DEPARTMENT OF ELECTRONIC AND TELECOMMUNICATION
UNIVERSITY OF MORATUWA

EN 2091: LABORATORY PRACTICES II

This is offered as a "EN 2090: Labaratory Procedures II" module's partial completion.

Five Band Audio Equalizer

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$26^{th} of February, 2023$

Abstract

This report describes how to design five band equalizer to separate the different frequency bands and amplify them separately. In this project we use capacitors, resistors and Op amps for the filters to separate the frequency bands. And we use +9V[5], -9V[2] as supply voltages.

Index terms: - Filtering, Adder circuit, Buffer, Amplifiers

Contents

1 Introduction

In this project, we were asked to build a five-band equalizer that can divide the audio spectrum into separate frequency bands and have amplifiers for each band. With these amplifiers, we can change the gain of each band. Here we used a buffer circuit to separate input and filters and we used an adder circuit to recombine the frequency spectrum. For the filters, we use op-amps, capacitors, and resistors. And we use a center tap transformer and Voltage regulators to get +9V, -9V as supply voltages. We use a 3.5mm audio jack for audio input and output. We use variable resistors to change the gain. With the help of an LED strip and a dot bar display driver IC[3], we can see the gain of each frequency band.

2 Methodology

[1]

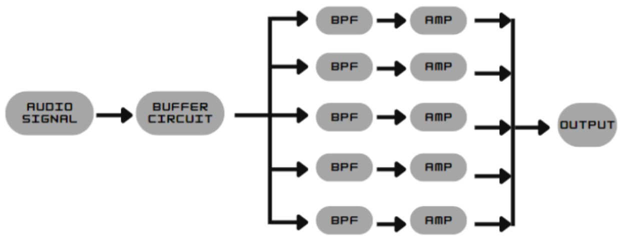


Figure 1: Flow Chart

2.1 Input Audio Signal

To input the audio signal that we need to process, we planned to use a 3.5mm audio jack. We choose this because, most people use this jack frequently, so this is very convenient for our customers.

2.2 Buffer Circuit

Here we use a buffer circuit to separate input and filters. That is to isolate filters from the input side. And also, Buffers prevent excessive current extraction from a signal source.

2.3 Filters

For the filters, at the beginning, we had two options which are active filters and passive filters. After a long discussion, we choose active filters. Because Active filters have a 0dB gain for the pass band, meaning that they do not attenuate the signals in the pass band. They have high input and low output impedance, which correspond to two advantages - one, the load impedance doesn't affect the performance of the filter, and two, cascading multiple filters do not cause a decrease in gain, or it mitigates inter stage loading. These characteristics make them perfect for cascading. Since they are active devices, they can be configured to boost the signals in the pass band. And the output of passive filters changes with the load, whereas active filters maintain their performance irrespective of the load connected. And also, Passive filters cannot apply additional gain to the signal, whereas active filters can. Even if no gain is applied, active filters maintain the signal amplitude, whereas passive filters attenuate the entire signal.

After we chose active filters, we had another problem. That is which active filters we are going to use. With the help of some resources, we found some types of active filters those are multiple feedback filters, sallen-key filters, biquad filters, Wein notch filters, etc. after some research we planned to use multiple feedback filters[6]. Because the multiple feedback topology is widely used as a band-pass filter because it offers a simple and reliable band-pass implementation, especially below a Q (quality factor) of 20 or so. This filter is a simple-looking design, but it is difficult to calculate the values for a given set of parameters. These filters are useful for equalization, analysis, and other tasks such as the Sound to Light converter or even a fully functional vocoder. In multiple feedback filters, we planned to implement fourth order filter. Because to get sharp edges for the filters we need to implement a higher order filter but if we choose more than four the implementation will very tough. That's why we choose forth order multiple feedback filter.

This is not the last problem that we faced. Another problem that we faced was which op amp we are going to use. In the market there are lots of variable op amps are available. Each op amp is used for a specific purpose. So, after a long search, we found an op amp for audio processing that LM4562[4].

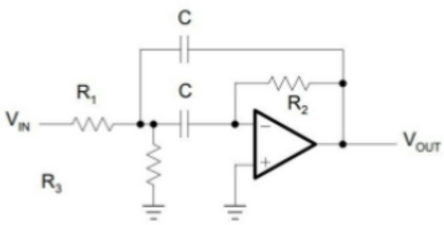
2.4 Why LM4562 ?

- **KEY SPECIFICATIONS :** ±17V. Over this supply range the LM4562's input circuitry maintains excellent common-mode and Power Supply Voltage Range: ±2.5V to ± 17V.
- Power supply rejection, as well as maintaining its low
- THD+N (AV = 1, VOUT = 3VRMS, fIN = 1kHz) input bias current. The LM4562 is unity gain stable.
- RL = 2kΩ: 0.00003% (typ) This Audio Operational Amplifier achieves.
- R outstanding AC performance while driving complex L = 600Ω: 0.00003% (typ) loads with values as high as 100pF.
- **Input Noise Density:** 2.7nV/Hz (typ).
- The LM4562 is available in an 8-lead narrow body. Slew Rate: ±20V/s (typ) SOIC, an 8-lead PDIP, and an 8-lead TO-99.

- **Gain Bandwidth Product:** 55MHz (typ) Demonstration boards are available for each
- **Open Loop Gain (RL = 600Ω):** 140dB (typ) package.
- **Input Bias Current:** 10nA (typ)
- **Input Offset Voltage:** 0.1mV (typ)
- **DC Gain Linearity Error:** 0.000009%

Figure 2: LM4562

2.5 Multiple Feedback Filters



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

Figure 3: Multiple Feedback Filters

2.6 Chosen Frequency Bands and their Mid Frequencies

Bands	Mid frequency
60 - 300	203.106
300 -1000	644.338
1000 - 4000	2652.96
4000 - 10000	6964.185
10000 - 16000	13434.8523

Figure 4: Freuency bands and Mid Frequencies

2.7 Amplifiers

We were asked to change the gain of each frequency band. For that, we implemented separate amplifiers for each band. For the amplifiers also choose LM4562. Because this is specified for audio processing. And we use a potentiometer to change the gain.Here we use inverting amplifiers because after this we need implement the adder. Adder will invert the phase of the audio signal that's why we use inverting amplifiers.

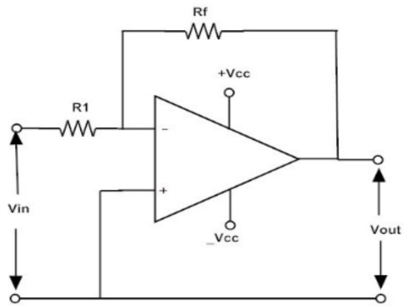


Figure 5: Amplifiers

2.8 Adder Circuit

After we process the frequency spectrum separately, we need to recombine the bands to get the audio signal. For that, we need to implement the adder circuit. For the adder circuit, we give each separate frequency band signal as the input. For this purpose also use LM4562 opamp.

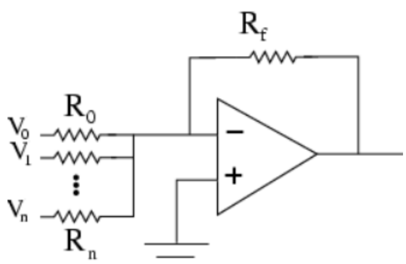


Figure 6: Adder Circuit

2.9 Power Supply Circuit

For the op amps we need ±9V. So, we decided to use center tap transformer to get those voltages. To regulate the voltage we Voltage regulator ICs.

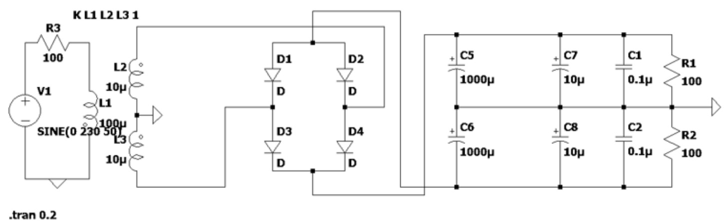


Figure 7: Power Supply

2.10 Output

We use LED stripes to show the gain of each band. With the help of this, we can adjust the gain of a particular band. And we use a 3.5 mm jack for audio output. We use LM3914 IC for the audio output to LED.

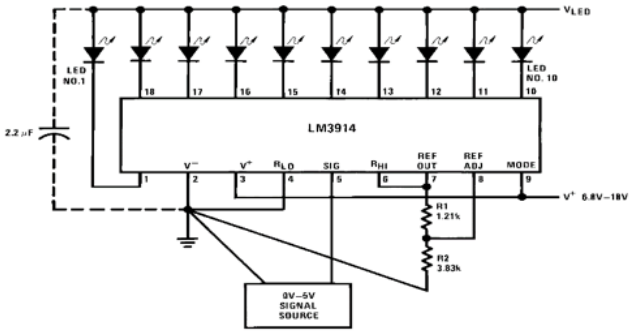


Figure 8: LM3914

2.11 Calculations

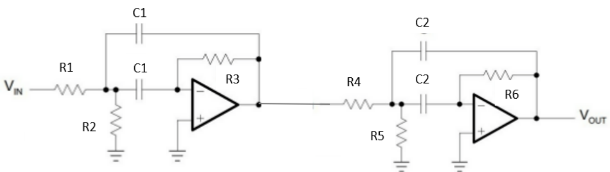


Figure 9: $4^{th} order filter$

Band	R1	R2	R3	R4	R5	R6	C1	C2
60 – 300 Hz	47K	68K	180K	15K	15K	47K	6.8nF	68nF
300 – 1000 Hz	1.5K	1K	47K	3.3K	2.2K	10K	100nF	100nF
1 – 4 kHz	1K	680	3.3K	1K	680	2.2K	33nF	100nF
4 – 10 kHz	1K	470	3.3K	1K	470	3.3K	15nF	33nF
10 – 16 kHz	33K	1.5K	100K	33K	2.2K	100K	1nF	1nF

Figure 10: Table of Components

Images and Tables attached on appendix separately.

To separate each band, we have to implement a band pass filter for each band. For that, we use software called analog filter wizard. With the help of this software and the transfer function, we decide the resistor values and capacitor values. We checked the values with the software. After that, we approximated the value to the market value.

3 Simulation and Testing

We use LTspice for our simulations first we tried with proteus and modelsim, but we can't simulate our circuit in real-time. So, we faced some errors in that software. So, some seniors instruct us to use LTspice for this project.Before we design the PCBs we implemented the circuit in the breadboard and checked it. After that, we use Altium for design our PCBs.

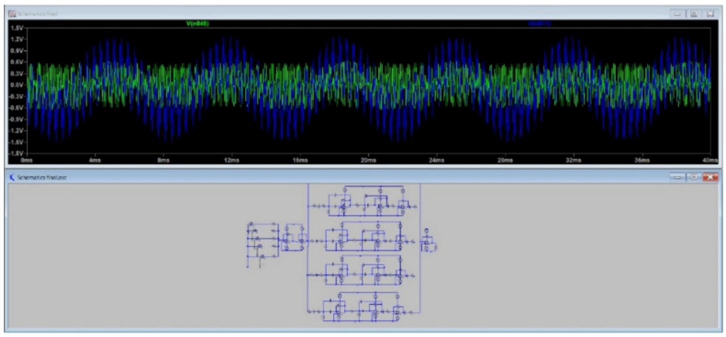


Figure 11: Simulation and Testing

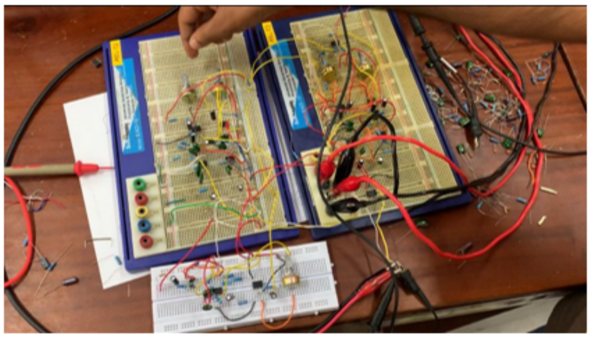


Figure 12: Bread Board Implementation

Figure 13: Bread Board Implementation

4 Results

For mid frequencies that we mentioned above we get a good amplifying. after the cutoff frequencies, we should get the gain as zero but here we use only forth order filters so we can't get accurate sharp edges. So, there are some errors in the cutoff frequencies. But that is not a big issue. In real life, we can't get a sharp edge.

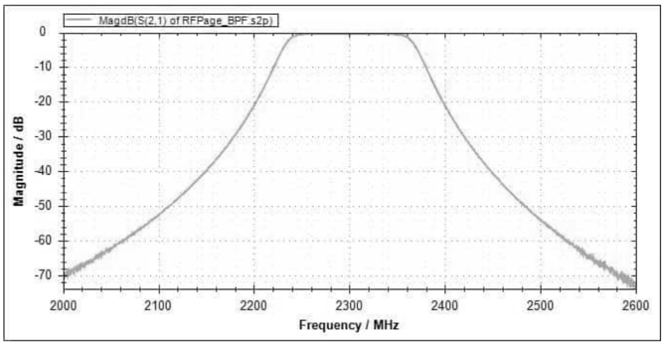


Figure 14: Real Life Bandpass filter

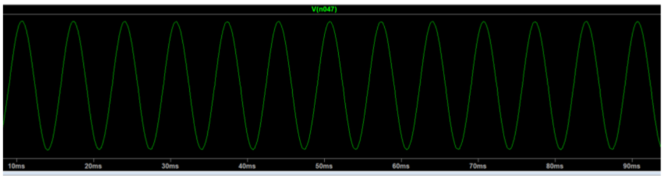


Figure 15: Results for Low Frequency

And with the help of the amplifiers, we can get a gain of 10. If we increase more than 10 we will get a distorted signal. That's why we are limited to 10. Because of the lower order and the IC, we chose we are getting some noise for high-frequency components. But we are getting a clear signal for low-frequency components. But we can't differentiate the noise so it shouldn't't a big issue.

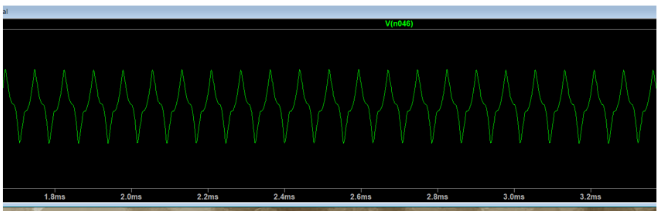


Figure 16: Results for High Frequency

We can get ±12V from the transformer. But there is a voltage drop in diode so we can't regulate it for 12 V so we use 7909 ICs to regulate the voltage as 9V. LM4562 can works in ±9V. So, this will not an issue.

5 Discussion

This project contains almost all the requirements that were asked. LM4562 can use almost all frequency components, but there are some issues with high frequency. But we can ignore it. We can't hear high frequencies. While we are designing filters we face some issues. The resistor and capacitor values that we got from the transfer function and software are not available in the market. So, we approximated the values. Because of that, there are slight changes in the cutoff and mid frequencies. But not that much.

First, we choose modelsim as simulation software. While we are using that software, we faced lots of issues. Because we can't use modelsim for real-time simulations. So, we have only one option LTspice. But, in LTspice also there is an issue in that we can't input an audio signal in LTspice. For that, we create a frequency spectrum using an adder circuit and use that as the input signal. Because of that, we can't get accurate results.

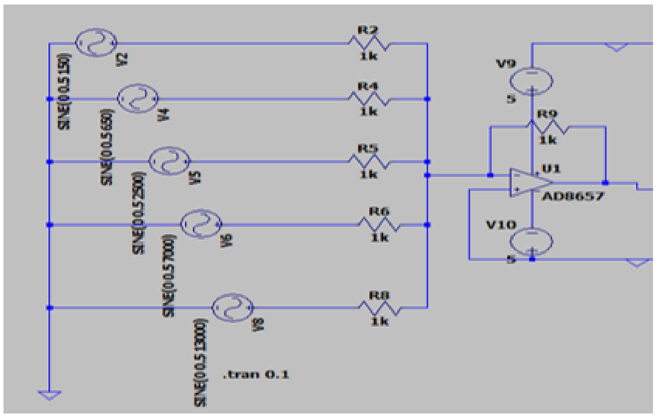


Figure 17: Adder Circuit in LTspice

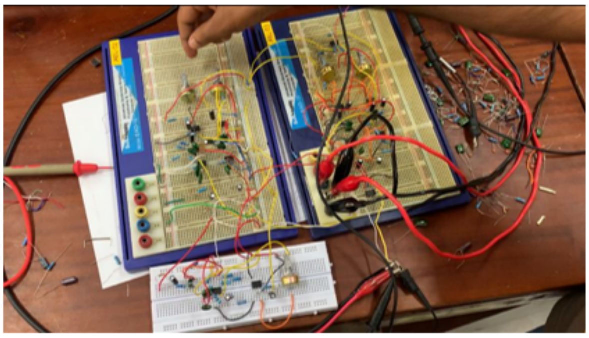


Figure 18: Bread Board Implementation of Input Circuit

6 Enclosure Design

We used solidworks for designing the enclosure for our project. After getting printed we joined them using screws. We created our enclosure in two parts one for lid and one for the base. Following figures show the images of the enclosure designs

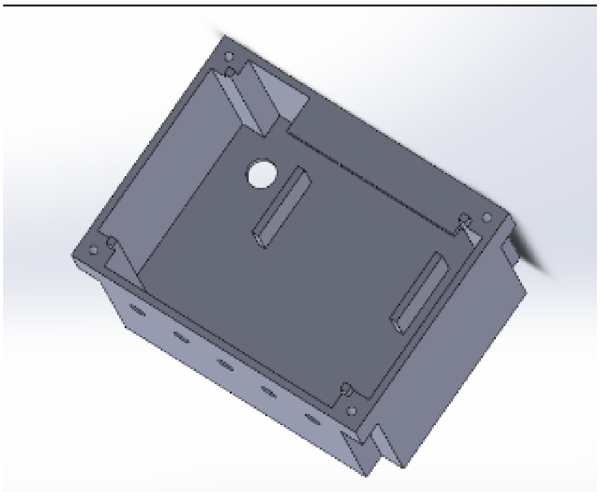


Figure 19: Lid

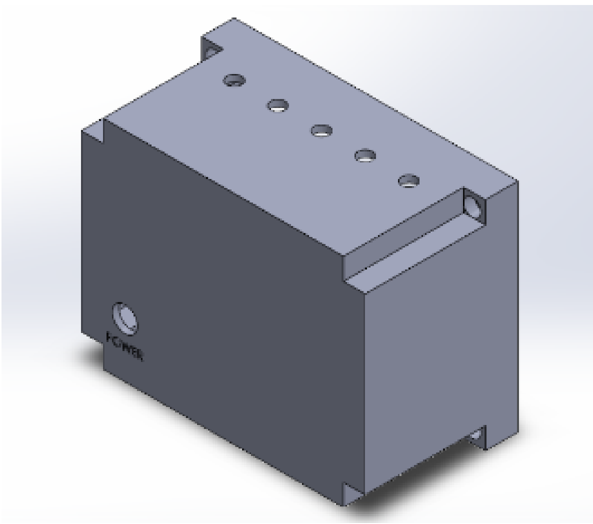


Figure 20: Lid

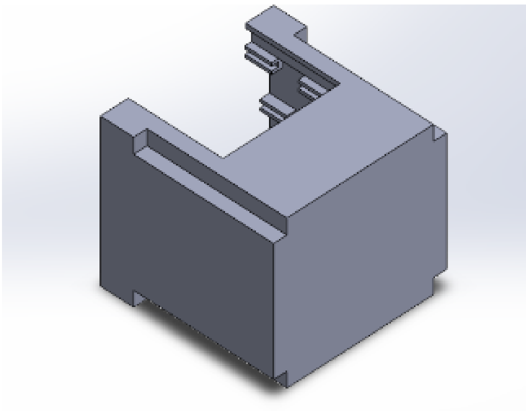


Figure 21: Base

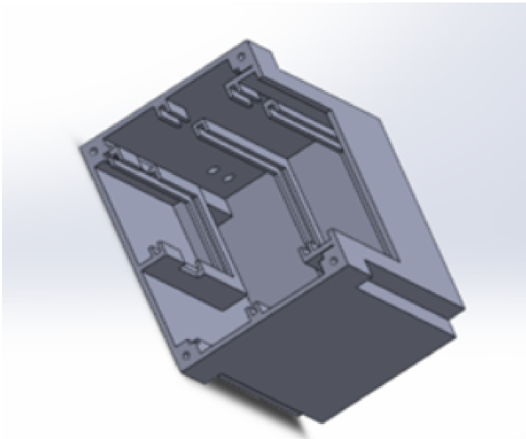


Figure 22: Base

Enclosure After assembling the Base and Lid

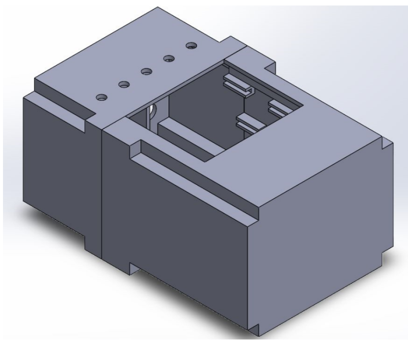


Figure 23: Assembled Enclosure

7 PCB Design

After checking the circuit with the bread board, we designed the circuit using Altium Designer. And we send the gerber files to the PCB printers to print the PCB. After we got the PCBs we soldered the components on it and checked it.

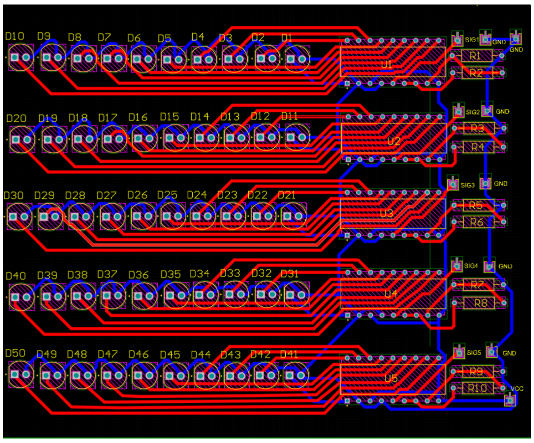


Figure 24: Output PCB

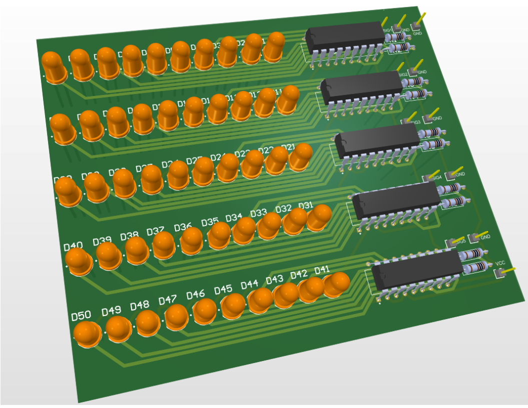


Figure 25: 3D design of Output PCB

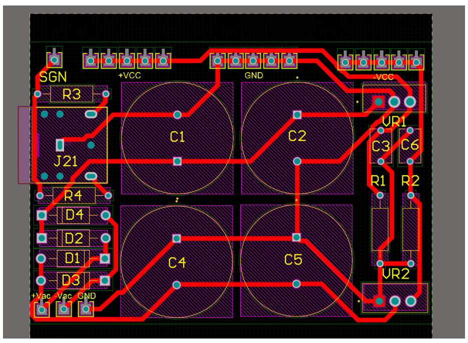


Figure 26: Power Supply PCB

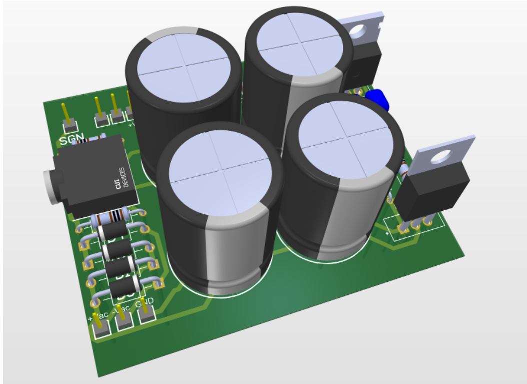


Figure 27: 3D design of Power supply PCB

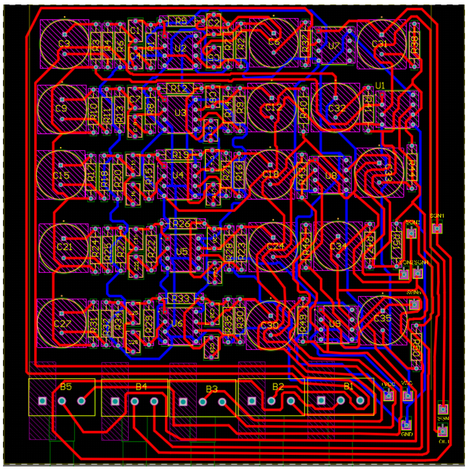


Figure 28: Filter PCB

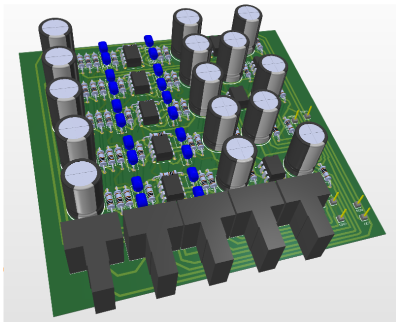


Figure 29: 3D design of Filter PCB

8 Final Product

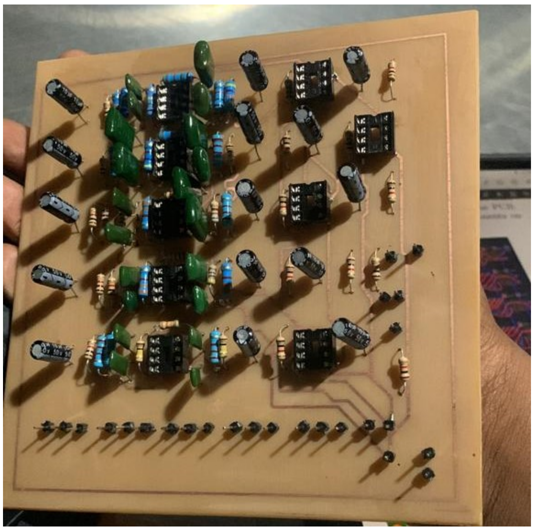


Figure 30: Fully Soldered Filter Cicuit

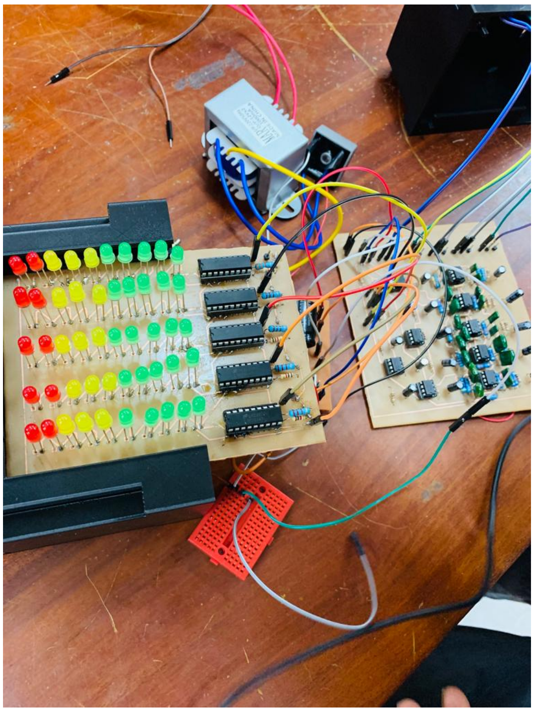


Figure 31: Fully Assembled Circuit

Figure 32: Assembled PCBs inside Enclosure

Figure 33: Fully Finished Product

9 Acknowledgement

First, we like to pay our gratitude to Dr. Perera M.T.U.S.K. Sampath. Because he guided us through this project. He gave us the initial idea about the project and help us to move on. After that, we have to thank Pahan Mendis who is our final-year senior. He spent more time with us and help a lot. Every week, he spent some time with us. We worked very freely with him. He treats us as his brothers and sisters. Particularly we should thank Dr. Prathapasinghe Dharmawansa. Because he gave a very clear idea about filters. With the help of the module that we taught in semester 3, we did this project. At last, we should thank Amal ayye who is our department's non-academic staff for helping for printing our PCBs.

10 Task Allocation

Member	Tasks
CHANDRASIRI Y.U.K.K.	Enclosure Design, Circuit testing and debugging
KANNANGARA N.V	Circuit testing and debugging, PCB Design
NIRUSHTIHAN B.	Documentation, PCB Setting and Soldering
VISHAGAR A.	Schematic design, Documentation

Figure 34: Task Allocation

11 Appendices

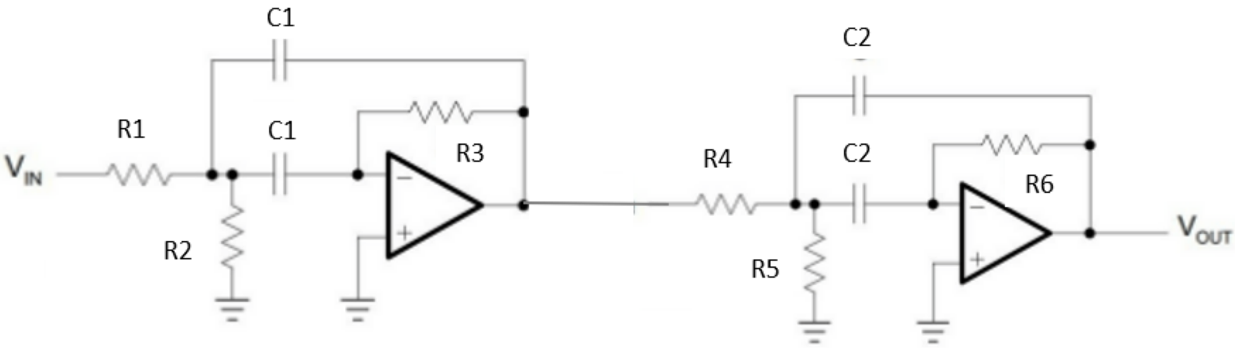


Figure 35: $4^{th} orderfilter$

Band	R1	R2	R3	R4	R5	R6	C1	C2
60 – 300 Hz	47K	68K	180K	15K	15K	47K	6.8nF	68nF
300 – 1000 Hz	1.5K	1K	47K	3.3K	2.2K	10K	100nF	100nF
1 – 4 kHz	1K	680	3.3K	1K	680	2.2K	33nF	100nF
4 – 10 kHz	1K	470	3.3K	1K	470	3.3K	15nF	33nF
10 – 16 kHz	33K	1.5K	100K	33K	2.2K	100K	1nF	1nF

Figure 36: Table of Components

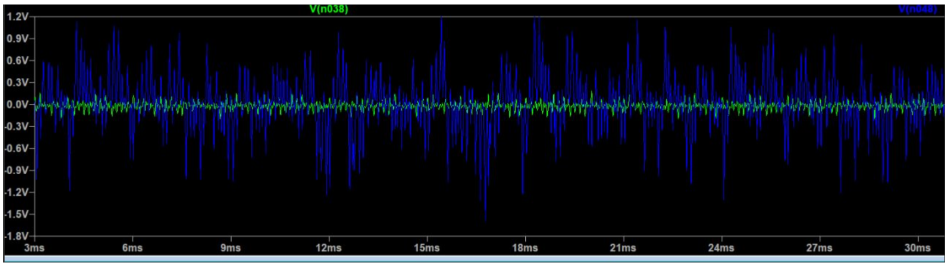


Figure 37: Output of the Simulation

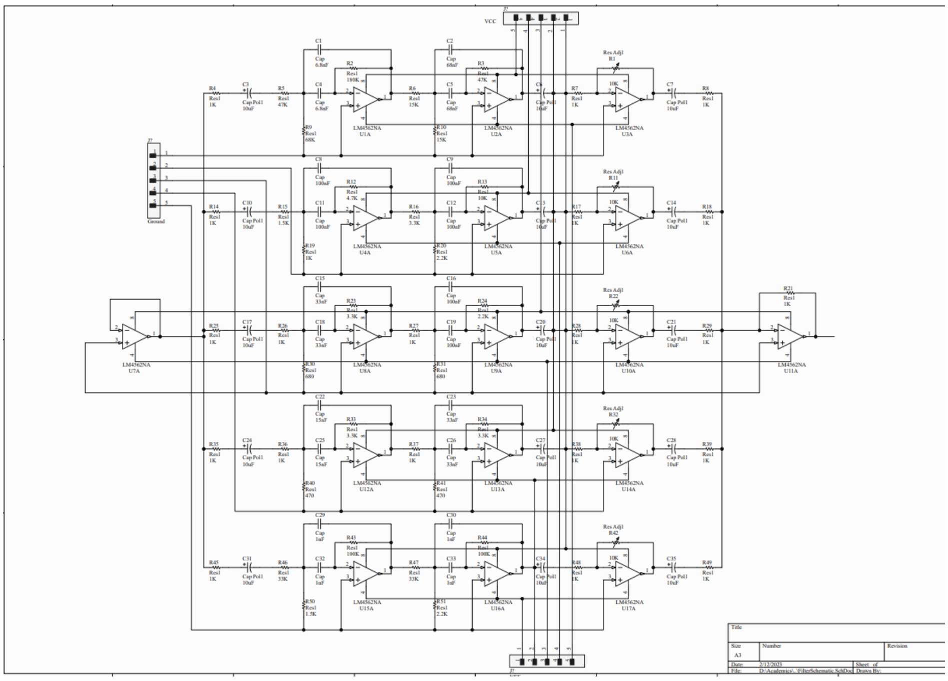


Figure 38: Schematic of the Filters and Amplifiers

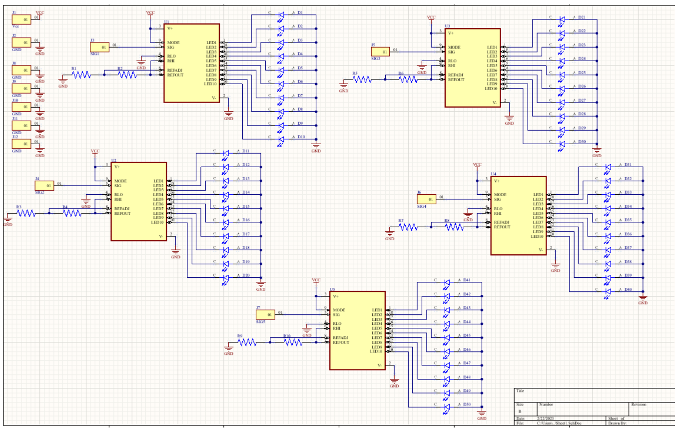


Figure 39: Schematic of the Output Circuit

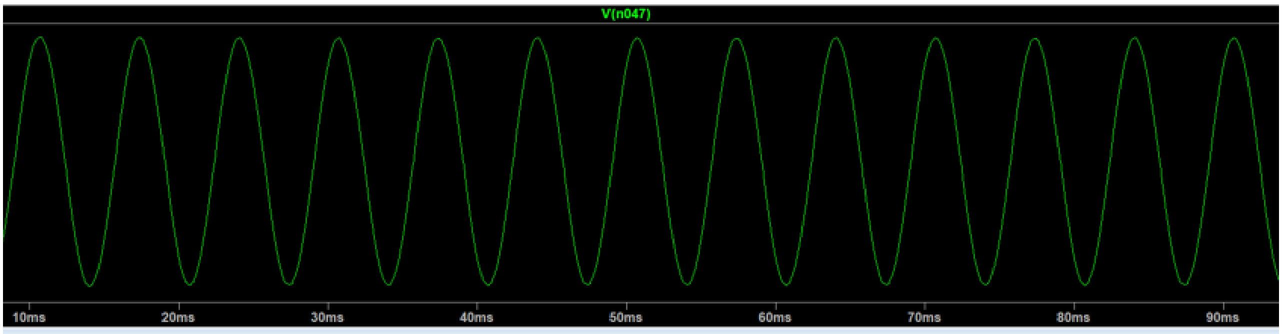


Figure 40: Results for Low Frequency

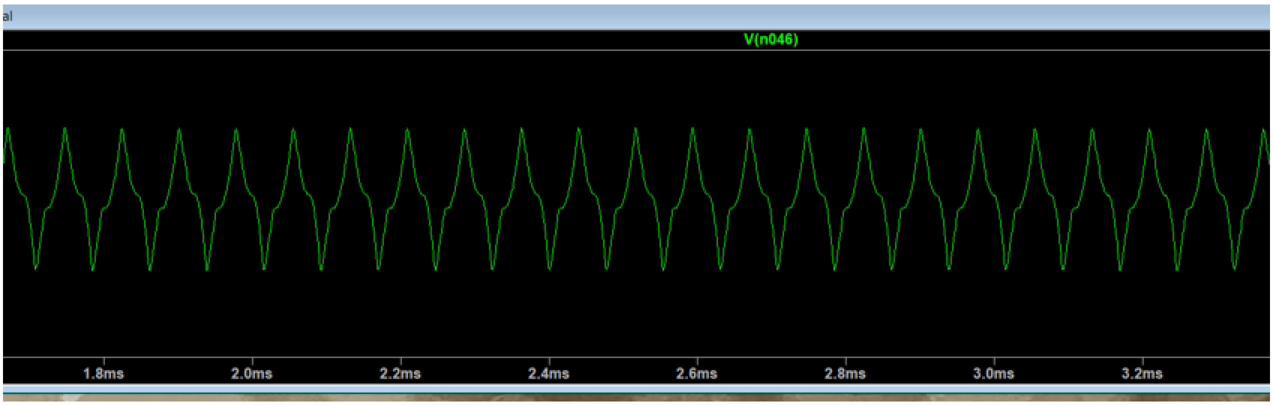


Figure 41: Results for High Frequency

12 Bibliography

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