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Five Band Equalizer

Name	Index	Contribution
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Abstract:This is a detailed report discussing the introduction, functionality description, system model (with design parameters), schematic, PCB design, enclosure design, individual contributions of each group member, simulation results, conclusion and future works, and references. EQs are primarily used to modify the frequency response of audio by increasing or decreasing the energy of various frequency ranges, or bands. To make up a standard graphic equalizer, many audio filters or amplifiers(which consist of the equalizer) are tuned to a different frequency in the audio spectrum.In this project, we have created a five band EQ which capable of applying a custom gain for each frequency band and displays the sound level by using a sound level indicator

Contents

1 Introduction

In sound recording and production studios, equalization is a widely used method. By employing an audio equalizer, we may adjust various audio frequency ranges by using linear filters. Simply put using equalizer we can adjust which frequency ranges to allow and which range to reject from the audio signal. An equalizer has ability to modify the phace, tone and other different aspects in an audio signal. We can devide EQ system into two categories.

1.1 Parametric Equalizer

This type of EQs offer more accuracy, they are frequently used in mixing and recording studios. They allow the user to adjust the central frequency, the bandwidth (also known as Q or quotient of change), and the frequency levels (the gain) using several knobs.

Figure 1 : Parametric Equalizer

1.2 Graphic Equalizer

It includes a fixed bandwidth and middle frequency, however there are many sliders that may be used to change the gain of a particular frequency band. The more sliders are present on this EQ imply that, the more control there is over a larger spectrum of frequencies.

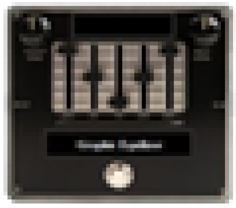

Figure 2 : Graphic Equalizer

In this project, we'll use audio filters to create a five-band graphic equalization circuit. To separate the low, high, and mid-range frequencies of the audio stream, it will thus contain low, high, and band-pass filter circuits. Our circuits contains operational amplifier-based active filters (op-amps) and has capable of adding gains in 5 frequency bands 20-300 Hz , 300 - 1 kHz, 1kHz - 4 kHz , 4kHz - 10kHz , 10kHz - 20kHz.

2 Functionality Description

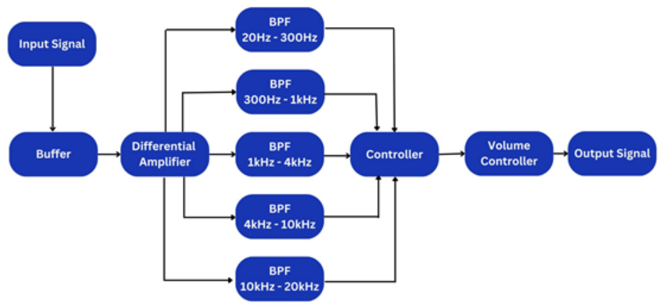

Figure 3 : Functional Block Diagram

2.1 Power Supply

We use two 9V batteries to get the necessary power supply for the five band equalizer.

Figure 4 : 9V Batteries

2.2 Input Signal

We supply the input audio signal to our circuit from smartphone through a 3.5mm audio jack cable.

Figure 5 : Audio Jack Cable

2.3 Buffer Circuit(For Circuit Isolation)

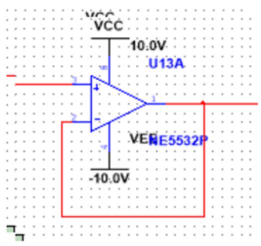

Figure 6 : Buffer Circuit

The figure shows an Op-Amp Follower, or Buffer. The buffer has an ability to produce the output that precisely mirrors the input (if the voltage rails are within reach). Therefore we may feel that it is useless at first. However, the buffer circuit helps to solve many impedance issues. The input impedance of the op-amp buffer is very high (close to infinity) and the output impedance is very low (few ohms). The buffer allows us to move from one circuit to another(circuit isolation) and maintain the voltage level without drive a current from the input signal.

2.4 Differential amplifier

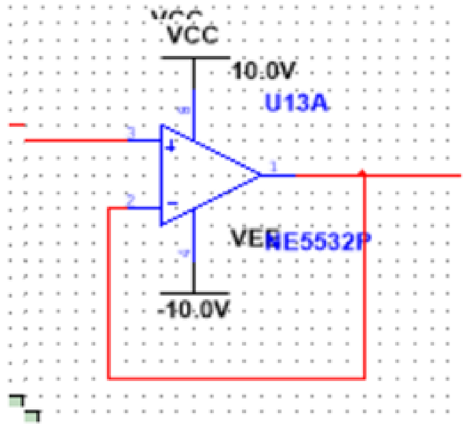

Figure 7 : Differential Amplifier

Differential amplifiers has ability to suppress noise. Differential noise and common-mode noise are the two main types of noise. They can simply suppressed with an op-amp. We can consider two main reasons for common-mode noise.
1. Noise is generated in the wires and cables, due to electromagnetic induction, etc., and it causes a

difference in potential (i.e., noise) between the signal source ground and the circuit ground.
2. A ground potential rise occurs when current from another circuit flows into a circuit's ground (noise). In either scenario, noise causes the ground potential, a circuit reference, to fluctuate. Common-mode noise is challenging to eliminate using conventional filters. Common-mode noise is reduced by using differential amplifiers.

2.5 Band Pass Filters

We need BPF since it allows signals within a chosen frequency range to be heard while blocking transmissions at unwanted frequencies.
The following points can be considered as the main reasons why we use active filters instead if passive filters.
• These filters are more responsible than passive filters
• No resonance issue
• Used for voltage regulation
• It provides reliable operation
• It can be designed to provide some passband gain
• The component used in the active filter is a smaller size as compared to the passive filter
We considered two active bandpass filter design topologies and we used both of them for our filter design.

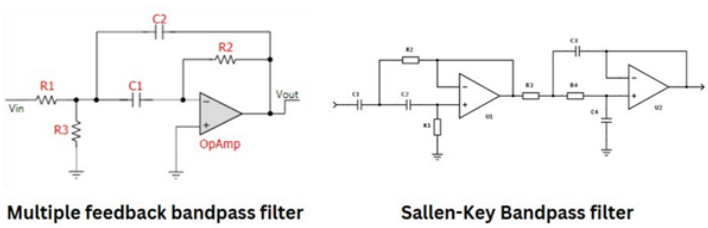

Figure 8 : Multiple Feedback and Sallen-key Bandpass Filters

Easy to cascade for higher order filters, even if we want to cascade high and low pass filters together with different Q factors, simplicity and understanding of their basic design can be considerd as the main advantages of sallen-key filter design. We can build an almost ideal sallen-key high pass filter by increasing the Q factor. But we need to careful about the Q factor of the filter. Because if it is too high, system become unstable and starts to occilate. Another disadvantage in sallen-key filter is the high frequency behavior. We can overcome high frequency behavior by building multiple feedback filters. We used fourth order filters based on the two factors of increasing the roll off rate and minimizing the number of components used.

2.6 Amplifier for adding a gain in each band

We have added a gain to the each paticular band of the input signal from 0 upto 10 by changing the variable resistor value.

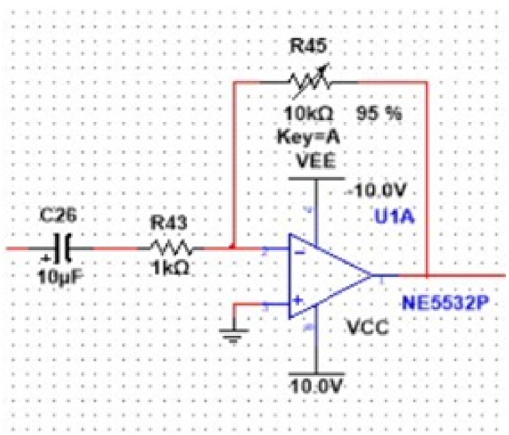

Figure 9 : Amplifier Circuit

2.7 Variable Scaling Adder Curcuit

This circuit adds 5 filtered outputs by producing a scaled(upto 10) sum of them. We can use this as volume controller.

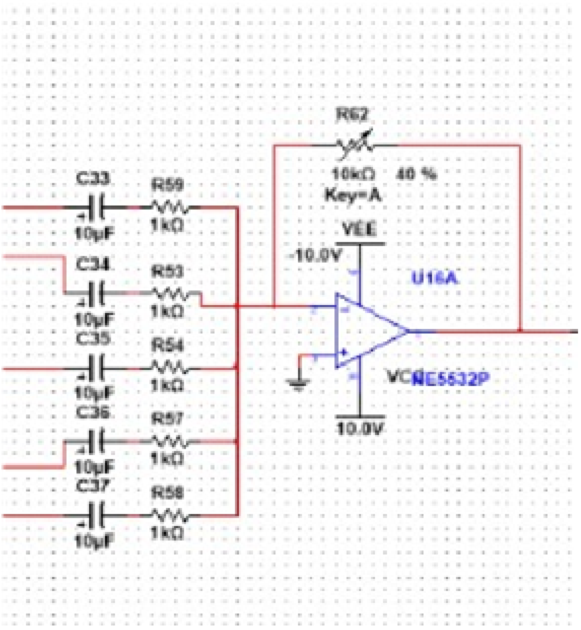

Figure 10 : Variable Scaling Adder Circuit

We used TL072CP IC for all the 5 filters and differential amplifier for reducing the noise. used this IC since,

- High Slew rate - 20 V/us
- Low Noise - 18 nV/Hz (type) at f = 1 kHz
- High power supply -40V to +40V
- Low power Consumption
- This IC is also capable handle higher noise input signals than NE5532 IC

We used NE5532P IC for amplifiers for all the amplifiers such as the amplifiers of each filter and power amplifier. we used this IC since,

- High slew rate - 9 V/us
- Wide power supply - -40V to +40V

2.8 LM3915 IC Visualized Audio Level Display

It is frequently necessary to measure the level of audio signals in a variety of applications. For instance, at discotheques, where it is necessary to determine how loud the music is, and in other locations, where it is necessary to determine how loud the noise is. Sound pressure level meters, which only compute the change in pressure of the sound signal, are one of the various ways to show the audio level of the signal. They are founded on the idea that sound impulses at various frequencies have varying pressure levels. A visual representation of the audio signal's loudness is an additional method. The display is often an array of sequentially lighting LEDs that show the volume of the audio stream. When we use the term "loudness," we refer to the amount of sound pressure that each person can tolerate after being exposed to it. Different frequencies have varying loudnesses for the same pressure.

LM3915

The LM3915 is a dot/bar display driver that uses the analog input to control a number of LEDs. In essence, it drives each neighboring LED in a logarithmic manner, in 3DB steps. It runs on a supply voltage range of 3 to 25 volts.

Pins 1, 10, and 18: The 3V to 20V supply is linked to the output LEDs' anodes. Pin 2: Usually linked to the ground, this pin is the negative analog voltage source. Pin 3: This pin serves as the positive voltage source and typically ranges from 3 to 20 volts. pin4:Typically grounded is pin number four. Pin 5: The audio signal input goes to this pin, which serves as the signal input pin. A short exists between pins 6 and 7. Each LED draws its own current based on the current flowing through pin 7. Pin 8: It is used to change the reference voltage. Pins 7 and 8 are connected by a resistance of 1.2 kohms, resulting in a 1.25 volt difference between them. In order to change the reference voltage, a potential divider is attached to the resistor. Pin 9: This pin, designated as the mode selector, is used to choose between the dot mode and the bar mode. The pin is directly linked to pin 3 for the bar mode, or the positive voltage supply. The pin is left unconnected and open for dot mode.

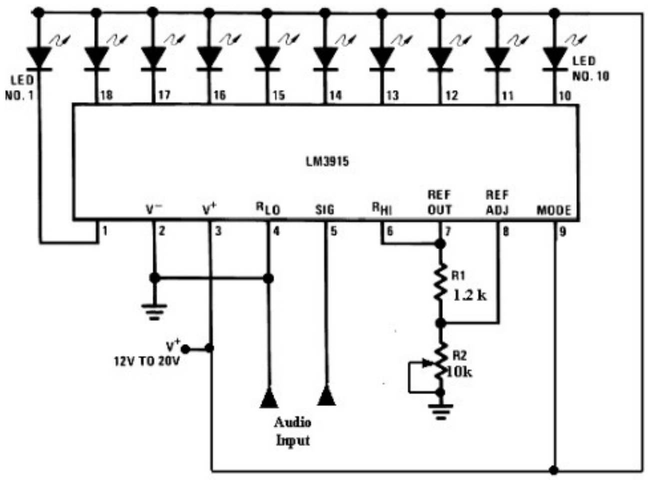

Figure 11 : LM3915 as an Audio Level Meter

3 System model (with design parameters)

In our circuit, the TL072CP IC has been used as a non-inverting amplifier. The input signal from the passive, high-pass filter is connected to the IC's non-inverting input pin (pin 3). A 10 kilo-ohm resistor(R1) is connected between the IC's pins 6 and 2, providing negative feedback. The inverting pin (pin 2) is grounded via a 1kilo-ohm resistor(R2). Therefore the gain can be calculated as follows:

$$Gain = \frac{R1}{R2}$$

$$= \frac{10}{1}$$

$$= 1$$

As a result, in comparison to the input audio signal, the high-frequency component of the audio signal is amplified ten times.
In order to calculate the resistance and capacitance values of each multiple feedback BPF manually, we can follow the below method.

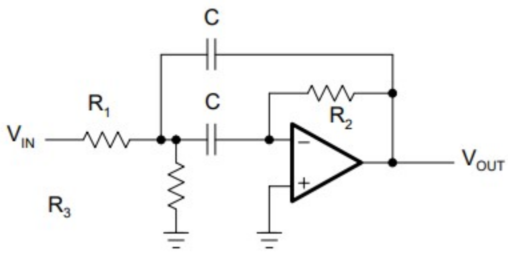

Figure 12 : Multiple Feedback Topology

Transfer function of the second order MFB filter can be considered as below.

$$A(s) = \frac{-\frac{R_2R_3}{R_1+R_3}C\omega_m \cdot s}{1 + \frac{2R_1R_3}{R_1+R_3}C\omega_m \cdot s + \frac{R_1R_2R_3}{R_1+R_3}C^2 \cdot \omega_m{}^2 \cdot s^2}$$

By conсidering the camparison of the coefficients, we can obtained below equations.

$$\text{mid-frequency: } f_m = \frac{1}{2\pi C}\sqrt{\frac{R_1 + R_3}{R_1 R_2 R_3}}$$

$$\text{gain at } f_m\text{:} \quad -A_m = \frac{R_2}{2R_1}$$

$$\text{filter quality:} \quad Q = \pi f_m R_2 C$$

$$\text{bandwidth:} \quad B = \frac{1}{\pi R_2 C}$$

In our circuit, we have choosen $4^{th} orderactiveBPF$.
In order to construct that we have cascaded two $2^{nd} orderactiveBPFs$.
Transfer function of that circuit can be obtained as below.

$$A(s) = \frac{\frac{A_{mi}}{Q_i}\cdot \alpha s}{\left[1 + \frac{\alpha s}{Q_1} + (\alpha s)^2\right]} \cdot \frac{\frac{A_{mi}}{Q_i}\cdot \frac{s}{\alpha}}{\left[1 + \frac{1}{Q_i}\left(\frac{s}{\alpha}\right) + \left(\frac{s}{\alpha}\right)^2\right]}$$

Ami:mid frequency, Qi: pole quality of each filter
α *and* $1/\alpha$ *are thefactors by which the*
mid frequencies of the individual filters, fm1 and fm2: can be calculated from the mid frequency, fm: overall bandpass.

We can find the α *value*
by substituting into the succesive approximation equation given below.

$$\alpha^2 + \left[\frac{\alpha \cdot \Delta\Omega \cdot a_1}{b_1(1+\alpha^2)}\right]^2 + \frac{1}{\alpha^2} - 2 - \frac{(\Delta\Omega)^2}{b^1} = 0$$

Below table contains the different
α *values for different Q factors.*

Bessel				Butterworth				Tschebyscheff			
a_1	1.3617			a_1	1.4142			a_1	1.0650		
b_1	0.6180			b_1	1.0000			b_1	1.9305		
Q	100	10	1	Q	100	10	1	Q	100	10	1
$\Delta\Omega$	**0.01**	**0.1**	**1**	$\Delta\Omega$	**0.01**	**0.1**	**1**	$\Delta\Omega$	**0.01**	**0.1**	**1**
α	1.0032	1.0324	1.438	α	1.0035	1.036	1.4426	α	1.0033	1.0338	1.39

Table 1 : Values of αFor
Different Filter Types and Different Qs

Futher we can use below equation to find resistance values of each filters.

$$f_{m1} = \frac{f_m}{\alpha} \qquad f_{m2} = f_m \cdot \alpha \qquad Q_i = Q \cdot \frac{(1+\alpha^2)b_1}{\alpha \cdot a_1} \qquad A_{mi} = \frac{Q_i}{Q} \cdot \sqrt{\frac{A_m}{B_1}}$$

$$R_{21} = \frac{Q_i}{\pi f_{m1} C} ; \quad R_{11} = \frac{R_{21}}{-2A_{mi}} \quad R_{31} = \frac{-A_{mi} R_{11}}{2Q_i^2 + A_{mi}} \quad R_{22} = \frac{Q_i}{\pi f_{m2} C}$$

$$R_{12} = \frac{R_{22}}{-2A_{mi}} \quad R_{32} = \frac{-A_{mi} R_{12}}{2Q_i^2 + A_{mi}}$$

Because of the long procedure of the above calculations, we have to to use the filter wizard to find resistance and capacitance values directly.

4 Final Schematic

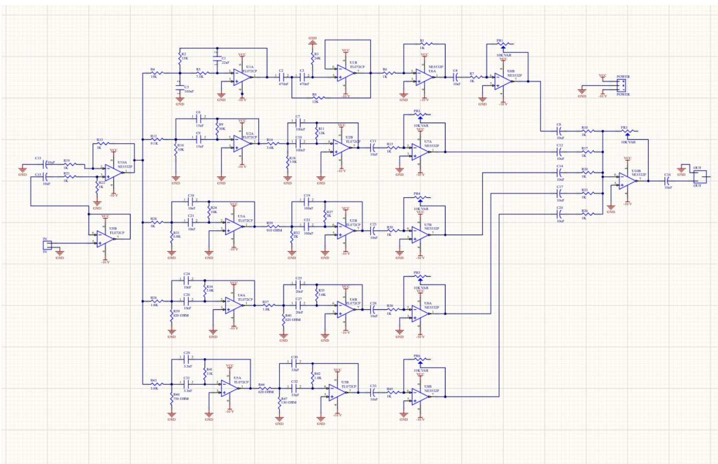

Figure 13: Final Schematic

This is the final schematic for a five band graphic equalizer built with TL072CP operational amplifier chips. The op-amp ICs are one of the best option since they are affordable and have output that is of reasonable quality. To construct this five band graphic equalizer, we will need 4 TL072CP ICs, each of which comprises a dual op-amp circuit.

We have considered the following points to get maximum audio performance.
• Regulated power supply circuit
• Use metalfilm type of resistor
• Use MKM type of nonpolar capacitor
• Use tantalum type of bipolar capacitor (electrolytic capacitor)

Typically, equalizer circuits partition the audio spectrum into distinct frequency bands, with independent gain controls for each band. An audio power amplifier receives the combined output of each band at IC4(A). Avoiding overlap in neighboring bands is necessary since doing so produces coloration into the audio stream. A suitable quality factor (Q) must be used. As the filters are highly sensitive and for best performance, the resistors should be metal-film type and the capacitors should be polyester type.

5 PCB Design

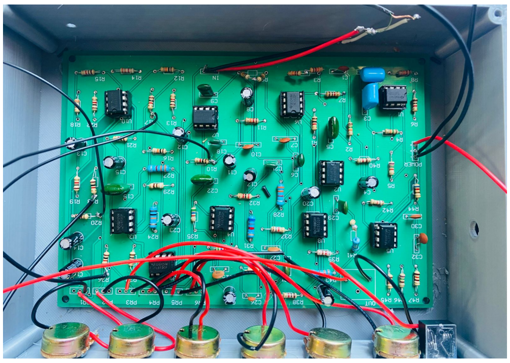

Figure 14: PCB Design

5.1 The Equalizer Circuits

The proposed equalizer is a 5-band circuit, the cut-off frequencies are at: 20Hz, 300Hz, 1kHz, 4kHz,10kHz,20kHz.
We used TL072CP ICs for all the 5 filters and NE5532P ICs for amplifiers such as amplifiers of each filter and power amplifier.

5.2 The Power Supply

This Circuit's power supply is symmetrical. i.e: 9V, 0V, -9V with DC voltage. The consumption current of the IC is 2.5mA maximum and 1.4 mA on average.

5.3 The Circuit

The 5-band Equalizer uses four Integrated Circuits operational amplifier TL072CP, each of which comprises a dual op-amp circuit. The capacitors control the frequencies; the lower the cutoff frequencies, the larger the capacitance of the capacitors. The pinout and setup of the TL072CP and NE5532P integrated circuits are displayed in the Figures below.

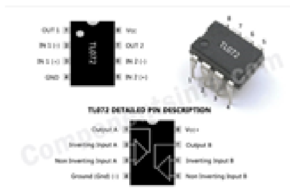

Figure 15: Pinout of TL072CP

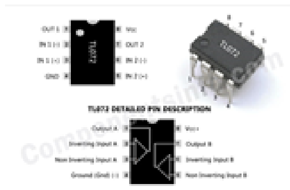

Figure 16: Pinout of NE5532P

6 Enclosure Design

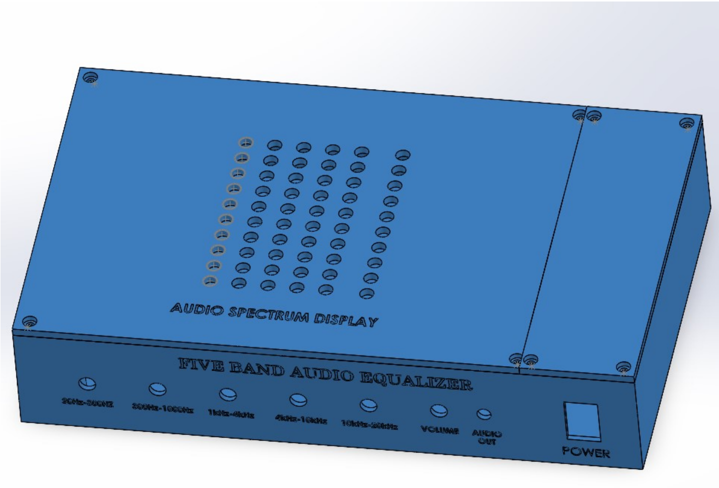

Figure 16: Enclosure Design

6.1 About this product

• EQ Band Center Frequencies: 20 Hz, 300 Hz, 1 kHz, 4 kHz, 10 kHz, 20 kHz allowed for flexible aux volume adjustment, each with a different Frequency control. A 9V DC power supply is used to operate this audio equalizer.In order to measure the level of the audio signal, our circuit used LM3915. The LM3915 is a dot/bar display driver that utilizes the analog input to operate a group of LEDs.

6.2 Product Specifications

• Model: EQ5
• Material: Abs Plastic
• Weight: 700g
• Size: 221 mm × 132 mm × 42 mm
• Power Supply: 9V DC
• Max Output Voltage: 8V RMS
• Leveling Range: ± 12dB

7 Simulation Results

The following figure shows the simulation related to the final output.

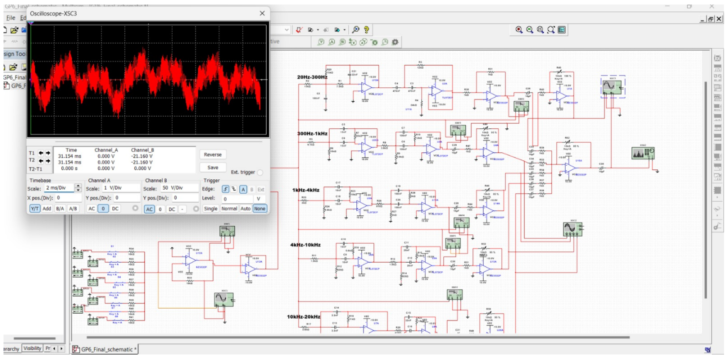

Figure 17: Simulation Results

8 Conclusion and future works

This paper describes an active equalizer ,featuring with simplified approach, fast balancing speed, and easy implementation compared with other current-controlled equalizers. The proposed equalizer is capable of adding gains in 5 frequency bands 20-300 Hz , 300 - 1 kHz, 1kHz - 4 kHz, 4kHz - 10kHz, and 10kHz - 20kHz. The equalizer circuit is built by assembling a power supply, audio source, buffer, differential amplifier, band pass filters, controller and volume controller. The gain for the different frequency bands is controlled through variable resistors. The frequency bands are then merged to create a single audio signal, which is then sent to a power amplifier and speaker. We can propose and implement two approaches, namely the VOT method and the VRM method, to manage the balancing current in order to address the issue of decreasing balancing current in the latter stage of equalization, which results in a protracted equilibrium period. The VRM approach allows for maximal energy transmission during the full equalization, whereas the VOT method may execute a continual balancing current. The battery equalizer's key objectives are high energy transfer efficiency, quick balancing, and user safety. Moreover, the main prerequisite is to accomplish stability quickly. The balancing current has a direct impact on the balancing speed. The buck-boost converter is employed as an equalization structure in the traditional control approach, keeping the duty cycle, or conduction time, of the primary switch constant. The voltage differential between nearby batteries gets reduced throughout the equalization process as the high-SOC battery's voltage steadily drops. So, if the conduction time does not adjust, the balancing current will decrease, adding a substantial amount of time to the equilibrium. As a result, the balance may be completed more quickly if the balancing current's intensity could be adjusted in accordance with the battery's voltage. In this circuit, the average current from the battery when the main switch is on can be calculated from the inductor current summed over one switching period. As a result, the balancing current may be adjusted by varying the duration of the conduction state.

9 Appendices

9.1 Appendix A - Our Product

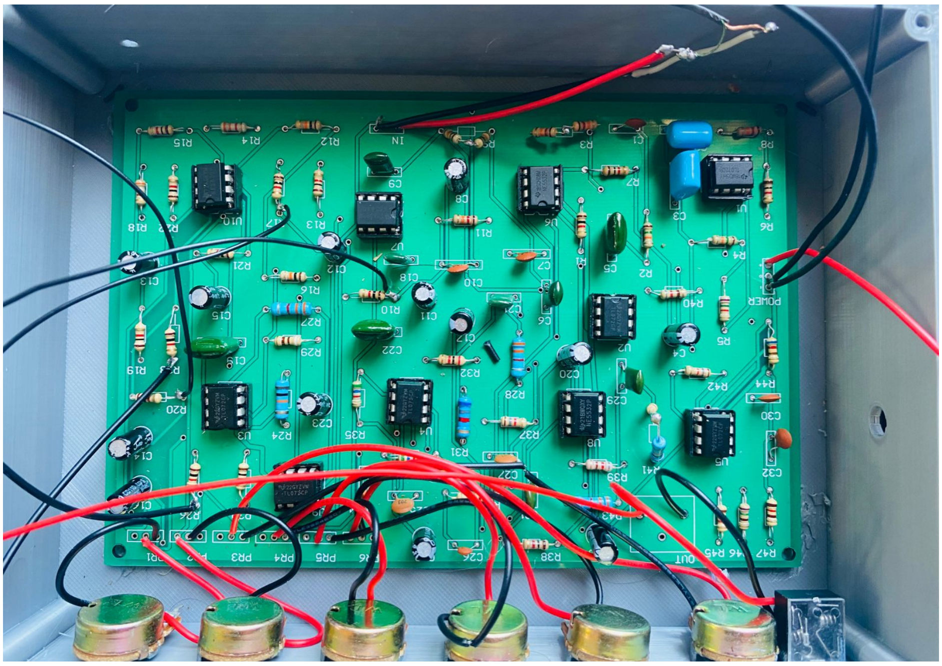

Figure 18.1: PCB Design

Figure 18.2: Final Product

9.2 Appendix B - Enclosure Design

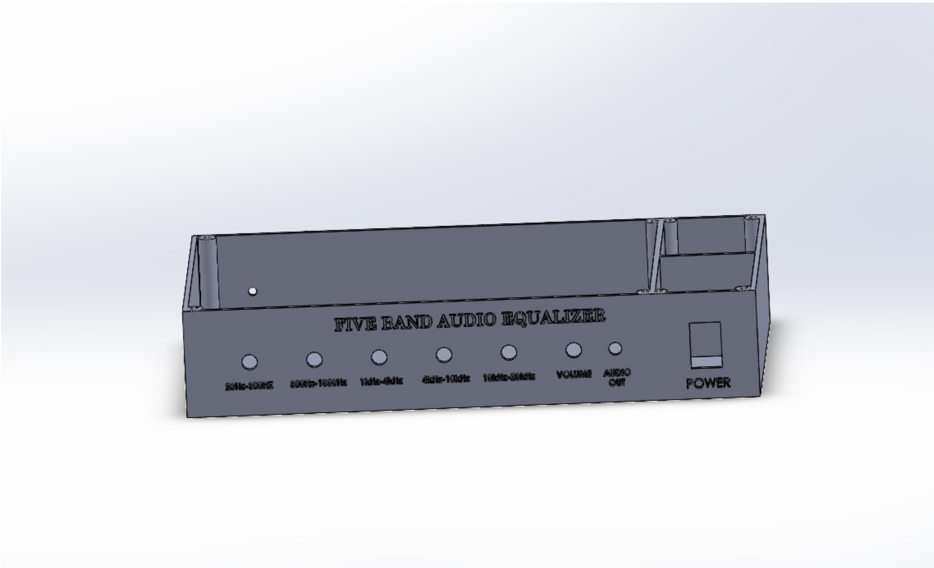

Figure 19.1: Enclosure

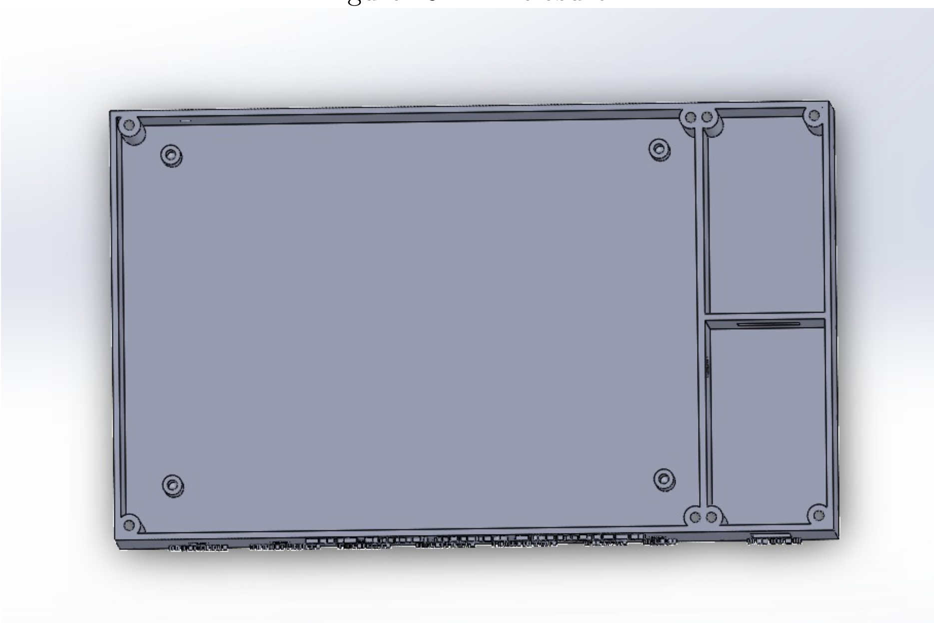

Figure 19.2: Enclosure

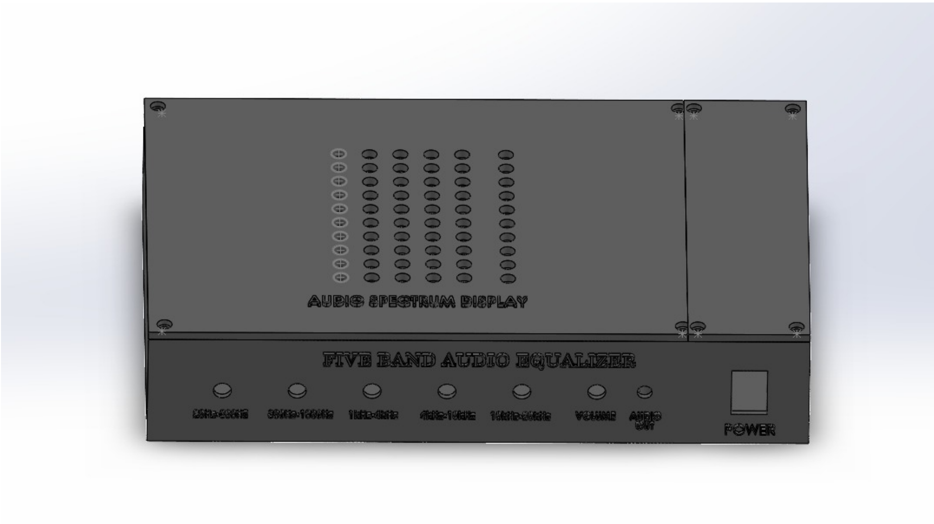

Figure 19.3: Enclosure

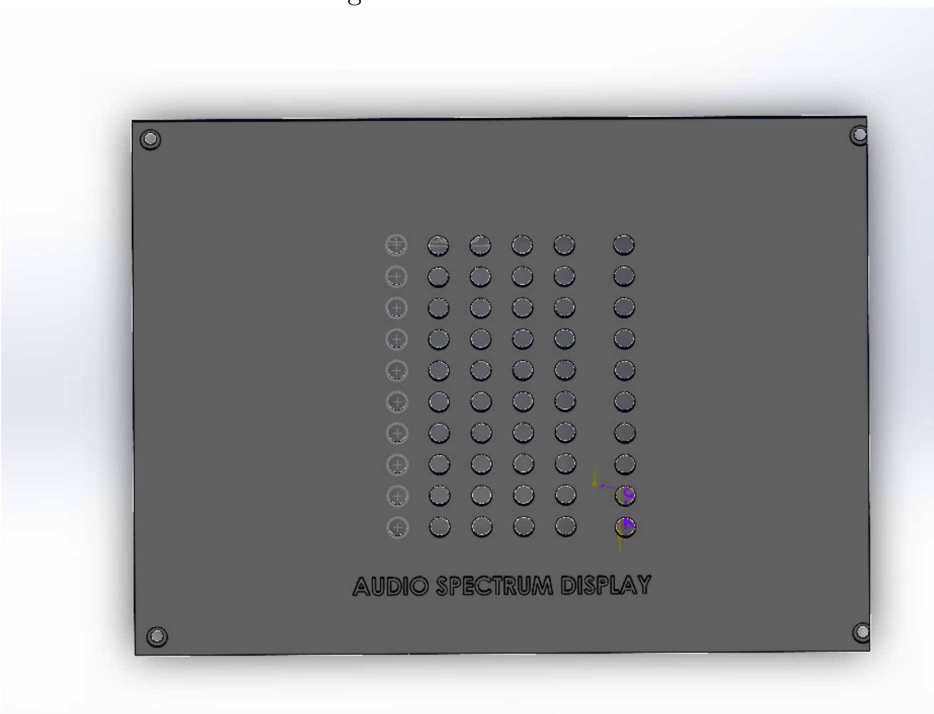

Figure 19.4: Enclosure

9.3 Appendix C - PCB

9.3.1 PCB schematic

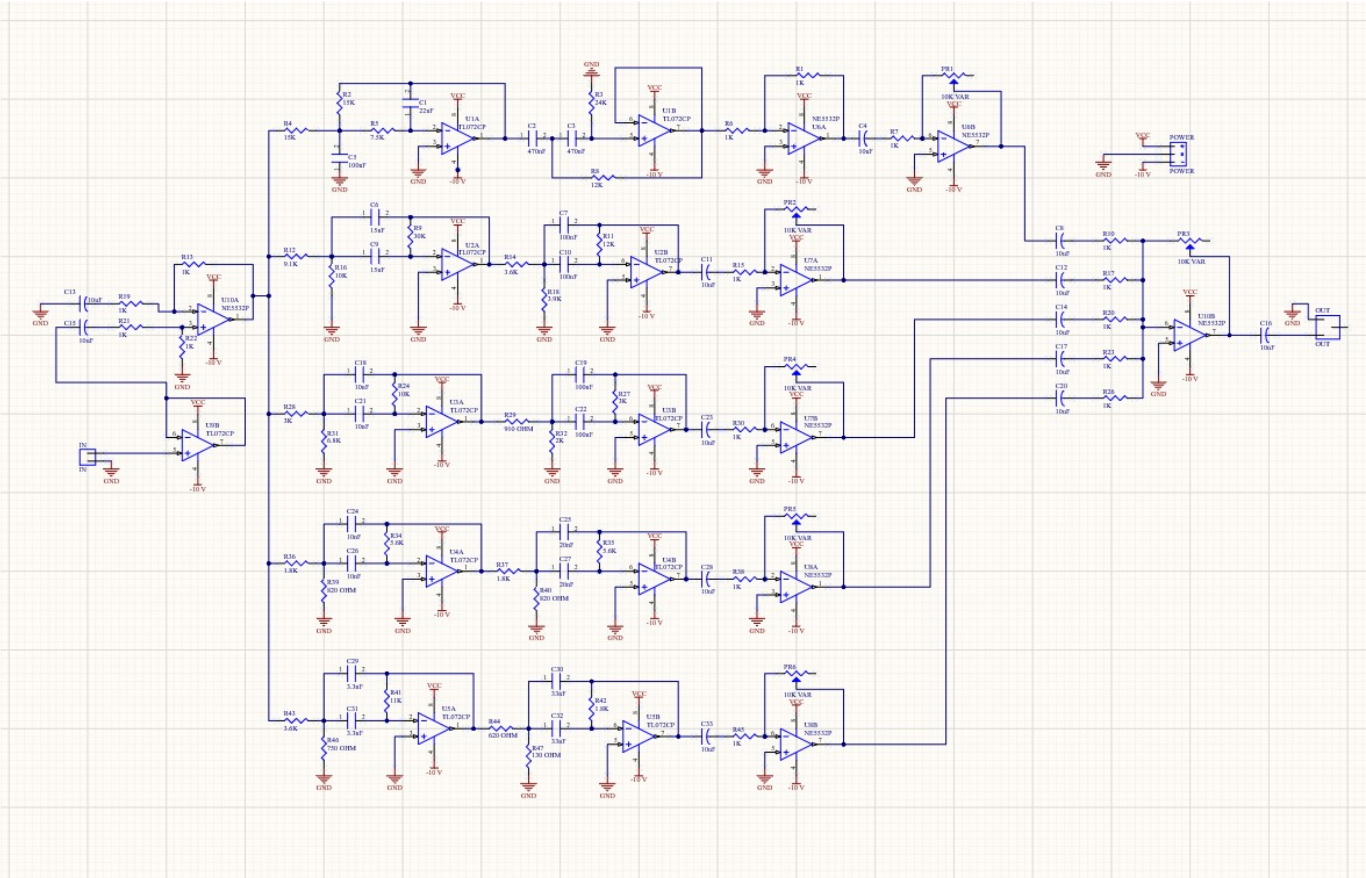

Figure 20.1: PCB schematic

9.3.2 PCB layout

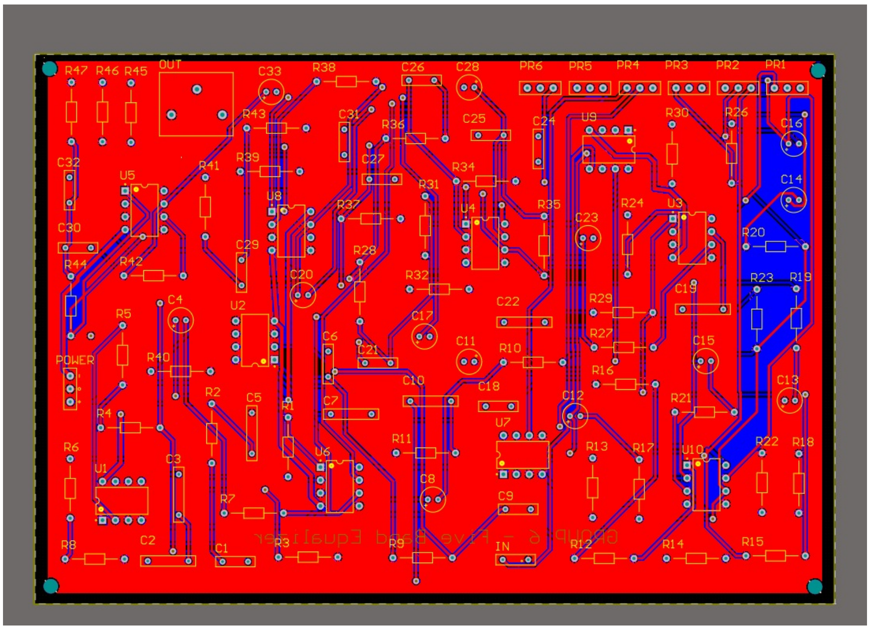

Figure 20.2.1: PCB Layout

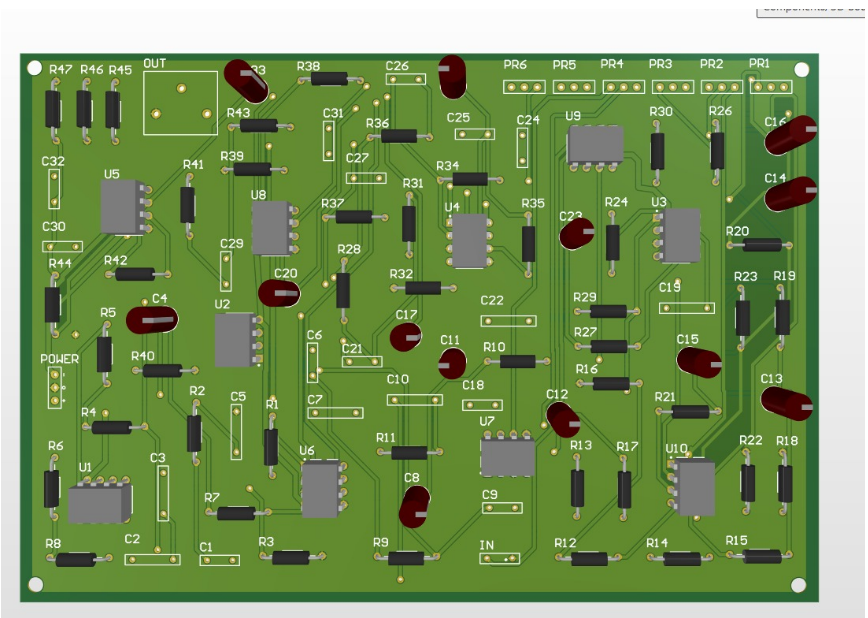

Figure 20.2.2: PCB Layout

Introduction[3]
Active and Passive Filters[2]
Implementation of audio Equalizer[1]

References

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[2] Zhe Chen, Frede Blaabjerg, and John Kim Pedersen. A study of parallel operations of active and passive filters. In *2002 IEEE 33rd Annual IEEE Power Electronics Specialists Conference. Proceedings (Cat. No. 02CH37289)*, volume 2, pages 1021–1026. ieee, 2002.

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