING AMPLIFIER

The summing amplifier merges left and right audio signals, removes DC offset, and ensures balanced amplification. A high-pass filter preserves low-frequency integrity, enhancing overall performance.

A. Design and Analysis



Fig. 2. Input Filter and Summing Amplifier Schematic

The overarching goal of the summing amplifier is to achieve a cutoff frequency of approximately 20Hz with resistor values of R1 and R2 greater than $10k\Omega$ and a linear gain of 3. The cutoff frequency is given by:

$$f_c = \frac{1}{2\pi RC}.$$
 (1)

In order to achieve these values with the given components in the lab kit, a ceramic capacitor of 0.1μ F was chosen first. Plugging this value as well as the desired cutoff frequency back into the equation, we get:

$$20 = \frac{1}{2\pi R (0.1 \cdot 10^{-6})}.$$
 (2)

$$R = \frac{1}{40\pi (0.1 \cdot 10^{-}6)} = 79577\Omega \approx 82k\Omega.$$
(3)

Using the closest resistor value in the lab kit, $82k\Omega$, and capacitor values of $C_5 = C_6 = 0.1\mu F$, we achieve a cutoff frequency of 19.4Hz

The output of the inverting op-amp is given by the equation:

$$V_{out} = \pm \frac{R_f}{R_{in}} V_{in}.$$
(4)

Because of the nature of inverting operational amplifiers, the node at the inverting terminal is a virtual ground. Therefore, using this fact and node analysis at the T-junction at the feedback loop, and solving for the output voltage, V_{mix} , we get:

$$\frac{V_T - 0}{R_3} + \frac{V_T - 0}{R_4} + \frac{V_T - V_{mix}}{R_5} = 0.$$
 (5)

$$V_{mix} = V_T \left(\frac{R_5}{R_3} + \frac{R_5}{R_4} + 1\right).$$
 (6)

Once again, using the fact that the node at the inverting terminal is a virtual ground, using R_{in} as $82k\Omega$ as calculated above, and performing a node analysis at the inverting terminal V_{in} and solving for V_T , we get:

$$\frac{0 - (V_L + V_R)}{R_{in}} + \frac{0 - V_T}{R_3} = 0.$$
 (7)

$$V_T = \frac{-R_3(V_L + V_R)}{R_{in}}.$$
 (8)

Plugging this equation back for V_T into the previous equation for V_{mix} and simplifying, we get:

$$V_{mix} = -(V_L + V_R)\left(\frac{R_5}{R_{in}} + \frac{R_3R_4}{R_4R_{in}} + \frac{R_3}{R_{in}}\right).$$
 (9)

In order to calculate component values that yield a linear gain of 3, we plug in values for R_{in} into the equation for gain, which is equivalent to dividing both sides by $(V_L + V_R)$ in the equation above. Plugging in 82k Ω for R_{in} and simplifying the equation, we get:

$$Gain = \frac{V_{mix}}{V_L + V_R} = \left(\frac{R_5}{R_{in}} + \frac{R_3R_4}{R_4R_{in}} + \frac{R_3}{R_{in}}\right) = 3.$$
(10)

$$246k\Omega = R_5 + \frac{R_3R_5}{R_4} + R_3. \tag{11}$$

For simplicity, if we choose R_4 and R_5 to have the same resistor value, we can further simplify the equation above to:

$$246k\Omega = R_5 + 2R_3.$$
(12)

Referring back to the given component values in the lab kit, it can be calculated through simple guess and check that the values $R_4 = R_5 = 180k\Omega$ satisfy this equation and requirement to achieve a gain of 3.

B. Simulations

Simulating the schematic shown above inFig. 2 with input sin waves of 0.5V and 1kHz for both the left-channel input and right-channel input through Multisim [1], Fig. 3 verifies the output, V_{Mixed} , as a sin wave of 3V and frequency of 1kHz, matching the desired goal of adding both inputs and a gain of 3. Running another simulation with input sin waves with frequencies of 15Hz resulted in a flat DC output, verifying the cutoff frequency of 20Hz as well.



Fig. 3. Summing Amplifier Multisim: Input waves of 0.5V and 1kHz for both Left-channel and Right-channel

C. Experimental Results

An audio signal of the song "Seven Nation Army" with separate left and right channels was plugged in as the input for the left-channel and right-channel using 3.5mm audio jacks. Using an AD2 board [2]to measure V_{mixed} on the pcb board, Fig. 4 shows the output of the input filter and summing amplifier, clearly showing both the left and right channels merging together and having a gain in amplitude.



Fig. 4. Summing Amplifier PCB Board: Output of Summing Amplifier with song input to left and right channels

III. HIGH AND LOW PASS FILTERS

The high-pass and low-pass filters separate the combined audio input into distinct frequency ranges, isolating lower and higher frequency components. This filtering is essential for ensuring that the subsequent peak detectors and LED drivers respond specifically to either bass or treble content, enabling accurate frequency-based visual output.



Fig. 5. Sallen-Key Low-pass Filter Schematic



Fig. 6. Sallen-Key High-pass Filter Schematic

To design the high-pass and low-pass filters, I referred to the TI analysis of the Sallen-Key architecture, specifically the document "Analysis of the Sallen-Key Architecture" by Texas Instruments [3]. In the standard derivation, the transfer functions include a factor K, which accounts for the feedback created by resistors in the amplifier's negative feedback loop. However, for this project's specific use of Sallen-Key filters, no additional feedback resistors were used, meaning the negative feedback resistors are effectively zero. As a result, the provided transfer functions were evaluated by setting the corresponding terms involving resistors in the negative feedback loop to zero, simplifying the expressions accordingly. The simplified transfer function for the Low-Pass Sallen-Key Circuit and the corresponding cutoff frequency and Q-value is represented by:

$$H(s) = \frac{V_{out}}{V_{in}} = \frac{1}{s^2 R_6 R_7 C_7 C_8 + s(R_6 C_7 + R_7 C_7)}$$

$$f_c = \frac{1}{2\pi \sqrt{R_6 R_7 C_7 C_8}}$$

$$Q = \frac{\sqrt{R_6 R_7 C_7 C_8}}{R_6 C_7 + R_7 C_7 + R_7 C_8}$$
(13)

Using the simplified transfer functions, resistor and capacitor values were selected from the available lab kit components to achieve the desired cutoff frequencies of approximately 650Hz for the low-pass filter and 2kHz for the high-pass filter, as well as a Q-value as close as possible to 0.707.

Running a MATLAB code (see Appendix 1) to find the best solution for each case, the resistor and capacitor values $R_6 = 330k\Omega$, $R_7 = 180k\Omega$, $C_7 = C_8 = 0.01nF$ were selected for the low-pass filter and $R_8 = 120k\Omega$, $R_9 = 56k\Omega$, $C_9 = C_{10} = 1nF$ for the high-pass filter. These components yielded $f_c = 653.0206Hz$, Q = 0.6770Hz for the low-pass filter, while the selected values for the high-pass filter resulted in $f_c = 1.941kHz$, Q = 0.7319.

B. Simulations

Performing an AC sweep on the schemtaics of both Fig. 5 and Fig. 6 on the input of the filters and a probe on VO_{lp} and VO_{hp} , the bode plots, shown in Fig. 7 and Fig. 6 verified that the low-pass filter started attenuating signals starting at 650Hz and the high-pass filter attenuates signals below 2000Hz.



Fig. 7. Sallen-key Low-Pass Filter Multisim AC Sweep Bode Plot



Fig. 8. Sallen-key High-pass Filter Multisim AC Sweep Bode Plot

C. Experimental Results

With the same input settings from Fig. 4, scopes were placed in J9 and J10 on the pcb board and were measured using the AD2. Fig. 9 and Fig. 10 show the results accurately representing how the low-pass filter only passes through low frequency components of the audio signal and the high-pass filter only passes through high frequency components of the audio signal.



Fig. 9. Low-pass Filter PCB: Scope at J9 Showing Output of the Low-pass filter



Fig. 10. High-pass Filter PCB: Scope at J10 Showing Output of the Highpass Filter

IV. PEAK DETECTOR

A precision rectifier and peak detector use a super diode to capture and hold peak voltages accurately for both outputs of the low-pass filter and high-pass filter. The diode, along with a resistor and capacitor, minimizes voltage drop and preserves the highest signal value for reliable amplitude measurement.



Fig. 11. Peak Detector Schematic

A. Design and Analysis

The goal of this sub-part is to design resistor and capacitor values in the negative feedback loop and a super diode to achieve a time constant τ between 50 ms and 80 ms. In order to design these values, choosing a ceramic capacitor of capacitance $0.1\mu F$ out of the given components, we can derive the equation $50 \cdot 10^{-3} \leq \tau = RC \leq 80 \cdot 10^{-3}$. Plugging in a value of $0.1\mu F$ and solving for the resistor value, we get $500 \cdot 10^3 \leq R \leq 800 \cdot 10^3$. Again, referring to the given components in the lab kit, a resistor value $680k\Omega$ comfortably satisfies this condition. Solving for the time constant, $\tau = RC = (680 \cdot 10^3)(0.1 \cdot 10^{-6}) = 68ms$.

B. Simulations

Simulating the schematic shown above in Fig. 11 with an input sin wave of 1V and 100Hz into the non-inverting

terminal of the op-amp and a probe at the inverting-terminal of the op-amp through Multisim, an output wave that holds the peak voltage of 1V and slowly decaying with a time-constant of 68ms was observed, verfifying the functionality of the circuit design.

C. Experimental Results

Placing a scope on J_{12} on the PCB board, the output of the peak detector, which controls the MOSFET that turns on the green LED, is shown in Fig. 12 to accurately hold the peak voltage from the output of the high-pass filter with a time-constant of $\approx 60ms$.



Fig. 12. Peak Detector Output for Analog LED Driver on J12 with Song Audio Signal as Input

V. ANALOG LED DRIVER

The purpose of the analog LED driver is to light up an LED when high-frequency sound is detected, providing a visual indication of audio activity.

A. Design and Analysis



Fig. 13. Analog LED Driver Schematic

The analog LED driver detects high-frequency audio signals and uses a MOSFET to control the state of an LED accordingly. To achieve this, the circuit uses a high-pass filter followed by a peak detector to extract the envelope of the high-frequency signal. Capacitor C_{11} acts as a DC block, allowing only the AC signal from the high-pass filter to influence the gate voltage. The peak detector converts the filtered signal into a smoothed DC value that holds the peak of the high-frequency signal. The biasing resistors R_{10} and R_{11} form a voltage divider that sets a baseline voltage at the gate of the MOSFET. The voltage at the gate of the MOSFET V_q is given by the equation:

$$V_g = V_{hpf} + \frac{R_{11}}{R_{10} + R_{11}} (V_{CC} - V_{EE}) + V_{EE}.$$
 (14)

In order to ensure that the gate-source voltage, V_{gs} remains in the MOSFET's cutoff region when no signal is present ($V_{hpf} = 0$), we set the equation above minus $V_s = -4.5V$ (when no current is flowing through the MOSFET) to get V_{gs} equal the threshold voltage $V_{th} = 1.7V$ of the MOSFET and choose resistor values that place the product of the voltage divider slightly below this value. Choosing resistor values of $R_{10} = 470k\Omega$ and $R_{11} = 100k\Omega$, we get $V_{gs} = 1.58V$ which is slightly below the threshold voltage.

Resistor R_{13} , placed at the source of the MOSFET, limits the maximum current through the LED by creating a voltage drop proportional to the current. It is chosen based on the LED's forward voltage and current rating to ensure safe operation. Referring to the LED datasheet [4] for the green LED, we find that the maximum voltage drop across the green LED is 1.9V and the maximum rated current is 7mA. For a safe approximation, the MOSFET is simplified to a short circuit, and using Ohm's law, we get the relationship $V_{LED} + I_{LED}R_{13} = V_{CC} - V_{EE}$ when the MOSFET is on. Solving for R_{13} using the maximum values for voltage and current and choosing the next largest possible resistor value, we get that $R_{13} = 1.2k\Omega$.

B. Experimental Results

Using visual verification, it was observed that the green LED accurately lit up whenever there was a clear presence of high pitch in the audio, such as snare drums or guitar solos. The brightness of the LED also dimmed and brightened accordingly to the volume of the input audio and turned off in the absence of any input audio.

VI. PWM LED DRIVER

The purpose of the PWM LED driver is to visualize lowfrequency audio signals by adjusting LED brightness using pulse-width modulation. A low-pass filter extracts the lowfrequency content, which is compared to a triangle wave. The result controls the LED's brightness, creating a visual representation of bass activity.

A. Design and Analysis

The PWM LED driver visualizes low-frequency content in an audio signal by modulating an LED's brightness using pulse-width modulation (PWM). The output of the low-pass filter, V_{lpf} , is compared to a reference triangle wave in a comparator to generate the PWM signal.



Fig. 14. Low-pass LED Driver using PWM

For the triangle wave oscillator, using the relationship $V_{triangle} = \frac{R_{18}}{R_{19}} |V_{square}|$, where V_{square} is the output of the Schmitt trigger oscillating between +4.5 V and -4.5 V, we solve for resistor values $R_{18} = 1k\Omega$ and $R_{19} = 4.7k\Omega$ in order to get a desired triangle wave amplitude of approximately 1 V. Next, using the equation $f = \frac{1}{4R_{17}C_{15}}\frac{R_{19}}{R_{18}}$, and a goal of achieving a frequency of approximately 1 kHz, we solve for resistor and capacitor values of $R_{17} = 10k\Omega$ and $C_{15} = 0.1\mu F$. This triangle wave provides a repeating reference against which the low-pass signal is evaluated. The comparator outputs a PWM signal whose duty cycle varies based on the amplitude of V_{lpf} , which directly controls the brightness of the LED.

Resistors R_{14} and R_{15} form a voltage divider that introduces a DC offset to V_{lpf} , allowing finer control over when the LED begins to turn on. This offset ensures the LED remains off when no low-frequency content is present and begins to brighten as the signal amplitude increases. The output of the peak detector, which is also the input into the non-inverting terminal on the comparator, is represented by the equation:

$$V_{+} = V_{lpf} + \frac{R_{15}}{R_{14} + R_{15}} (V_{CC} - V_{EE}) + V_{EE}.$$
 (15)

Because we want the just below the triangle wave generator when there is no low-frequency content, we set this equation for $V_+ < -1$. Solving for resistor values that place V_+ right under the triangle wave oscillator when $V_{lpf} = 0$, we get $R_{14} = 470k\Omega$ and $R_{15} = 270k\Omega$. These resistor values result in $V_+ = -1.2V$ when there is no low-frequency content, which is right below the triangle wave oscillator.

Similar to the analog LED driver, R_{20} is the source resistor that limits the current flow through the red LED when the MOSFET is on. Referencing the datasheet [5], we know that the maximum voltage drop across the green LED is 2.4V and the maximum rated current is 30 mA. Using these values and a similar relationship shown in the analog LED driver, we get that $R_{20} = 220\Omega$.

B. Experimental Results

Again, using visual verification, it was observed that the red LED brightly lit up everytime there was a beat on the bass of the audio. The red LED accurately dimmed and brightened based on the strength of the bass as well as the volume of the input audio. The red LED was also observed to be completely off in the absence of any input audio.

VII. SYSTEM-LEVEL DESIGN AND TESTING

To test the complete system, two separate 3.5mm audio input jacks were used to independently simulate the right and left audio channels from "Seven Nation Army" by The White Stripes. The summing amplifier successfully merged the two inputs into a single-channel signal, removing any DC offset and providing balanced amplification for further processing.

This combined signal was fed through both the high-pass and low-pass filters, which effectively separated the audio into bass and treble components based on their frequency ranges. The low-frequency content, dominated by the iconic bass line, triggered the red LED through the analog LED driver. Visually, the red LED actively pulsed in sync with the strong beats of the bass, demonstrating that the lowpass filter and peak detector accurately captured the low-end energy of the song.

Simultaneously, the high-frequency content, such as snare drum hits and sharp guitar elements during the chorus, caused the green LED to respond. The PWM LED driver modulated the brightness of the green LED in real time, reflecting changes in the amplitude of the high-frequency components.

The system's behavior — with the red LED tracking bass hits and the green LED tracking snares and highpitched sounds — confirmed that each subsystem worked as intended, and that the overall design provided a clear, real-time visual representation of the different frequency components of the music.

VIII. CONCLUSIONS AND THOUGHTS FOR FUTURE CLASSES

This project demonstrated how analog circuit building blocks can be combined to form a functional system that visually responds to audio signals. By designing and testing each stage—including the summing amplifier, frequency filters, peak detectors, and LED drivers—the project highlighted the importance of both individual subsystem performance and overall system integration. The final result showed that the red and green LEDs accurately responded to different parts of the audio signal, confirming that the design goals were met.

Future students are encouraged to begin early and to test each part of the circuit separately before assembling the full system. Identifying problems is much easier when each section is known to be working on its own. In addition, updating the project report immediately after completing each subsystem is highly recommended. Keeping the report up to date throughout the process makes the final submission much easier and avoids the difficulty of recalling design choices and results at the end.

A strong understanding of how all the subsystems work together is also crucial. Developing an overarching view of the full system allows for more efficient debugging during integration, since many issues arise not from individual components, but from how they interact with each other.

The project also provided valuable hands-on experience with soldering and assembling printed circuit boards, building practical skills that are essential for working with hardware systems. Furthermore, this project emphasized the importance of professional documentation. Careful attention to formatting, clarity, and organization in writing the project report proved to be just as important as technical performance.

Careful attention should be paid to component values, circuit layout, and signal connections, as small errors in these areas can cause large issues. Staying organized, maintaining clean notes, and comparing simulation results to experimental measurements are key habits that lead to successful project completion. Overall, this project provides a valuable opportunity to apply circuit concepts to a real-world application and to develop strong habits in testing, troubleshooting, documentation, hardware assembly, and system-level thinking.

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APPENDIX

(1) The following MATLAB script was used to select resistor values for the Sallen-Key filters to achieve the desired cutoff frequency and quality factor.

```
5.6e3, 6.8e3, 8.2e3, 10e3, 12e3, 15e3, 18e3,
6
           22e3, 27e3, 33e3, ...
      39e3, 47e3, 56e3, 68e3, 82e3, 100e3, 120e3,
          150e3, 180e3, 220e3, ...
      270e3, 330e3, 390e3, 470e3, 560e3, 680e3, 820
          e3, 1e6];
  % Constants
10
% 1 nF
               % 1 nF
13 target_fc = 950; % Target cutoff frequency (Hz)
14 tolerance_fc = 50; % Tolerance for cutoff
      frequency (Hz)
15 target_Q_min = 0.677; % Minimum Q
  target_Q_max = 0.8; % Maximum Q
16
  k = 1; % Unity gain
p = pi; % pi constant
18
19
20
  % Initialize
  best_designs = [];
21
  % Search over resistor combinations
24
  for r1 = resistor_values
      for r2 = resistor_values
25
          fc = 1 / (2 * p * sqrt(r1 * r2 * C1 * C2))
26
               ;
          Q = sqrt(r1 * r2 * C1 * C2) / (r2 * (C1 + C2))
               C2) + r1 * C2 * (1 - k);
28
          if abs(fc - target_fc) <= tolerance_fc &&</pre>
29
               r1 >= 10e3 && r2 >= 10e3
               if Q >= target_Q_min && Q <=</pre>
30
                   target_Q_max
                   best_designs = [best_designs; r1,
                       r2, fc, Q];
              end
32
          end
      end
34
  end
35
```

Listing 1. MATLAB code used to select resistor values for Sallen-Key design



Fig. 15. Full Schematic of the Audio Analyzer Circuit