# Bass Guitar Amplifier



## Senior Design I - Final Draft

**Group 26 Fall 2022**

Kris Edstrom *(Computer Engineering)*
Armon Eghbali *(Electrical Engineering)*
James Howell *(Electrical Engineering)*
Jeremy Nelson *(Electrical Engineering)*

# Table of Contents

# 1.0 Executive Summary

A bass amp header is a piece of equipment which allows a user to output an electric bass guitar through a speaker. It performs this by using a preamp to bring the signal from the bass up to a line level, equalizing this signal, and then using a poweramp to increase the wattage of the signal. At this point the signal is able to drive a loudspeaker. Most amp headers consist of only these components, and give the user basic control over volume, and adjusting the bass, mid and treble frequencies. Applying effects to the signal requires separate devices which drives up the price of an already expensive field.

Effects can either be acquired through individual pedals, or by using multi-effect processing units. These units are used in conjunction with an amp, either modifying the signal prior to entering the amp, or an amp may use an effect loop, which can apply an effect to the signal after a voltage gain has been applied by the preamp. This is primarily determined based on the effect, some of which are user preference, since whether the effect is applied before or in the effect loop changes the sound.

This means picking up the bass guitar as a hobby or profession can be an expensive and confusing task. This project will design a product which solves this problem by creating an all in one unit that will apply digital and analog effects to a signal, as well as providing the basic functionality of an amp header to drive a speaker.

This device will be simple to use, and provide the previously stated features at a competitive price. The product will consist of an amplifier's basic components, a preamp, equalization stage, and power amp, a digital signal processing unit to apply digital effects to the sound, as well as circuitry to apply analog effects to the signal. The user will interface with the device through knobs, as well as a touchscreen, which will be handled by a microcontroller unit. The device will also handle power management from a wall outlet to all of its individual components.

The product will provide an approachable and affordable way for anyone to begin playing the bass guitar, while providing enough features to interest experienced players as well.

The scope of the project provides the group with a variety of technologies to be researched, and ample room to learn and use the skills gained over the course of its development. It also provides a modular design which allows for expanding the scope to include more effects, or increasing the amplification to drive a larger speaker.

# 2.0 Project Description

## 2.1 Motivation

Music is enjoyed by many on a daily basis, both by people who listen or play music. Aside from people's general intimidation of learning an instrument, the cost to enter this field acts as another deterrence. This holds especially true with electric instruments such as an electric bass guitar which requires an amplifier to produce aurally appealing sounds. However, these amp headers can cost an absurd amount of money, leading some to never venture into the world of music. This problem is worsened by the need to use pedal effects, which does enhance the playing experience, but still cost a substantial amount of money. We desire to make music more accessible and enjoyable to those of all ages by reducing the financial risk associated with leaping into the world of music.

## 2.2 Project Goals and Objectives

Our project and objective is to make an affordable bass amp header. We aim to make an efficient, yet quality amp header that does not have unnecessary engineering and features to ramp up the price. This amp header would have multiple outputs such as a headphone jack, a speaker output, as well as bluetooth output. An inevitable part of advancing your techniques on an electric bass guitar would be the use of pedals and effects. Instead of leaving these features off, only for the customer to later go on and buy it themselves, they will be implemented in the amp head.

Some standard effects, such as distortion, do not have a lot of variability so it will be implemented using analog circuitry. Whereas other effects will be implemented using digital circuitry, allowing for customizability and unique effects to be implemented in the header. The user would be able to control these effects through a touch screen interface built into the amp header.

The touch screen interface would be user intuitive and would allow a large degree of customization over how the effects are applied. The ability to interface between digital effects, generated by a DSP, and analog effects, from analog circuitry, at the same time would be the main feature of the project. Providing these advanced effects at a low cost would help satisfy the goal of creating an affordable bass amp.

- Touch screen interface capable of controlling digital effects
- Analog effects such as fuzz, distortion, overdrive
- Digital effects such as reverb, delay, and other custom effects implemented with digital circuitry.
- Ability to output to a speaker or headphones

## 2.3 Project Specifications

### 2.3.1 Project Specifications Analysis

Table 2.3.1.a details the specifications that were targeted for this project. These specifications were targeted in order to make our amplifier competitive, with other options on the market. The specifications have also considered the realistic constraints placed on the project.

| Attribute | Description |
|---|---|
| Size | 24" x 12" x 12" (W x H x D) |
| Weight | 12lbs |
| Output Power Rating | 100W @ 8Ω |
| Input Power Rating | 120VAC 60Hz |
| Frequency Response | 31Hz - 4.5kHz |
| THD (Total Harmonic Distortion) | 0.5% |
| Input Impedance | 1MΩ |
| Output Impedance | 8Ω |
| SNR (Signal to Noise Ratio) | 80dB |
| Analog Effects | 4 |
| Digital Effects | 4 |
| Inputs | ¼" (6.35mm) Audio Input Jack<br>¼" (6.35mm) Effect Input Jack |
| Outputs | ¼" (6.35mm) Speaker Pass Through<br>⅛" (3.5mm) Aux Output<br>Bluetooth 5.0 |

Table 2.3.1.a - Project Specifications

## 2.4 House of Quality

### 2.4.1 House of Quality Analysis

Figure 2.4.1.a is the House of Quality for this project. This illustrates the comparison of customer requirements to the actual engineering specifications behind these needs. For each of these comparisons the relationship is labeled as a strong, moderate or weak relationship. So for financial cost, a strong relationship is labeled for the analog and digital effects, which would require more development time and circuitry as more effects are implemented, while it has a weak relationship to target weight, since the cost of the project will not inherently determine anything about its weight. For reliability, all of the relationships are labeled weak, since none of the engineering requirements specifically target the reliability of the product, instead it will just come down to the build quality and design as a whole. The safety of the system will primarily be down to the output power and impedance, since if this were to be poorly managed, it is what would put the user at the greatest risk of receiving a shock from the device. The safety is also impacted by the weight, since too heavy of a unit could cause strain when moving it, or injury if it fell.

Next, we have the direction of improvement for each of the engineering requirements, the first being the output power rating, which improves by maximizing it, since having a larger output power will allow the amp to drive larger speakers. Other requirements such as the input and output impedance are specifically targeted at that value. The weight and total harmonic distortion are requirements which can be minimized, since a lighter product with less distortion is desirable. The number of effects and the signal to noise ratio can also be maximized, since a higher signal to noise ratio means a clearer signal, and more effects means more features for the device.

Finally, we can look at the correlations between the engineering requirements, which is read by looking at a square in the top triangle section, and seeing which requirement numbers line up diagonally with that spot. Some examples are that the peak of the triangle shows a positive correlation between the output power and the target weight, which shows that if the output power goes up, the weight will go up since increasing the power will require more components and a larger Power Amplifier. It can also be seen that increasing the number of effects will be correlated to an increase in the signal to noise ratio and the total harmonic distortion.
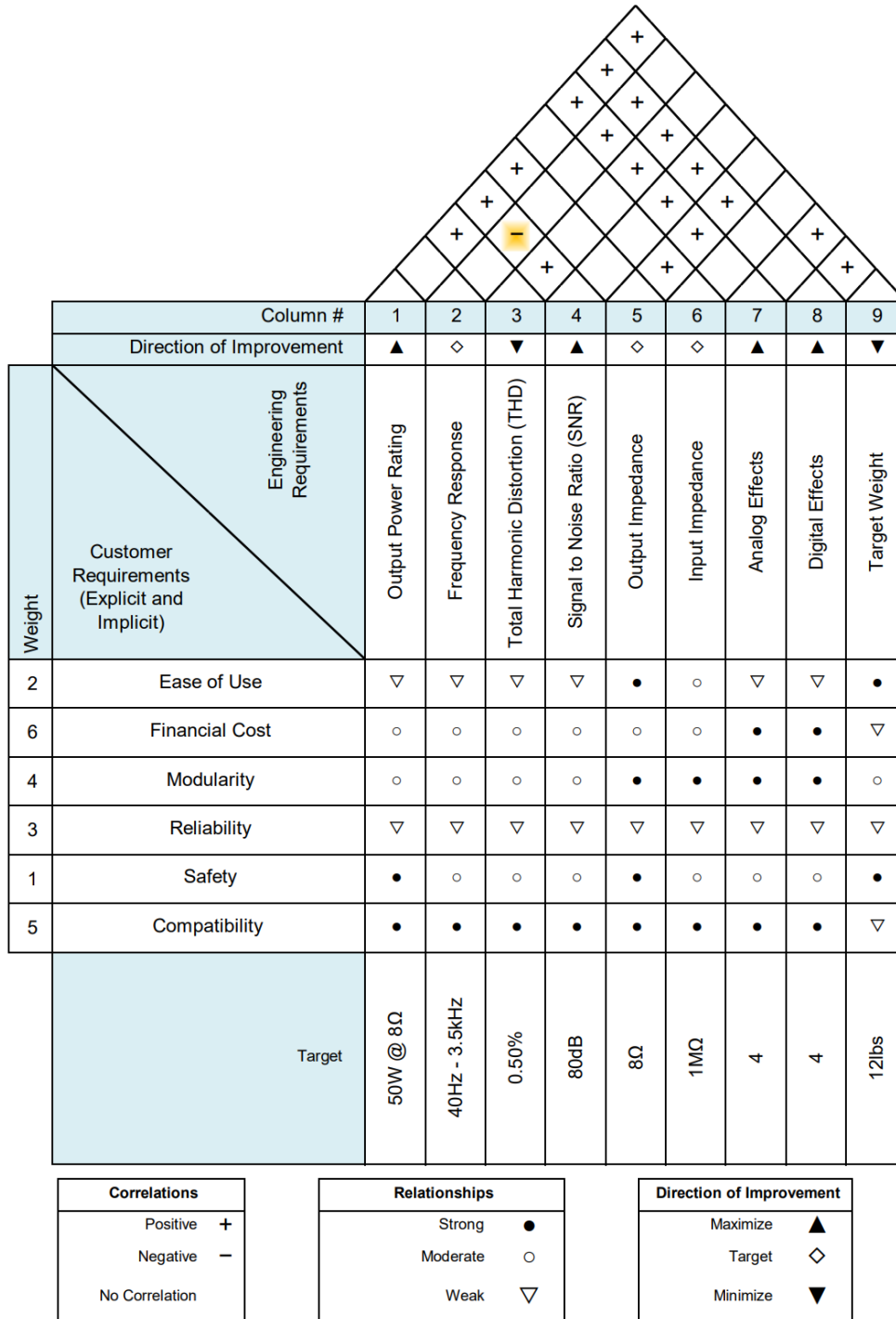
Figure 2.4.1.a - House of Quality

| Column # | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 |
|---|---|---|---|---|---|---|---|---|---|
| Direction of Improvement | ▲ | ◇ | ▼ | ▲ | ◇ | ◇ | ▲ | ▲ | ▼ |
| Engineering Requirements → | Output Power Rating | Frequency Response | Total Harmonic Distortion (THD) | Signal to Noise Ratio (SNR) | Output Impedance | Input Impedance | Analog Effects | Digital Effects | Target Weight |
| Weight / Customer Requirements (Explicit and Implicit) | | | | | | | | | |
| 2 — Ease of Use | ▽ | ▽ | ▽ | ▽ | ● | ○ | ▽ | ▽ | ● |
| 6 — Financial Cost | ○ | ○ | ○ | ○ | ○ | ○ | ● | ● | ▽ |
| 4 — Modularity | ○ | ○ | ○ | ○ | ● | ● | ● | ● | ○ |
| 3 — Reliability | ▽ | ▽ | ▽ | ▽ | ▽ | ▽ | ▽ | ▽ | ▽ |
| 1 — Safety | ● | ○ | ○ | ○ | ● | ○ | ○ | ○ | ● |
| 5 — Compatibility | ● | ● | ● | ● | ● | ● | ● | ● | ▽ |
| Target | 50W @ 8Ω | 40Hz - 3.5kHz | 0.50% | 80dB | 8Ω | 1MΩ | 4 | 4 | 12lbs |

**Correlations**
Positive +
Negative −
No Correlation

**Relationships**
Strong ●
Moderate ○
Weak ▽

**Direction of Improvement**
Maximize ▲
Target ◇
Minimize ▼

## 2.5 Project Block Diagram

### 2.5.1 Project Block Diagram Overview

Figure 2.5.1.a details the Project Block Diagram of the Bass Guitar Amplifier. This diagram was developed by considering the interactions between the necessary components of a bass guitar amplifier. The diagram considers the motivations, goals, and objectives outlined in previous sections.
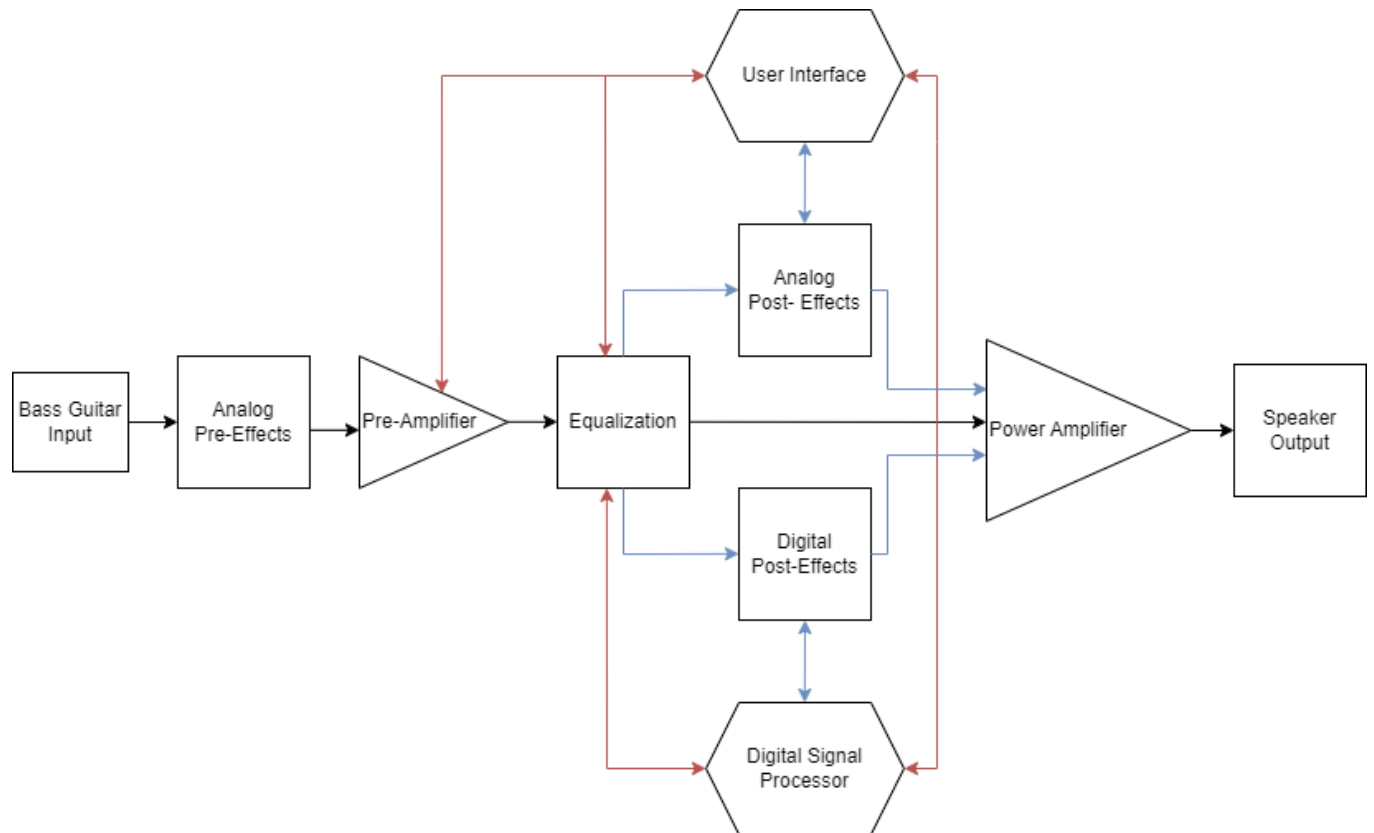


Figure 2.5.1.a - Project Block Diagram

| Block Name | Person Assigned | Description | Inputs/Outputs |
|---|---|---|---|
| Bass Guitar Input | | Bass Guitar | Inputs: None<br>Output: Analog Pre-Effects |
| Analog Pre-Effects | James Howell | Apply Analog Effects to the signal | Inputs: Bass Guitar<br>Output: Pre-Amplifier |
| Pre-Amplifier | James Howell | Raise input to line voltage | Input: Analog Effects, User Interface<br>Output: Equalization |
| Equalization | James Howell | Equalize the signal | Input: Pre-Amplifier, User Interface<br>Output: Analog Post-Effects, Digital Post-Effects, and Power Amplifier |
| Analog Post-Effects | James Howell | Apply Analog Post-Effects to the signal | Input: Equalization, User Interface<br>Output: Power Amplifier |
| User Interface | Kris Edstrom | Interface with the Components | Input: User<br>Output: All |
| Digital Post-Effects | Armon Eghbali | Apply Digital Effects to the signal | Input: Equalization, User Interface, DSP<br>Output: Power Amplifier |
| Digital Signal Processor | Armon Eghbali | Create the Digital Effects | Input: User Interface<br>Output: Digital Post Effect |
| Power Amplifier | Kris Edstrom | Amplify Signal for Speaker output | Input: Post-Effects<br>Output: Speaker |
| Power Supply | Jeremy Nelson | Generate Power for Components | Input: User<br>Output: All |

Table 2.5.1.b - Project Block Diagram Breakdown

# 3.0 Research

## 3.1 Pre-Amplifier

### 3.1.1 What is a Pre-Amplifier

A Pre-Amplifier, or preamp for short, is a type of amplifier that amplifies the input signal to line level. As the name suggests, the preamp comes before the power amplifier in an amplifier stack. Typically, a preamp only applies a voltage gain to the signal and allows the power amplifier to deal with current amplification. This voltage gain allows the signal to be more easily processed later in the stack.

### 3.1.2 Why Do You Need a Pre-Amplifier

Especially on an instrument amplifier, a preamp is necessary because the output of a bass guitar is in the range of 10mV to 100mV. This range will be different depending on the pickups being used, but the general range still shows that output amplitude is low. Being able to bring this low signal to line level, around 1V to 2V, allows the signal to be less susceptible to noise and more susceptible to different types of signal processing.

### 3.1.3 Ideal Pre-Amplifier

The ideal preamp has many important characteristics. The most important include:
1)      High Input Impedance
2)      Low Output Impedance
3)      Flat Frequency Response
4)      Low Noise

A high input impedance allows the preamp to see the signal without placing a load on it. This is also known as Impedance Bridging or a Bridge Amplifier in audio applications. The low output impedance allows the now amplified voltage to not be affected when a load is placed on the output. The flat frequency response is the most sought-after characteristic and the hardest to obtain. The voltage gain applied to the signal must be constant over the operating frequency. Low Noise is an important characteristic of most components in audio applications, but especially important in a preamp. If the signal has a lot of noise added on this stage, the final output after the power amplifier stage will be unusable.

### 3.1.4 Pre-Amplifier Topology

Due to these necessary characteristics, not all types of amplifier topologies are suitable for this application. Bipolar Junction Transistors, BJTs, have very high gain in its linear operating region and introduces very little noise to the signal, but has more drawbacks than positives. One drawback is the low input impedance due to the nature of how a BJT operates. Metal Oxide Semiconductor Field Effect Transistors, MOSFETs, have less voltage gain in its linear region than BJTs, but offer almost infinite input impedance. The downsides are a high output impedance and a high junction capacitance.

Next up are vacuum tube amplifiers. Being very sought-after for their sound quality in the audio world, tube amplifiers can offer high input impedance and low output impedance. There are reasons why tube amplifiers are not used anymore. Depending on which tube is used, the supply voltage needed is anywhere from 50V to 500V. They also require a tube heater which draws more power. They are also big and fragile. For those reasons, tube amplifiers will not be used. Junction Field Effect Transistors offer all characteristics needed for a preamp. Due to the structure of a JFET, the gate operates as a diode. Unlike on a MOSFET, where the gate operates like a capacitor. JFETs are commonly used when low noise is an important characteristic. For these reasons, the design of the Pre-Amplifier will be using a JFET topology.

### 3.1.5 Integrated Circuits or Discrete Stages

In today's world, the usage and availability of Integrated Circuits, ICs, has skyrocketed. One of the most common types of ICs are Operational Amplifiers, op amps. Based on solid-state or vacuum tubes, op amps are primarily used as voltage amplifiers with very high gain. Op amps can be configured to meet the other characteristics as well. Other characteristics of an op amp IC include:

1) High Common Mode Rejection Ratio, CMRR
2) Low Noise
3) Low Power Consumption
4) Low Cost
5) Broad Availability

Discrete Stages can still be used in preamp design, but the positives of an op amp greatly outweigh the positives of discrete designs. Due to the goals of this project, the design of the Pre-Amplifier will use Op Amps with a JFET architecture.

### 3.1.6 Op Amp Selection

There are thousands of op amps available for the design. Not all of them meet the criteria needed for the design. The characteristics that are needed are listed below and in Tables 3.1.5.a and 3.1.5.b:

1) Common Mode Rejection Ratio, CMRR
2) Gain Bandwidth Product, GBP
3) Slew Rate
4) Total Harmonic Distortion + Noise, THD + N
5) Voltage Noise Density
6) Cost

| Part | CMRR (dB) | GBP (MHz) | Slew Rate (V/us) | THD + N (%) | VNoise Density (nV/√Hz) | Cost |
|---|---|---|---|---|---|---|
| AD8510 | 100 | 8 | 20 | 0.00005 | 8 | $3.85 |
| LT1792 | 105 | 5.6 | 3.4 | 0.00005 | 8.3 | $11.64 |
| OPA1642 | 126 | 11 | 20 | 0.00005 | 5.1 | $2.43 |
| OPA1652 | 110 | 18 | 10 | 0.00005 | 3.8 | $2.75 |
| AD711 | 88 | 4 | 20 | 0.00005 | 45 | $11.66 |

Table 3.1.5.a - Op Amp Options and Specifications

| Part | Minimum Supply Voltage (±V) | Maximum Supply Voltage (±V) | Quiescent Current (mA) |
|---|---|---|---|
| AD8510 | 4.5 | 18 | 2.5 |
| LT1792 | 4.5 | 20 | 4.2 |
| OPA1642 | 2.25 | 18 | 1.8 |
| OPA1652 | 2.25 | 18 | 2 |
| AD711 | 4.5 | 18 | 2.5 |

Table 3.1.5.b - Op Amp Power Specifications

Based on these specifications, the competition was narrowed down to either the OPA164x series or the OPA165x series. The low cost and statistically better options from Texas Instruments was the deciding factor. Due to the slightly higher CMRR and Slew Rate of the OPA164x series, it was determined that the Pre-Amplifier would be designed using this op amp.

## 3.2 Overload Detector

### 3.2.1 What is an Overload Detector?

An Overload Detector is a circuit that detects if the input signal amplitude exceeds a controllable limit. When this detector is triggered, it will notify the user that the voltage is above the safe limit. Usually, this is in the form of an automatic shutoff or a led.
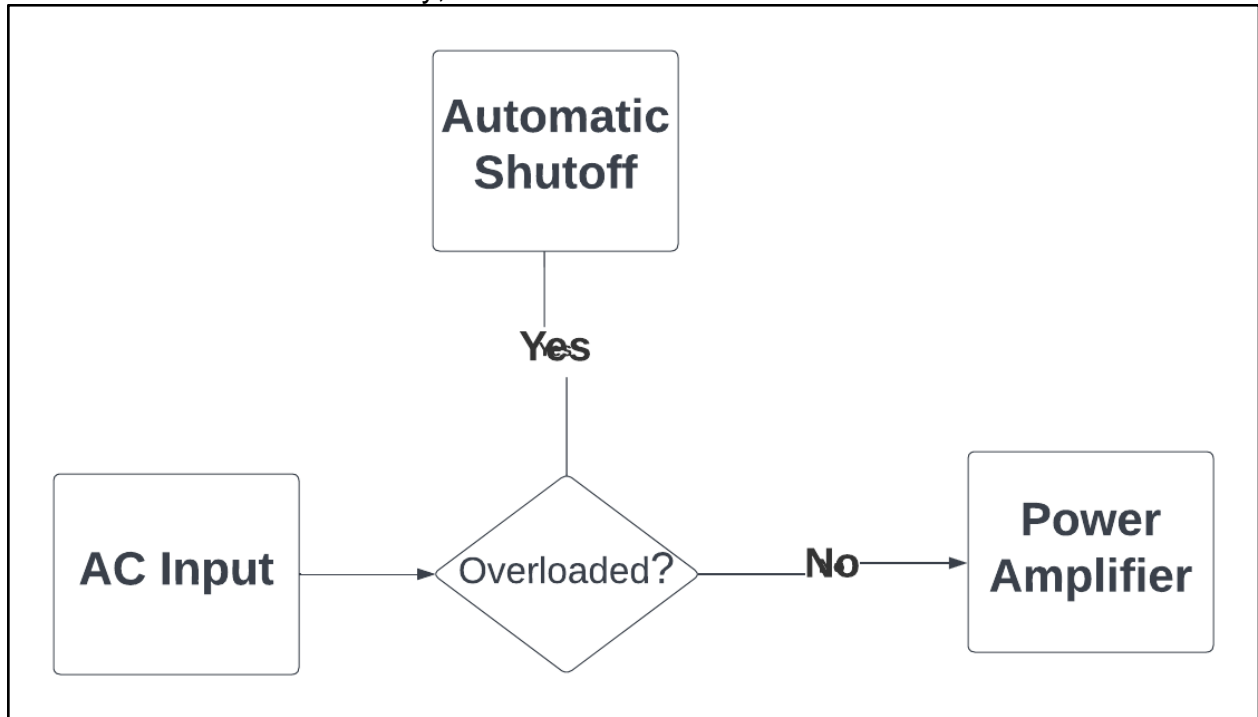


Figure 3.2.1.a – Overload Detector Flowchart

### 3.2.2 Why Do You Need an Overload Detector?

When dealing with any type of power amplification, it is important that it is done in a safe, controllable manner. Complying to safety standards is pertinent to the design being viable. Even though the gain on this stage is not as high as it will be in the power amplification stage, it is still important to detect if there is a problem now and be able to fix it.

### 3.3.3 Overload Detector Topology

There are many ways to design a voltage overload detector. Voltage Comparators, Zener diodes, or even dedicated IC's can be used in this situation. For simplicity, it was decided to use a LM393 Voltage Comparator. This comparator will be set to detect a line voltage over 5V and turn on a red led. This lets the user know that the voltage is over the safe limit and needs to be adjusted.

## 3.3 Equalization

### 3.3.1 What is Equalization

Equalization, commonly known as just EQ, is the application of filters in order to adjust the gain of specific frequencies. This is done so that the frequency response of the amplifier can be adjusted for the flat response or personal preference. EQ, on a bass amplifier, is very important for achieving the tone and sound quality preferred by the genre of music. EQ also happens to be commonly overlooked on commercial products and tends to be lackluster, without a standalone unit.

### 3.3.2 Why Do You Need an Equalizer

Arguably the most important component of a bass guitar amplifier, the Equalizer is a must-have in any amplifier design. The ability to adjust the frequency response to match the desired sound is the reason for this importance. EQ is usually the first step modifying the input signal and is the most notable part in finding the desired sound.

### 3.3.3 Parametric or Graphic Equalization

Parametric Equalization gives the ability to continuously control the equalization parameters, while the input signal is active. The adjustable parameters are gain, frequency, and bandwidth of the filter. This is in contrast to Graphic Equalization that does not allow continuous control. The only controllable parameter on the Graphic EQ is the gain. For this design, Graphic EQ was chosen to be used. Graphic EQ is simpler to design and cheaper to manufacture. Parametric would allow a greater ability to control the sound, but comes in at a much higher cost.

### 3.3.4 Equalizer Component Selection

Having many of the same ideal characteristics as the Pre-Amplifier, the component selections for the Equalizer led to very similar outcomes. Due to the operating characteristics of the op amp, it was decided that the main source of voltage amplification and attenuation would come from op amps. This time, the cost of individual op amps was weighed heavier because of the amount that will be used. Looking for lower cost, but still similarly high performance, led to Table 3.3.4.a. Table 3.3.4.a showcases the specifications of potential op amp choices.

| Part | GBP (MHz) | Slew Rate (V/us) | THD + N (%) | VNoise Density (nV/√Hz) | Cost |
|------|-----------|------------------|-------------|--------------------------|------|
| OPA1642 | 11 | 20 | 0.00005 | 5.1 | $2.43 |
| NE5532 | 10 | 9 | 0.002 | 5 | $0.57 |
| LM833 | 15 | 7 | 0.002 | 4.5 | $0.62 |
| TL971 | 12 | 5 | 0.003 | 4 | $0.75 |

Table 3.3.4.a - Op Amp Options and Specifications

These specifications led to the decision to use the NE5532 op amp from Texas Instruments. The low cost and comparable performance was the driving force behind this decision. Other components used, such as resistors, capacitors, and potentiometers will be of the highest quality. These standards will help keep the simulated performance accurate to the practical performance.
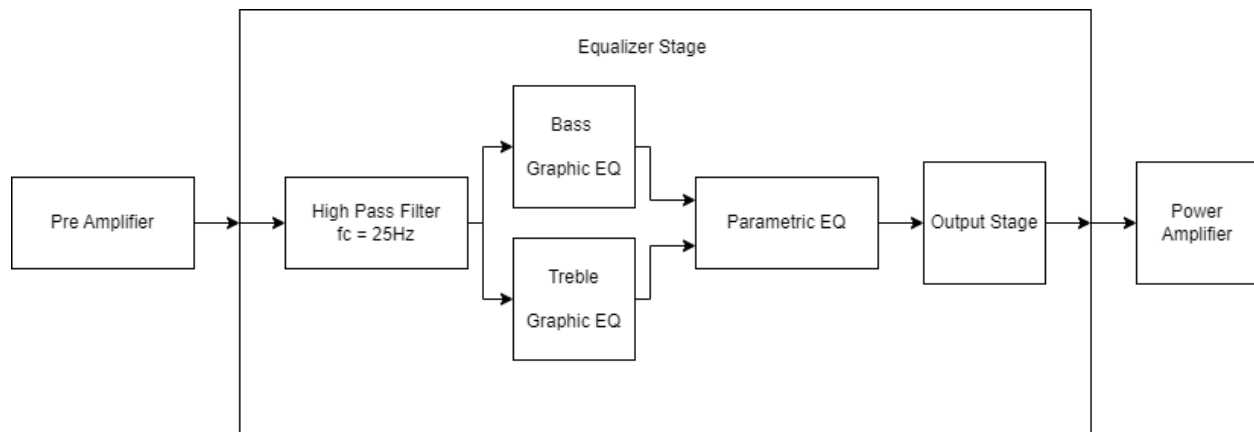


Figure 3.3.4.b - Equalizer Block Diagram

### 3.3.5 Active Filter Topology

There are many different topologies for active filter design. For the application of an equalizer, band pass filters are commonplace. The specific architectures of band pass filters, that were considered for the design, are as follows: Bessel, Butterworth, Chebyshev, and Gyrator Based.

Bessel band pass filters are very good at preserving the phase response of the input. This is an important characteristic of an equalizer, but the drawbacks outweigh the positives. Some drawbacks are a relatively low filter quality, Q, when compared to the

other filter types and a higher number of stages required. This increases the cost and design complexity too much for the desired effects.

Butterworth filters are a great option for when an almost perfectly flat frequency response in the pass band is required. Due to the nature of the filter, the pass band bandwidth and center frequency are able to be implemented exactly. The Q factor is slightly higher than the Bassel. The total number of components and stages required drive the complexity and cost up significantly. Other than that, the Butterworth filter is a great option for the equalizer.
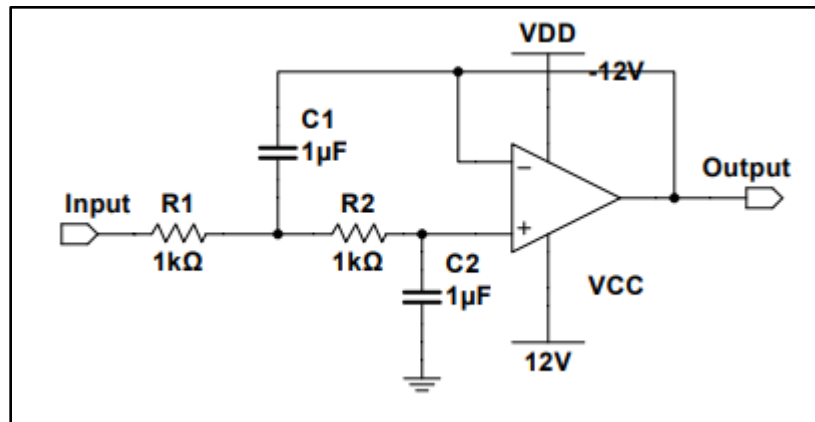


Figure 3.3.5.a - Sallen-Key Butterworth Filter

Chebyshev filters are very similar in construction to that of the Butterworth filter. The Chebyshev filter has a much higher stopband attenuation, but has a passband ripple. This is undesired for the application, but could be mostly eliminated with a stage increase. The filter quality of a Chebyshev is significantly higher than the Bessel or Butterworth.
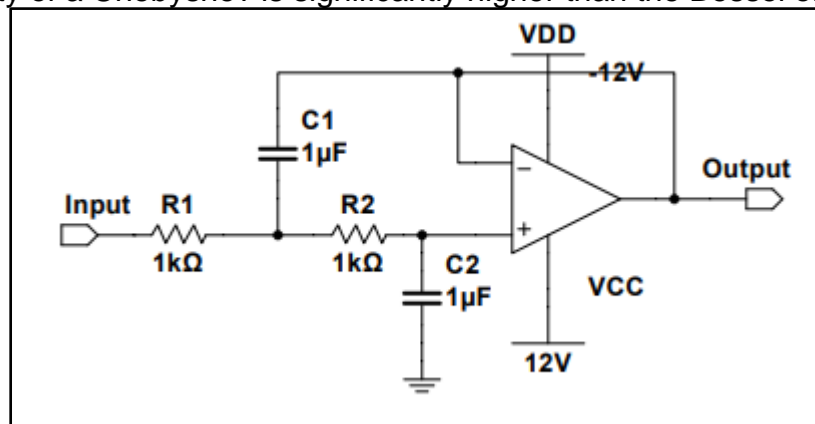


Figure 3.3.5.b - Chebyshev Filter

The last filter architecture in consideration is the Gyrator bandpass filter. A Gyrator filter has the ability to simulate the characteristics of an inductor, but only using resistors, capacitors, and op amps. The op amp can be substituted for transistors, but the architecture of the design calls for op amps. Gyrator bandpass filters are simple in design

14

complexity and cost-effective. They are also extremely common in other products featuring graphic equalization. The Gyrator filter was chosen for the design.
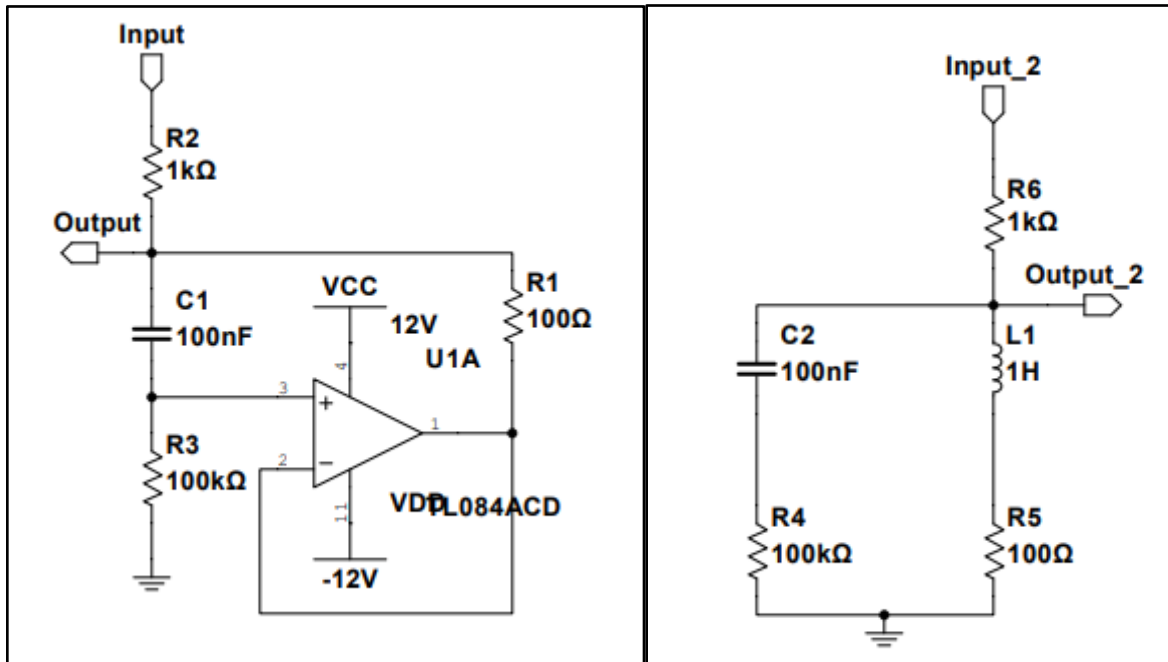


Figure 3.3.5.c - Equivalent Gyrator and Inductor Filters
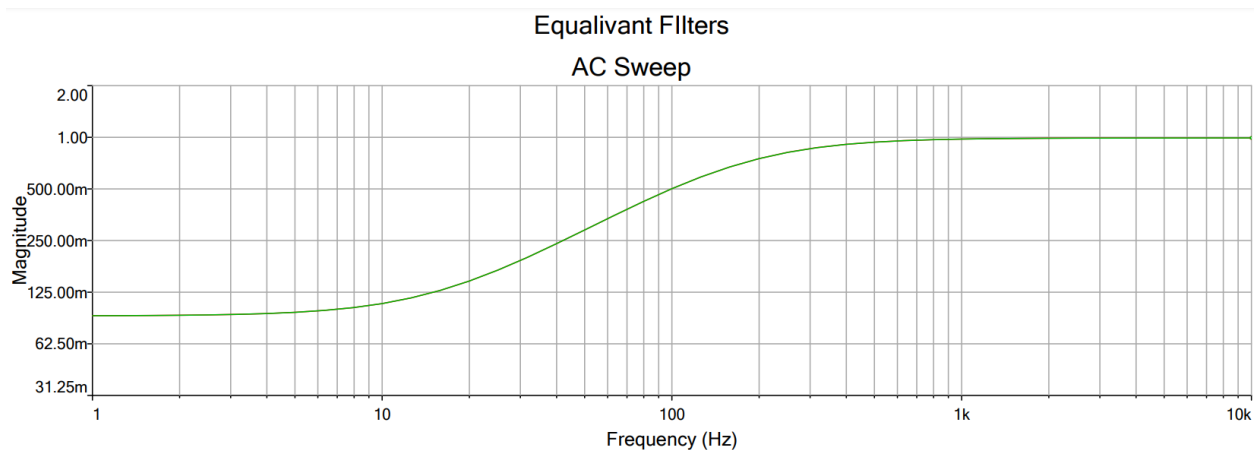


Figure 3.3.5.d - Frequency Response of Gyrator and Inductor Filters

## 3.4 Power Amplifier

### 3.4.1 What is a Power-Amplifier

A power-amplifier, or poweramp, receives the preamps signal which has gone through equalization. While a preamp typically only applies a voltage gain to the circuit, the poweramp applies a current gain to the circuit. This increases the power of the signal, which is what makes it able to drive the speaker.

15

### 3.4.2 Why Do You Need a Power Amplifier

On the bass guitar amplifier, the poweramp is necessary because the preamp has brought the voltage of the signal up to line level, the current is still low. The goal of the poweramp is to take this signal and increase the current in order to drive a large speaker. This is done through current amplification, in order to preserve the integrity of the waveform produced by the preamp. One of the primary reasons this is separated into a separate stage is because the current amplification to increase the wattage generates heat which needs to be dissipated.

### 3.4.3 Power Amplifier Parts

Table 3.4.3.a below shows the different possible parts which could be used for the power amp.

| Part | Slew Rate (V/us) | Output Power | THD + N (%) | Cost |
|------|------------------|--------------|-------------|------|
| LM3875 | 11 | 56W @ 8Ω | 0.06 | $9.33 |
| TDA7294 | 10 | 100W @ 8Ω | 0.01 | $10.45 |
| TPA3156D 2 | 10 | 50W @ 8Ω 70W @ 4Ω | 0.1 | $2.05 |
| TPA3221 | 10 | 170W @ 2Ω | 1 | $2.11 |

Table 3.4.3.a - Power Amplifier Options and Specifications

### 3.4.4 Cooling

The current amplification done by the preamp to drive the speaker ends up having a high thermal output, which needs to be dissipated. If not, some of the components may surpass their effective operating range, or the increase in heat could change the effective value of a component, changing the behavior of the circuit, which would cause an issue with the sound quality. In an extreme case, some components may become so damaged they are no longer usable.  There are two primary techniques of cooling, passive and active cooling.
Passive:

Passive cooling means there are no moving parts, the chip is cooled by the environment around it. This is usually improved by using heatspreaders or heatsinks. A thermally conductive material such as a thermal pad or thermal paste would be applied to the chip before a heatsink was attached to it. When the heatsink is attached, the heat from the chip will dissipate to the heatsink, which should cool off more quickly with a larger amount of surface area to be in contact with the air.  Heatsinks are designed with fins along the surface, which increase the surface area, which improves the cooling capabilities of it.
Active:

Active cooling means the device actively works to cool itself. This is done primarily through two ways, water and air cooling. Water cooling works by mounting a block to the IC which has tubes connected to it through which water can flow. The water will flow through a radiator, which can then have fans blowing on it to cool the water more effectively, which is pushed through this loop by a pump. This is more expensive and typically needed for higher thermal applications. Air cooling works similar to passive cooling, by attaching a heatsink to the chip. Rather than letting the ambient temperature of the air cool the heatsink, a fan is directed at it which pushes fresh air over the heatsink, which is able to dissipate more heat than it can passively. Table 3.4.x.a shows some of the options that can be used in this configuration.

| Part | Noise (dB) | Airflow (CFM) | Dimensions | Cost |
|---|---|---|---|---|
| 418-MC655HEATSINK KIT | N/A | N/A | N/A | $4.97 |
| V5619A | N/A | N/A | N/A | $0.36 |
| FAD1-06020DSHW11 | 31.0 | 16.4 | 60x60mm | $2.65 |
| EE80251S2-1000U-999 | 30.0 | 37.0 | 80x80mm | $3.82 |
| EE92251S2-1000U-999 | 32.0 | 45.0 | 92x92mm | $4.02 |
| CFM-9225V-117-177 | 17.7 | 28.5 | 92x92mm | $5.57 |

Table 3.4.4.a - Cooling Solutions

## 3.5 Power Supply

### 3.5.1 Overview

A power supply is, as it sounds, a means of providing power for the whole system. A power supply in this context includes: a transformer to reduce the 120V AC from a typical wall outlet down to a smaller more workable voltage, a AC to DC rectifier from the AC signal to the DC voltages that we will be using in this project, and lastly a voltage regulator to maintain the carefully chosen values and prevent any fluctuations using active components such as the LM317 adjustable regulator.

Correct power supply design is critical in order to ensure that all components of the bass amplifier are functioning within the boundaries of their specifications and working at their highest possible efficiency.
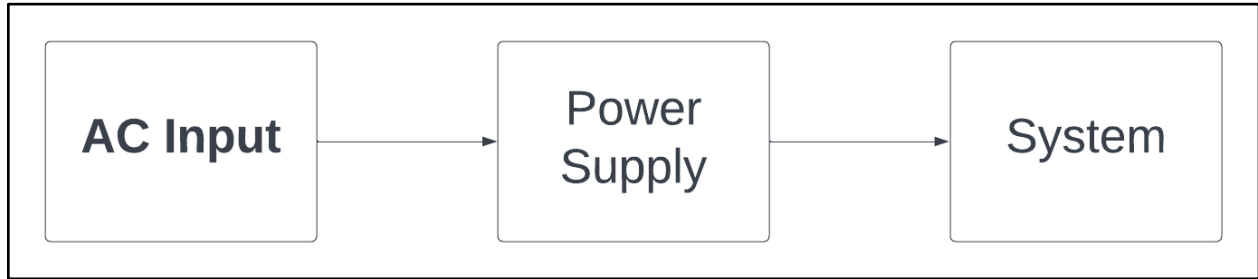
Figure 3.5.1.a - Overview

### 3.5.2 Types of Power Supplies

There are three major types of power supplies, *unregulated*, *linear regulated*, and *switching*. In the next sections I will go into detail about each type and the advantages or disadvantages of each. I will also discuss our choice and the reasoning behind that choice.

### 3.5.3 Unregulated Power Supplies

The unregulated power supply is the simplest to construct of the three major power supply designs. It consists of most of the components laid out in the overview (section 3.5.1) and nothing more. It is a Transformer leading into either a Full-wave or Half-wave rectifier, leading into a low-pass filter to filter out some of the high frequency AC noise. The advantages of such power supplies are their very cheap components, simplistic design, and also their rather high efficiencies. The simple design allows it to be very durable and unlikely to have components break down, assuming everything is within its specification temperature, voltage, and current. Unregulated power supplies tend to have an efficiency around 60% in most applications (iqsdirectory.com) which is not spectacular, but pretty good considering the cheap components.

The disadvantages of unregulated power supplies is, as its name implies, it doesn't have a voltage regulator. This can cause a number of issues in some applications (ours includes) such as: sudden and large fluctuations in voltage and lots of unfiltered AC noise. This is of course because any change in the AC input directly affects the DC output of the power supply due to the lack of voltage regulation.

The situations in which one would use an unregulated power supply is almost always in low-power applications and where minor modulations in frequencies or voltage are not noticeable or damaging. Such situations include if you were building a lamp, or a fan. The lamp is a fine application because any small changes in voltage result in a slight increase or decrease in luminosity. Especially since there are no moving parts in a lamp and it is very unlikely that any components would be damaged with such small modulations. For desk fans or any other type of fan would only slightly increase or decrease in speed which is not a problem for fans especially those with multiple speed settings. Of course, these are all cases of magnitude. If there was a severe enough

deviation from the baseline voltage, something that could occur in a power surge, this could be enough influx current to damage some components, but with correct design this would be avoided.
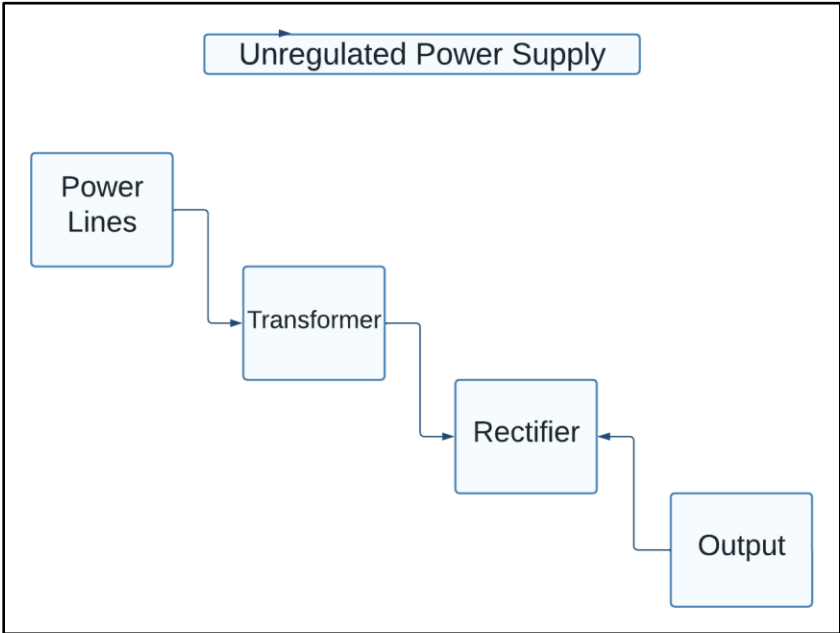


Figure 3.5.3.a – Unregulated Power supply block diagram
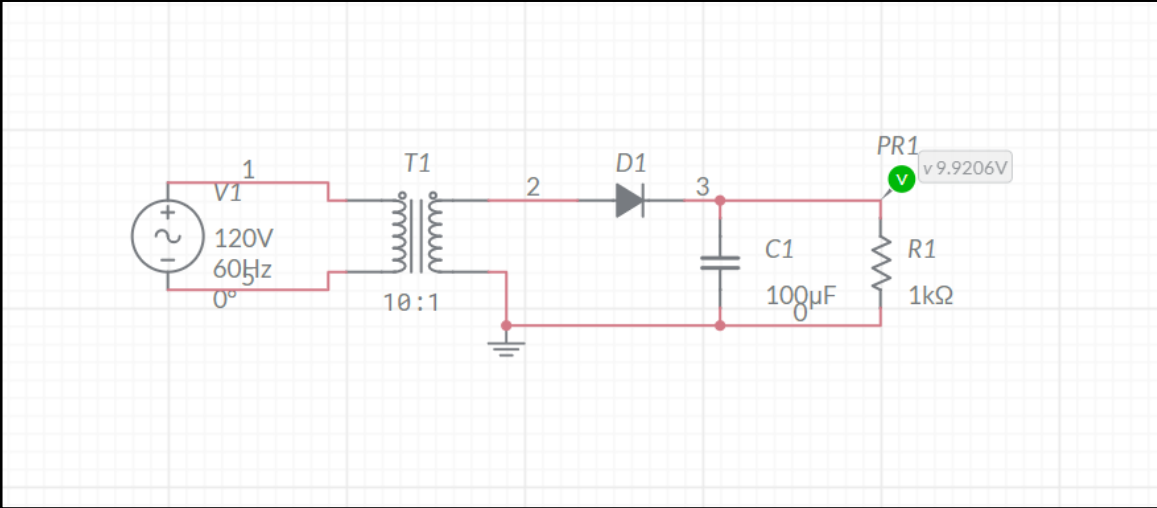


Figure 3.5.3.b – Example Unregulated Power Supply Schematic
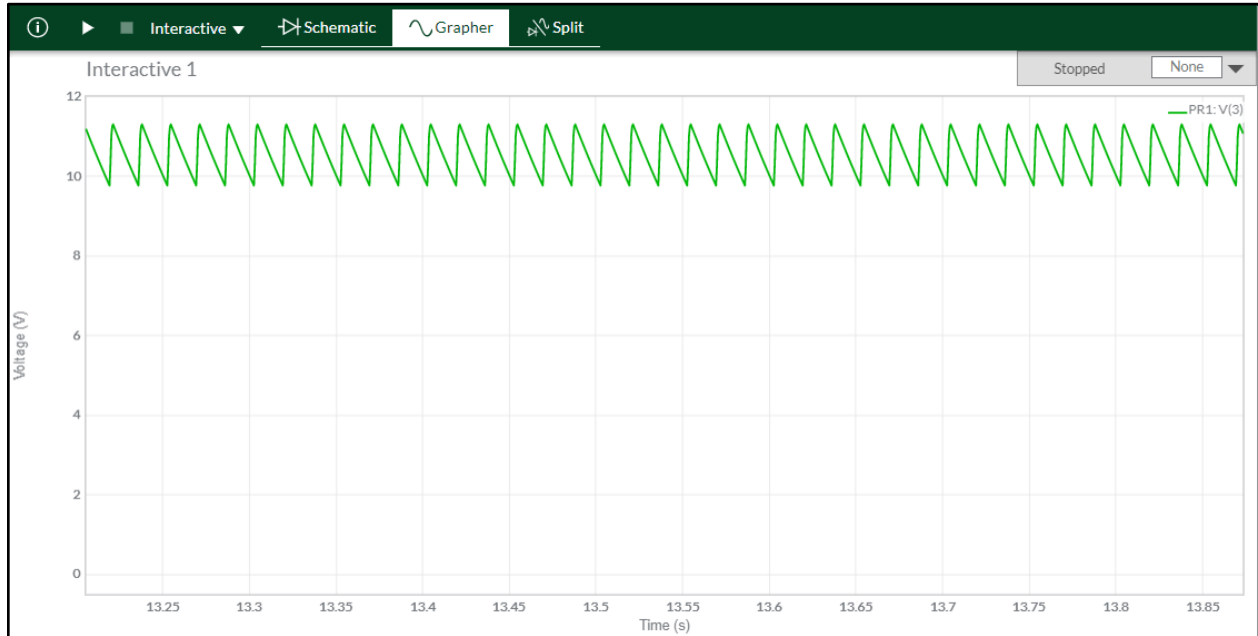
Figure 3.5.3.c – Unregulated Power Supply Output

As you can see from the output graph, this design does not exactly provide DC power. However, it is close enough to a single voltage that some devices, like lamps, relays, or other loads that don't require a perfectly stable input voltage. The rectifying method used in this design is also a single diode rectifier as opposed to the full-bridge rectifier circuits that will be used for the other designs. I chose to use a single diode because it seemed like it was more in the spirit of the design, a cheap way to provide power at approximately the correct voltage, but a full-bridge rectifier could have just as easily been used.

We will not be using the unregulated power supply as the power supply for this project because of the significant drawbacks that have been demonstrated throughout this section. It is simply not worth the small cost savings to put all of the expensive components of the project at risk of permanent and irreparable damage.

### 3.5.4 Linear Regulated Power Supplies

Linear regulated power supplies consist of an unregulated power supply followed by a transistor, either MOSFET or BJT, operating in its linear or active mode. The transistor functions as an active voltage divider, setting the output to a fixed voltage based on the ratio of the parallel resistors. Most linear regulators also use op-amps to drive the base or drain pin on the transistor to keep the voltage at the inverting input equal to the voltage at the non-inverting input.

Some advantages of linear regulated power supplies include the capability to have a varying input voltage/current with a constant output voltage/current. Their components are also still rather cheap, not as cheap as unregulated power supplies, but still very cheap; since Op-amp chips and transistor chips tend to be a bit more expensive than simple passive components like resistors and capacitors. Linear regulators also offer low output noise as a consequence of the filter design.

These power supplies do have some drawbacks however. Most significant of which is the reduced efficiency. The transistor inside the regulator acts as a variable series resistance. Therefore, especially if there is a high voltage disparity between the input and output voltage, there is a lot of power dissipated and lost in the form of heat. Linear regulators typically have some form of heat sink for this very issue. The second drawback directly stems from the first. Because linear regulators are typically operating at higher temperatures, their components can suffer much lower efficiencies or even altogether go into a thermal shutdown. The latter extreme can be avoided with diligent and careful design, but is still certainly a risk that results from the use of a linear regulator.
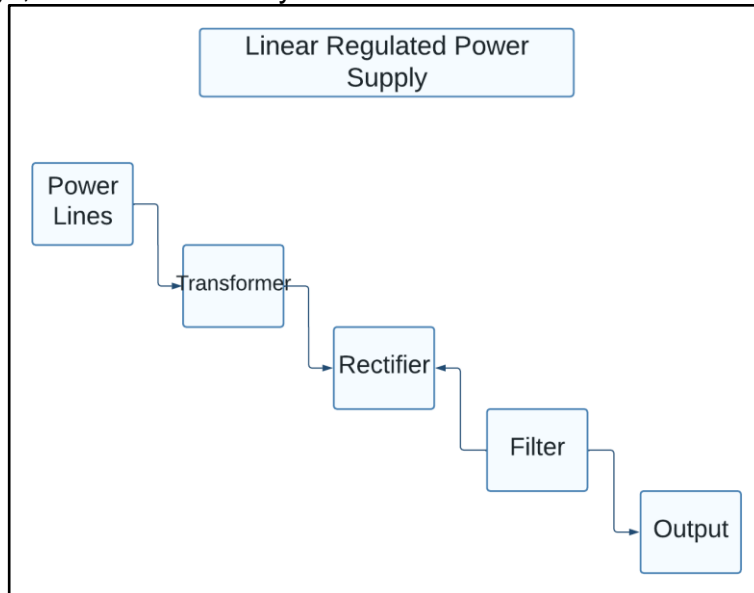


Figure 3.5.4.a – Linear Regulated Power Supply Block Diagram
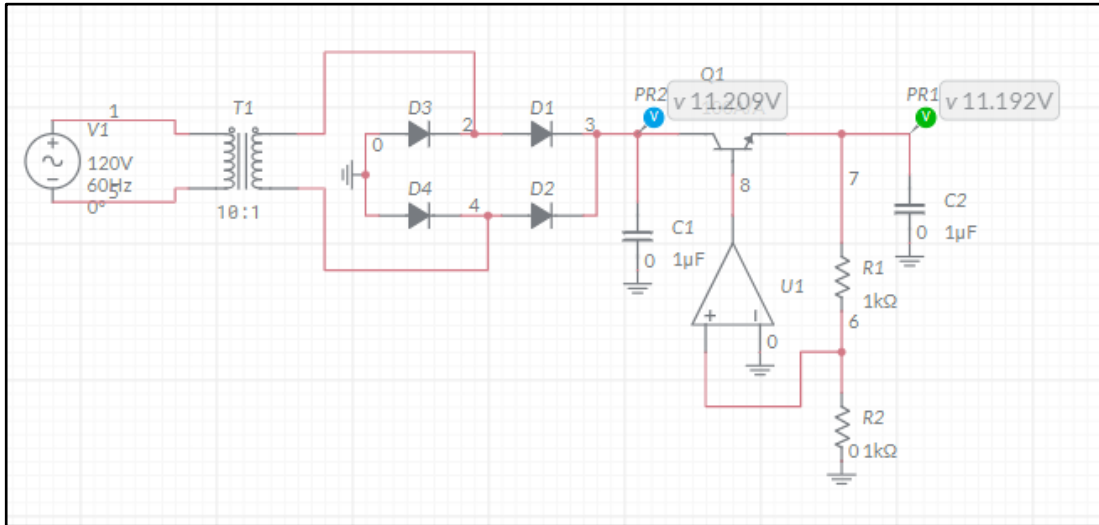
Figure 3.5.4.b – Example Linear Regulated Power Supply Schematic



Figure 3.5.4.c – Linear Regulated Power Supply Output

As you can see from Figure 3.5.4.c the voltage at the output of the bridge rectifier and the voltage at the output of our active components are virtually identical with only a small amount of loss from the voltage divider circuit. This gives the impression that linear regulators are very efficient, but this is not the case and is a simple consequence of Multisim not being able to account for all real-world factors such as heat.

We plan on implementing a linear regulated power supply to be the power supply of this project. The information that has been presented in this section demonstrates their viability in audio applications and in other situations.

### 3.5.5 Switching Power Supplies

Switching regulated power supplies were created in an effort to have the same benefits of the previous two power supply designs: Simple, small, efficient, cheap, and with clean/stable output voltage. Switching power supplies work by taking the power line AC voltage and rectifying it to high voltage DC. The DC voltage is then converted back to square wave AC using an array of transistors operating as switches. Then, step that AC up or down using a transformer before finally rectifying back to DC and filtering any fluctuations with a capacitor. Voltage regulation is handled differently than in linear regulators. It is achieved by changing the duty cycle of the square wave DC on the primary side of the transformer by either using a PWM (pulse width modulation) signal, or a PFM (pulse frequency modulation) signal.

One additional advantage of these power supplies, along with their higher efficiency over linear regulators and better voltage regulation than unregulated designs, is their flexibility with respect to their input voltage. Switching power supplies are sometimes called "universal power supplies" because of their ability to work on nearly every power system in the world. Being a universal power supply will not necessarily help in making a bass amplifier as our power source will have a relatively stable input from the city power, and any fluctuations would be well within the ability for a linear regulator to handle. The most significant advantage would certainly be the efficiency, as we certainly don't want a massive amount of excess heat to be produced by our amplifier.

There are, however, some drawbacks to this design. Firstly, its design is more complicated than the previous examples. And in order to maintain an output voltage with minimal ripples you will need to spend more on more precise components. Some cheap switching power supplies even have the same, if not worse, ripple voltage on their outputs than the unregulated design. The heat dissipation could also be a major problem if left untreated this could truly damage some components within our amplifier. For example, if we were to have only an efficiency of 40% for our 50W bass amp we would be dissipating 20 watts of heat directly into the heart of our amplifier. After only 1 hour of use, this would raise the temperature by up to 69 degrees Fahrenheit. There is also of course the risk of the switching power supply causing interference with some of the audio effects.
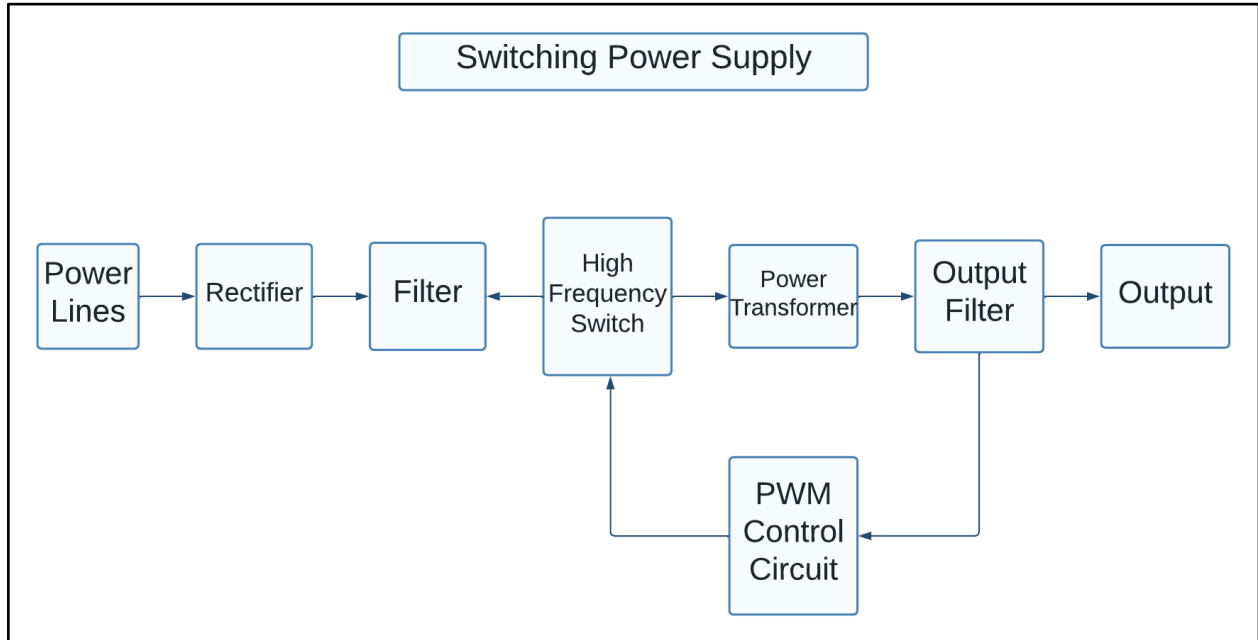
Figure 3.5.5.a - Switching Power Supply Block Diagram

### 3.5.6 Switching Power Supplies in Audio Amplifier Applications

Typically switching power supplies are not used in audio applications because of a myth that they produce a lot of electromagnetic noise and can even affect the sound quality of the amplifier. While this can happen, it is not true that switch-mode power supplies in and of themselves cause increased noise or magnetic hum in audio applications.

Switch-mode power supplies can and have been used in the design of bass-amplifiers. However, according to most literature, Switch-mode powered amps are harder to design because of the high amount of AC interference from a switch oscillating at many kilohertz or even megahertz in some applications. If these challenges can be overcome though, the high efficiency and stable voltage output can be a powerful advantage. An article by benchmarkmedia.com goes into more detail about the myth and is linked in the references section.

### 3.5.7 Conclusion

Simply put, the *unregulated power supply* is the cheapest and most efficient, but has an inconsistent voltage output. The *linear regulated power supply* has the most consistent voltage output, but can be inefficient. And the *switching power supply* is very efficient and has a consistent voltage output, but it is very expensive to have both of these qualities at once.

| Power Supply Type | Efficient | Cheap | Consistent Voltage Output |
|---|---|---|---|
| Unregulated | ✓ | ✓ | ☒ |
| Linear Regulated | ☒ | ✓ | ✓ |
| Switching | ✓ | ☒ | ✓ |

Table 3.5.6.a – Power Supply Relationships

## 3.6 Analog Effects

### 3.6.1 What are Analog Effects

Analog Effects are specially designed filters, placed in series in the amplifier stack, that apply specific changes to the input signal. These changes can affect the amplitude, frequency, or the phase of the signal. Analog Effects, specifically, use analog circuitry to accomplish this. In the world of bass guitars, effects are not as common as guitar effects, but still have their place.

### 3.6.2 Why do you Need Analog Effects

As stated in 3.6.1, bass guitar effects are not as common as guitar effects. They are much less necessary to achieve the wanted characteristics. There are still important effects that many famous bassists swear by. The next few paragraphs outline some of these more common effects, detailing the physics behind them.

### 3.6.3 Compressor

In the world of Bass Effects, a Compressor is one, if not the most, commonly used effect. A Compressor works by attenuating the input signal if the amplitude passes a certain minimum value. In an easier to understand form, a compressor reduces the perceived volume of the loud notes, so that overall output volume is uniform. While using a Compressor, the tone of the Bass is better sustained, more even across all the strings, and may have a mild overdrive effect.

The amount of attenuation is typically referred to as the Compression Ratio. The Compression Ratio is the input signal, that is above the threshold, compared to the output signal, over the threshold. The Compression Ratio is measured in decibels, dB. A 1:1 Compression Ratio means no compression, while a ratio of 5:1 would mean that for every 5dB the input signal is above the threshold, the output would only be 1dB above the threshold. Figures 3.6.3.a and 3.6.3.b, graphically, what these compression ratios would look like.
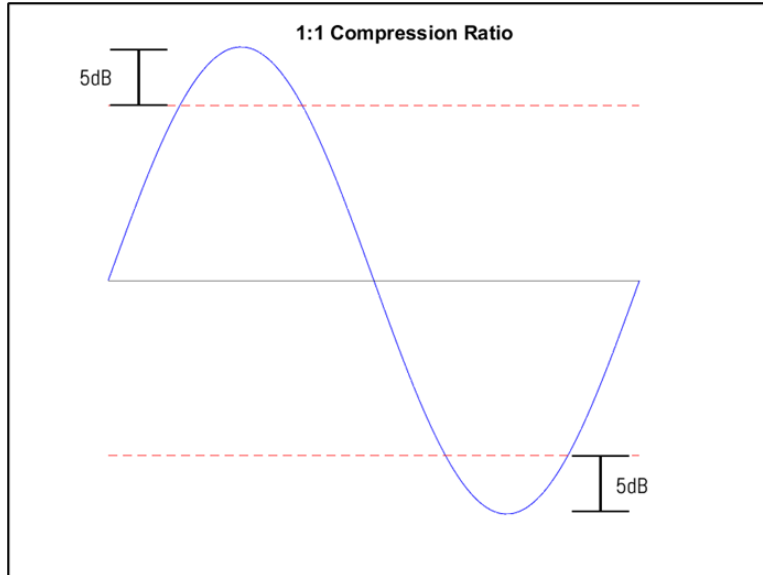
Figure 3.6.3.a – 1:1 Compression Ratio
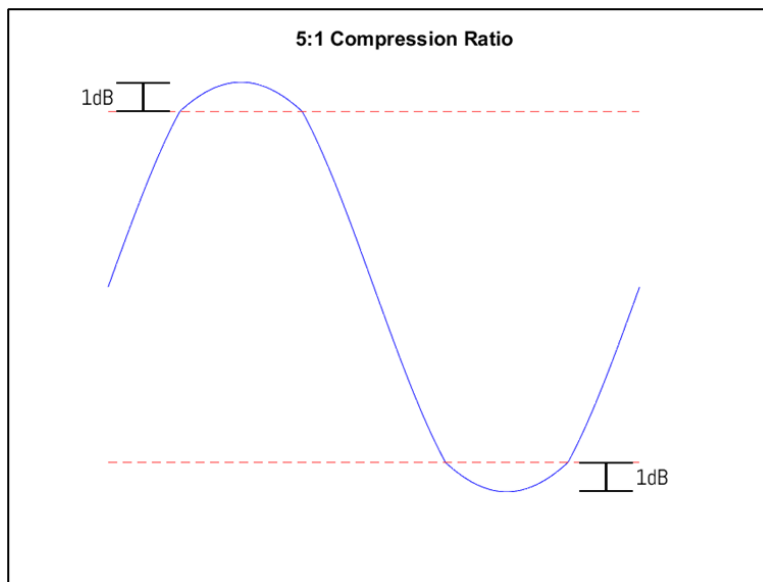


Figure 3.6.3.b – 5:1 Compression Ratio

The Compression Ratio is usually controlled by a potentiometer on the pedal itself. Other common controls include Make-Up Gain, Attack, Release, and Sustain. Make-Up Gain refers to an amplifier used to bring the modified signal back to unity-gain. Attack refers to how fast the compressor will compress the signal when the threshold is reached. This is also known as the slew rate. The Release refers to how fast the compressor will stop compressing the signal when it goes below the threshold. Very similar to Attack and Release, Sustain refers to how long the signal will stay compressed after the input has changed.

### 3.6.4 Overdrive

Another very common Bass Effect is the Overdrive. An Overdrive Effect attempts to simulate the clipping and distortion that occurs when an amplifier is used to amplify a signal above its maximum gain. Generally, the desired effect is that of a vacuum tube amplifier that is in the saturation region. Overdrive effects apply a "soft clipping" to the input to get the intended distortion. This modification has a mild effect on the tone of the bass making it sound "warm". Figure 3.6.4.a shows types of clipping found in Overdrive, Distortion, and Fuzz Effects.
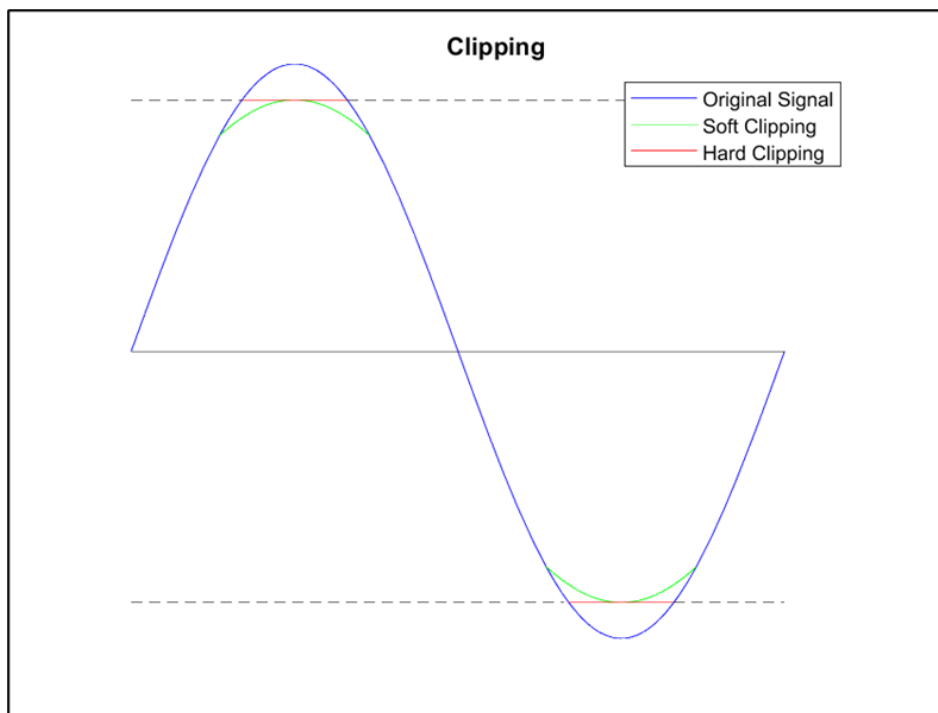


Figure 3.6.4.a - Clipping

Common controls on an Overdrive Pedal are Level, Drive, and Tone. Level refers to volume of the effect. Drive refers to the gain of the overdriven tube. Tone refers to how much the input, or dry, signal and the modified signal, or wet, are mixed together.

### 3.6.5 Distortion

While the concept of distortion might seem similar to that of overdrive, a Distortion Effect has a very harsh effect on the waveform. A Distortion Effect applies heavy clipping, phase shifts, noise, changes in the frequency response, and other non-linear transformations of the input signal. Another characteristic that sets the Distortion Effect apart from Overdrive or Fuzz, is that the amount of distortion produced is the same at any volume level. Distortion Pedals are commonly used in heavy metal and hard rock but are not used outside of those genres.

Common controls on a Distortion Effect include Level, Tone, and Distortion. Level refers to the overall volume of the effect. Tone refers to how much the dry and wet signals are blended in the output. Distortion refers to the gain of the distortion creating components.

### 3.6.6 Fuzz

The Fuzz Effect uses similar techniques to the Distortion and Overdrive Effects but takes it a step further. A Fuzz Effect uses hard clipping to transform input into imperfect square waves and further enhances it with overtones. Unlike Distortion or Overdrive, a Fuzz Effect is not meant to simulate a vacuum tube amplifier. The sound produced has more buzz and is harsher compared to the others. Using frequency multipliers, harmonic or inharmonic overtones can be added to the signal to make the sound warmer or harsher, respectively.

### 3.6.7 Vibrato

Vibrato is a modulation effect that modifies the dry signal variations in pitch. Unlike other effects, Vibrato only deals with the frequency of the signal. Being one of the oldest effects, the circuitry behind Vibrato is very simple. The dry signal is fed into an analog delay, which delays the signal, and then outputs it. The analog delay is controlled by an oscillator circuit, operating at a low frequency. Effectively, the delay is stretching and compressing the signal, changing the frequency of it. Typically, there is no feedback in this system.

The common controls on the Vibrato Effect are Speed and Depth. The Speed control changes the operating frequency of the oscillator circuit. The Depth control determines how much the Speed control will affect the oscillator.

### 3.6.8 Tremolo

Another example of a modulation effect, Tremolo, modifies the dry signal's amplitude so that there is a periodic change in volume. Tremolo and Vibrato are very commonly, incorrectly, used in place of each other. Tremolo only affects the amplitude, or the volume. While Vibrato only affects the frequency, or the pitch. This misconception creates confusion as to how to modulate the signal, to achieve the desired effect. Classic Tremolo Effects use an optocoupler, as a variable resistor, to change the amplitude of the signal. An oscillator circuit is used to turn an incandescent bulb on and off, changing the resistance in the optocoupler. The oscillator would generate either a sine or a triangle wave, based on the intended tone. Newer Tremolo Effects still use this same technology to create Tremolo.

The common controls on the Tremolo Effect are, once again, Speed and Depth. The Speed control changes the operating frequency of the oscillator circuit. The Depth control determines how much the Speed control will affect the oscillator.

**3.6.9 Chorus**

Operating on the same technology as Vibrato, Chorus is a modulation effect that modulates the dry signal with pitch variations and adds it back onto itself. The main difference between Chorus and Vibrato is that Chorus does not fully cut out the dry signal when it outputs. This creates a rich, deep sounding effect that attempts to emulate the sound produced by a chorus of singers. The correct amount of pitch variation is vital to the effect producing a "good" sound. If there is too much variation, the waves will not sync with each other and produce an out of tune sound. When correctly tuned, the multiple waveforms act like a singular wave, and produce a chorus-like effect. Being in the modulation effect group, the common controls are the Vibrato or Tremolo Effects.

## 3.7 Digital Effects

Digital effects are widely used in the world of guitars and bass guitars. Digital effects use some sort of DSP to discretize the incoming input audio signal, apply some sort of modulation or filter to alter the soundwave, and output it. This occurs through an audio codec that is present on the DSP. The audio codec contains an ADC and DAC which are running on the same clock and supports communication protocols such as I2S, I2C, and SPI.

Digital sound effects can be classified based on their effect on auditory perception. These characteristics include loudness, time, pitch, spatial hearing, and timbre.

Loudness simply affects the perceived intensity of the sound. This loudness ranges on a scale from pianissimo to fortissimo. Modulation of a sound on this scale is called tremolo, which is a popular musical sound effect. Time is altering the duration of a time signal or gaps. Pitch involves changing the frequency of signal, affecting the sharpness of the perceived sound. Through filtering and delays, the spatial hearing of a sound can be manipulated. In doing so, the location and direction of sound are perceived differently, as observed by effects such as reverberation and echo. The final aspect that is relevant to audio effect characteristics is timbre. Timbre is what would be associated with quality of sound. This is the phenomenon where the same note can be played on two different instruments, or on the same instrument with the same loudness and pitch, yet it still sounds different.

**3.7.1 DSP Selection**

Choosing the correct DSP microcontroller is crucial for audio effects. In choosing a board, the most important comparison to make is cost versus performance. Being that the objective of this project is to make a cost effective bass amplifier, something that meets in the middle of these two criteria would be optimal.

One consideration is the bit size data that the microcontroller can process. The higher bit processing that the controller can do , the more the cost increases. There are 8-bit,16-bit, 24-bit, and 32-bit DSP microcontrollers. In general, audio processing uses 16-bit buses.

The memory of a DSP board should also be considered. Having adequate non-volatile RAM is essential in an application like this, since the DSP microcontroller is not being flashed everytime it starts up. It is going to be in a device that should be usable at any moment. Having more memory also allows more of the audio file to be stored, allowing for proper retention of quality and the ability to have more precise modifications to it.

Clock frequency is something important to be considered. The higher frequency clock is not always the best. For instance, between boards of the same family, the higher frequency would be better since the architecture is similar. A faster clock allows for instructions to be executed faster. Whereas devices with different architectures cannot be compared just off clock frequency because differently bult devices require different numbers of instructions to complete a task.

| Part # | Frequency | Flash memory | ADC resolution | Core size | Price | MIPS |
|---|---|---|---|---|---|---|
| TMS320F 28377D | 20MHz | 1024kB | 12-16 bit | 32-bit | $12.04 | 80 |
| TMS320V C5402AZ WS16 | 20MHz | - | - | - | $17.31 | 160 |
| ADSP- BF536BB CZ-3A | 300MHz | 100kB | 16-bit | - | $29.74 | - |

Table 3.7.1.a - DSP Options and Specifications

### 3.7.2 Audio Codec

An important consideration in choosing a DSP microcontroller for audio processing is an audio codec. These can be bought as a separate piece of hardware from the microcontroller itself. They are useful for encoding and decoding audio data and saving space. It is much more effective at storing audio files without losing the quality of them, which it does through the I2S communication protocol and uses I2C to write and read from the codec's registers.

There are three main kinds of audio codecs: lossy, lossless, and uncompressed. In lossy, the audio file is compressed, but does not restore its original qualities when decompressed. Although cheaper, would not be a good option for this project since audio quality is a pivotal quality in speakers. Uncompressed audio codecs do not compress the audio file. While maintaining the original sound quality, it takes up much too much disk space, which is especially concerning for a microcontroller due to the limited memory. Lossless would be best for this project since it can compress an audio file to save space in the memory, but can still retain good sound quality when decompressed.

Something important to consider when choosing a suitable codec is the sampling rate. There is a well known fundamental rule called Nyquist's rule, which says to accurately preserve information of a signal, that signal needs to be sampled at twice its own frequency. Sampling less than this irretrievably loses information. A bass guitar fundamentally has a frequency of 40Hz-400Hz, with harmonics reaching anywhere from 4KHz-5kHz. Having a codec capable of sampling at higher frequencies allows for more preservation of information , but drives the cost higher. In general, a sampling rate of 44.1kHz is ideal for audio recordings. Below is a list of potential audio codecs varying in price and performance. Selection will be heavily based on price, performance, and availability. As seen in the table below, there are very affordable options, but it does highly restrict the potential performance of the system. The more expensive audio codecs do not necessarily have better performance than the mid-range cost ones, it is mainly the additional DACs and ADCs that drive the cost. For this project, it seems more sense to stay within a $5-$7 range for audio codecs.

| Manufacturer Part # | Price | Resolution | Max Sampling rate | Interface type | # of DACs | # of ADCs |
|---|---|---|---|---|---|---|
| MAX98090AEWJ+T | $6.75 | 24-bit | 96kHz | I2S | 1 | 1 |
| TLV320AIC34IZASR | $10.10 | 24-bit | 96kHz | I2S | 4 | 4 |
| MAX9860ETG+T | $3.37 | 16-bit | 48kHz | I2C | 1 | 1 |

Table 3.7.2.a - Audio Codec Options and Specifications

### 3.7.3 Important Digital Components

There are some important digital circuit components that are used throughout digital effects

One such component is the comb filter. The comb filter is especially used in creating reverberation. The comb filter's purpose is to filter out certain frequencies and its harmonics. The waveform looks like a comb. It is quite a simple circuit where the signal is added to itself with a short delay.

Comb filtering is a natural phenomena where a sound constructively interferes with itself successively within short time periods. These time periods can vary from 1ms to 25ms. Comb filtering can only occur when successive signals are within 10dB from each other.

There are two types of comb filters
1. Feedforward comb filter

2. Feedback comb filter

The feedforward comb filter would be what is considered an finite impulse response (FIR) filter. This is due to the lack of feedback. When a signal is passed through the circuit, the signal will eventually die off. These types of filters will always be stable, and provide linear phases. While these are the benefits of FIR filters, they also have their drawbacks. FIR filters are more computationally intensive and require more memory. This is because they require more filter coefficients to produce proper frequency cutoffs. Memory is an important consideration since a DSP microcontroller is being used. Microcontrollers have less computational power and memory than a typical laptop or desktop.
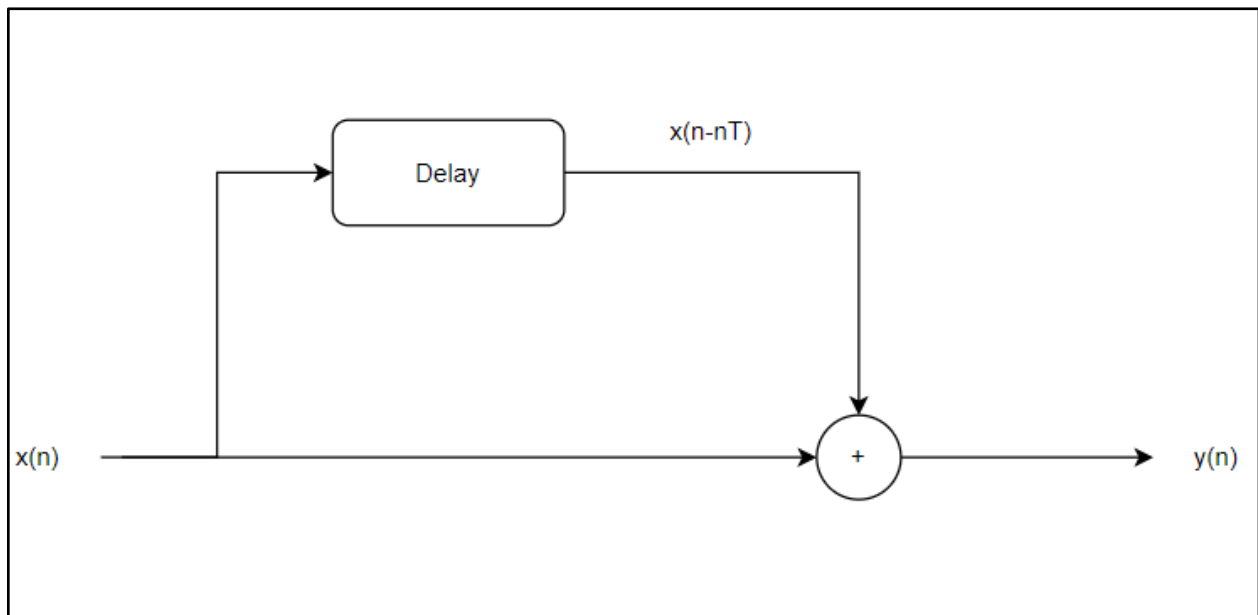


Figure 3.7.3.a - FIR Comb filter

The other kind of comb filter is an IIR comb filter. This means the filter is recursive, which is caused by the feedback. As seen, if the feedback coefficient was greater or equal to one, the input signal would continuously grow. This means the current input is always being added with a previous sample, which will then again be delayed. IIR filters require less memory and thus are easier to implement. They also have lower latency.

The problem with IIR filters is that they are not inherently stable and produce non-linear phases, making it significantly harder to analyze an IIR filter as opposed to a FIR filter
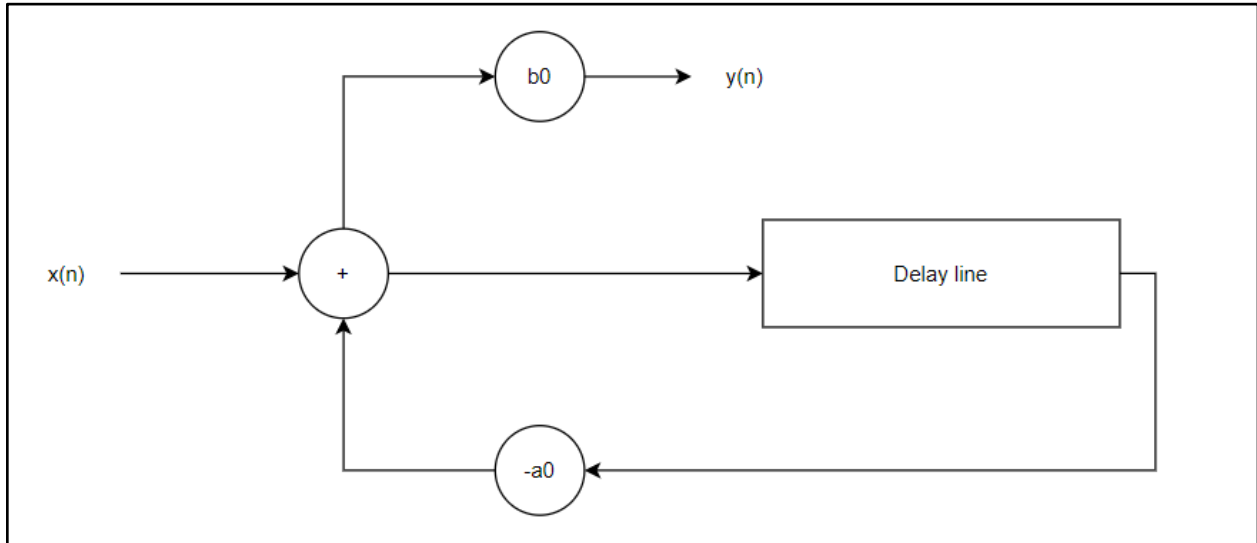
Figure 3.7.3.b - IIR Comb Filter

$$y(n) = x(n - K) + x(n)$$

Equation 3.7.3.c - FIR Comb filter difference equation

$$y(n) = b0(x(n)) - aM * (yn - M)$$

Equation 3.7.3.d - IIR Comb filter difference equation

Another important filter is an allpass filter. The allpass filter is used to pass signals of all frequencies, but shifts the phase of certain frequencies by different amounts. They are particularly useful in correcting phase shifts caused by equalizers.

They also serve as a useful tool in creating artificial reverberation. The two methods of artificial reverberation are the more rudimentary one of Schroeder's and the improved version of Schroeder's reverb which is Moorer's reverb. Both of these use nested allpass filters, which is really good at simulating the late reverberations of a room.
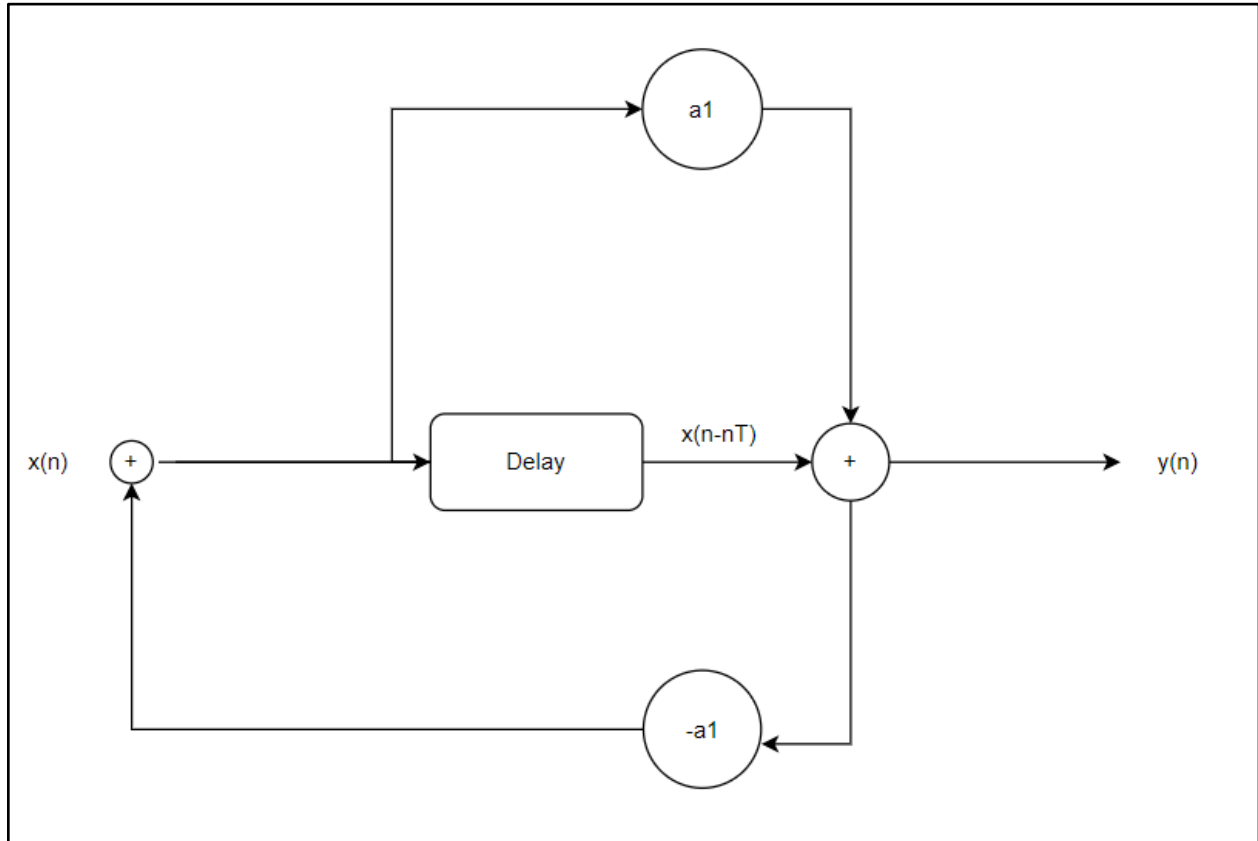
Figure 3.7.3.d - Allpass filter

### 3.7.4 Wah

The wah-wah effect is one that mainly affects timbre and pitch of a sound. When passed through a certain circuit, it creates a sound spectrum close to that of human speech, producing the "wah-wah" sound. The wah-wah effect fundamentally works by shifting the center frequency. Traditionally done through a foot pedal, where pushing the pedal engages some sort of filter, either a bandpass filter or a lowpass filter. A knob on the pedal is connected to a variable resistor, which when tuned moves the center frequency of the bandpass or the cut-off frequency for a lowpass. This in turn adjusts the Q-factor of the filter. A higher Q-factor will make a more restrictive filter, increasing the peak resonance, giving a sharper sound.

Although an effect typically done through a pedal, there are variations of this effect that do not require one, which will be an auto-wah. Using a DSP, the incoming audio signal can be mixed with a low frequency oscillator (LFO) to produce the auto-wah. Ultimately replacing the need for a manual changing of the center frequency. There are other versions of this circuit as well like the M-tap wah. It is fundamentally like the auto-wah, except it has amplitude variation in the oscillating modulating wave.
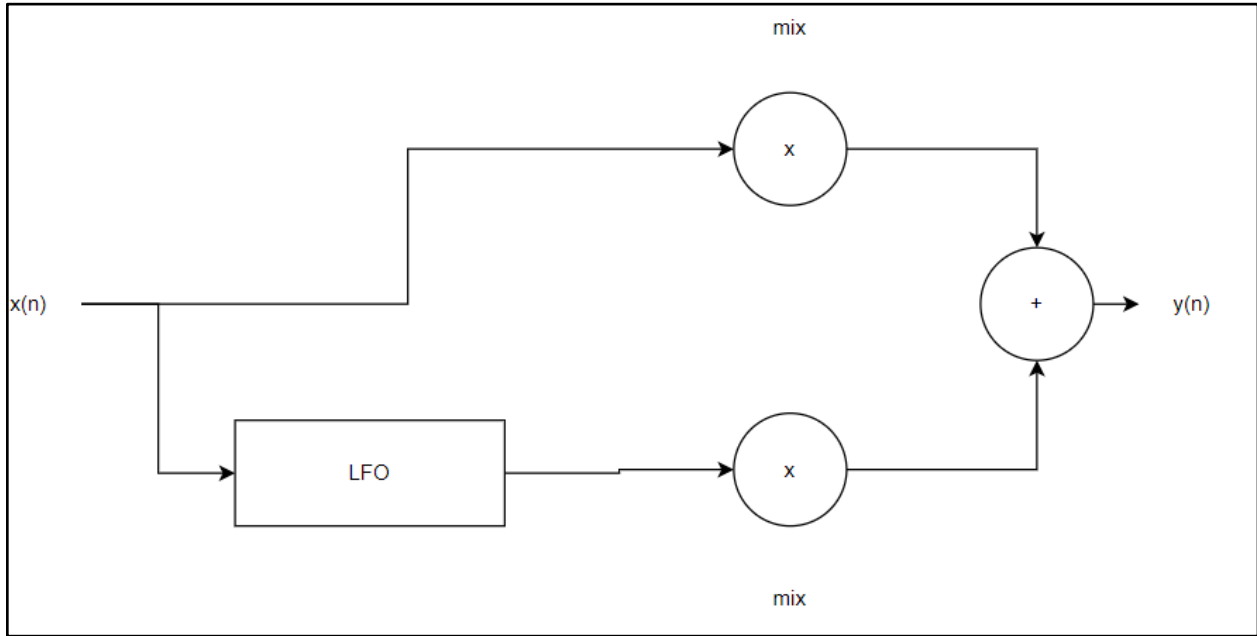
Figure 3.7.4.a - Auto Wah
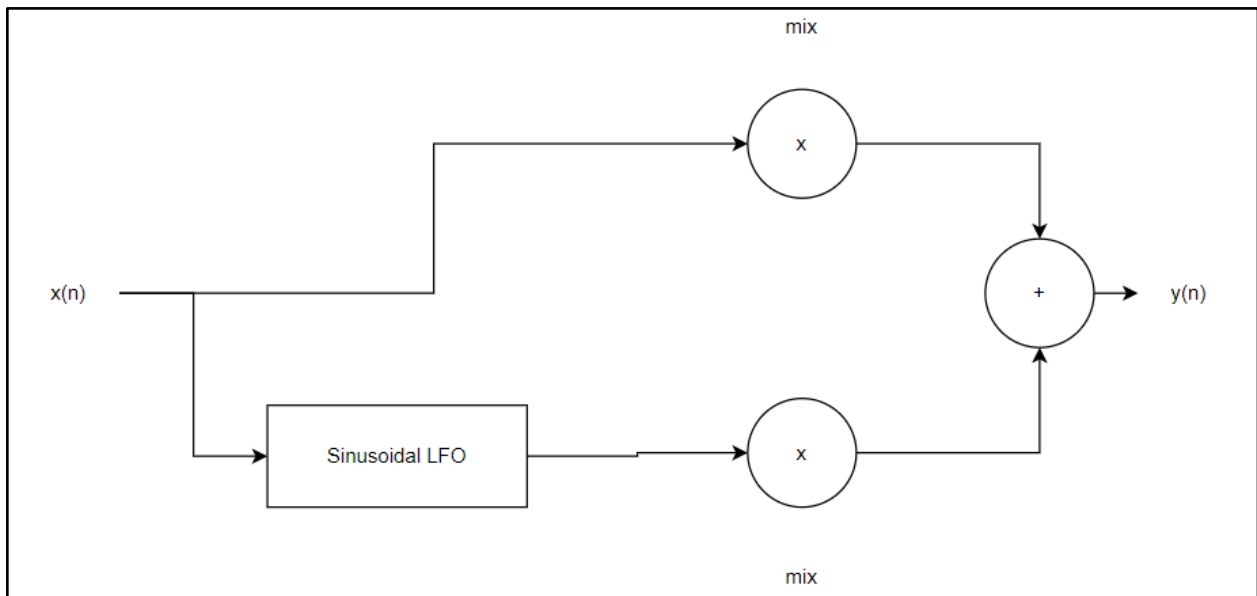


Figure 3.7.4.b - M Tap Wah

### 3.7.5 Echo

Echo is part of a group of effects produced by time delay. A delay of greater than 50 ms causes the echo effect. Echo does not distort the delayed sound signal, it should sound similar to the original audio but with a fading effect. This is typically done by sending the input signal through a delay line with an attenuator.
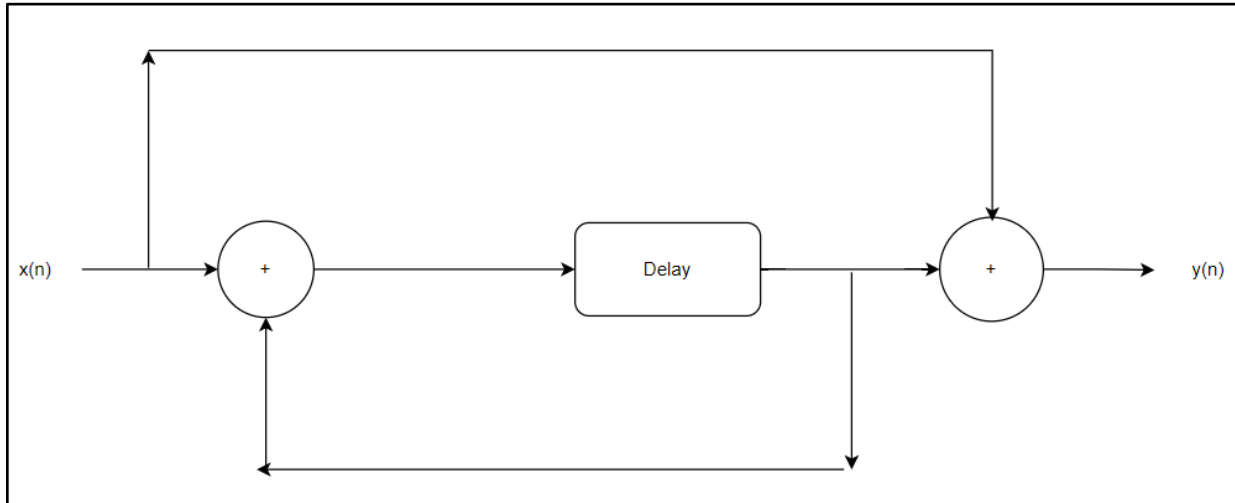
Figure 3.7.5.a - Echo Block Diagram

### 3.7.6 Flanger

Flanger is another type of delay based audio effect that can be observed with a delay between 0-15 ms and when the input signal is modulated with a sine wave to produce a phase shift. The sine wave generally has a varying delay of 0-3ms at a frequency of 1Hz. This effect can be relatively easily realized by mixing the input signal with a low frequency oscillator with a delay.
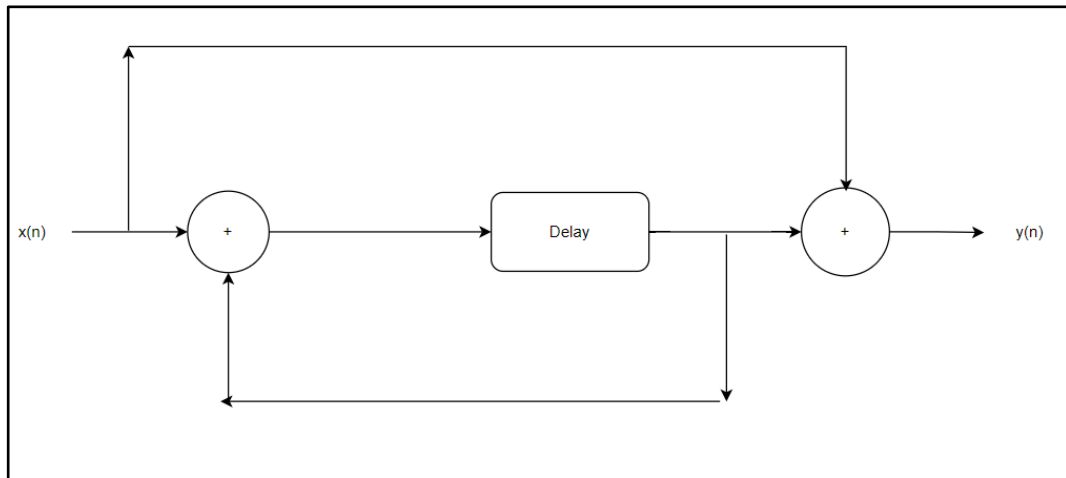


Figure 3.7.6.a - Flanger Circuit

### 3.7.7 Reverberation

Reverberation is one of the most used guitar and bass sound effects used. Naturally, when a sound is produced it propagates in a wavefront pattern. There is the shortest path the sound takes, which is what reaches the perceivers ear first. Then there are also sound waves which take longer paths. Along those paths the wave can encounter objects where it reflects back to the observer. This occurrence is reverberation and affects

the spatial hearing of listeners. This is not to be confused with echo. Echo requires a relatively large time delay between signals to produce its effect.
The delayed signal sounds the same, due to a fixed time interval of the delayed signal.

In reverberation, there are three distance periods for a sound wave. The first is direct sound, which is where the sound reaches the ear via the shortest path. Then comes the early reflections, which are generally more ordered reflections which are responsible for relating the sound to the space and size of a room. After that are the late reflections, which are more random in terms of time of arrival. These late reflections produce sounds discernable from the direct sound.

Reverberation can be simulated in one of two ways, a filter bank/ delay line method, or convolution method. In the former method, there are two circuit configurations used to mimic reverberation. A rather primitive method is Schroeder's Reverberation, which passes an input signal through parallel comb filters then through series allpass filters. Schroeder's reverberation lacks the exponential decay that natural reverb has. The improved version of this is Moorer's reverberation which builds upon the base parallel comb filters fed to the allpass filter, but a proceeding delay line is present. The input signal also goes through a tap delay line which is used to simulate the early reflections seen in natural reverberation. The importance of comb filters is that they create a decreasing magnitude spectrum for a wave, however it also creates it with rather uniform spacing, which is unlike real reverb. The allpass filters correct this by increasing the signal density which is similar to the late reflection seen in reverberation.
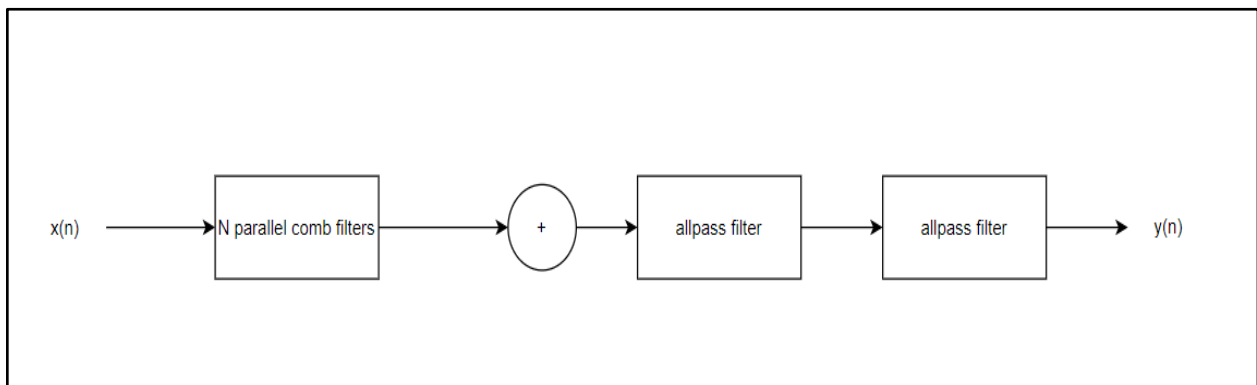


Figure 3.7.7.a - Schroeder's Reverb
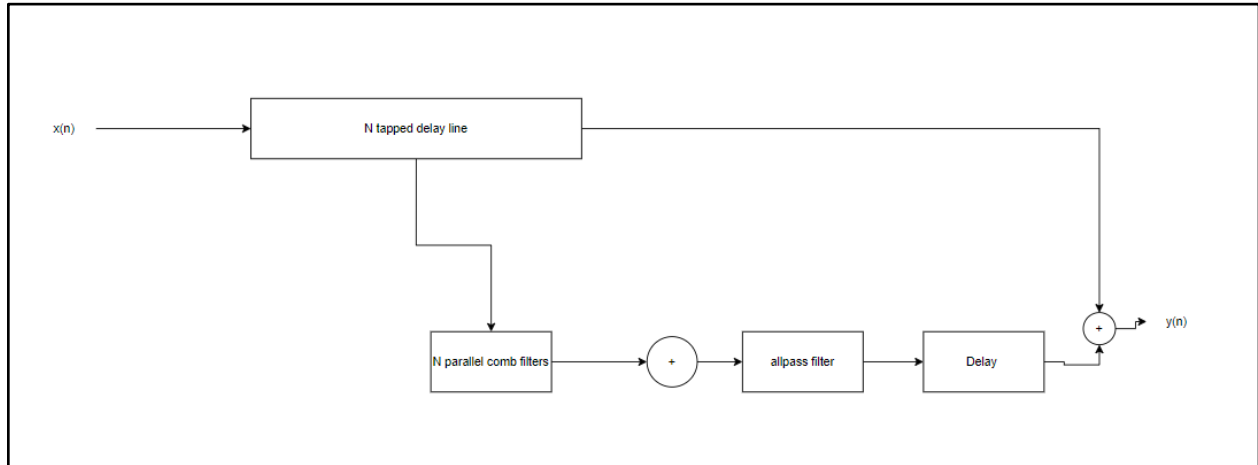
Figure 3.7.7.b -  Moorer's Reverb

The other method of creating reverberation is called convolution reverb. This method requires the impulse response of a room to be captured. This is usually done by recording some loud and sharp noise. Oftentimes, the trigger sound of whatever event caused the impulse response needs to be removed through deconvolution. Once that occurs, the input audio signal can be convolved with the impulse response of the room through software to generate reverberation.

There are several types of reverberation. The first kind of reverberation is called hall reverb. Hall reverb is meant to mimic the space of a large concert hall, where sound waves have a long decay time. This gives the effect of a large room. Another kind is room reverb, which has a very fast decay and gives the impression of being in a regular sized, filled room. Similarly, room reverb would occur from the impulse response of a regular sized room.

 Spring reverb was invented in a time where reverb was trying to be made through analog devices. The sound would be sent through a steel spring in an attempt to simulate reverb. This method did not have a good way of implementing the early reflections seen in natural reverb.

Gate reverb is implemented with a noise gate. As opposed to natural reverb which naturally dies out, gated reverb ends abruptly with the use of a noise gate.

Plate reverb is one of the earliest attempts at artificial reverb. It was created using a metal sheet. A magnetic driver sends vibrations through a metal sheet and a microphone picks up the resulting sound. The EMT-140 was a German made plate reverberator and the first ever plate reverb. It was widely used by bands such as the Beatles and Pink Floyd.

### 3.7.8 FPGA and Digital Effects

Using FPGA instead of DSP cores or analog components can be a cheaper, easier, and more modular way of implementing digital effects into projects such as these.

FPGAs are semiconductor devices that are based around CLBs or configurable logic blocks that can be connected or disconnected in a highly customizable fashion.

This is of course ideal for our bass amp because we would prefer not to spend a lot of money on many different highly specialized DSP cores and would like a board that can handle multiple effects that we could even change later if there was some mistake or we simply wanted to add more effects. Modularity is something that we've aimed for specifically in this project as we attempt to build this bass guitar amp as cheaply as possible.
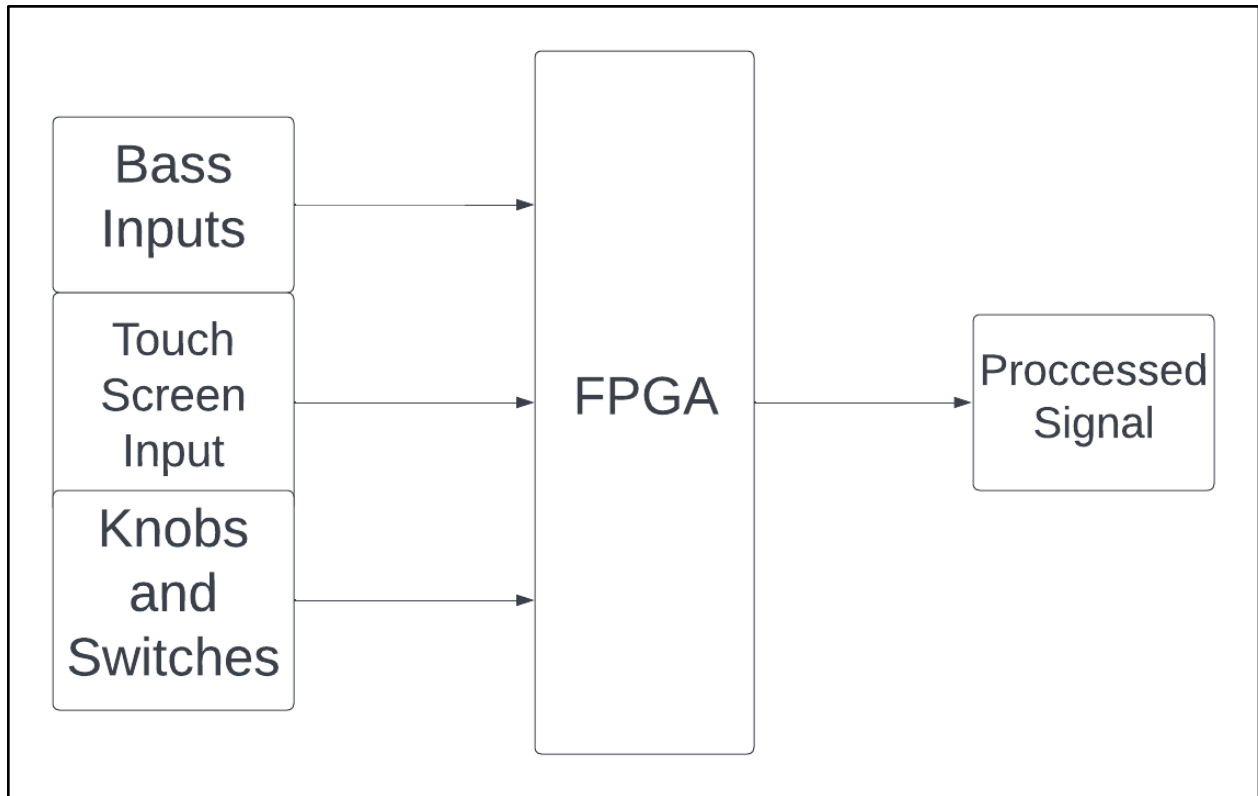


Figure 3.7.8.a – High-Level View of FPGA Effects

Figure 3.7.8.a offers a high-level view of what the FPGA board would do in our project, from a variety of inputs, both analog and digital we will be able to output a processed signal that can be taken directly to the speaker and played as audio.

## 3.8 Interface

### 3.8.1 Touch Screen Technology

The touch screen will be one of the primary methods the user interfaces with the device, so it is vital to choose a screen that will be serviceable, without increasing the overall price of the unit by a significant margin. There are five major types of touch screen technologies that will be looked at.

Resistive:

Resistive touchscreens use an overlay over the screen which is composed of a flexible top layer, usually plastic, and an inflexible bottom layer. Each layer has a thin metallic layer along the inner surface, and there are spacers, which hold the spacers apart. When there is an input, the layers are pushed together which causes the metallic layers to touch, which allows the voltage to be measured to determine an X-Y coordinate. The advantages of this method include being able to actuate the touchscreen with most objects, lower cost to produce, as well as lower power use. The disadvantages would be the reduction in image quality because of the overlay, and the outer plastic layer being prone to scratches and even possibly being punctured.

Surface Capacitive:
Surface capacitive touchscreens use an overlay consisting of an electrode film on top of a glass layer, all underneath a protective cover. The electrode film is reactive to the current that can flow through a person, by using four sensors to measure where the capacitance of the film decreases, which pinpoints the X-Y coordinate.  The advantages of this method is an improved image quality, as well as a durable protective layer which makes it resistant to scratching.  The disadvantage is the use of the electrode film requires the input to be from a bare finger, or a specific stylus made to interface with this type of screen.  It also is susceptible to electromagnetic and radio interference.

Projected Capacitive:
Projected capacitive touchscreens use a similar overlay to surface capacitive, but instead of the single electrode film, it has two interlaced layers of electrode film, which is able to generate an electrostatic field above the screen, which is sensitive to the changes in currents when a finger comes in contact with the screen, it is also able to detect multiple simultaneous input. The advantages of this screen would be clear image quality, sensitivity to simultaneous inputs, and a scratch resistant screen. The disadvantages would be the requirement for a bare finger, as well as still being sensitive to electromagnetic and radio interference.

Surface Acoustic Wave:
Rather than some of the more multifaceted overlays from the previous technologies, the surface acoustic wave screen just has a glass panel with piezoelectric transmitters and receivers along the edge that send a wave along the glass overlay. Touching the panel results in a finger absorbing some of the wave, which is measured by the receivers. This allows the receiver to pinpoint the X-Y coordinate of the finger.  The advantage of this is that with only a glass overlay, the image integrity is very well retained. The disadvantages are that anything spilled on or stuck to the screen can inhibit the function of the touchscreen.

Infrared Touch Screen:
Infrared touch screens do not have an overlay, they instead use infrared emitters paired with sensors. When one of the infrared beams is blocked by a finger, it is able to pinpoint the location based on the sensor no longer receiving the beam which was blocked. The advantage of this is there is nothing obscuring the screen so the integrity of the image is maintained, as well as the "overlay" not able to be scratched since it is simply

an infrared grid, so the durability of the screen depends on what screen it is paired with. The disadvantages of this technology is a high cost, as well as being sensitive to materials stuck to the screen.

Weighing options:
      The technology best suited for our device is the resistive touch screen. The most important aspect is the lower cost and power consumption, since the screen does not need to have the best picture quality, text simply needs to be legible, and the plastic layer should have a low risk on a stationary device. This makes the main downsides of a resistive touchscreen nonfactors, while each of the other technologies has downsides for usage with this device.  Both capacitive screens could be disrupted by other components in the amplifier, and the higher cost of the surface acoustic wave or infrared screen is not worth the tradeoff for better image quality or durability.

Comparison of Resistive touchscreens:
      Resistive touchscreens can come in 4-wire and 5-wire variants. The 4-wire touch screen operates as explained previously, using two layers with a metallic film along the inner portion of each layer, one layer with electrodes in the X direction and one in the Y direction, and when the layers are pushed together it uses this to determine the X-Y coordinate. The 5-wire screen only sends voltage across the bottom layer, and uses the top layer as a voltage probe which can be used to pinpoint a coordinate when pressed. This makes it slightly more resilient since minor damages to the top layer may leave the screen with its functionality, but for this application the increased cost is not worth the durability, making the 4-wire variant the best candidate.

Parts:
      When looking for screens, they can come with an overlay already attached, and a ribbon cable with all the necessary connections on it, or a screen and overlay can be purchased separately, and the overlay can be placed over the screen. This usually will result in a cheaper overall price, but requiring slightly more work to manage the different connections. Table 3.8.1.a weighs some of the possible options.

| Part | Diagonal Size | Resolution | Screen/Overlay | Cost |
|---|---|---|---|---|
| ER-TP070-1 | 7 inch | N/A | Overlay | $9.39 |
| 2544-AFK800480A0-7.0N12NTM-R-ND | 7 inch | 800x480 | Both | $36.04 |
| AFY800480B0-5.0N12NTM-R | 5 inch | 800x480 | Both | $31.62 |

| | | | | |
|---|---|---|---|---|
| ATM0500D27-T | 5 inch | 800x480 | Both | $36.39 |
| ER-TFT070-2 | 7 inch | 800x480 | Screen | $17.09 |
| ER-TP050-2 | 5 inch | N/A | Overlay | $2.96 |
| ER-TFT050-3 | 5 inch | 800x480 | Screen | $16.52 |
| MSP4022 | 4 inch | 480x320 | Both | $11.83 |

Table 3.8.1.a - Touch Screen Options and Specifications

## 3.8.2 Processor:

A microcontroller unit is an integrated circuit that includes a processor, memory, I/O, and more depending on the unit. Examples of this would be an MSP430 board by Texas instruments. The units are typically used in embedded systems, and have less processing power. A system on a chip usually combines a variety of systems, such as an MCU, DSP, or other integrated circuits. This gets used in more complicated situations. An example of this type of chip would be an Arduino or a Raspberry Pi.

Although a system on a chip would likely make for easier interfacing and control of the touchscreen LCD, an MCU should be sufficient for the task, so it would be overspending to utilize a system on a chip for this application. Table 3.8.2.a below weighs different microcontroller options which could be used.

| Part | Core Size | RAM | # of I/O | Cost |
|---|---|---|---|---|
| MSP430G2553IPW28R | 16-bit | 512 B | 24 | $3.18 |
| ATSAM4LC4AA-MUR | 32-bit | 32 KB | 27 | $6.36 |
| PIC24FJ256DA210T-I/PT | 16-bit | 96 KB | 84 | $11.35 |
| MK70FN1M0VMJ15 | 32-bit | 128 KB | 128 | $30.94 |
| MSP430FR6989IPZR | 16-bit | 2 KB | 83 | $12.12 |
| ESP-WROOM-32 | 32-bit | 520 kB | 39 | $5.99 |

Table 3.8.2.a - Microcontroller Options and Specifications

## 3.9 Housing

There are two main approaches to be made when considering housing options, building a unit, or buying a commercial pc/electronics case. The listed specifications of the overall unit are 2 feet wide by 1 foot tall by 1 foot deep and less than 12 pounds, this would be the upper limit. As components are designed the dimensions of the housing should decrease to be just large enough to fit all of the components. The main benefits to building a housing unit is being a cheaper product, as well as being completely customizable to fit our needs. The benefits of purchasing a case would be better quality than a homemade box, already integrated with some front I/O, at a higher price, and requiring modification to fit some of our interfacing requirements.

### 3.9.1 Building a Unit

With no one in the group having a background in woodworking or some other construction based hobby, any unit built will end up being somewhat primitively designed and constructed. Since the design will end up being the same for each unit built by the group, the main difference would be the choice of material. The main options being MDF board, wood panels, or metal panels. Metal panels provide possibly the highest rigidity and best aesthetic, at a higher cost and difficulty of working with, while wood and MDF would be easier to work with and have a lower cost. One advantage of building the unit is that if any I/O changes, we can simply swap a panel with the correct holes while if a purchased unit has had any modifications to it, it would require purchasing an entirely new unit. Different possible materials as well as their cost and cost per square foot are shown below in table 3.9.1.a.

| Type of Material | Size | Area | Cost | Cost/ft^2 |
|---|---|---|---|---|
| MDF Board | ¼'' x 24'' x 48'' | 8ft^2 | $12.49 | $1.56 |
| Red Oak Plywood | ¼'' x 24'' x 48'' | 8ft^2 | $18.99 | $2.37 |
| Aluminum Composite ACM | ⅛'' x 24'' x 48'' | 8ft^2 | $44.04 | $5.50 |
| Aluminum | .019'' x 36'' x 36'' | 9ft^2 | $34.47 | $3.83 |
| Stainless Steel | .02'' x 6'' x 12'' | 0.5ft^2 | $6.47 | 12.94 |

Table 3.9.1.a - Material Options for Enclosure

## 3.9.2 Buying a Unit

Purchasing a unit would provide the best aesthetic and structural quality, but at the cost of a high price, and if any mistakes are made while modifying the unit, an entirely new unit would have to be purchased. There are a few types of units to be considered for purchase.  The first being a generic metal enclosure as shown below in Figure 3.9.2.a. This would be the cheapest type of enclosure that can be purchased as is, and has no I/O already built into it. This makes it the lowest cost if mistakes are made, and as shown in the picture, it has removable panels, which means if a mistake or change was made to the front panel, only the front panel would have to be replaced.



Figure 3.9.2.a - Metal Enclosure for DIY Project

The next type of enclosure to be considered is a PC case, which would include some front I/O, a higher build quality, and rear I/O where the motherboard usually is. The main type of cases to consider would be a mini-ITX case or a micro-ATX case for their small form factor. Figure 3.9.2.b below shows an example of a mini-ITX case which could be used for the product. The main drawbacks to this is a higher cost, and if any mistakes are made, an entirely new case would need to be purchased and modified.

Figure 3.9.2.b - Mini-ITX PC case

Another possibility would be a mountable enclosure, which could be mounted to a server rack or audio stand that some people have in their home setups. This enclosure would require the same modifications as the previously mentioned types of enclosures to allow for I/O.  This enclosure comes at a higher price, but compatibility with existing audiophiles system towers.



Figure 3.9.2.c - Mountable Enclosure

One possible problem with all of these units or ones similar, is that they do not exactly fulfill size requirements like if an enclosure to size, instead something simply within margin needs to be found, so building an enclosure would allow us to create the smallest possible footprint. Table 3.9.2.d below shows some of the possible enclosures that could be used for the product.

| Part | Size | I/O | Cost |
|------|------|-----|------|
| Golden Field N-1 Mini ITX Case | 12.91'' x 10.24'' x 7.95'' | 2 x USB, 1 x Headphone, 1 x Microphone, 1 x Power Button, 1 x Reset Button | $54.99 |
| Zulkit Junction Box | 12.6'' x 10.6'' x 4.7'' | N/A | $32.99 |
| Rack Mount Case Enclosure (37-2U) | 17.44'' x 11.81'' x 3.26'' | N/A | $71.03 |
| Cooler Master MasterBox Q300L | 15.39'' x 9.06'' x 15'' | 2 x USB, 1 x Headphone, 1 x Microphone, 1 x Power Button, 1 x Reset Button | $69.99 |
| GETTTECH Mid Tower Computer ATX Case | 13'' x 6.9'' x 16'' | 2 x USB, 1 x Headphone, 1 x Microphone, 1 x Power Button | $41.12 |

Table 3.9.2.d - Enclosure Options and Specs
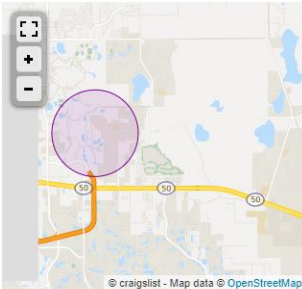
### 3.9.3 Recycling a Unit

In addition to buying, or making a housing unit, there is always the option of recycling another amplifier housing unit. The similar shape doesn't even have to come from specifically a bass amplifier either, we could use the old housing of regular guitar amps, old PCs, cable boxes, or other sound equipment for a much cheaper but similarly effective housing.

Below are some examples of electronics that are being sold on Craigslist that could be gutted and used as a housing for our amplifier.
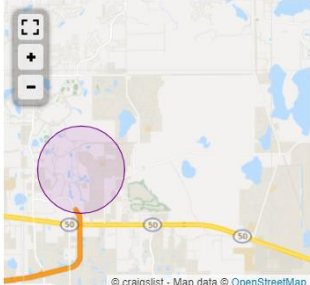
This receiver and track player has a similar size to the dimensions we laid out earlier in this document, such that we could likely quite easily salvage the housing and use it in our project. The knobs and buttons could also save us some cost so that we wouldn't have to buy those components separately.



Similarly to the previous example, this receiver could be salvaged as a housing or just for the buttons and knobs on the front.



Figure 3.9.3.a - Examples of Recycled Units

The advantages of recycling an already used unit, even if it was completely nonfunctional, are the multitude of components that could be salvaged. Lots of components in this project are small and relatively inexpensive, but since we are only building a single bass amplifier and not an assembly line of thousands of them we are not buying in bulk which is much cheaper per component. Recycling a housing as well as some knobs and switches could save lots of money.

# 4.0 Constraints and Standards

## 4.1 Constraints

| Constraint | Description |
|---|---|
| Cost | Maximum limit of $500 spread across the team. This project will be self-funded so being below budget would be acceptable. |
| Time | The deadline for SD1 is approximately 12 weeks away. The SD2 Final report is due 31 weeks. Budgeting the group's time will be pertinent to the success of the project. |
| Product Availability | Due to global events, many electronic components are on backorder. Being able to order the components and have them in-hand before certain deadlines will constrain the project. |
| Social | The project is not constrained by any social factors. |

Table 4.1.a - Constraints

### 4.1.1 Economic and Time

The economic constraints of the project stem from it being a student-led project without the funding of a company or a grant behind it. This makes budgeting, and smart utilization of resources a primary concern for the development of the project. If surpassing the decided budget becomes inevitable, the group will reassess the state of the project, as well as the financial state of individual members to decide what needs to be done to keep the project moving forward.

### 4.1.2 Environmental, Social, and Political

This project does not have any major environmental, social, or political constraints. If it advanced into a large-scale operation, it could focus on replacing materials with recyclable or reusable ones, in order to lessen the impact of the product filling up landfills at the end of its owner's usage or the product lifespan.

There are no social or political constraints involved with developing a bass guitar amp header, as it does not have a noticeable impact on the social or political climate.

### 4.1.3 Ethical, Health, Safety

There are no ethical constraints on the development of the project, but each team member should be keeping up with their work so no one becomes individually burdened.

The health and safety constraints on the project stem from working with managing power from a wall outlet, and signal amplification taking place. Improper calculations or safety precautions could result in harm to a member of the team.

### 4.1.4 Manufacturability and Sustainability

The manufacturability constraint is focused on a modular design, which could replace certain components such as the preamp or digital effects unit if they fail, rather than replacing the whole system.

The sustainability of the project would only be considered if it were moved forward with production, but it would be based around designing a product with a low cost to produce, which could be sold at a reasonable profit margin while still being competitive on the market.

## 4.2 Standards

| Standard | Description |
|---|---|
| Bluetooth | Would comply with Bluetooth 5.0 Standards (IEEE 802.15.1) |
| RoHS Compliant | All components will be compliant with the RoHS standard for hazardous materials |
| IEC 61984 | Neutrik Chassis Connector |
| NEMA 5-15 | 3 Prong Plug |
| IPC-2221 | Ensure design reliability |
| IPC-2222 | Rigid PCB |
| IPC-2223 | Flex PCB |

Table 4.2.a - Standards

# 5.0 Design

## 5.1 Pre-Amplifier

### 5.1.1 Technical Design

The Pre-Amplifier design features two, cascaded, op amp stages. There are two voltage gain stages focused on applying a specified gain to the signal. There is a dip switch control to set the mode to either high or low gain. After these stages there is an output stage connecting to an overload detector and an output to the equalization stage. These stages will be detailed later. A signal flow diagram of the design is shown in Figure 5.1.1.a below.
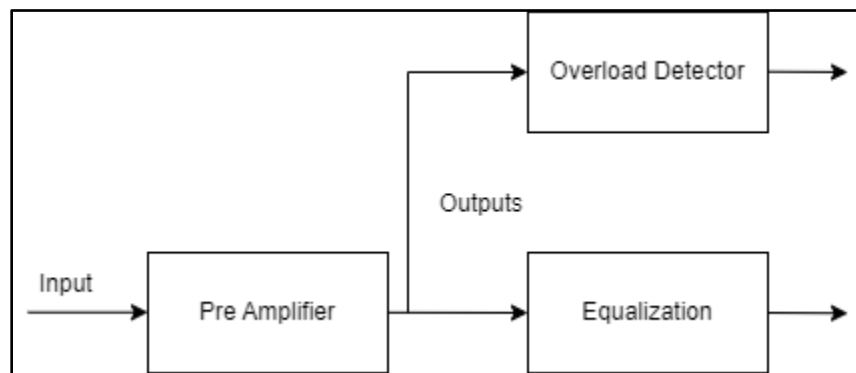


Figure 5.1.1.a - Signal Flow Diagram of Pre-Amplifier

Both gain stages use the OPA1642 op amp in the non-inverting configuration. The stages are set up in a way to work off each other, while being able to be adjusted. The first stage is a simple non-inverting voltage amplifier. Resistor 1, R1, is a current limiting resistor. Ideally, there is no current coming into or leaving an op amp terminal, but in practice, there is a small current. The value of the resistor should be large enough to prevent too much current entering, but still small enough to not place a load on the signal. The value of 220Ω was chosen for R1. Next, R3 sets the input impedance of the amplifier. This needs to be a very large value so that there is no current on either of the op amp terminals. 1MΩ is a standard value for this application. The gain of a non-inverting configuration is set by either Formula 5.1.1.b or Formula 5.1.1.c.

$$A_V = 1 + \frac{R2}{(R4 + \frac{1}{j\omega C_1} + R_6)//R_5}$$

Formula 5.1.1.b - Low Gain Equation

$$A_V = 1 + \frac{R2}{(R4 + \frac{1}{j\omega C_1})//R_5}$$

Formula 5.1.1.c - High Gain Equation

Formula 5.1.1.b applies when the switch S1 is open. This is the low gain mode of the amplifier. Formula 5.1.1.c applies when S1 is closed. This is the high gain mode of the amplifier. The low gain mode applies a lower gain to the signal. This mode is useful when the input signal does not need a high gain or when the input is noisy. The high gain mode creates more noise but also amplifies the input signal mode. The capacitor C1 does affect the phase response of the system, but this is not meaningful because the equalization stage will have more of an effect on the phase response. C2 is a filter capacitor. The value of 10uF was calculated to have the least effect on the phase response, while still providing a filter against DC voltages. R7 is a dual-gang potentiometer. The other side of this potentiometer is R11. This is the main gain control of the system. Commonly confused as a volume knob, R7 and R11 will be the gain control knob on the front of the housing. The value of 10kΩ was chosen to limit the small amount of current the op amp is outputting. These two potentiometers being connected allows the easy configuration of two gains, as well as being a safety measure. If the gain is set too high, the preamp will not function correctly and be a safety hazard.

Stage two of the preamplifier is another simple, non-inverting configuration. The gain is set by Formula 5.1.1.d.

$$A_V = 1 + \frac{R10}{R9}$$

Formula 5.1.1.d - Gain of Stage Two

R8 is a current limiting resistor, working in series with R7. The total gain of stage two is 10 V/V. C3 is a filter capacitor. 10uF was chosen using the same math as C2. R11 was talked about in the previous paragraph. Finally, the diode D1 is used to isolate the output of the preamp to the Overload Detector. The specific diode, 1N4148, was chosen for its very fast switching speed and its reverse saturation current properties.

The finished design and schematic shown in Figure 5.1.1.e. The Bill of Materials, BOM, is shown in Table 5.1.1.f.
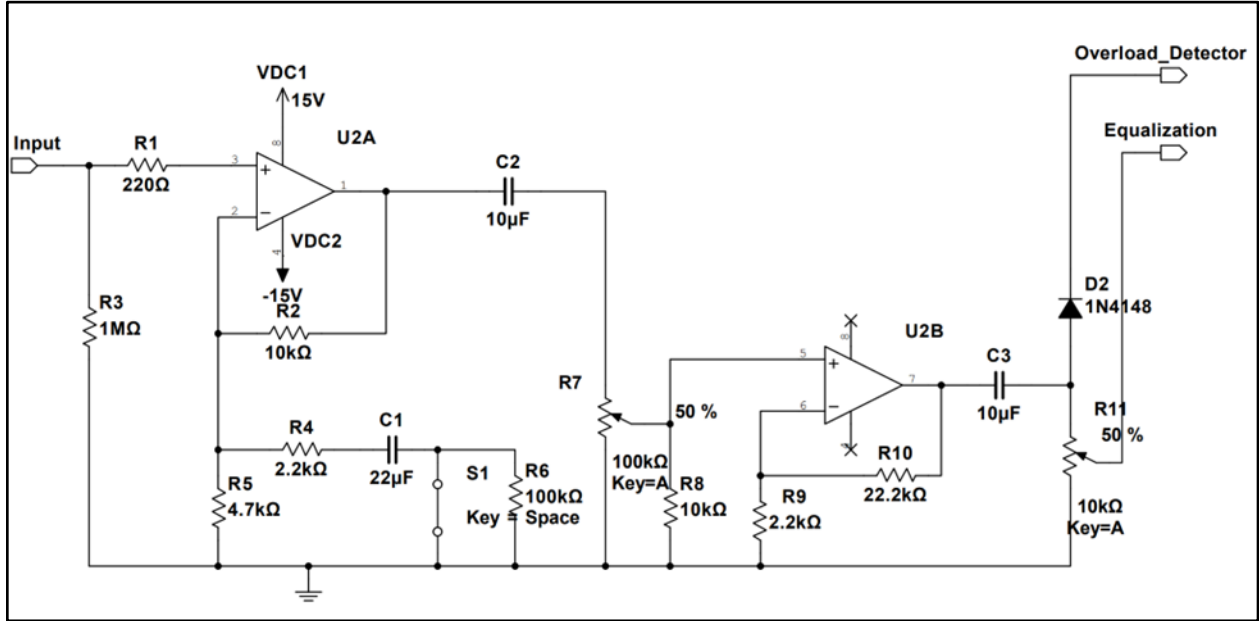
Figure 5.1.1.e - Pre-Amplifier Design Schematic

| Qty | Parts | Value | Description |
|---|---|---|---|
| 4 | C1, C2, C3, C4 | 10u | CAPACITOR |
| 1 | D1 | 1N4148 | DIODE |
| 1 | IC1, IC2 | OPA1642 | OP AMP |
| 1 | R1 | 220 | RESISTOR |
| 1 | R7, R11 | 10k | DUAL-GANG POTENTIOMETER |
| 1 | R2 | 1M | RESISTOR |
| 1 | R3 | 10k | RESISTOR |
| 2 | R4, R8 | 2.2k | RESISTOR |

| | | | RESISTOR |
|---|---|---|---|
| 1 | R5 | 4.7k | |
| 1 | R6 | 100k | RESISTOR |
| 2 | R9, R10 | 22.2k | RESISTOR |
| 1 | S1 | | DIP SWITCH |

Table 5.1.1.f - BOM for Pre-Amplifier Schematic

## 5.1.2 Simulation Testing

The Pre-Amplifier design was tested using Multisim 14.3. The OPA1642 PSpice model was downloaded from Texas Instruments' website. The potentiometers were set at 50% of their value. The input was a 25mV peak sine wave at a frequency of 1kHz. This provides a good middle ground for what could be an input to the preamp. Figure 5.1.2.a shows the output versus the input of the preamp when it is operating in the low gain configuration. The red sine wave is the output and the green sine wave is the input. This is a constant throughout all of the figures. This was done for visibility purposes. Figure 5.1.2.b shows the output versus the input, but now the preamp is in the high gain mode.

Next is the frequency and phase response of the Pre-Amplifier. Figure 5.1.2.c shows the overall frequency and phase response. The ideal design would have a completely flat frequency response. For the frequency specification of the project, 31Hz through 4.5kHz, the frequency response is very much comparable to ideal. Figures 5.1.2.d,e, and f show the frequency response broken up into bass, mid, and treble frequencies. This was done to more thoroughly show the frequency response. In this stage, the phase response does not impact the performance of the preamp.

Finally, Figure 5.1.2.g details the total harmonic distortion and the signal to noise ratio. A THD of 0.001% satisfies the requirements of the design. The SN ratio of 98.38 dB also satisfies the requirements of the design.
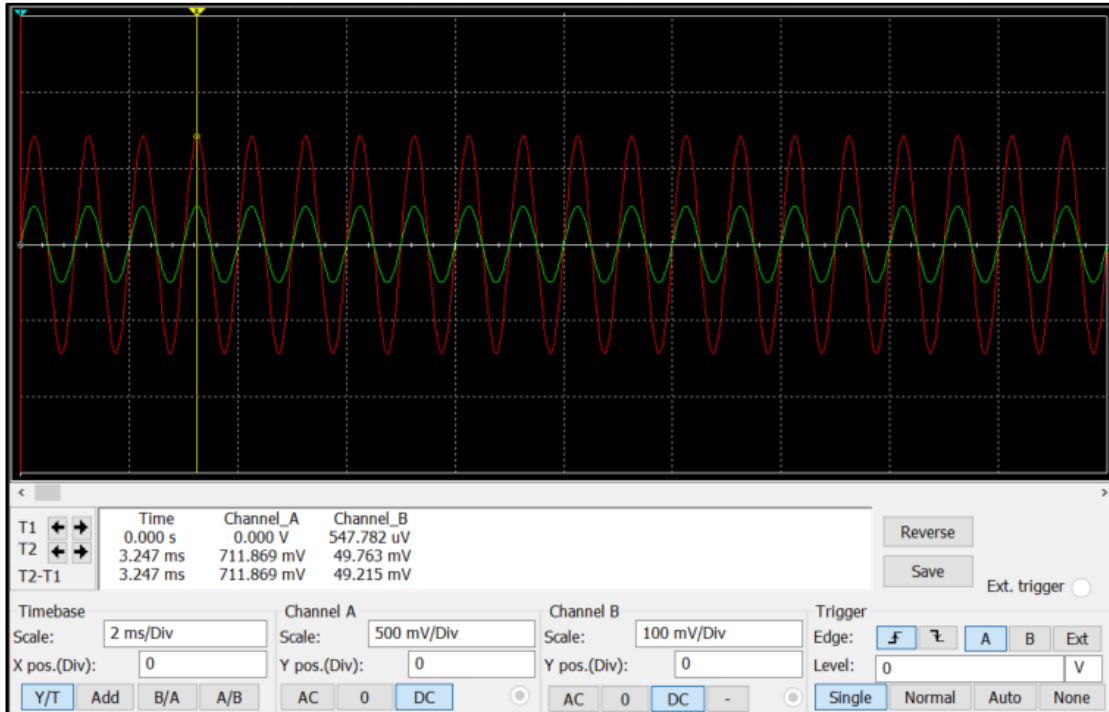
Figure 5.1.2.a - Vout vs. Vin Low Gain
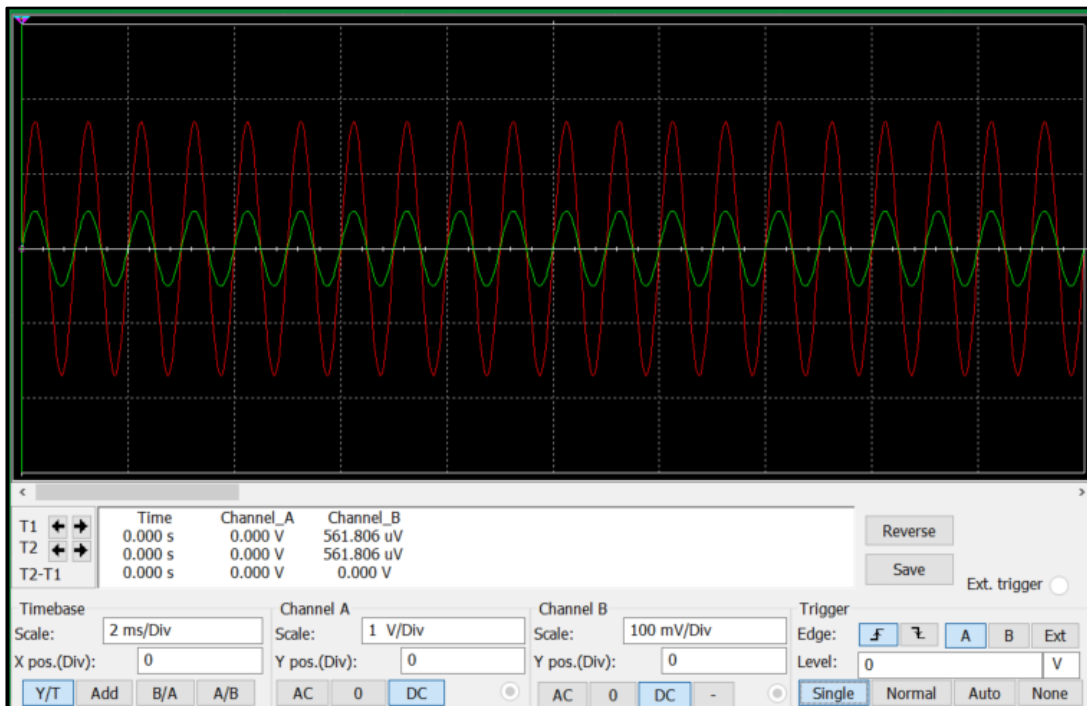


Figure 5.1.2.b -  Vout vs. Vin High Gain
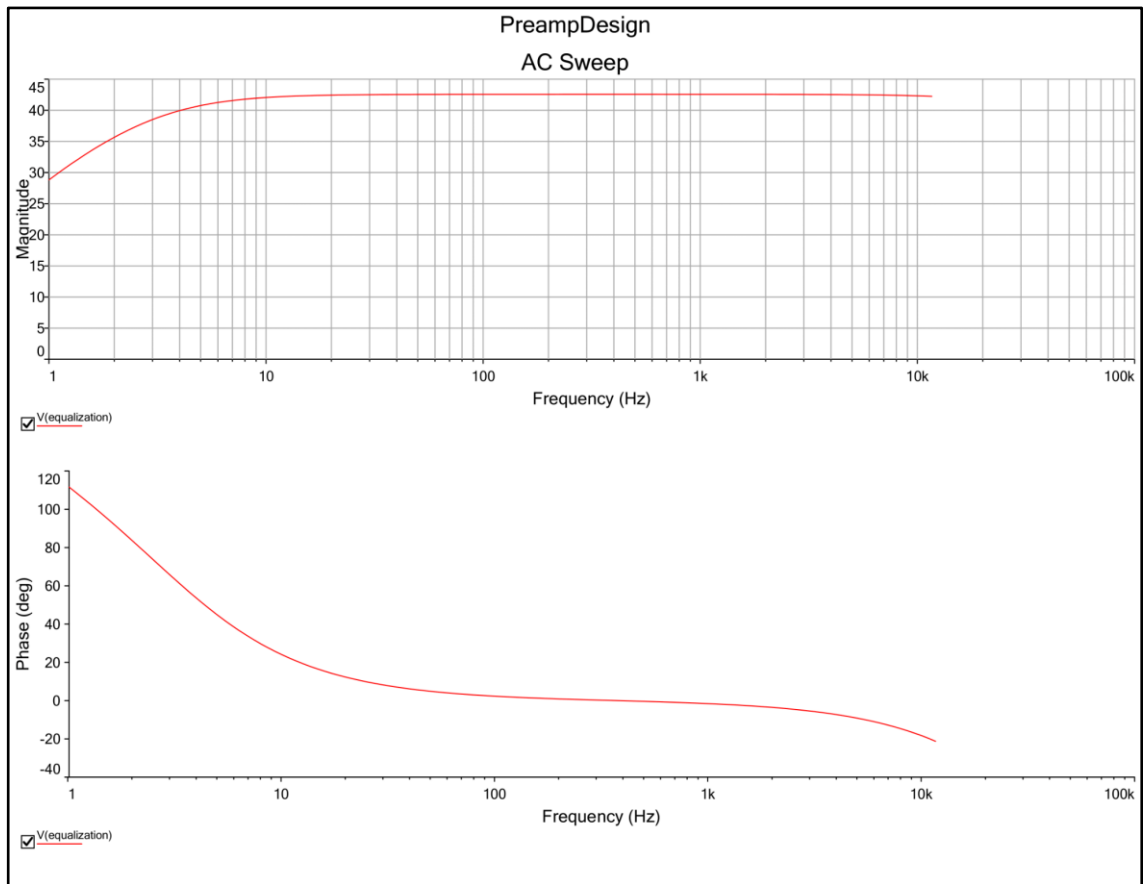
Figure 5.1.2.c - Overall Frequency Response
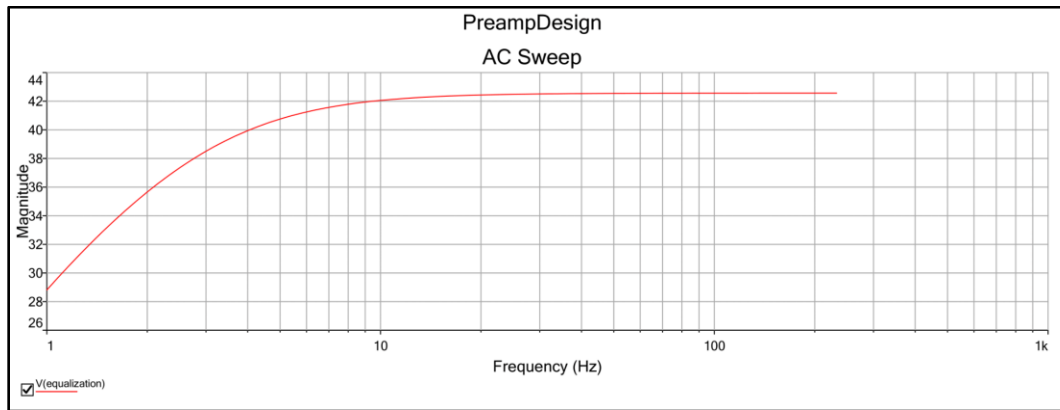


Figure 5.1.2.d - Bass Response
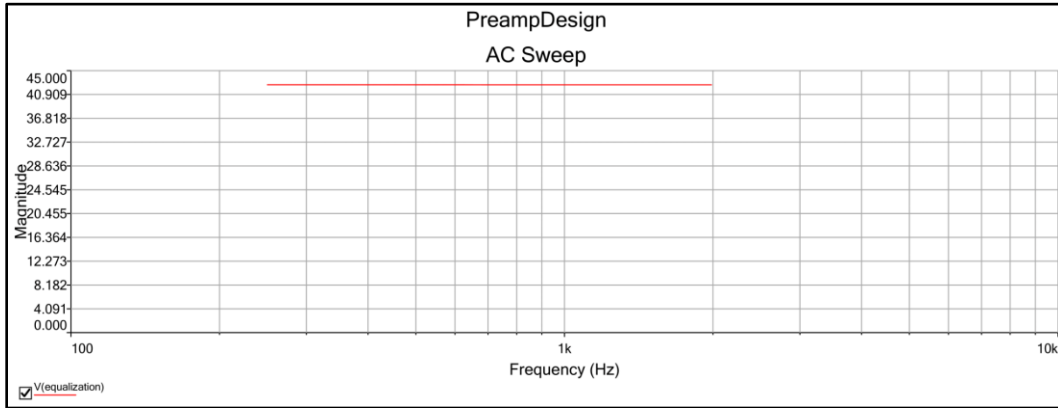
Figure 5.1.2.e - Mid Response
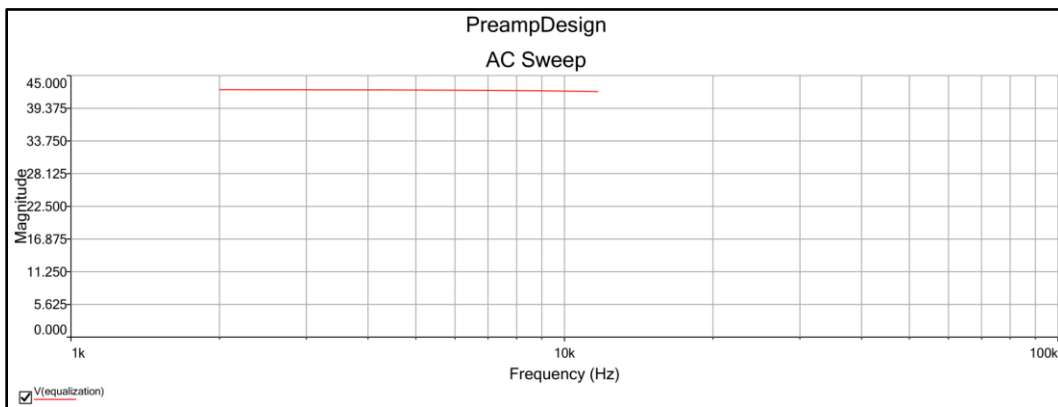


Figure 5.1.2.f - Treble Response
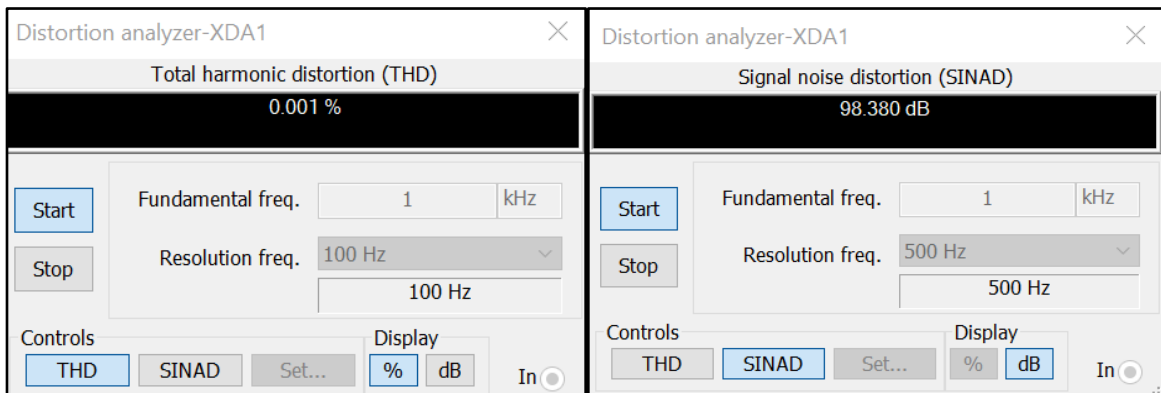


Figure 5.1.2.g - Total Harmonic Distortion and Signal to Noise Ratio

## 5.2 Overload Detector

### 5.2.1 Technical Design

The Overload Detector design is very simple, featuring a LM393 comparator. The design also features a BS170 MOSFET to drive the red led. From Figure 5.2.1.a, the voltage divider between R1 and R3 set the reference voltage at 5V. So if the input voltage from the input is greater than 5V, the detector would apply a positive voltage to Q1. Due to the instability of comparators in general, hysteresis is used to create an upper and lower threshold for the output to change. R2 and R4 bias the positive terminal so that the upper threshold is 5.15V and the lower threshold 4.85V. The use of a BS170, Q1, was to be able to drive current to the led. The output of the comparator is not enough for this use case. R7 was used to tie the input to ground when there is no input. If left untied, this may present issues with accurate readings.
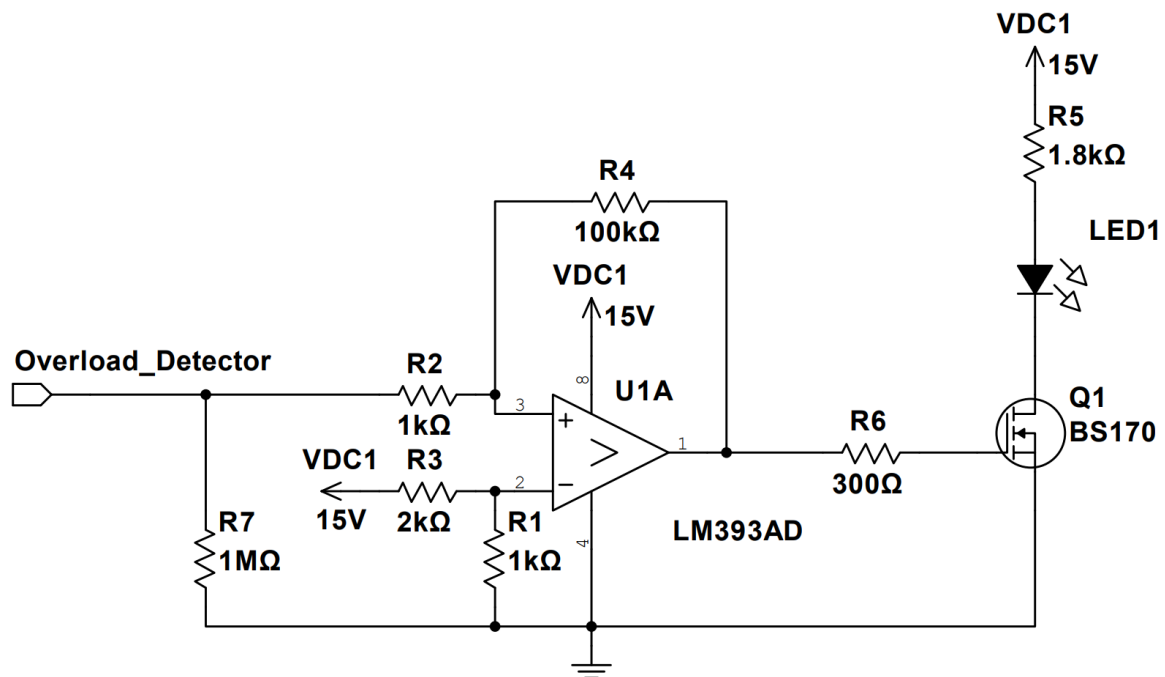


Figure 5.2.1.a - Overload Detector Schematic

| Qty | Parts | Value | Description |
| --- | --- | --- | --- |
| 2 | R1, R2 | 1k | RESISTOR |
| 1 | R3 | 2k | RESISTOR |
| 1 | R4 | 100k | RESISTOR |
| 1 | R5 | 1.8k | RESISTOR |
| 1 | R6 | 300 | RESISTOR |
| 1 | R7 | 1M | RESISTOR |
| 1 | UA1 | LM393AD | VOLTAGE COMPARATOR |
| 1 | Q1 | BS170 | N-CHANNEL MOSFET |
| 1 | LED1 | Red | LED |

Figure 5.2.1.b - BOM for Overload Detector Schematic

## 5.2.2 Simulation Testing

The testing of this design was done in Multisim 14.3 using standard components. If the design is functional, the led should turn on when the input is above 5V and turn off when it is below. Due to the hysteresis, the lower threshold is 4.85V and the upper is 5.15V. Figure 5.2.2.a, the output versus input graph, shows the design is functioning properly and meets the design requirements.
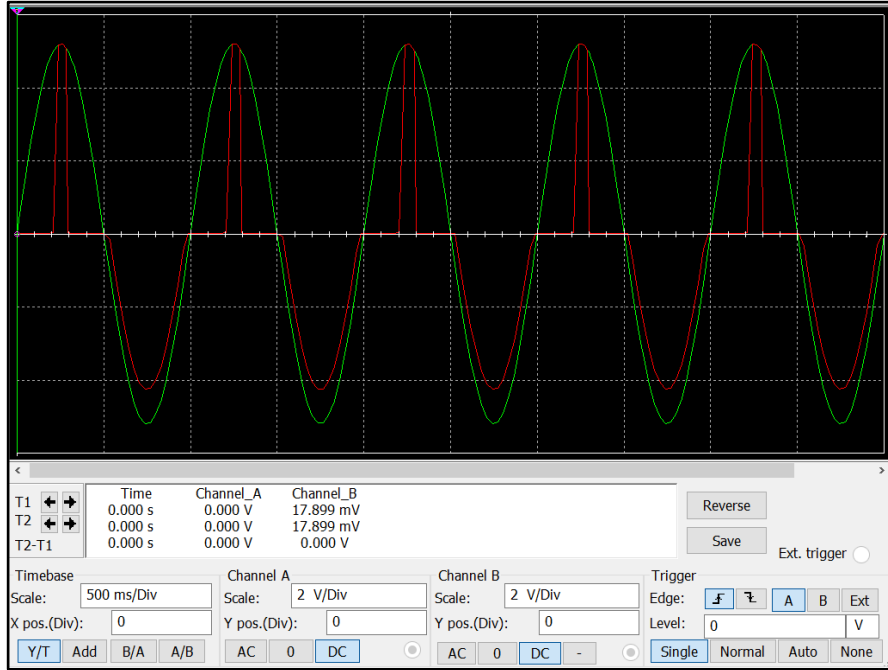
Figure 5.2.2.a - Output vs. Input Graph

## 5.3 Equalization

### 5.3.1 High Pass Filter

The first stage of the EQ design is a basic 2nd order active high pass filter. This is necessary to attenuate the very low frequency noise and other sources of interference. The lowest fundamental frequency of a 4-string bass guitar is 40Hz and 31Hz on a 5-string. Any frequencies below 31Hz are hard to drive and are unwanted. Anything below 20Hz is inaudible to humans, thus not important to the design. The design of the high pass filter has a corner frequency of 25.26Hz. Figure 5.3.1.a shows the schematic of the designed high pass filter.
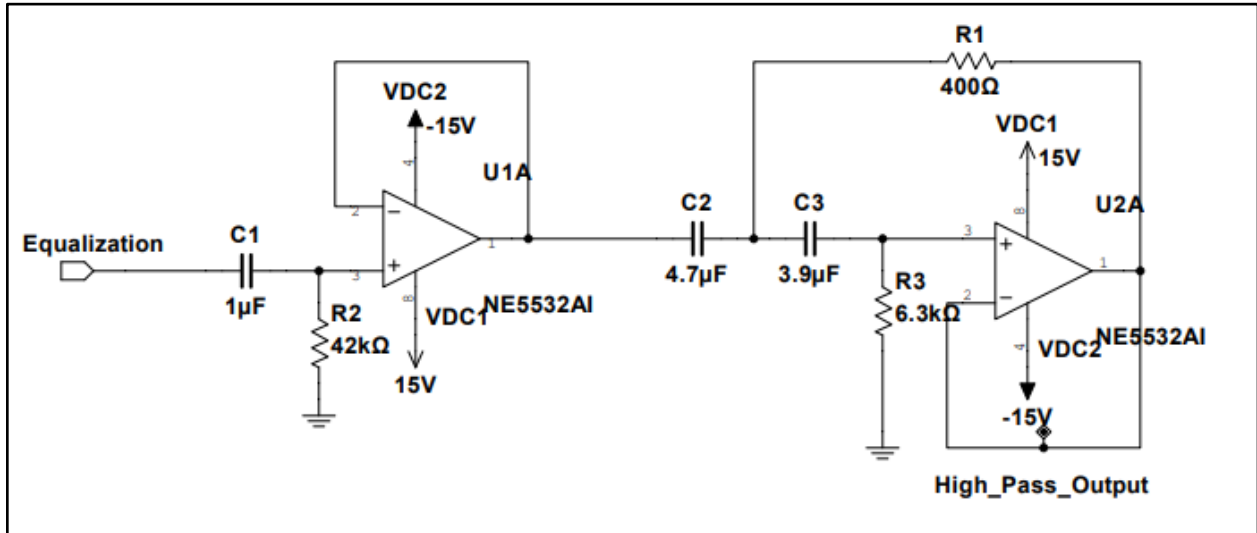
Figure 5.3.1.a - High Pass Filter Schematic

## 5.3.2 High Pass Filter Performance
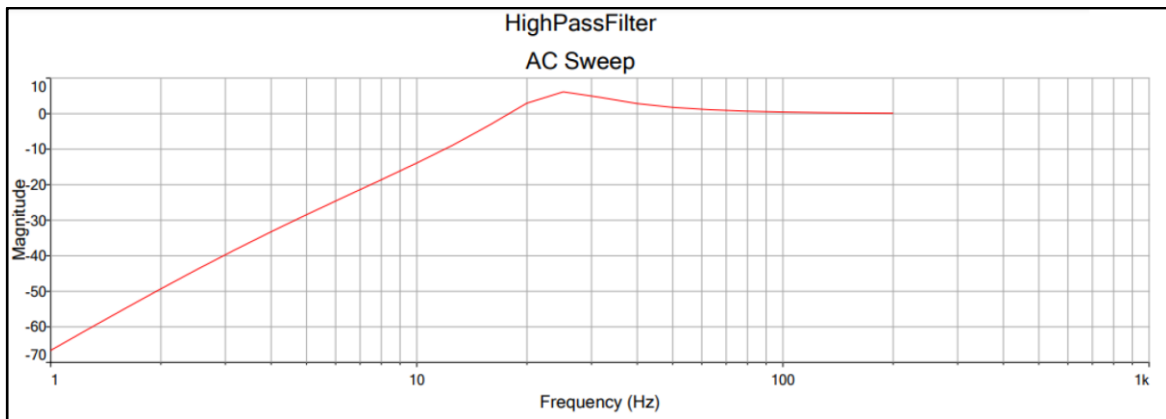


Figure 5.3.2.a - High Pass Filter Performance

## 5.3.3 Graphic Equalization

As referenced before, the design of the Graphic Equalizer was done using Gyrator Filters. The specific filter is known as a Gyrator Notch Filter. A notch filter is a bandstop filter, with a narrow stopband. The use of notch filters allows for a near 0dB insertion loss, due to frequency response at resonance. The impedance of the filter, at resonance, acts like a short circuit. The formula to calculate the resonance frequency of a gyrator notch filter is shown in Formula 5.3.3.a.

$$f = \frac{1}{2\pi\sqrt{R_1 * R_2 * C_1 * C_2}}$$

Formula 5.3.3.a - Gyrator Filter Resonance Frequency

Using Formula 5.3.3.a, the filters were designed with the targeted frequencies in mind. The targeted frequencies are: 80, 250, 500, 1500, 3000, and 5000 Hz. The calculations for the resonance frequency, based off of the designed filters are shown in Table 5.3.3.b.

| Name | Targeted Frequency (Hz) | Calculated Frequency (Hz) |
|---|---|---|
| Bass | 80 | 80.09 |
| Upper-Bass | 250 | 253.84 |
| Low-Mid | 500 | 491.56 |
| Mid | 1500 | 1504.81 |
| High-Mid | 3000 | 2948.32 |
| Treble | 5000 | 4997.09 |

Table 5.3.3.b - Resonance Frequency Calculations

Figure 5.5.3.c details the specific arrangement of components to obtain the notch filter with the calculated resonance frequency. All op amps were NE5532. The reasons for this component choice were detailed in Section 3.3.4. The Input and Output Stage, shown in Figure 5.3.3.d, utilizes an inverting summing op amp to combine each of the filters' outputs. The inverting nature of this stage allows the notch filters to keep the wanted characteristics of a notch filter, while applying a boost or cut to the signal.

**Bass**

C7
1µF

C8
94nF

VDC1
15.0V

R13
560Ω

U5A

3 +

2 −

1

8

4

VDC2

NE5532AI

R14
75kΩ

-15.0V

**250**

C1
820nF

C2
15nF

VDC1
15.0V

R5
470Ω

U2A

3 +

2 −

1

8

4

VDC2

NE5532AI

R6
68kΩ

-15.0V

**500**

C3
400nF

C4
8.2nF

VDC1
15.0V

R7
470Ω

U3A

3 +

2 −

1

8

4

VDC2

NE5532AI

R8
68kΩ

-15.0V

**1500**

C5
100nF

C6
3.5nF

VDC1
15.0V

R10
470Ω

U4A

3 +

2 −

1

8

4

VDC2

NE5532AI

R11
68kΩ

-15.0V
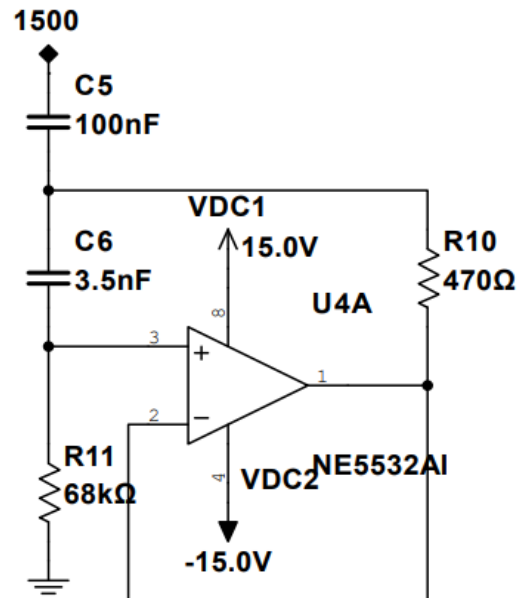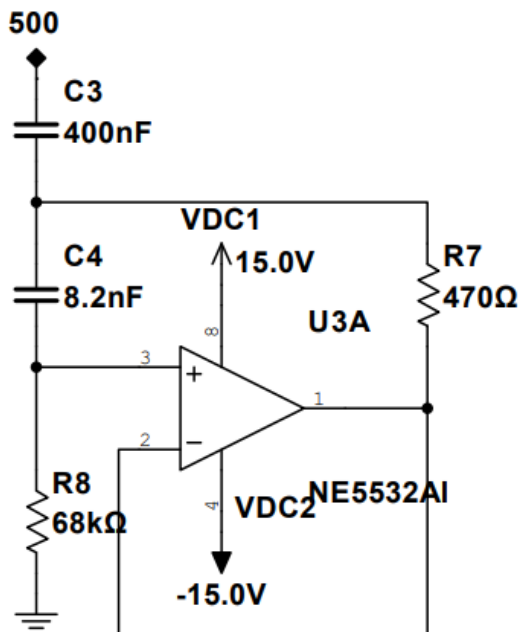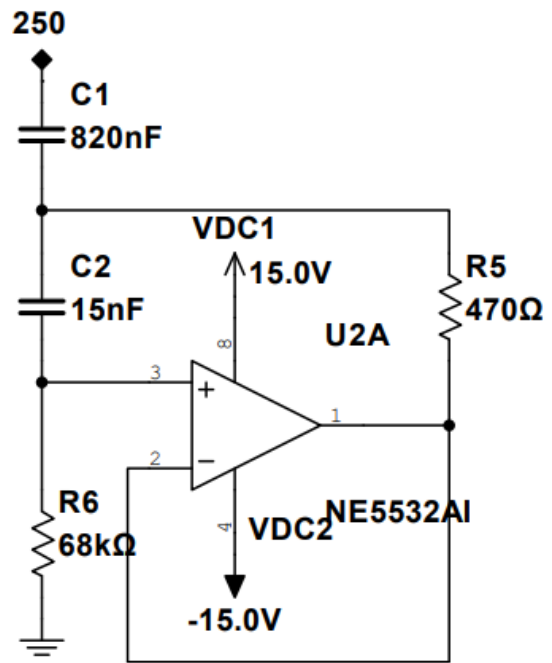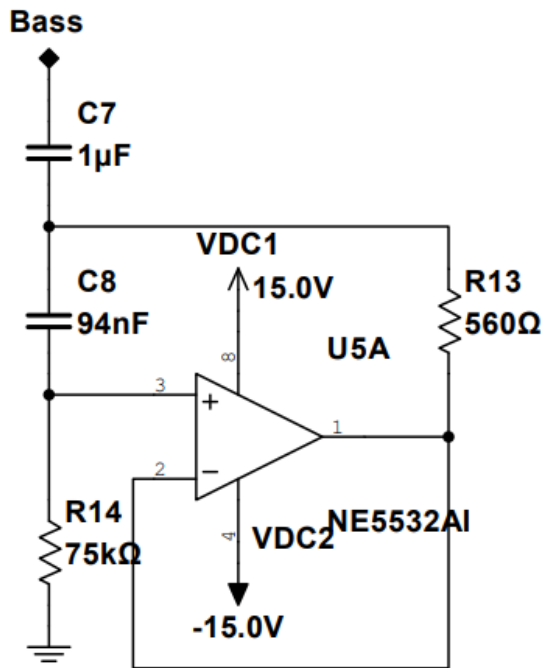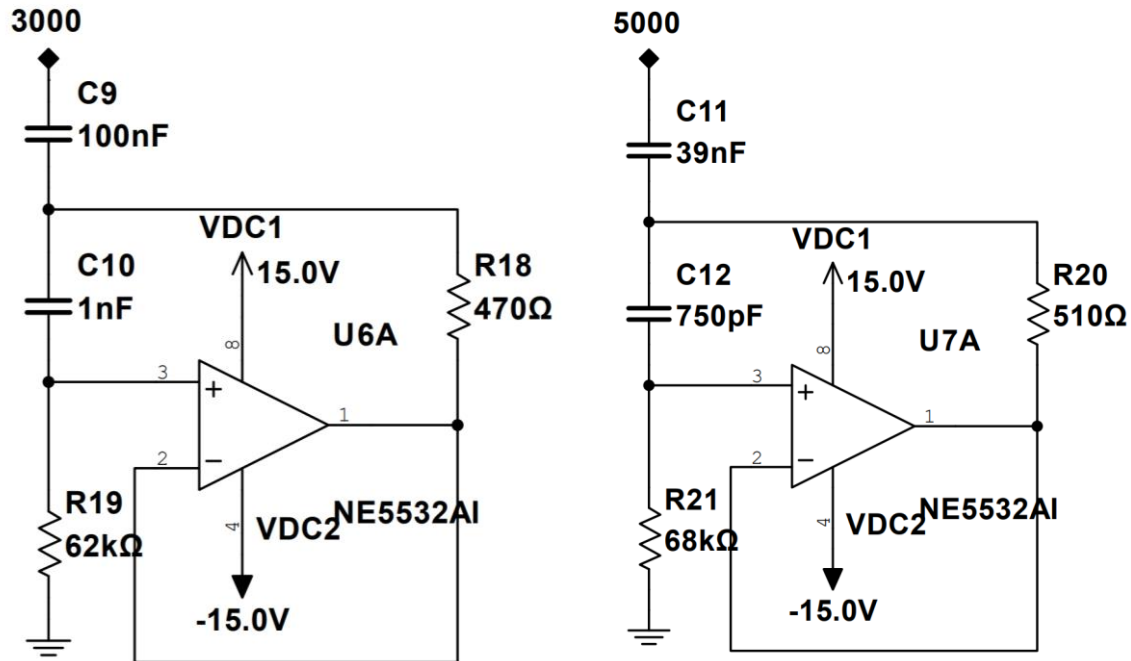
Figure 5.3.3.c - EQ Filter Design
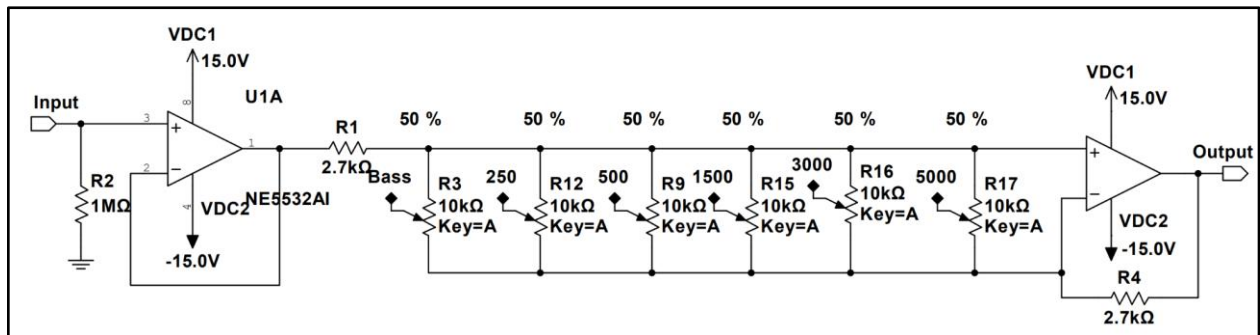


Figure 5.3.3.d - Input and Output Stage

## 5.3.4 Graphic Equalizer Performance

The performance characteristics of the designed Graphic Equalizer are shown in Figure 5.3.4.a. The performance for this Figure was tested with all potentiometers set at half. Figure 5.3.4.b details the maximum boost or cut that can be applied in the design. The maximum boost was measured at 12dB and the maximum cut was measured at -12dB.
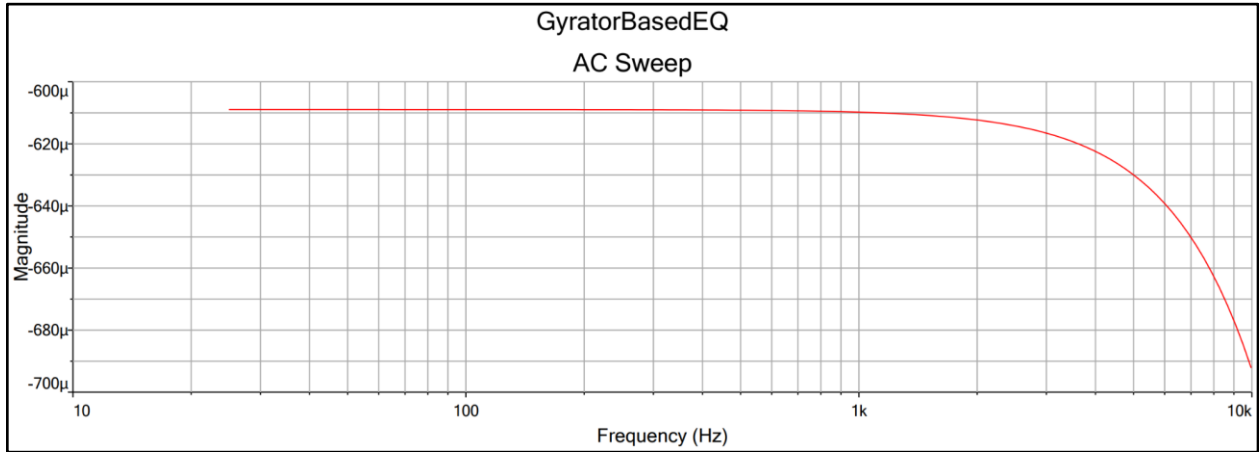
Figure 5.3.4.a - Frequency Response of Graphic Equalizer
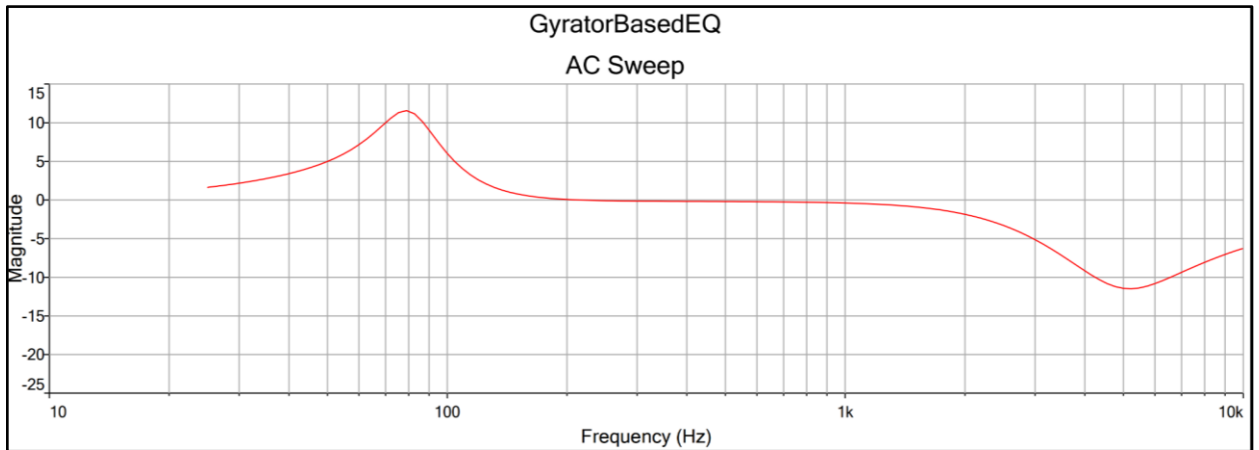


Figure 5.3.4.b - Maximum Boost and Cut Specification

## 5.4 Power Amplifier

The schematic below shown in Figure 5.4.a is the design for the power amplifier. The design is centered on the usage of an Integrated circuit for the poweramp, of which we selected the TDA7294. The data sheet provides a reference circuit to construct a poweramp utilizing the IC, and provides directions for choosing component values to achieve certain characteristics. Using this the design below was constructed.
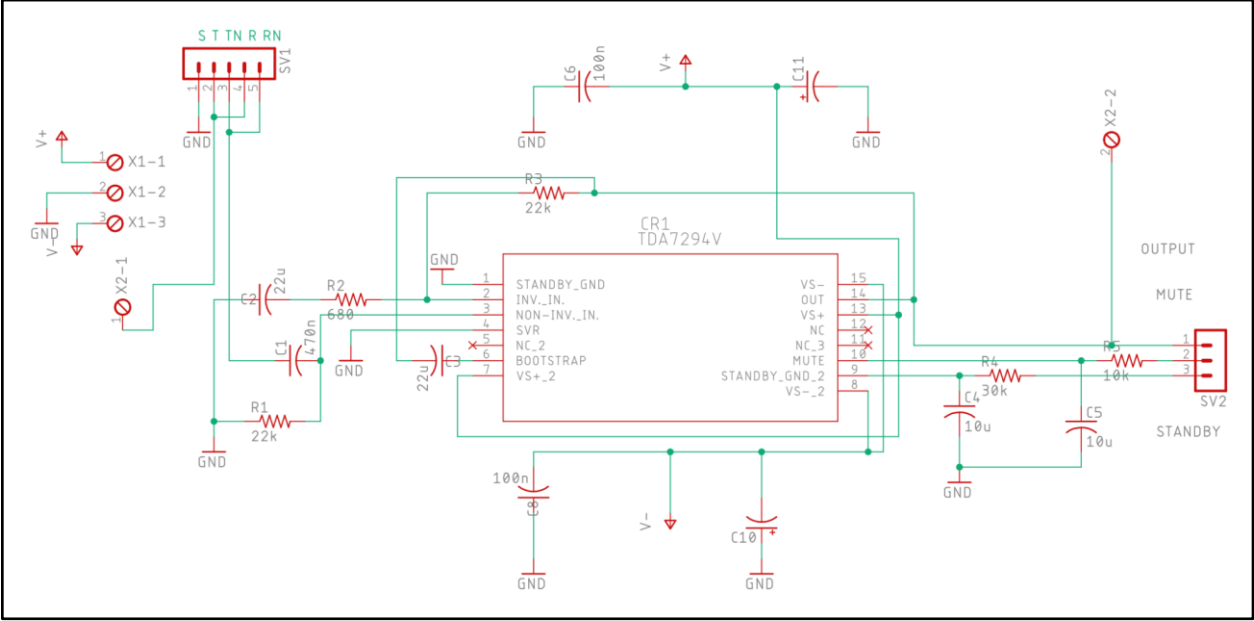
Figure 5.4.a - Power Amplifier Schematic

## 5.5 Power Supply
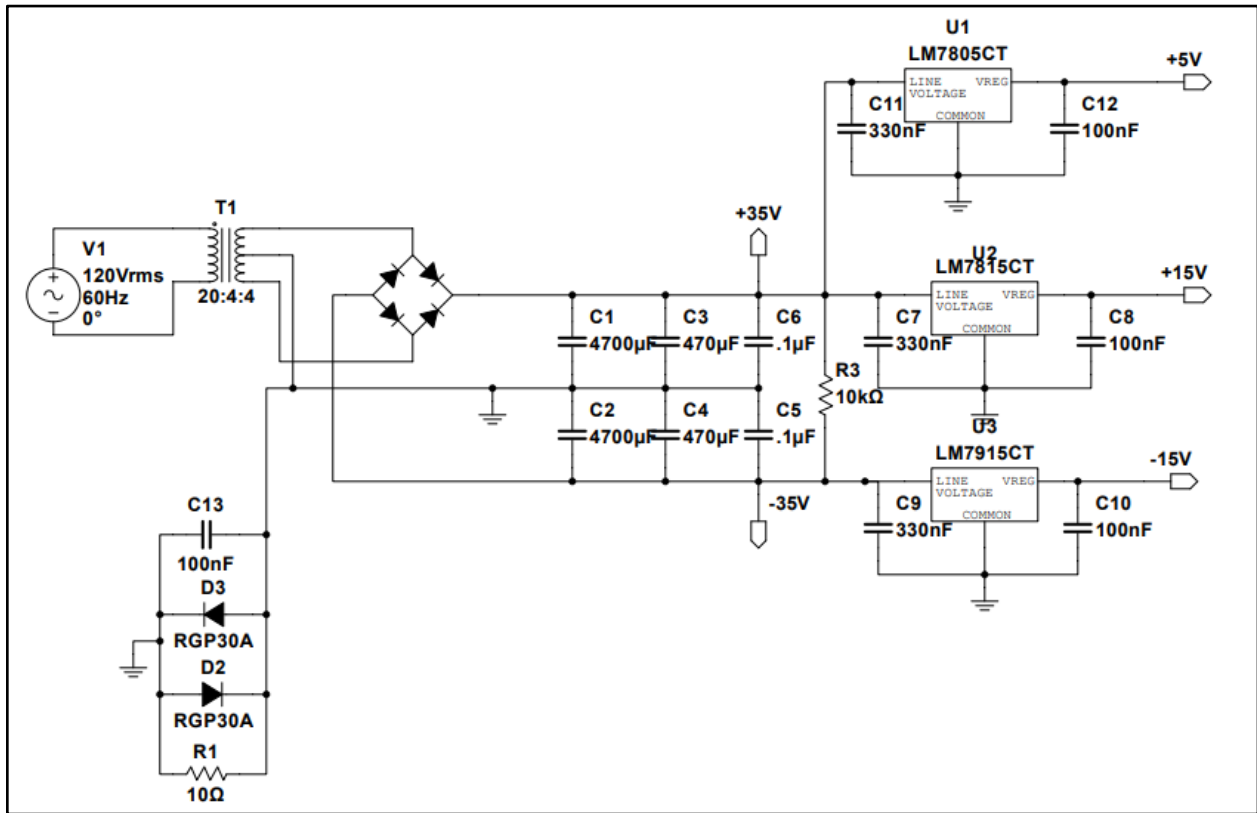


Figure 5.5.a - Power Supply Schematic

### 5.5.1 Transformer

In order for the Power Amplifier and the rest of the circuitry to not be current limited, the minimum power rating would be 150VA. The formulated design uses a 175VA, center-tapped transformer capable of outputting 36V at 4.8A. The specific model is the 185G36 from Hammond Manufacturing. This transformer has dual input voltages of 115 and 230VAC to support both North American and European power standards. Being based in North America, only the 115VAC input will be used.

### 5.5.2 Bridge Rectifier

The Bridge Rectifier chosen for this design is a through-hole, 10A rated glass passivated bridge rectifier. The 10A rating was chosen to allow for headroom. The total power supply should not draw close to 10A, so this part was chosen with headroom in mind.

### 5.5.3 Filters

The Power Supply Filters are a simple array of capacitors ranging from 4700uF to 0.1uF. Filter Capacitors are needed to deal with the ripple voltage created by the bridge

rectifier. All of the capacitors are rated for at least 50V. The maximum voltage that would be rectified is 35V, but due to ripple or other inconsistencies, a rating of 50V was chosen so that the capacitors do not malfunction.

The second stage of the Filters is a bleeding resistor. A bleeding resistor discharges the capacitors when there is no input. The bleeding resistor has been shown to have a moderate effect on the frequency response of the power supply. The overall frequency response of the power supply is not a huge concern, but the resistor value of 10k was chosen to limit this response, while still discharging the capacitors.

### 5.5.4 Regulators

The required regulated voltages for the design are +15V, -15V, and +5V. The information on regulators in the audio world tends to stay away from switching regulations. The high frequency noise generated from the switching action might lead to the interference of audio signals. For this reason, linear regulators were chosen to be used. The LM78xx Series of linear voltage regulators are commonplace in all regulation areas. The LM7815 is used to regulate +35V into +15V at 1A. The LM7915, a negative voltage regulator, regulates -35V into -15V at 1A. The LM7805 regulates +35V into +5v at 1.5A.

### 5.5.5 Power Supply Performance

The Power Supply was designed to output +35V, -35V, +15V, -15V, and +5V. The design was simulated in Multisim 14.3. Using the multimeter tool, the voltage at the output nodes was measured and recorded below in Figure 5.5.5.a.

Figure 5.5.5.a - Power Supply Performance

Another important performance metric of a power supply is the ripple voltage. The ripple voltage was measured to be 16.71mV at a maximum. Figure 5.5.5.b shows the output voltage and ripple voltage calculations.
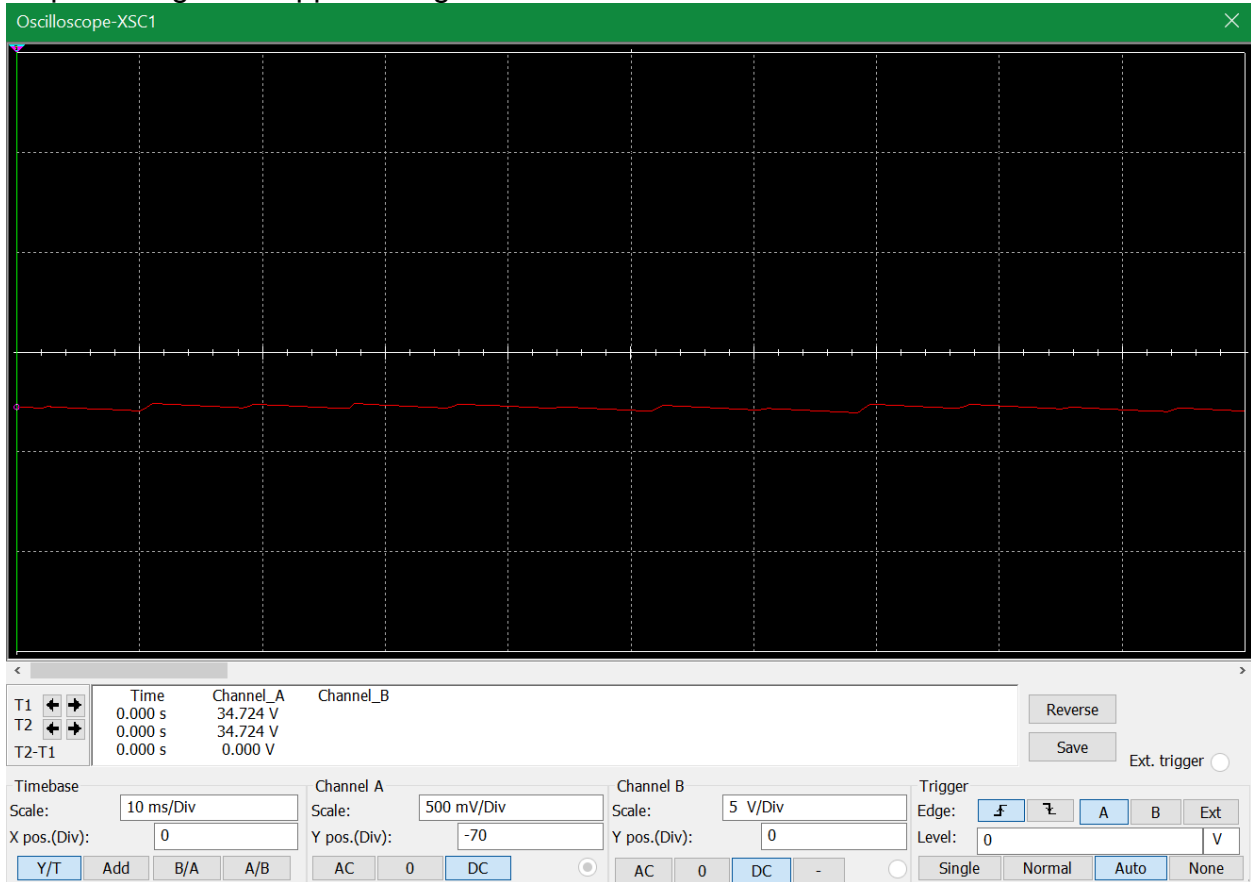


Figure 5.5.5.b - Power Supply Ripple Voltage

## 5.6 Analog Effects

### 5.6.1 Compressor Design

Most effect pedals are based on a tried and true design that has proved itself over the years. This compressor is based on the MXR Dyna Comp pedal adapted to our use case. The main components of the circuit include a power section which has a 9 volt battery along with a voltage divider to give a 2.54V rail for some of the components. The next section is an emitter follower which acts as an input buffer to the rest of the circuit. The next section is an OTA amp which is what actually compresses the signal to the desired threshold. It does this by applying a gain to the signal based on the input it receives from the next stage, the envelope detector. When the signal is weaker, it will provide a larger gain, while stronger signals will have a small or negative gain. The next section is an envelope detector, which as stated before, sends a signal back to the OTA

amplifier which decides how much gain will be applied to the signal. The final stage is another emitter follower, which acts as an output buffer for the signal.
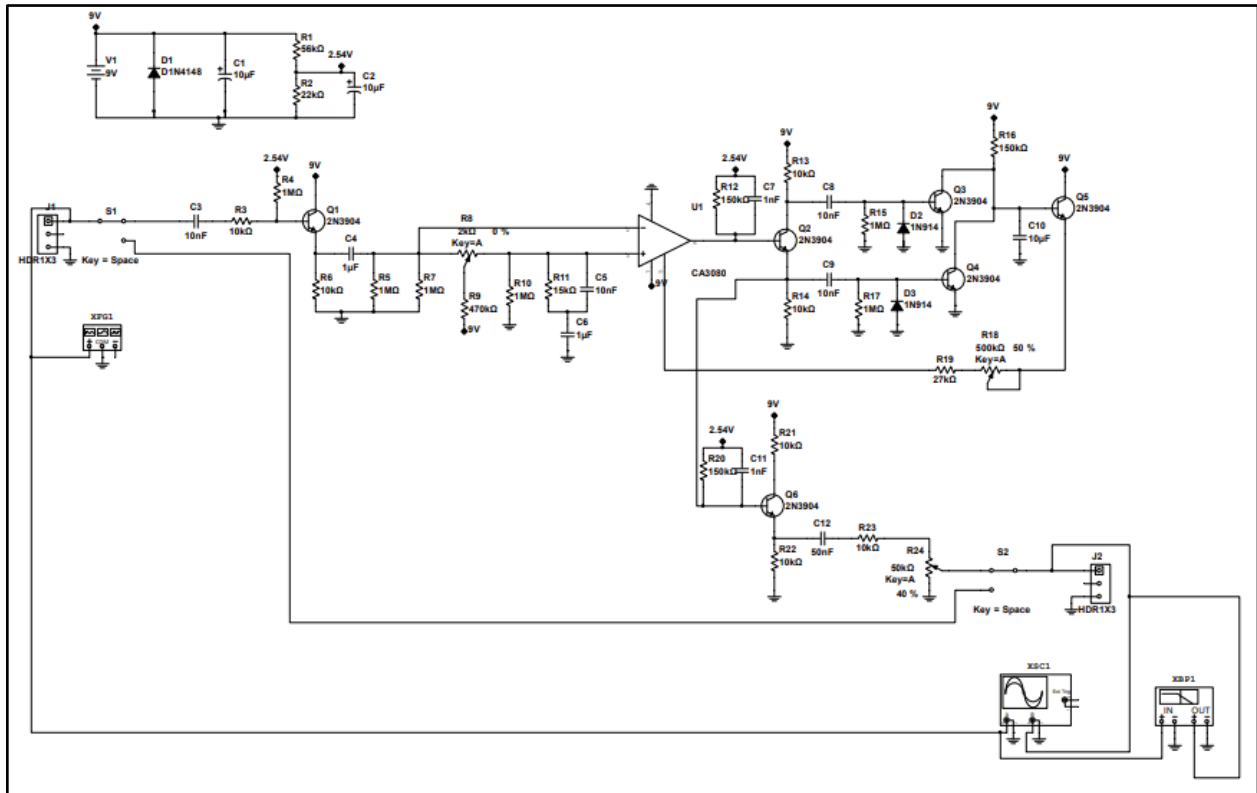


Figure 5.6.1.a - Compressor Schematic

| Qty | Parts | Value | Description |
|---|---|---|---|
| 7 | R3, R6, R13, R14, R21, R22, R23 | 10k | RESISTOR |
| 1 | R11 | 15k | RESISTOR |
| 1 | R2 | 22k | RESISTOR |
| 1 | R19 | 27k | RESISTOR |
| 1 | R1 | 56k | RESISTOR |
| 3 | R12, R16, R20 | 150k | RESISTOR |
| 1 | R9 | 470k | RESISTOR |
| 6 | R4, R5, R7, R10, R15, R17 | 1M | RESISTOR |

| 1 | R8 | 2k | POTENTIOMETER |
|---|---|---|---|
| 1 | R24 | 50k | POTENTIOMETER |
| 1 | R18 | 500k | POTENTIOMETER |
| 2 | C1, C2 | 10u | ELECTROLYTIC CAPACITOR |
| 2 | C7, C11 | 1n | CAPACITOR |
| 4 | C3, C5, C8, C9 | 10n | CAPACITOR |
| 1 | C12 | 50n | CAPACITOR |
| 2 | C4, C6 | 1u | CAPACITOR |
| 1 | C10 | 10u | CAPACITOR |
| 1 | D1 | D1N4148 | DIODE |
| 2 | D2, D3 | 1N914 | DIODE |
| 6 | Q1, Q2, Q3, Q4, Q5, Q6 | 2N9304 | NPN TRANSISTOR |
| 1 | U1 | CA3080 | OP AMP |
| 2 | J1, J2 | HDR1X3 | 3 PIN HEADER |
| 2 | S1, S2 | SPDT | SWITCH |

Figure 5.6.1.b - BOM for Compressor Schematic

## 5.6.2 Compressor Performance

As can be seen in Figure 5.6.2.a, the oscilloscope output of the compressor circuit, the effect is operating as expected. The compression level can be controlled with the potentiometer, and when the input signal is too large, it will compress the signal without clipping or any other effects that can change the sound quality besides the amplitude of the signal.
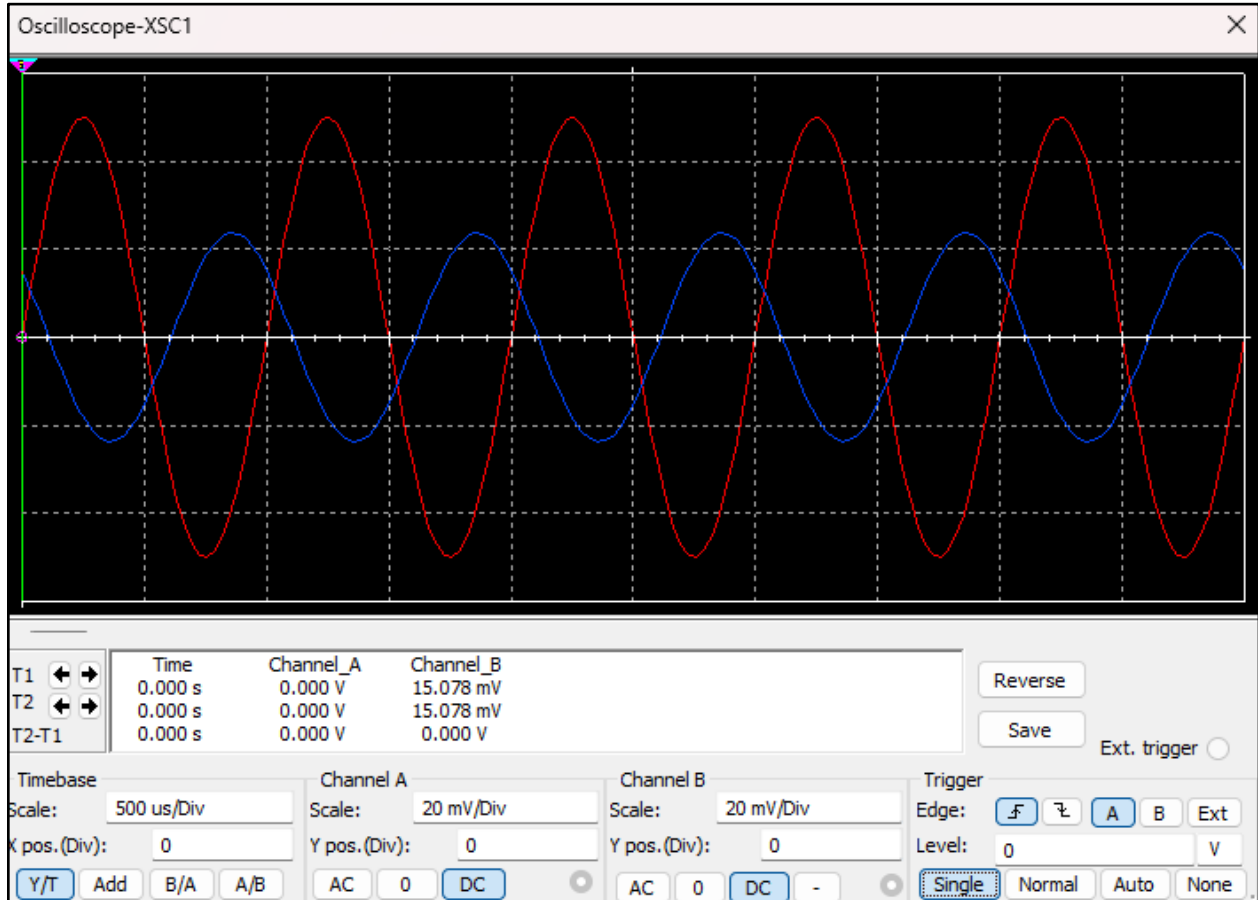
Figure 5.6.2.a - Vout vs. Vin Compressor Schematic

### 5.6.3 Fuzz Box Design

As stated before, many effect pedals are clones of older pedals. In the case of the fuzz box design, the design chosen was a clone of the 60's era Fuzz Face. There are some changes to bring the design up to today's standards. The original Fuzz Face used NKT275 germanium transistors. While not being in production anymore, the NKT275 is highly sought after for effect pedals, and therefore, commands a high price. In order to minimize cost, the AC128 PNP germanium transistor was chosen as the replacement part. It's also rumored that Arbiter, the original manufacturer of the Fuzz Face, used the AC128 in place of the NKT275 occasionally. The other design choices tried to keep the familiar sound quality of the Fuzz Face in mind.
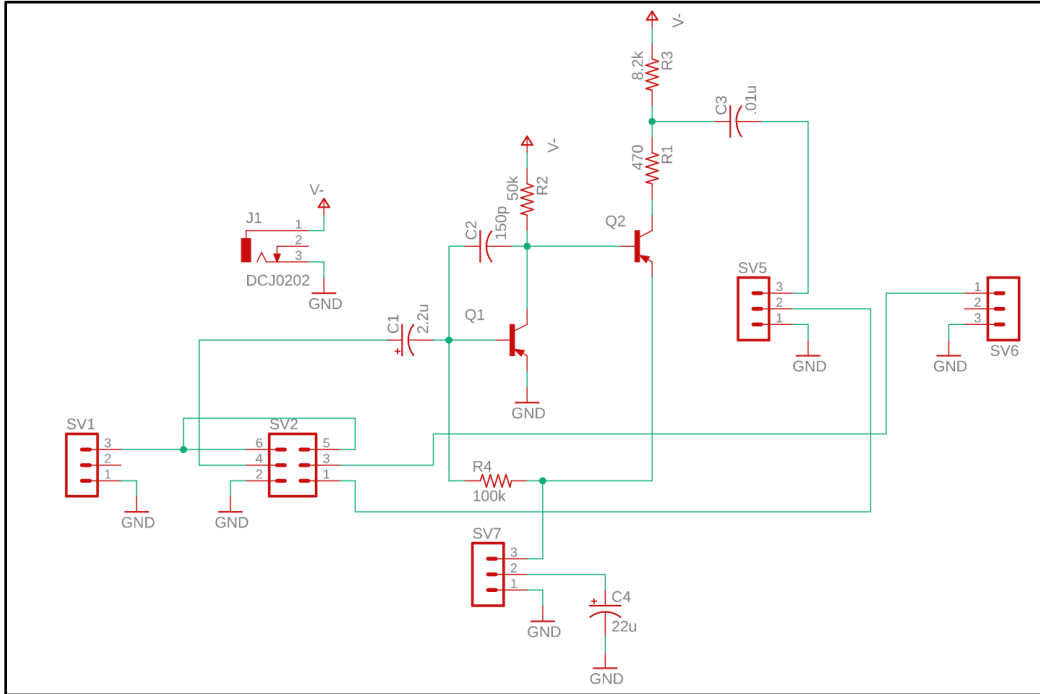
Figure 5.6.3.a - Fuzz Face Schematic

### 5.6.4 Fuzz Box Performance

Figure 5.6.4.a details the output versus input performance of the designed Fuzz Box. The performance is in line with the expected results of the design and other Fuzz Box competitors.
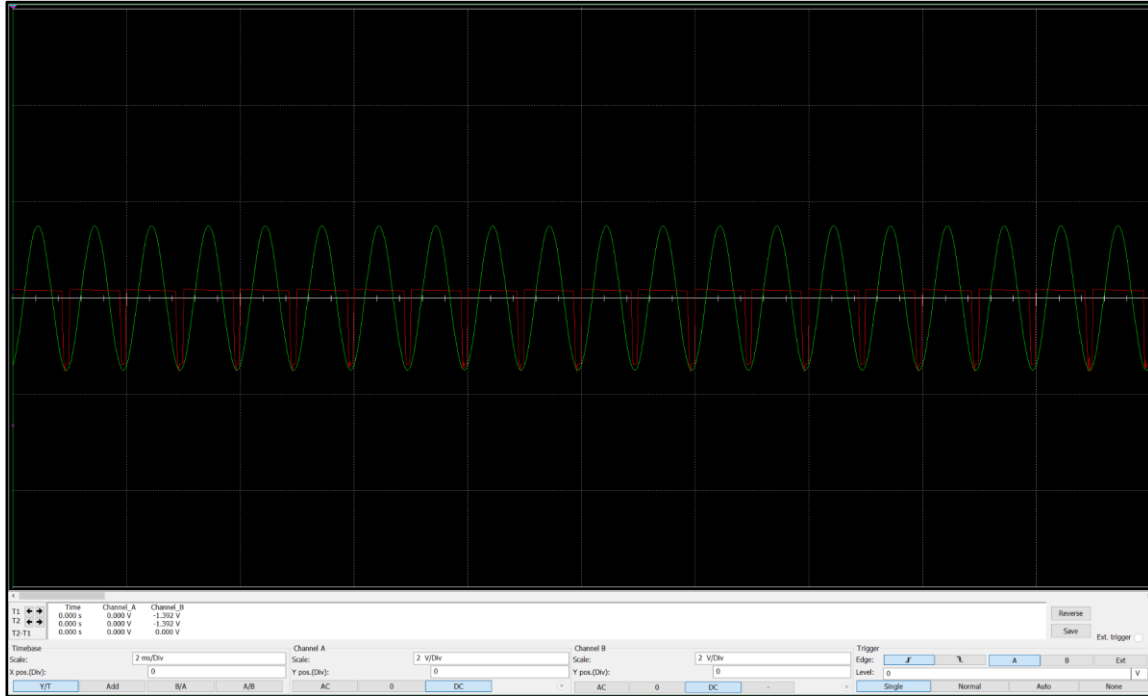
Figure 5.6.4.a - Vout vs. Vin Fuzz Box Schematic

### 5.6.6 Distortion Design

Previously mentioned was the fact that most effect pedals are cloned off of older models. The same follows for the distortion schematic below being based off the Boss DS1 adapted for our use. The main components of the circuit include a 9 volt battery with a voltage divider so there is a 9 volt and 4.5 volt rail to use throughout the circuit. The next section is an emitter follower, which acts as an input buffer. Next there is an emitter amplifier, which applies a high gain to the signal and clips the signal. Following that is the op amp block which applies further gain to the signal, and uses diodes D3 and D4 to hard clip the signal, which causes the stereotypical distortion effect. Potentiometer R14 controls the gain at this stage, or the distortion of the signal. The next potentiometers, R19 and R20 control the tone and level respectively of the output signal. Finally there is an emitter follower which acts as an output buffer. The other main component additions include two SPDT switches which allow the signal to pass through rather than applying an effect to it, and two 3 pin headers, which will allow the ¼ inch jacks to be wired to the board.
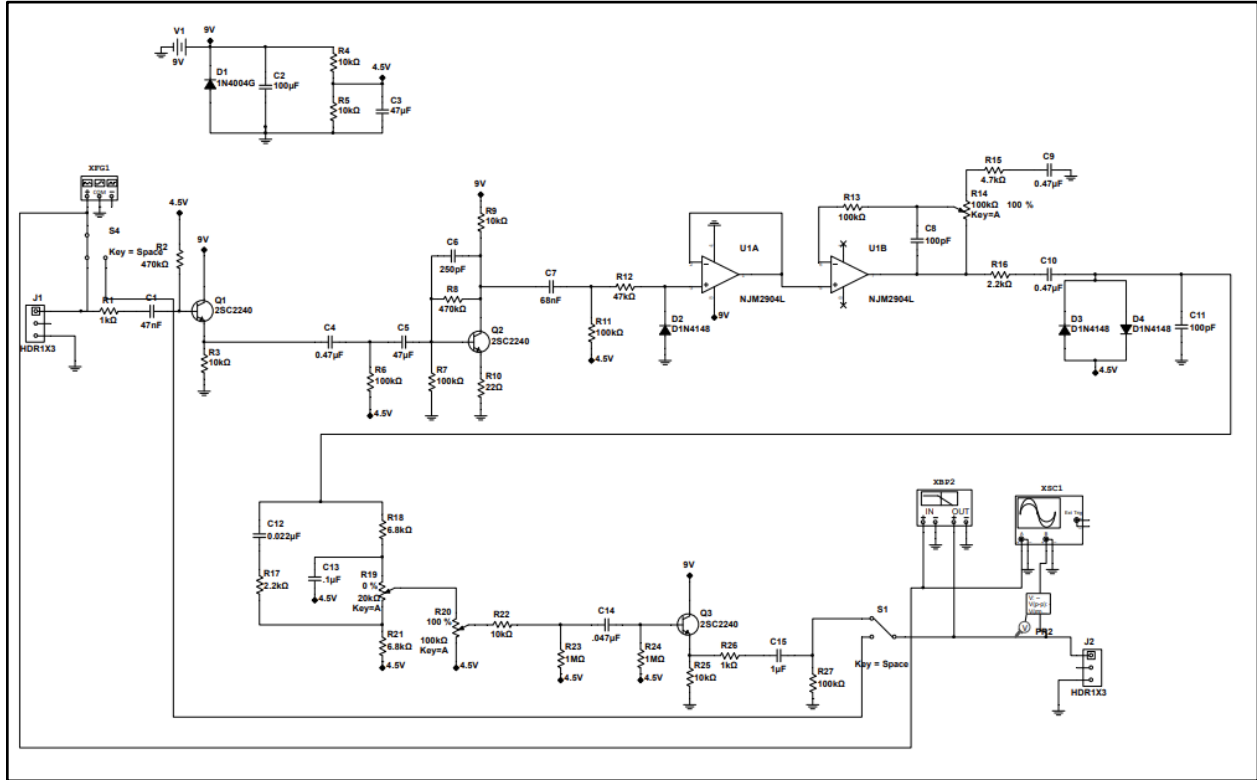
Figure 6.5.5.a - Distortion Schematic

| Qty | Parts | Value | Description |
| --- | --- | --- | --- |
| 1 | R10 | 22 | RESISTOR |
| 2 | R1, R26 | 1k | RESISTOR |
| 2 | R16, R17 | 2.2k | RESISTOR |
| 1 | R15 | 4.7k | RESISTOR |
| 1 | R18 | 6.8k | RESISTOR |
| 6 | R3, R4, R5, R9, R22, R25 | 10k | RESISTOR |
| 1 | R12 | 47k | RESISTOR |
| 5 | R6, R7, R11, R13, R27 | 100k | RESISTOR |
| 2 | R2, R8 | 470k | RESISTOR |
| 2 | R23, R24 | 1M | RESISTOR |
| 1 | R19 | 20k | POTENTIOMETER |

| | | | |
|---|---|---|---|
| 2 | R14, R20 | 100k | POTENTIOMETER |
| 2 | C8, C11 | 100p | CAPACITOR |
| 1 | C6 | 250p | CAPACITOR |
| 1 | C12 | 22n | CAPACITOR |
| 1 | C1 | 47n | CAPACITOR |
| 1 | C7 | 68n | CAPACITOR |
| 1 | C13 | 0.1u | CAPACITOR |
| 4 | C4, C9, C10, C14 | 0.47u | CAPACITOR |
| 1 | C15 | 1u | CAPACITOR |
| 2 | C3, C5 | 47u | CAPACITOR |
| 1 | C2 | 100u | CAPACITOR |
| 3 | Q1, Q2, Q3 | 2SC2240 | TRANSISTOR |
| 1 | D1 | 1N4004G | DIODE |
| 3 | D2, D3, D4 | D1N4148 | DIODE |
| 1 | U1A, U1B | NJM2904L | DUAL OP AMP |
| 2 | J1, J2 | HDR1X3 | 3 PIN HEADER |
| 2 | S1, S2 | SPDT | SWITCH |

Table 6.5.5.b - BOM for Distortion Schematic

### 5.6.8 Distortion Performance

The distortion pedal is operating as it is expected to, applying a high gain to the signal, as well as a harsh clipping. As can be seen in Figure 5.6.6.a, which is the oscilloscope output for the Distortion schematic, a high gain is applied to the input signal, and then the signal is harshly clipped, which is able to apply the stereotypical distortion effect most commonly associated with hard rock or metal.
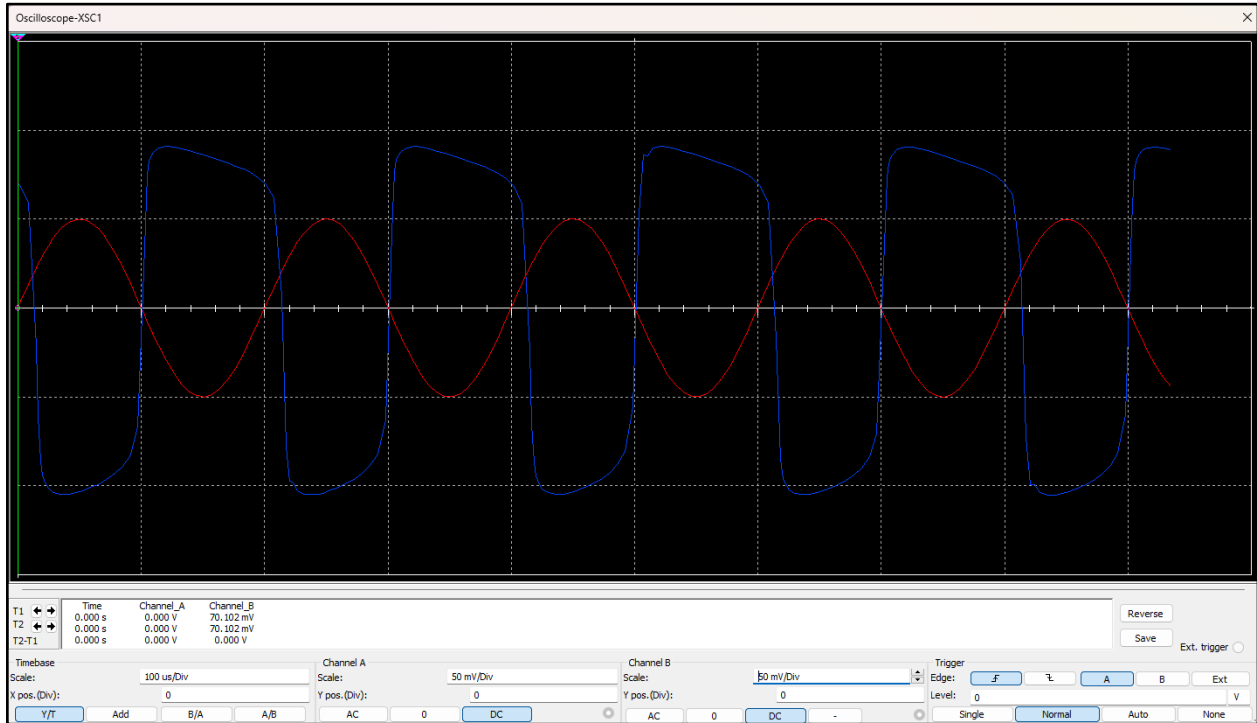
Figure 5.6.6.a - Vout vs Vin of Distortion Schematic

### 5.6.9 Phase 90 Schematic

Previously mentioned was that most pedal effects are cloned from an older model. This one being based on the MXR Phase 90. The primary sections are the power section, which consists of a 9V battery with a 9V and 5.1V rail, as well as a bias rail which is controlled by a potentiometer to change the bias on the transistors. The next section is an op amp which acts as an input buffer for the circuit. The next section is the sequence of op amps where the phase shifting takes place. Next is the low frequency oscillator section of the circuit which affects the bias of the transistors so it controls the phase shift. Finally the last stage is an emitter amplifier, which is able to blend the shifted and non-shifted signal to create the phase audio effect.
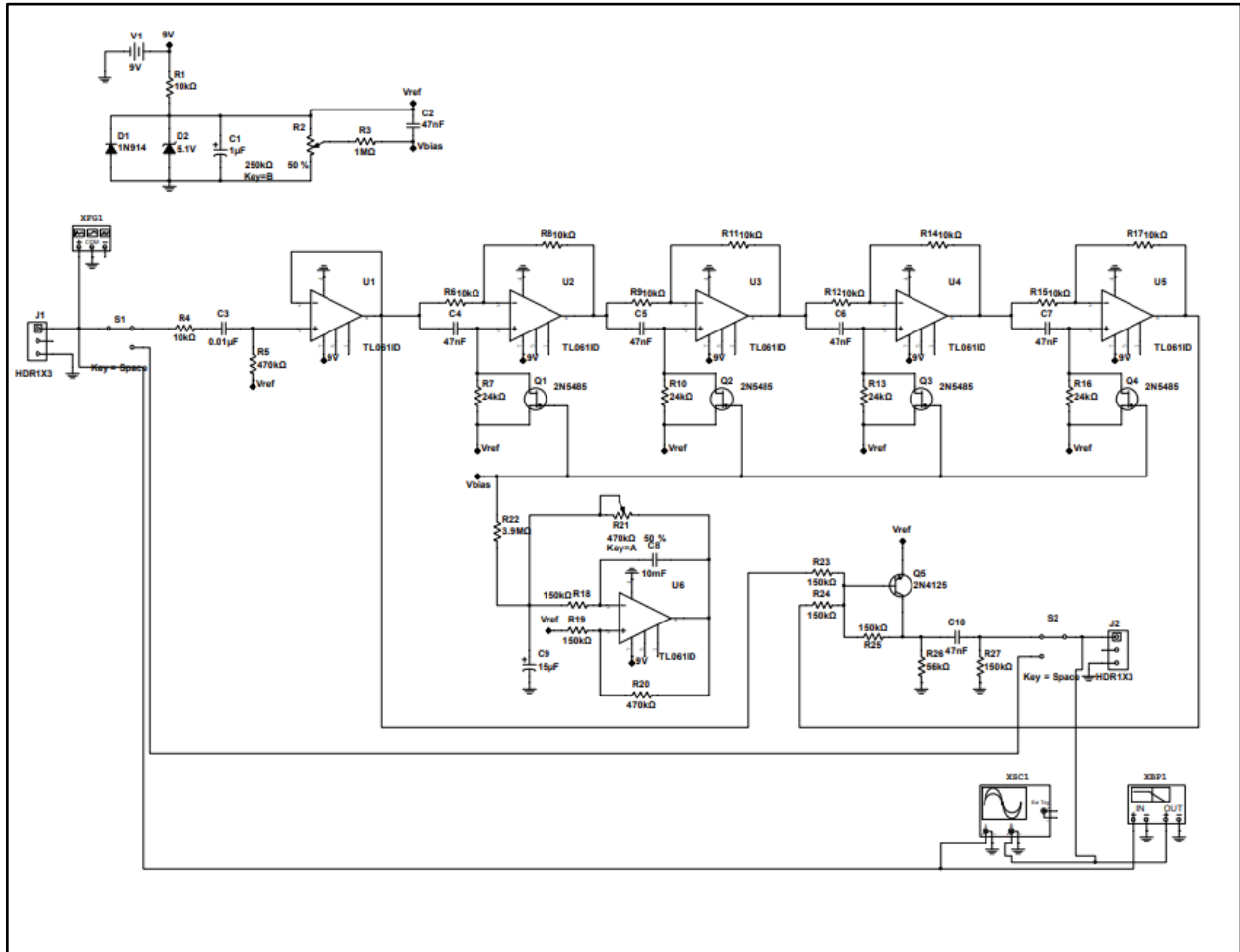
Figure 5.6.9.a - Phase 90 Schematic

| Qty | Parts | Value | Description |
|---|---|---|---|
| 10 | R1, R4, R6, R8, R9, R11, R12, R14, R15, R17 | 10k | RESISTOR |
| 4 | R7, R10, R13, R16 | 24k | RESISTOR |
| 1 | R26 | 56k | RESISTOR |
| 6 | R18, R19, R23, R24, R25, R27 | 150k | RESISTOR |
| 2 | R5, R20 | 470k | RESISTOR |
| 1 | R3 | 1M | RESISTOR |
| 1 | R22 | 3.9M | RESISTOR |
| 1 | R2 | 250k | POTENTIOMETER |

| | | | |
|---|---|---|---|
| 1 | R21 | 470k | POTENTIOMETER |
| 1 | C1 | 1u | ELECTROLYTIC CAPACITOR |
| 1 | C9 | 15u | ELECTROLYTIC CAPACITOR |
| 1 | C3 | 0.01u | CAPACITOR |
| 6 | C2, C4, C5, C6, C7, C10 | 47n | CAPACITOR |
| 1 | C8 | 10m | CAPACITOR |
| 1 | D1 | 1N914 | DIODE |
| 1 | D2 | 5.1V | ZENER DIODE |
| 6 | U1, U2, U3, U4, U5, U6 | TL061ID | OP AMP |
| 4 | Q1, Q2, Q3, Q4 | 2N5485 | N-CHANNEL JFET TRANSISTOR |
| 1 | Q5 | 2N4125 | BJT PNP TRANSISTOR |
| 2 | J1, J2 | HDR1X3 | 3 PIN HEADER |
| 2 | S1, S2 | SPDT | SWITCH |

Table 5.6.7.b - BOM Phase 90 Schematic

## 5.6.11 Phase 90 Performance

The design was able to work as expected, as can be seen in Figure 5.6.8.a, the magnitude response of the signal has two notches in it, caused by having 4 stages shifting the signal in the core of the circuit. These notches can be adjusted using the potentiometers in the circuit. The oscilloscope output of the signal shows the 90 degree phase shift of the output signal, which shows the pedal is operating as it is expected to.
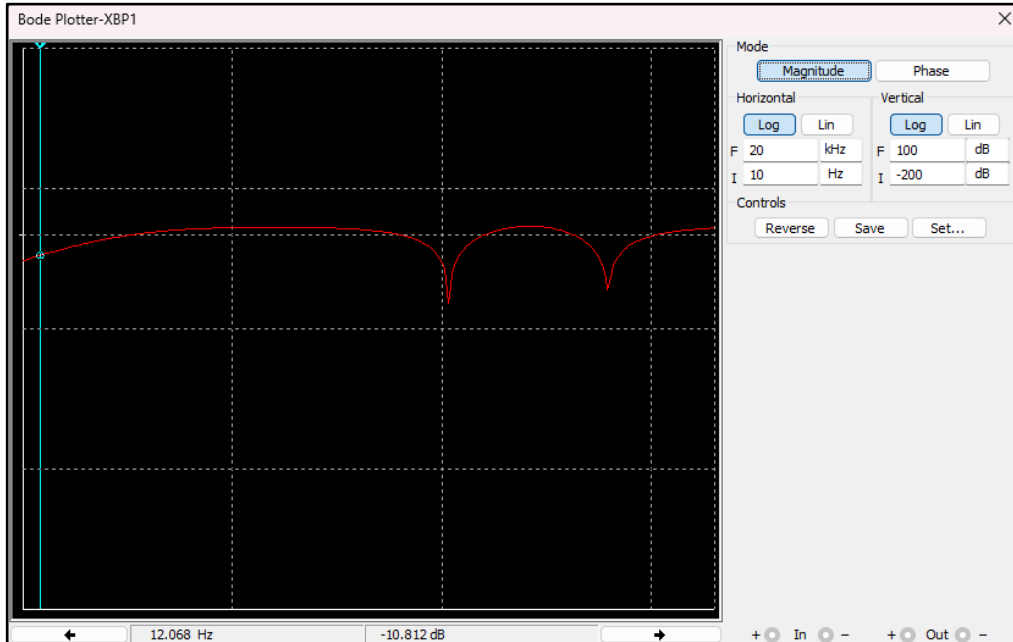
Figure 5.6.8.a - Magnitude Response of Phase 90
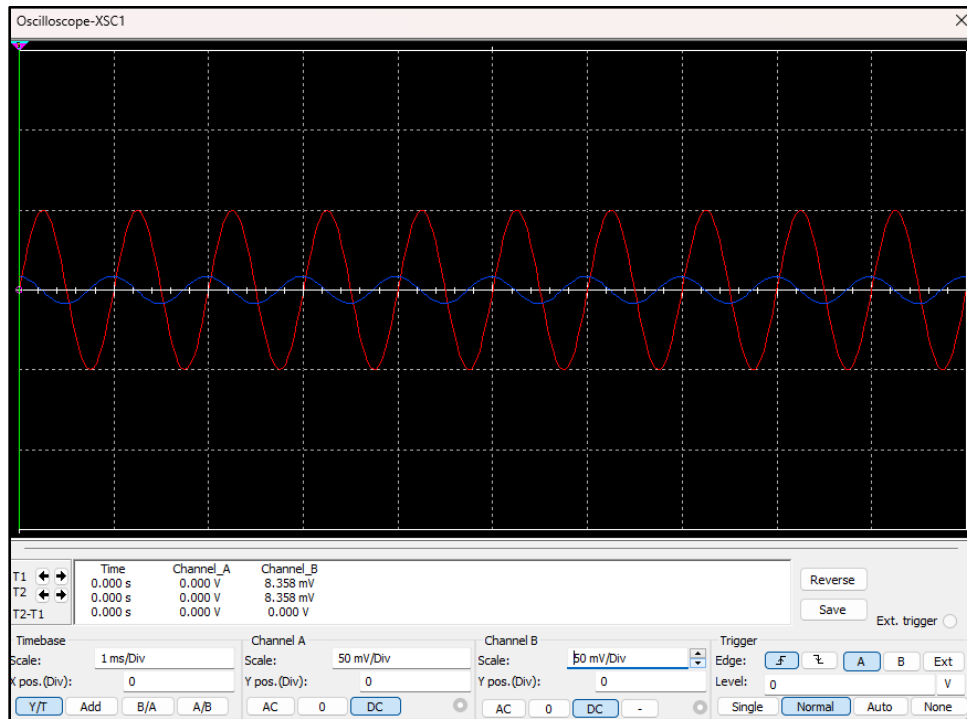


Figure 5.6.8.b - Vout vs Vin Phase 90

## 5.7 Digital Effects

### 5.7.1 General Digital Effects Design Information

The digital circuits will be designed using MATLAB in order to produce the proper waveforms that would be expected. After verification that the design of the circuit produces the desired results, the corresponding C code will need to be implemented to actualize the desired digital effects. The C code may not be what is actually implemented, but rather just pseudocode since writing proper code would require knowledge of the specific hardware to make proper references to the addresses and registers of the hardware. Another additional piece of information is that a generally good sampling frequency for audio recordings is 44.1kHz.

A sample bassline arpeggiation is loaded into MATLAB for testing purposes. For all effects, the audio will be loaded using the audioread() function. This function allows the user to read and load an audio file into MATLAB. In doing so, it places samples of the audio file into a Nx1 matrix.
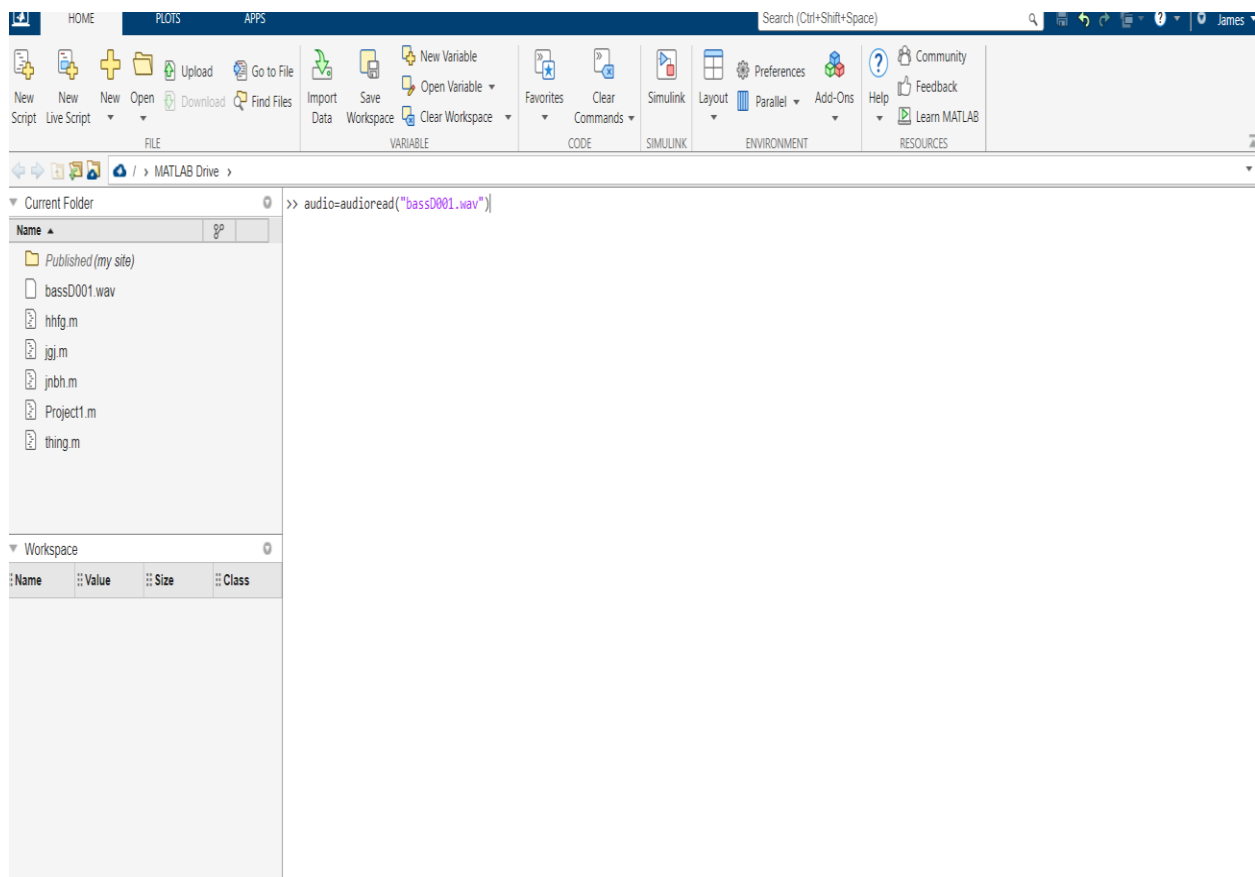


Figure 5.7.1.a - Command for audio import into MATLAB
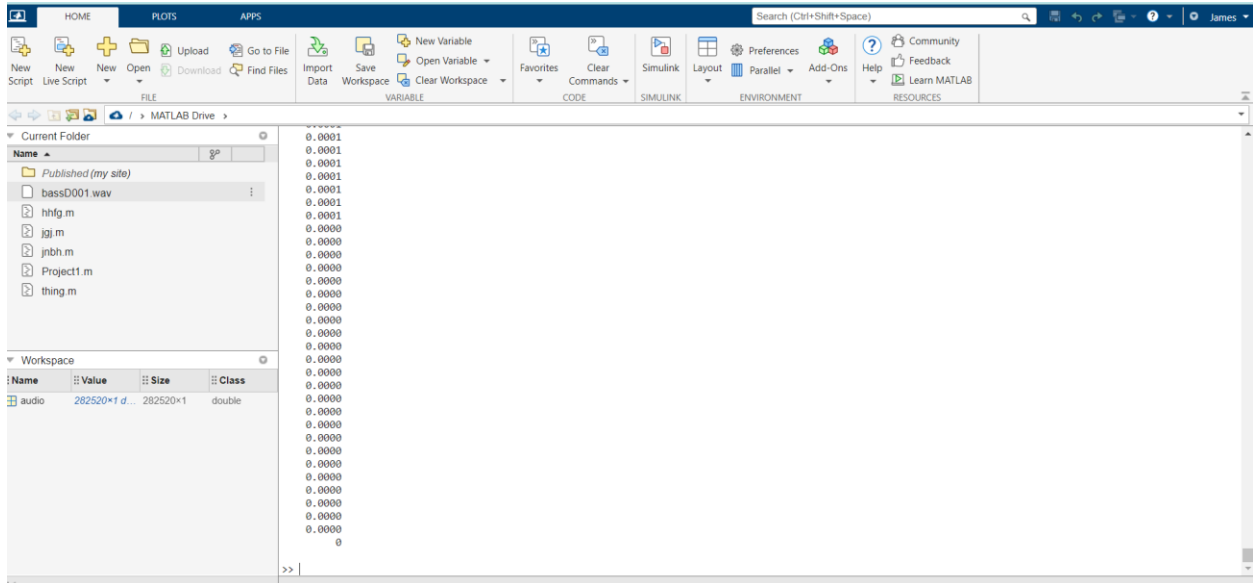
Figure 5.7.1.b - Audio file sampled and placed into matrix

Alternatively to the top method, a script can be written to load the audio file. The audioread() function discretizes the '.wav' file and provides the sampling rate, both denoted by 'x' and 'Fs' respectively. This method is more convenient than using commands in the command window constantly. These will act as the parameters for the functions that will be established in the .m files for each individual effect. These are simple designs, but the functions inside individual effect files can be parameterized further to give more customizability to the effects. In order to fundamentally design the effects, certain parameters will be kept constant.
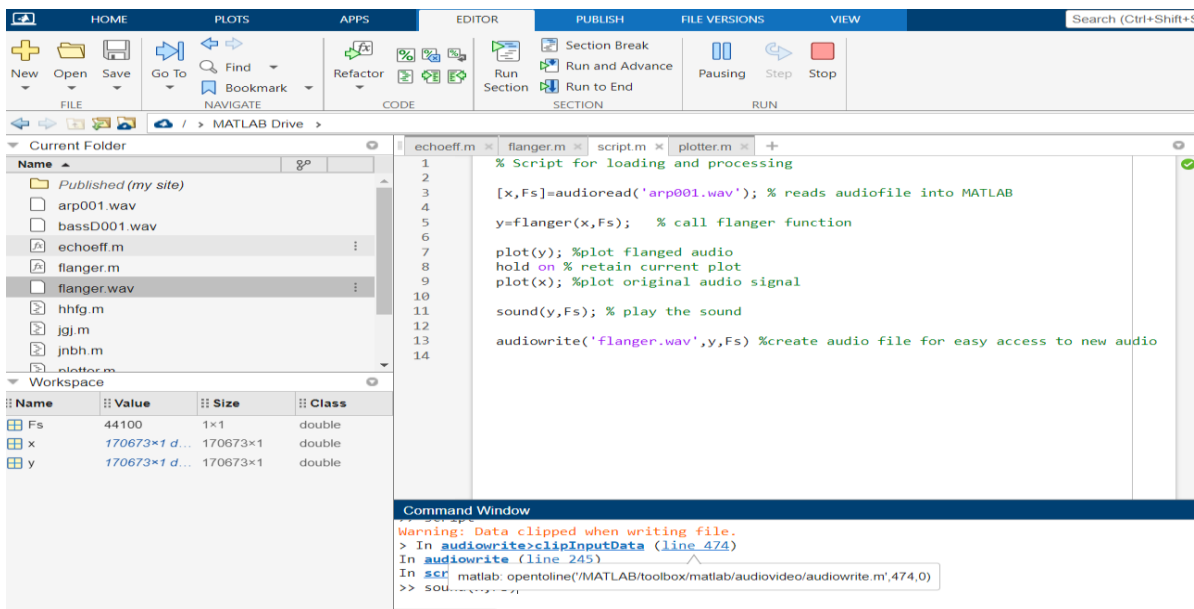


Figure 5.7.1.c - Load audio script

The plotter is used for verification purposes. Using an actual instrument produces waveforms that are not ideal, and thus in seeing it makes it difficult to see if an effect is working correctly on a mathematical level. The plotter.m file has a sampling frequency of 44.1kHz, two sine waves of arbitrary frequency, and then a third signal of the two previous signals mixed together. The signals are passed through the effect file in question and produce waveforms that should indicate the correctness of the effect.

```matlab
clear;             %clear all previous variables to begin test
close all;         %close all open figures
Fs = 44100;        %sampling frequency
Fw1 = 500;         %sine1 frequency
Fw2 = 5000;     %sine2 frequency
N = 1024;            %number of samples

for i = 1:N
    swave1(i) = sin(2*pi*(i-1)*Fw1/Fs);
    swave2(i) = sin(2*pi*(i-1)*Fw2/Fs);
    swave3(i) = (sin(2*pi*(i-1)*Fw1/Fs) + sin(2*pi*(i-1)*Fw2/Fs));
end


figure;
subplot(3,1,1);
plot(flanger(swave1(1:N),Fs));
title('Original Sines 500 Hz, 5K Hz, Flanged');
subplot(3,1,2);
plot(flanger(swave2(1:N), Fs));
subplot(3,1,3);
plot(flanger(swave3(1:N), Fs));
```

Figure 5.7.1.d - Plotter

## 5.7.2 Comb Filter

## 5.7.3 Flange

Flange requires a delay of anywhere from 0-15ms and modulated with a sine of depth 0-3ms and a frequency of 0-1Hz. A sine wave delay of 3ms at 0.5Hz is chosen. There are different ways of implementing the flanging effect, but this code is modeled off of the Dattaro standard effect structure. In this, it is recommended to have gains of 0.7 for the blend, forward feed, and feedback. This was problematic in my modeling of the flanging effect as it would cause clipping, and thus a heavy distortion. A lower gain of 0.5 was arbitrarily chosen then.

The continuous delay needs to be converted into a corresponding discrete amount. This is done by multiplying the continuous time delay by the sampling frequency. Multiplying these two values together gives a value of 132.3 samples. Using the MATLAB

82

round function, it rounds it to 132. Next an empty output array is created for space allocation. The for loop is responsible for assigning the modulated values into the output array. It does this through a range of the sample delay plus one, and the length of the audio sample. In digital signal processing, sinusoidal functions are characterized by pi, frequency of the signal divided by the sampling frequency, and the index of time. In the lfowave variable, it is seen that sine is a function of the iterative variable 'i'. The absolute value is taken of that and then multiplied by the total delay samples. This creates a varying sine wave from 0-3ms. The final equation is the summation of the current signal with the 0.5 gain and the input sample minus the amount of delay samples. This produces a flanging effect. A feedback statement was tried and implemented at first, but made no noticeable difference in the audio qualities and characteristics. Furthermore,

```
plotter.m ×   script.m ×   flanger.m ×   +
1    function [y] = flanger(x,Fs)
2        delay=0.003; % sine wave delay
3        lfofreq=.5; %frequency of 1Hz sine wave for modulation
4        gain=0.5; % gain for forward feed and blend
5
6        sampdel=(delay*Fs);
7        sampledelay=(round(sampdel)); % number of samples neeed for delay
8
9        y=zeros(length(x),1); % empty output array
0        y(1:sampledelay)=x(1:sampledelay);
1        for i= (sampledelay+1):length(x) % modulate for the number of samples needed for delay
2            lfowave=(sin(2*pi*i*(lfofreq/Fs))); % create array for sine wave modulation
3            abs_sin=abs(lfowave);
4            sin_delay=ceil(abs_sin*sampledelay); %% creates delay
5
6            y(i)=gain*x(i)+gain*(x(i-sin_delay)); %% feedforward and blend
7            %x(i)=x(i)+gain*(x(i-sin_delay)); %% create feedback
8        end
9    end
0
1
```

Figure 5.7.2.a - Flanger Code

```
plotter.m ×   script.m ×   flanger.m ×   +                        Figure 2 ×   +
1        % Script for loading and processing
2
3        [x,Fs]=audioread('bassEm009.wav'); % reads audiofile into MATLAB
4
5        y=flanger(x,Fs);    % call flanger function
6
7        figure;
8        subplot(3,1,1);
9        plot(y,'blue'); %plot flanged audio
10       subplot(3,1,2);
11       plot(x,'green'); %plot original audio signal
12       subplot(3,1,3);
13       plot(y,'blue');
14       hold on;
15       plot(x,'green');
16
17       sound(x,Fs); % play original audio
18       pause(5);   % wait 5 seconds
19       sound(y,Fs); % play the sound
20
21       audiowrite('flanger.wav',y,Fs) %create audio file for easy access to new aud
22
```

Command Window
ew to MATLAB? See resources for Getting Started.
> script

83

The blue waveform at the top represents the flanged audio, and the green waveform below it represents the original audio sample. As seen, the blue waveform lags slightly compared to the green one. The third figure is an overlay of the two previous waveforms simply to demonstrate that there is a delay between the new audio and the original audio.
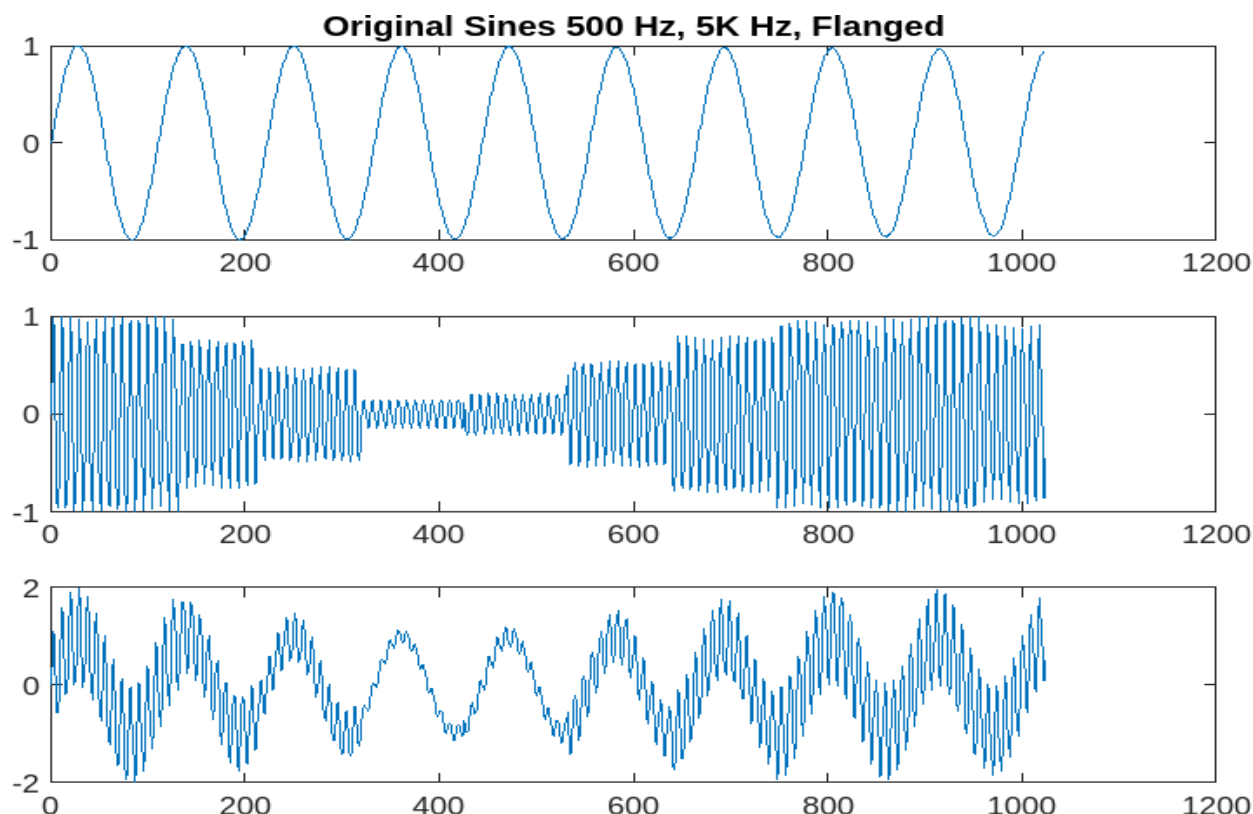


Figure 5.7.2.c- Flanger ideal waveforms

The second waveform and third waveform is what is expected to be seen given a sine wave input. As seen, since flanger is a phase modulated effect.

### 5.7.4 Echo

The echo sound effect can be realized by the following difference equation

$$y(n) = x(n) + \alpha y(n - N)$$

Equation 5.7.3.a - Echo Difference Equation

The code for echo functions similarly to that of flanger, being that they are both time delay based effects. Echo, unlike reverb, does not require any wave modulation. Its only requirement is a delay larger than 50ms. There is no real indication on what the delay

gain should be, smaller values give it a less aggressive echo while larger echoes really amplified the feeling of a large empty room. Echo requires more samples to be held since a greater delay is to be created. Using the same formula as flanger, the amount of samples needed to create the proper delay is the sampling frequency (44.1kHz) multiplied by the delay. This yields a rounded amount of samples needed to 3528. The same bounds for the for loop are used. The difference equation is then implemented in the for loop. This effect still needs to be modified, as the echo is not as prominent as it should be. Other sources say to apply a gain of less than one to the input sample, this may play a factor in creating the desired echo effect.

```
echoeff.m ×    plotter.m ×    script.m ×    flanger.m ×    +

 1        %%%%%%%%echo
 2 □    function [y]=echoeff(x,Fs)
 3            delay=0.07;
 4            gain=0.8; % gain for forward feed and blend
 5
 6            sampdel=(delay*Fs);
 7            sampledelay=(round(sampdel)); % number of samples neeed for delay
 8
 9            y=zeros(length(x),1); % empty output array
10            y(1:sampledelay)=x(1:sampledelay);
11 □        for i= (sampledelay+1):length(x) % modulate for the number of samples needed for delay
12
13                y(i)=x(i)+(gain*(y(i-sampledelay))); %% feedforward and blend
14                %x(i)=x(i)+gain*(x(i-sin_delay)); %% create feedback
15            end
16      end
17
```
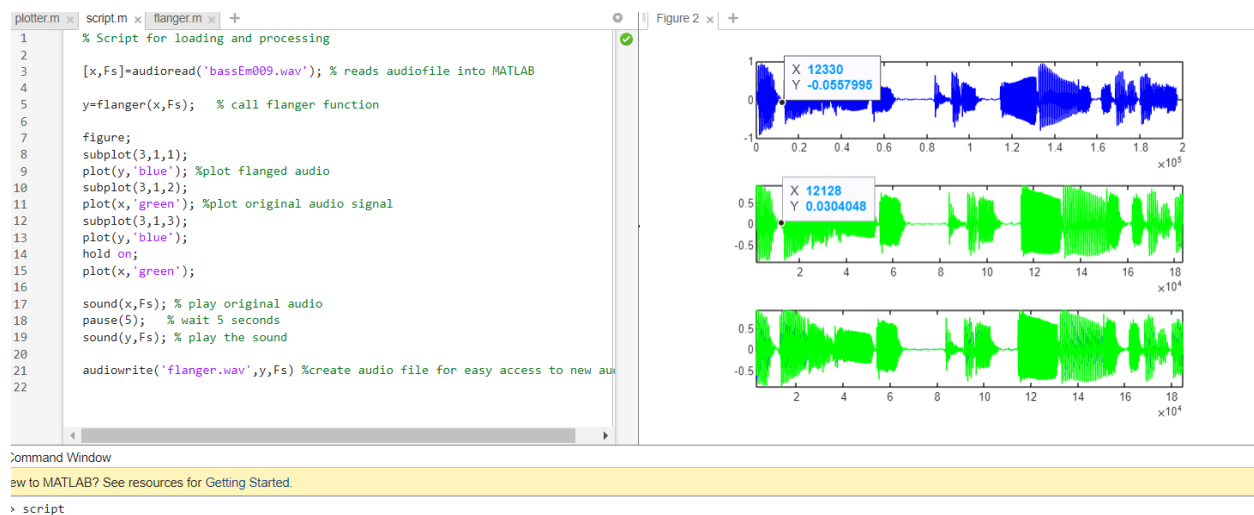
Figure 5.7.3.b - Echo Code
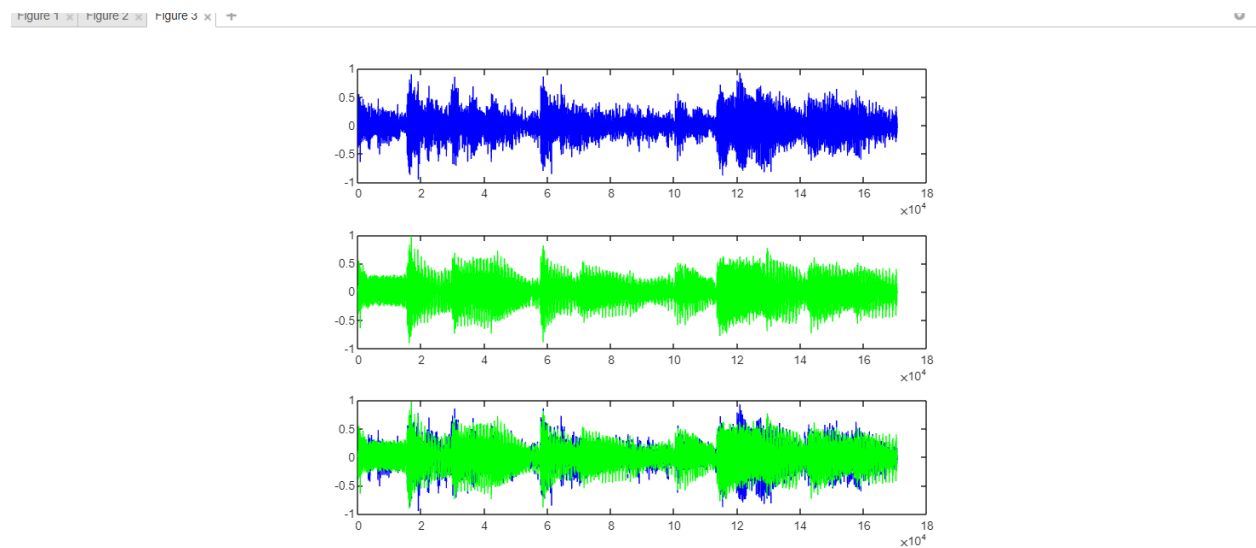
Figure 5.7.3.c - Echo Effect Waveform

The waveforms shown here are from sending the bassline audio sample through the echo effect. The blue waveform is the echo audio sample and the green waveform is the original input audio sample. When overlaid on each other as seen in the bottom plot, the green waveform is slightly ahead of the blue waveform. This makes sense since the echo effect is supposed to be delaying the output.
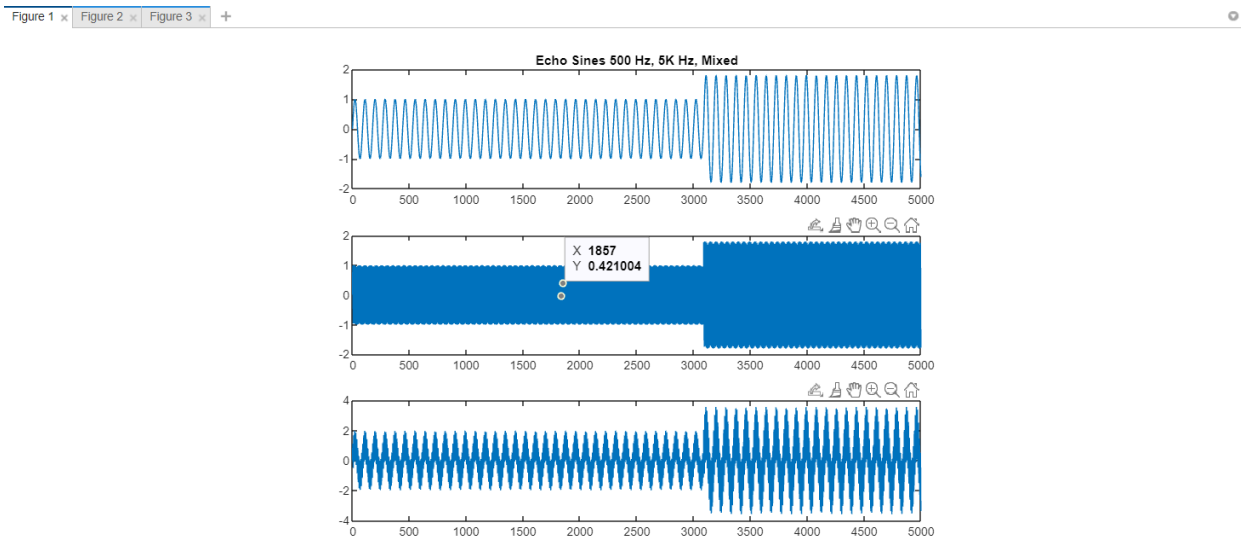


Figure 5.7.3.d - Echo Waveforms with Sine Waves

## 5.7.5 Reverb

It is particularly useful to implement convolution integral when the impulse response of the room is known to give a more authentic reverberation. This will be the case in implementing hall and room reverb. Sample .wav files of various concert halls and rooms can be used and convolved with the sample audio input signal to create true sounding reverberation.

The script for the convolution reverb is slightly altered. It reads the impulse response along with the sample bass input. Convolution is not reliant on the sampling frequency, thus, omitted from the calculations.

```
1       % Script for loading and processing
2
3       [x,Fs]=audioread('free002.wav'); % reads audiofile into MATLAB
4       [h,Fh]=audioread("1a_marble_hall.wav"); % read impulse response
5
6   ☐   %y=echoeff(x,Fs); % call effect function
7   └   %y=flanger(x,Fs);
8       y=reverb(x,h);% use this to call reverb, sampling frequency not needed
9       %convolution
10  ☐   %{
11      figure;
12      subplot(3,1,1);
13      plot(y,'blue'); %plot flanged audio
14      subplot(3,1,2);
15      plot(x,'green'); %plot original audio signal
16      subplot(3,1,3);
17      plot(y,'blue');
18      hold on;
19      plot(x,'green');
20      %}
21      % plots for reverb
22      figure;
23      subplot(4,1,1);
24      plot(y,'blue'); %plot flanged audio
25      subplot(4,1,2);
26      plot(x,'green'); %plot original audio signal
27      subplot(4,1,3);
28      plot(y,'blue');
29      hold on;
30      plot(x,'green');
31      subplot(4,1,4);
32      plot(h,'red')
33
34
```

Figure 5.7.4.a - Convolution Reverb Script

The blue shows the input signal, green is the output signal, and red is the concert hall impulse response. Looking at the output, it seems like the reverberation is working, as the sound signals are drawn out due to the mimicked waves bouncing back. The drawn out length of the output waveform shows that a large hall was simulated, since it is expected that some sort of auditorium would have longer decay times. When the input and output are overlaid, it can be seen that the decay of the output signal is exponentially decaying. This is expected due to the decaying nature of the impulse response.
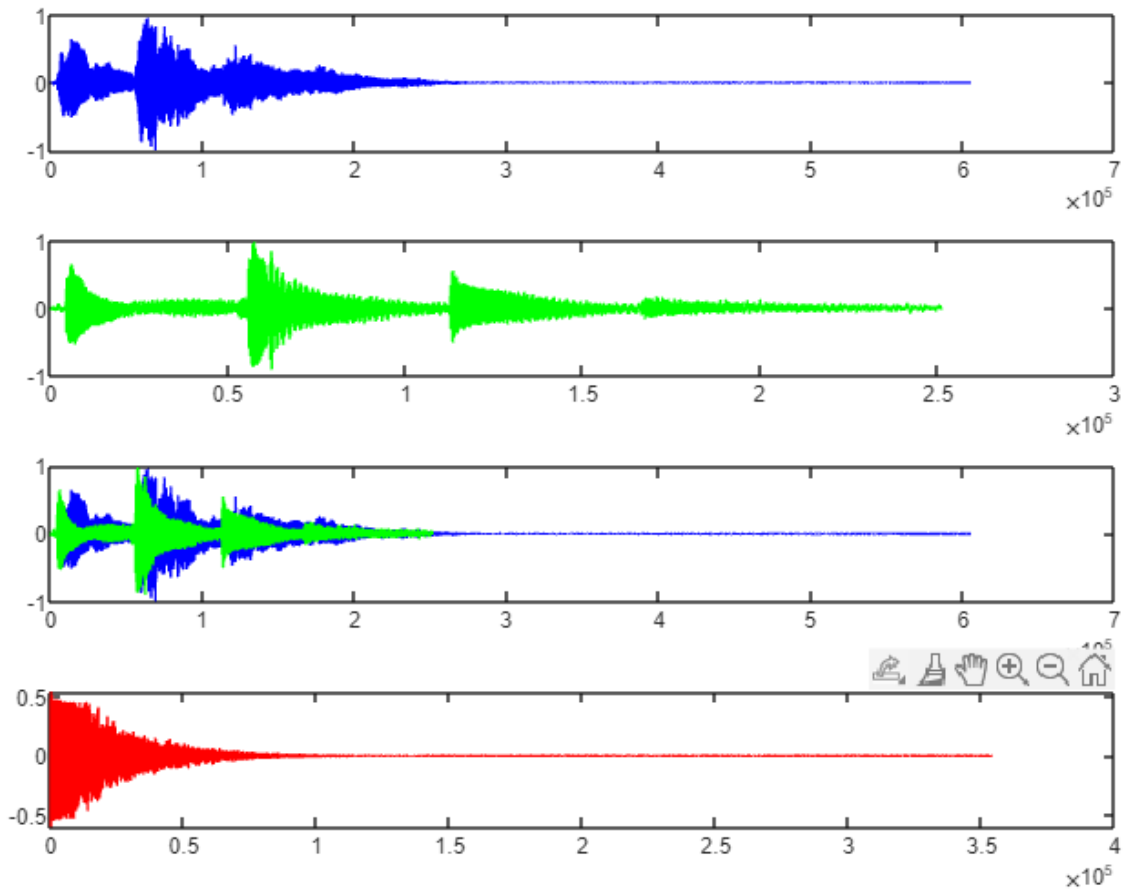
Figure 5.7.4.b - Hall Reverb Waveforms

The code for convolution reverberation is rather simple. The output matrix will need a length of the input and impulse combined. The length of the fourier transformed signals will be determined by Length2. It is desirable to have the signal matrices as power of two since it makes computation times faster. Although this is not entirely necessary for testing purposes, it is still good to implement. The transformed input and impulse signals are then multiplied together, since the convolution of two waves in the time domain is equivalent to the multiplication of them in the frequency domain. The inverse fourier transform is taken of the convolved wave, and the real part is extracted using the real() function in MATLAB. The normalized output wave is then given by dividing by the largest convolved value of the real part.

```
function[y]= reverb(x,h)
%%%%hall reverb
Length=length(x)+length(h)-1; %
Length2=pow2(nextpow2(Length)); % power 2 makes fft easier computationally
X=fft(x, Length2); % fft of input
H=fft(h, Length2); % fft of impulse response
Y=X.*H; % convolve
y=real(ifft(Y, Length2));
y=y(1:1:Length);
y=y/max(abs(y));

end
```

Figure 5.7.4.c - Convolution Reverberation Code

The same procedure can be used for the room reverb. In this instance, the impulse response of a classroom is used. The same script is used to read the audio files. This impulse response was captured in the middle of the classroom, as will be denoted by the filename.

```
[x,Fs]=audioread('free002.wav'); % reads audiofile into MATLAB
[h,Fh]=audioread("30x20y.wav"); % read impulse response
```

Figure 5.7.4.d - Reading Room Impulse Response

Seeing the waveforms for the room reverb, there are clear differences when compared with the hall reverb. The room impulse response is much shorter than that of the hall. This will be indicative that the reverberation time in a room is smaller than that of a large hall or auditorium.
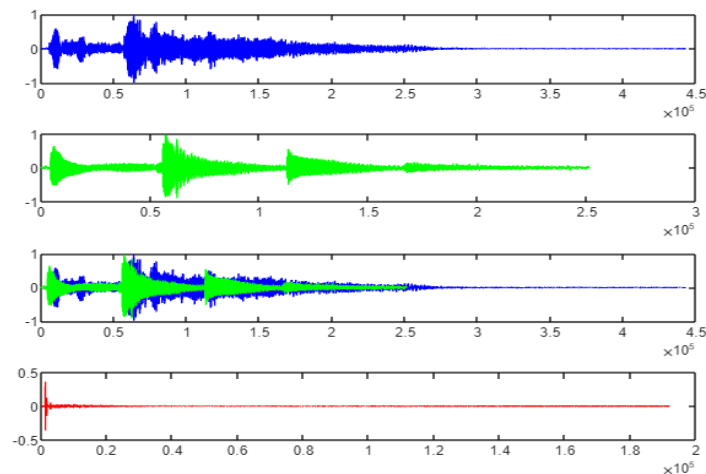
Figure 5.7.4.e - Room Reverb Waveforms

To further illustrate the concept of room reverbs, a second impulse response was used, this time of a smaller room. The expected outcome would be even shorter reverberation times than when the classroom impulse response was used. When this occurred, a much more natural sound was produced than the other two reverberations.

```
% Script for loading and processing

[x,Fs]=audioread('free002.wav'); % reads audiofile into MATLAB
[h,Fh]=audioread("impulse_room.wav"); % read impulse response

%y=echoeff(x,Fs); % call effect function
%y=flanger(x,Fs);
y=reverb(x,h);% use this to call reverb, sampling frequency not neede
%convolution
%y=wah(x,Fs);
%{
figure;
subplot(3,1,1);
plot(y,'blue'); %plot flanged audio
subplot(3,1,2);
plot(x,'green'); %plot original audio signal
subplot(3,1,3);
plot(y,'blue');
hold on;
plot(x,'green');
%}
% plots for reverb
figure;
subplot(4,1,1);
plot(y,'blue'); %plot flanged audio
subplot(4,1,2);
plot(x,'green'); %plot original audio signal
subplot(4,1,3);
plot(y,'blue');
hold on;
plot(x,'green');
subplot(4,1,4);
plot(h,'red')
```

Figure 5.7.4.f - Script for Reading and Plotting Room Reverb
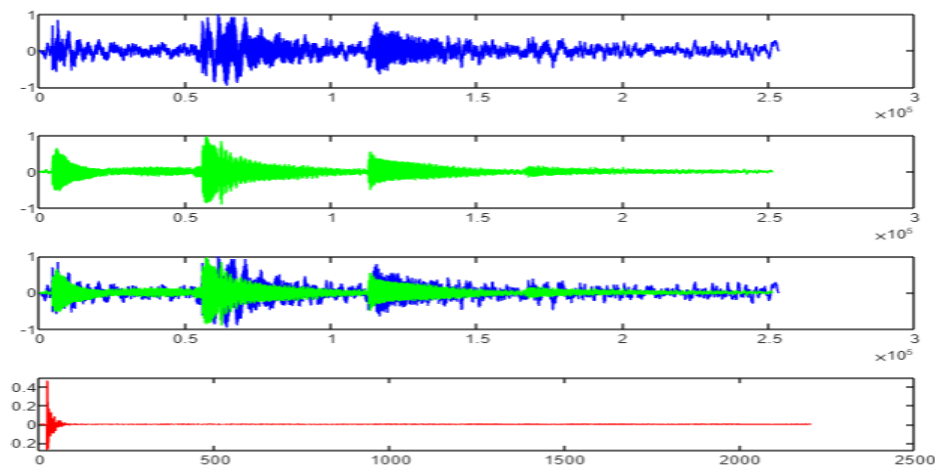


Figure 5.7.4.g - Room Reverb Waveforms

90

Impulse responses from the EMT-140 will be used to design the plate reverb. The EMT-140 impulse response will be convolved with the input signal.

```matlab
[x,Fs]=audioread('free002.wav'); % reads audiofile into MATLAB
[h,Fh]=audioread("emt_140_medium_2.wav"); % read impulse response
```

Figure 5.7.4.h - Reading EMT-140 IR

Looking at the waveforms for the plate reverb, it seems like the impulse response captured through a metal sheet is similar to that of the hall.
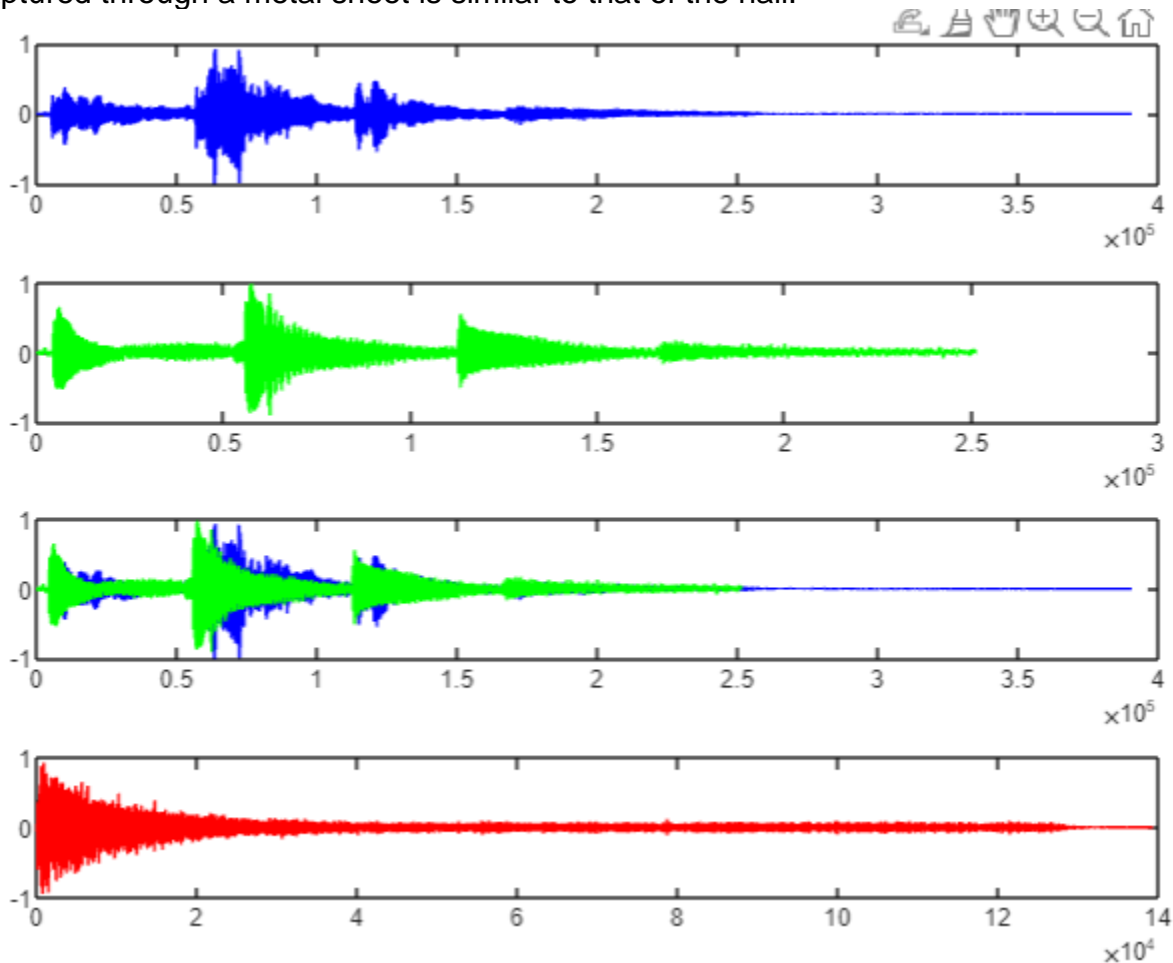


Figure 5.7.4.i - Plate Reverb Waveform

It just has slightly quicker reverberations. Among trying with different audio samples, it could be heard that there was a slight vibrating metallic feel to the sound.

Another reverb is spring reverb. The spring reverb's impulse response is recorded in a similar fashion to that of the plate reverb, with the difference being the medium being a spring.

```
[x,Fs]=audioread('free002.wav'); % reads audiofile into MATLAB
[h,Fh]=audioread("IR_Roland_RE201_spring_high-drive.wav"); % read impulse
```

Figure 5.7.4.j - Reading Spring Impulse Response

As one may expect, the resulting sound is springs, as in a jumpy kind of sound. This is to be expected when the impulse response of the spring is seen.
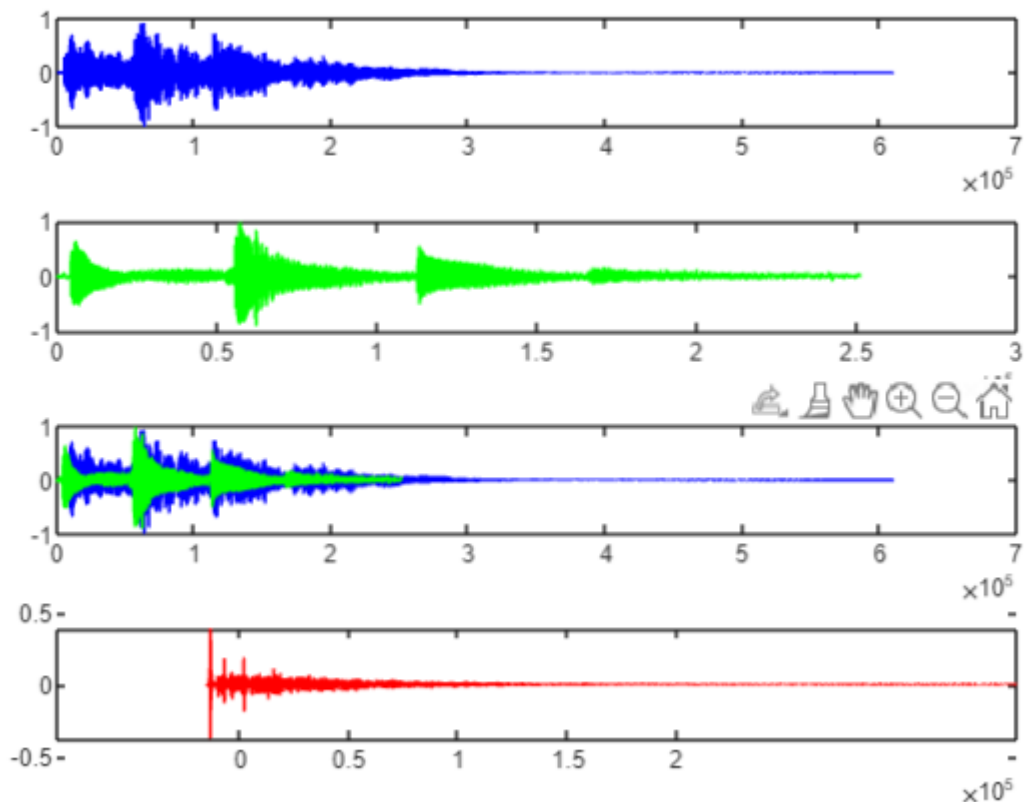


Figure 5.7.4.k - Spring Reverb Waveforms

## 5.7.6 Wah

Implementing the wah-wah filter requires the use of a bandpass filter. This can be implemented by using both a high pass filter and low pass filter together. The bandpass filter should have a narrow passband, and a common damping factor to use is 0.05. In order to make the wah-wah effect, the bandpass needs to sweep across the frequency range. This is accomplished by modulating the input signal with a triangle wave, which allows for very fast change in center frequency.

The difference equations for a low pass filter and high pass filter are needed, and the outputs of each can be summed together to mimic a bandpass filter. A common configuration to achieve this is through a state variable filter. The state variable can be represented by the following diagram. Q represents the Q factor of the filter. Q is calculated by the damping factor times two. F is the triangle wave to be modulated with

the dry signal. The difference equations for each stage of the state variable filter can be derived.
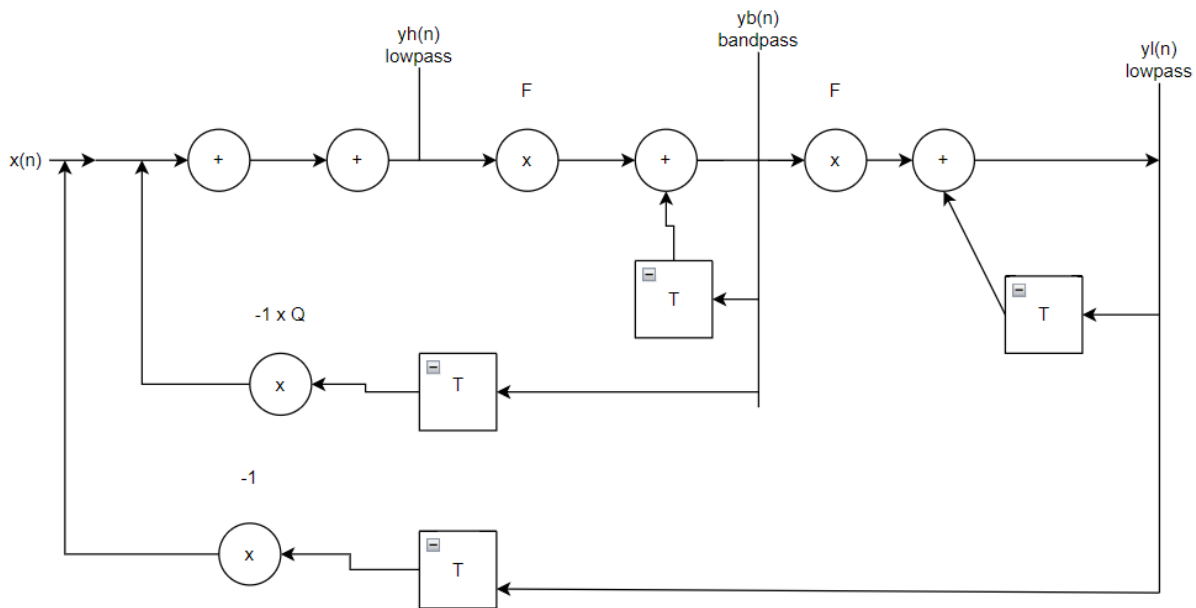


Figure 5.7.5.a - State Variable Filter Block Diagram

$$yh(n) = x(n) - yl(n-1) - Q1 * yb(n-1);$$

Equation 5.7.5.b - High Pass Filter Difference Equation

$$yb(n) = F1 * yh(n) + yb(n-1);$$

Equation 5.7.5.c - Bandpass Filter Difference Equation

$$yl(n) = F1 * yb(n) + yl(n-1);$$

Equation 5.7.5.d - Low Pass Filter Difference Equation

```matlab
function[y]=wah(x,Fs)
% audio = audioread("free002.wav");

z = 0.05; % rec damping factor
minf=250; %min frequency acepted
maxf=5000; %max frequency passed
Fw = 1000; % change in center frequency every sample
delta = Fw/Fs;
%create triangle wave
Fc=minf:delta:maxf;
while(length(Fc) < length(x) )
Fc= [ Fc (maxf:-delta:minf) ];
Fc= [ Fc (minf:delta:maxf) ];
end
%match the length of input and triangle wave
Fc = Fc(1:length(x));
% create oscillating triangle waave
F = 2*sin((pi*Fc(1))/Fs);

Q = (2*z); % calculation for q factor
yh=zeros(size(x)); % create emptly out vectors
yb=zeros(size(x));
yl=zeros(size(x));
```

Figure 5.7.5.e - Wah Code Part 1

```matlab
F = 2*sin((pi*Fc(1))/Fs);

Q = (2*z); % calculation for q factor
yh=zeros(size(x)); % create emptly out vectors
yb=zeros(size(x));
yl=zeros(size(x));

yh(1) = x(1);
yb(1) = F*yh(1);
yl(1) = F*yb(1);

for n=2:length(x)
yh(n) = x(n) - yl(n-1) - Q*yb(n-1);
yb(n) = F*yh(n) + yb(n-1);
yl(n) = F*yb(n) + yl(n-1);
F = 2*sin((pi*Fc(n))/Fs);
end
%normalize output
maxyb = max(abs(yb));
yb = yb/maxyb;
y=yb;
%sound(y,44100);
end
```

Figure 5.7.5.f - Wah Code Part 2

In the code, basic parameters are initialized. The 0.05 damping factor is a recommended value. For the minimum and maximum frequencies for the modulated

signal. These can be modified to allow for a wide or narrow range of modulation. The triangle wave is created by initializing an array whose bounds are the selected frequencies, incrementing by the delta variable. The delta variable is the change in frequency per sample.

A while loop is then implemented to create a full array for the frequency varying triangle wave. Rather than changing the frequency of the triangle waveform itself here, the frequencies are represented by the amplitude of the triangle wave.

The oscillating triangle wave is then created and represented using the variable 'F' , where it is represented by the digital equation of a sine wave. Where the frequency is the changing triangle wave frequency divided by the sampling frequency.

The difference equations are then implemented and the output is normalized.

## 5.8 Interface

The primary goal of our interface will be an intuitive experience, which anybody with some knowledge of instrumental effects should be able to quickly grasp and begin using, rather than reading a manual first. This will be accomplished by clearly labeling all hardware interfaces, as well as developing a simple GUI for the touchscreen with self evident controls.

### 5.8.1 Analog Effects

Our analog effects will be controlled by potentiometers, covered by knobs to interface with the user. This allows for a basic control scheme which, when properly labeled, will be intuitive to any user. The system will also receive input from pedals, which will allow the user to control the severity of an effect while playing. Potentiometers will also be used to control the basic inclusions of amp headers, including volume, gain, bass, mid, and treble.

### 5.8.2 Digital Effects

Our Digital Effects will be controlled by a touch screen, which will be connected to a microprocessor. The microprocessor will handle displaying to the screen, as well as registering any of the touch inputs to the screen. The GUI should offer the user the list of digital effects, with each effect having a toggle button, as well as a slider which controls the severity of the effect. It will also include a settings menu, which will allow the user to control basic features of the screen and GUI, such as brightness and text size. Figure 8 shows the interface, with each effect listed, a toggleable button, as well as a slider to control the severity.
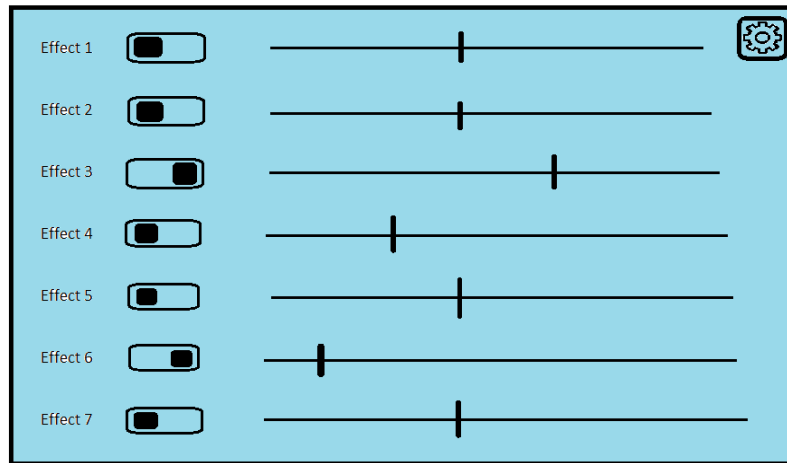
Figure 5.8.2.a - Touchscreen Interface

The microprocessor selected to accomplish this is the ESP32-WROOM-32 since it meets the requirements of running a touch screen, as well as having a small form factor and a low cost. The best touchscreen option is the MSP4022 as it includes the screen as well as a resistive overlay, and it comes pre-attached to a pcb with an integrated circuit to facilitate SPI communication with the screen and overlay. The connections that will be made are shown below in Figure 5.8.2.b.
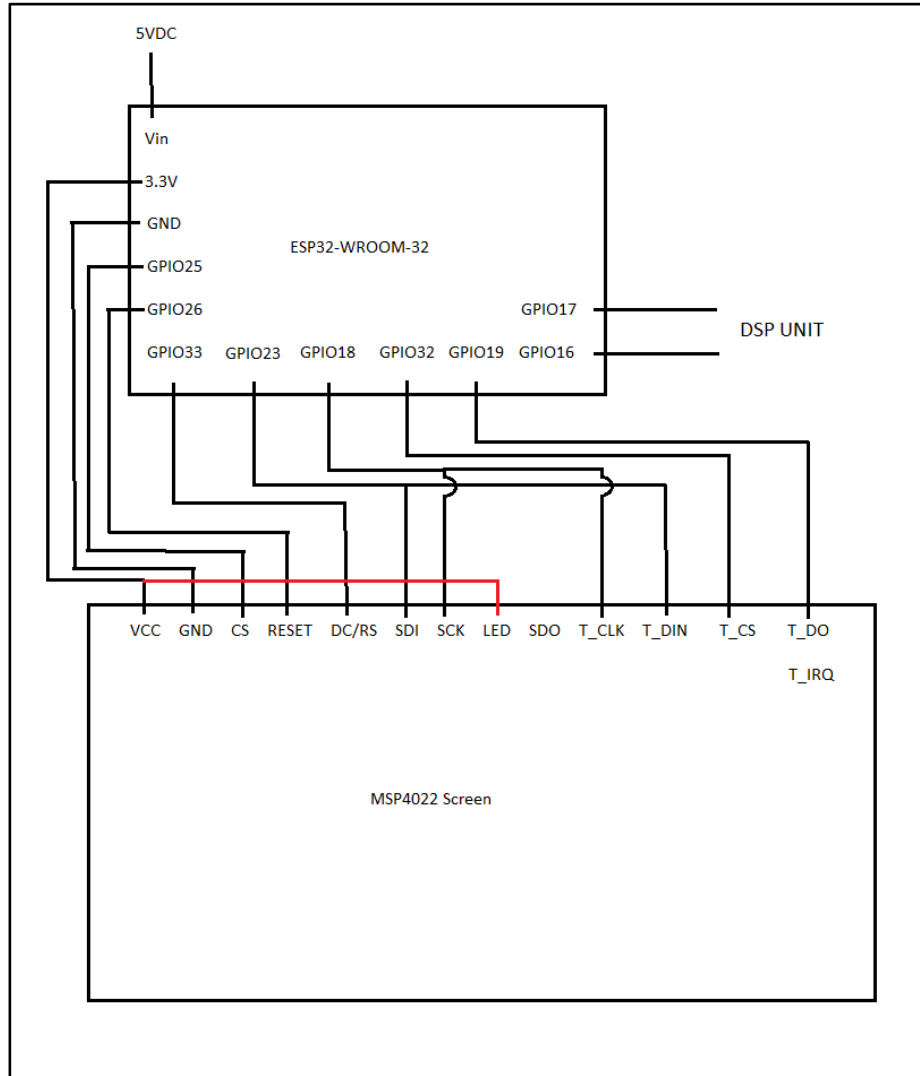
Figure 5.8.2.b - MCU-Screen Connections

These connections were made based off of the datasheet for the ESP32-devkit, as well as the datasheet for the MSP4022 screen. The microcontroller takes a 5V DC input for power, and the development kit has a voltage regulator which provides 3.3V output to power any accessory devices. This 3.3V pin provides the DC voltage to power the screen. The CS, RESET, and DC/RS pins are connected to arbitrary GPIO pins. The SDI pin is the serial data in for the LCD, which is connected to GPIO23, which is the MOSI pin for SPI communication on the ESP32. The SCK, or the SPI clock for the LCD is connected to GPIO18, or the SPI clock for the microcontroller. The LED pin on the screen controls the backlight, which can be set to the 3.3V signal to set the backlight on. The SDO pin is unused since no information is needed from the screen, data is only sent to it. The T_CLK pin is connected to GPIO18, since the touch controller and LCD are both set up as slaves to the microcontroller, it runs on the same SPI clock. The T_DIN pin is connected to GPIO23, since that is the MOSI pin for SPI communication on the microcontroller. The T_CS is the chip select control for the touch panel which is connected

to an arbitrary GPIO pin.  The T_DO pin is connected to GPIO19, which is the MISO pin for SPI communications on the microcontroller, so it receives the data from the touchscreen when presses are made.  The T_IRQ pin is an interrupt signal which does not need to be connected in this use case.  GPIO16 and GPIO17 are the UART transmission and receiving pins, which is how inputs the user makes, which are received by the microcontroller, will be sent to the DSP unit to control which digital effects are active as well as their strength.

The flowchart below shows the characteristics of the code which will be implementing the touchscreen.  The microcontroller will sit at the main menu waiting for an input to be detected.  Once there is an input, it will check whether one of the buttons to toggle a digital effect on or off was pressed, or if one of the sliders is being dragged. If one of the toggle buttons is pressed, it will check the current state of that effect, change the state, then send the new state to the DSP so an effect can be toggled on or off. If one of the sliders is being dragged, it will record the new value of the slider and check the current state of that effect. If the effect is currently on, the new value will be sent to the DSP so the strength of the effect can be changed in real time.  If the effect is toggled off, the DSP does not need to be updated, and the value will be sent next time the effect is toggled on.
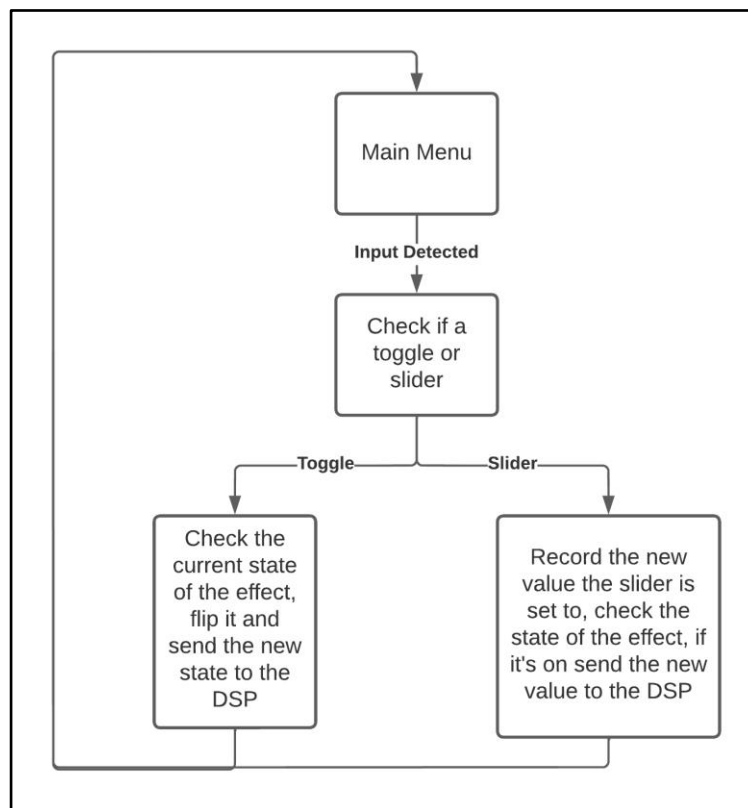


Figure 5.8.2.c - Flowchart of Touchscreen Logic

# 6.0 Prototyping

## 6.1 Pre-Amplifier/OL/HPF/EQ

The design of the Pre-Amplifier, Overload Detector, High Pass Filter, and the Graphic Equalizer was put into one pcb to save on cost. The drawback of this method is the added complexity of the pcb itself.
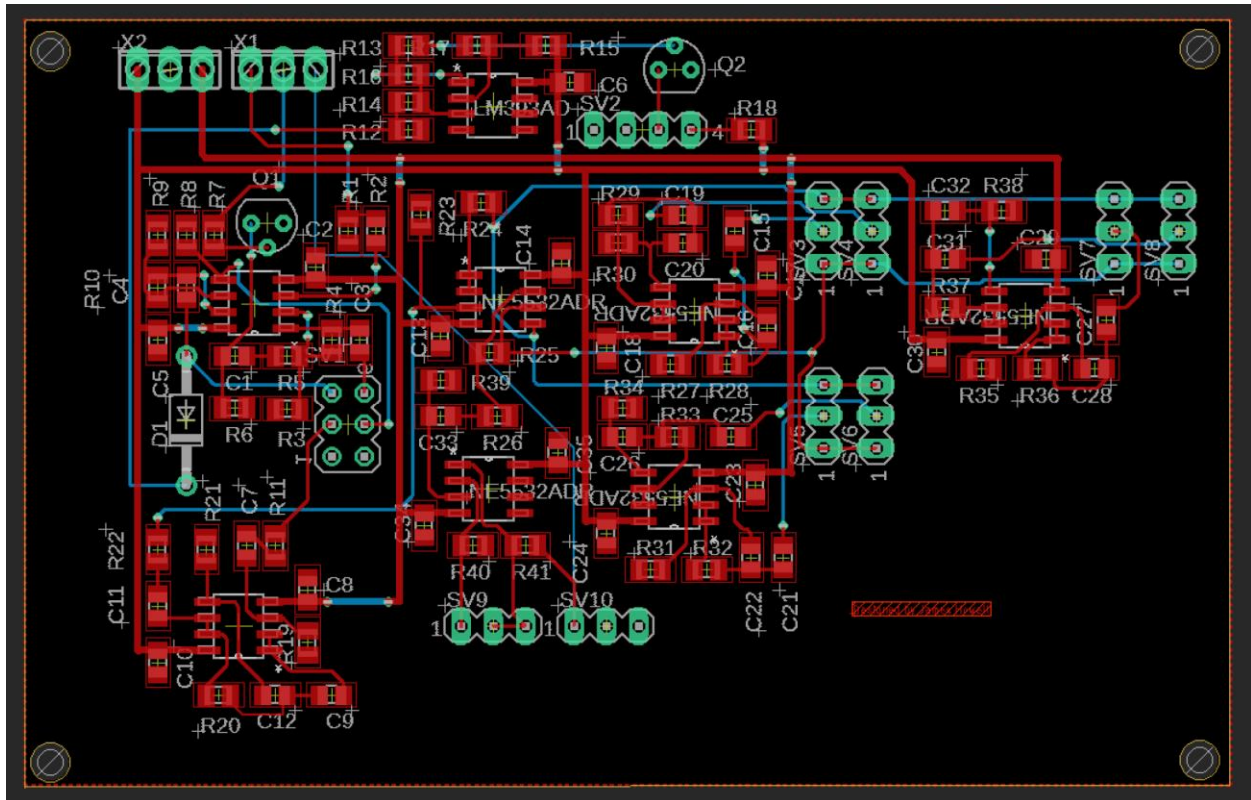


Figure 6.1.a - Pre-Amplifier PCB Schematic

| Qty | Value | Parts | Description | MPN |
|---|---|---|---|---|
| 17 | 100n | C2, C5, C6, C8, C10, C13, C14, C16, C17, C18, C23, C24, C25, C29, C30, C34, C35 | CAPACITOR | EEV106M035A9BAA |
| 9 | | SV3, SV4, SV5, SV6, SV7, SV8, SV9, SV10, SV11 | PIN HEADER | |
| 6 | NE5532ADR | U3, U4, U5, U6, U7, U8 | OP AMP | NE5532D8R2G |
| 4 | 470 | R29, R32, R33, R36 | RESISTOR | MC01W08051470R |
| 3 | 68k | R30, R31, R34 | RESISTOR | NTCS0805E3683FXT |
| 3 | 2.7k | R24, R25, R40 | RESISTOR | CR0805-JW-272ELF |
| 3 | 1u | C7, C15, C33 | CAPACITOR | 885012207051 |
| 3 | 1M | R1, R12, R23 | RESISTOR | ERJ-P06D1004V |
| 3 | 110 | R11, R26, R41 | RESISTOR | ERJ-HP6D2202V |
| 2 | BS170 | Q1, Q2 | N-CHANNEL MOS FET | BS170 |
| 2 | 62k | R35, R38 | RESISTOR | MCWF08R6202BTL |
| 2 | 22.2k | R8, R10 | RESISTOR | ERJ-HP6D2202V |
| 2 | 2.2k | R5, R9 | RESISTOR | ERA-6VEB2201V |
| 2 | 1k | R14, R17 | RESISTOR | CR0805-FX-1001ELF |
| 2 | 10u | C3, C4 | CAPACITOR | CL21A106KOQNNNE |
| 2 | 100k | R6, R16 | RESISTOR | CR0805AFX-1003ELF |
| 1 | OPA1642AIDR | U1 | OP AMP | OPA1642AID |
| 1 | LM393AD | U2 | Comparator | LM393ADR |
| 1 | 8n | C22 | CAPACITOR | 08055C802JAT2A |
| 1 | 820n | C19 | CAPACITOR | C0805C824K4RACTU |
| 1 | 75k | R27 | RESISTOR | RC0805FR-0775KL |
| 1 | 750p | C32 | CAPACITOR | C0805C751J5GACTU |
| 1 | 7.5k | R39 | RESISTOR | ERA-6ARB752V |
| 1 | 63n | C31 | CAPACITOR | MC0805B683K160CT |
| 1 | 6.3k | R21 | RESISTOR | RC0805FR-076K2L |
| 1 | 560 | R28 | RESISTOR | ERJ-6ENF5600V |
| 1 | 510 | R37 | RESISTOR | CRG0805F510R |
| 1 | 42k | R19 | RESISTOR | CR0805-FX-4222ELF |
| 1 | 400n | C21 | CAPACITOR | C0805X394K5RACTU |
| 1 | 400 | R20 | RESISTOR | ERJP06F1001V |
| 1 | 4.7u | C9 | CAPACITOR | 08053D475MAT2A |
| 1 | 4.7k | R3 | RESISTOR | ERJ-HP6D4701V |
| 1 | 330 | R13 | RESISTOR | CMP0805AFX-3300ELF |
| 1 | 30k | R4 | RESISTOR | ERJ6ENF3002V |
| 1 | 3.9u | C12 | CAPACITOR | C0805C395K8PACTU |
| 1 | 3.5n | C26 | CAPACITOR | GRM2165C1H362JA01D |
| 1 | 2n | C27 | CAPACITOR | C0805C202J5GACTU |
| 1 | 2k | R15 | RESISTOR | RC0805FR-072KL |
| 1 | 22u | C1 | CAPACITOR | GRM21BR61E226ME44L |
| 1 | 220 | R2 | RESISTOR | ERJ-U6RD2200V |
| 1 | 1N4004 | D1 | DIODE | 1N4004-E3/54 |
| 1 | 1n | C28 | CAPACITOR | C0805C102J1GACTU |

| | | | |
|---|---|---|---|
| 1 1K | R7 | RESISTOR | ERJ-U6RD1001V |
| 1 15n | C20 | CAPACITOR | C0805C153K5RACTU |
| 1 100u | C11 | CAPACITOR | GRM21BR60J107ME15L |
| 1 100 | R22 | RESISTOR | CRCW0805100RJNEA |
| 1 1.8k | R18 | RESISTOR | CMP0805AFX-3300ELF |
| 1 | SV1 | PIN HEADER | |
| 1 | SV2 | PIN HEADER | |
| 1 | SV12 | PIN HEADER | |
| 1 | X1 | PHOENIX CONNECTOR | |
| 1 | X2 | PHOENIX CONNECTOR | |

Figure 6.1.b - Pre-Amplifier BOM

## 6.2 Power Amplifier

The Power Amplifier pcb was designed based on the schematic in previous sections. The pcb falls in line with all specifications and requirements determined earlier in the project. Figure 6.2.a details the PCB designed.
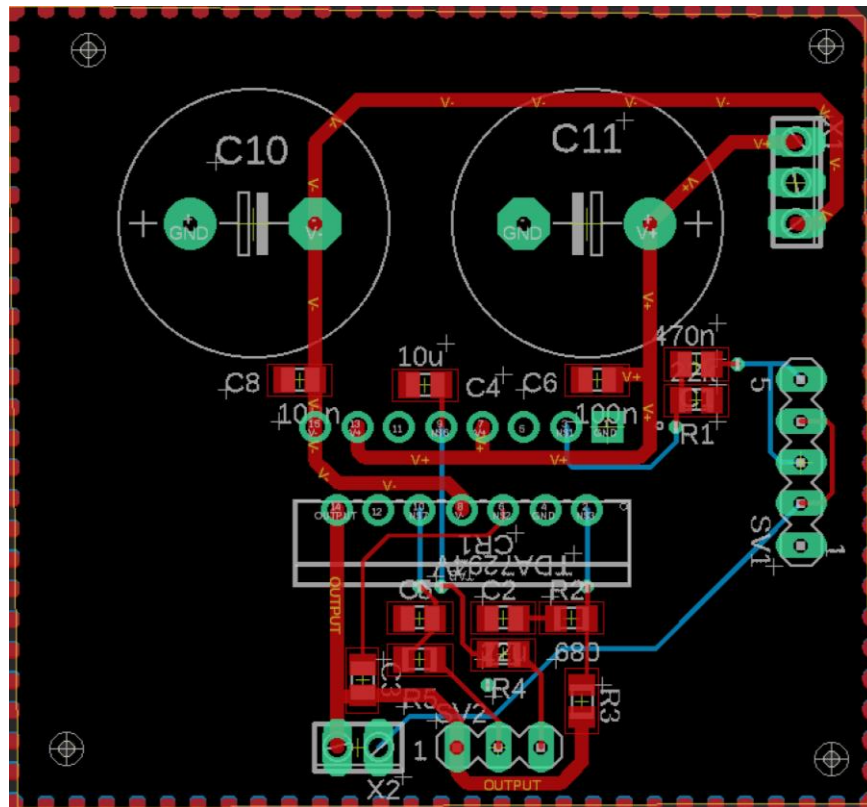


Figure 6.2.a - Power Amplifier PCB Schematic

| Qty | Value | Parts | Description | MPN |
|---|---|---|---|---|
| 1 | TDA7294V | CR1 | 100W Audio Amplifier | TDA7294V |
| 1 | 10k | R5 | RESISTOR | NCP15XH103F03RC |
| 1 | 680 | R2 | RESISTOR | MF50 680R |
| 1 | 30k | R4 | RESISTOR | MF25 30K |
| 2 | 22k | R1, R3 | RESISTOR | MF25 22K |
| 2 | 22u | C2, C3 | CAPACITOR | GRM188R61A226ME15D |
| 2 | 10u | C4, C5 | CAPACITOR | GRM188C81C106MA73D |
| 1 | 470n | C1 | CAPACITOR | GRM022R60J474ME15L |
| 2 | 100n | C6, C8 | CAPACITOR | CGA2B3X7R1V104K050BB |
| 2 | 1F | C10, C11 | POLARIZED CAPACITOR | |
| 1 | | SV2 | PIN HEADER | |
| 1 | | SV1 | PIN HEADER | |
| 1 | | X2 | PHOENIX CONNECTOR | |
| 1 | | X1 | PHOENIX CONNECTOR | |

Figure 6.2.b - Power Amplifier BOM

## 6.3 Power Supply

The Power Supply pcb was designed based on the Power Supply Schematic from previous sections. The pcb was designed with the standards and specifications of the project in mind. Figure 6.3.a details the designed pcb.
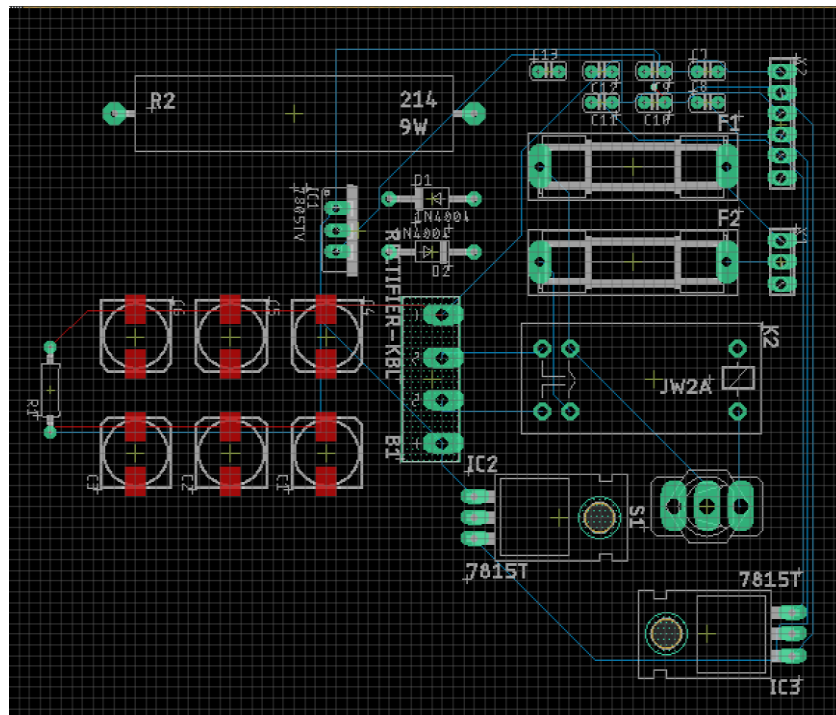


Figure 6.3.a - Power Supply PCB Design

## 6.4 Analog Effects

### 6.4.1 Compressor PCB
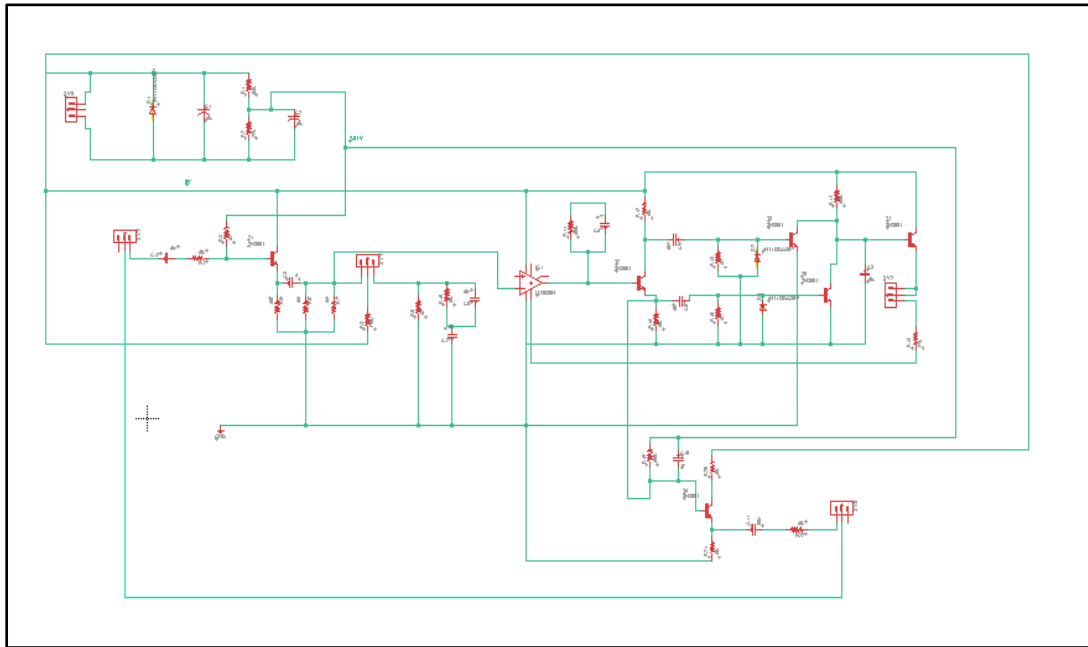


Figure 6.4.1.a - Compressor Schematic

The compressor PCB Schematic uses all of the values of the schematic from multisim but incorporated into the Eagle PCB schematic software. Some of the components needed slightly different footprints to others despite them being the same type of component. For example, in this schematic there are some electrolytic capacitors and some ceramic capacitors, these have different footprints. Some capacitors had a different footprint because of their capacitance. For example, there is a significant size difference between a capacitor with a capacitance of 50nF and a capacitor of 10uF.

The work in progress PCB is shown below:
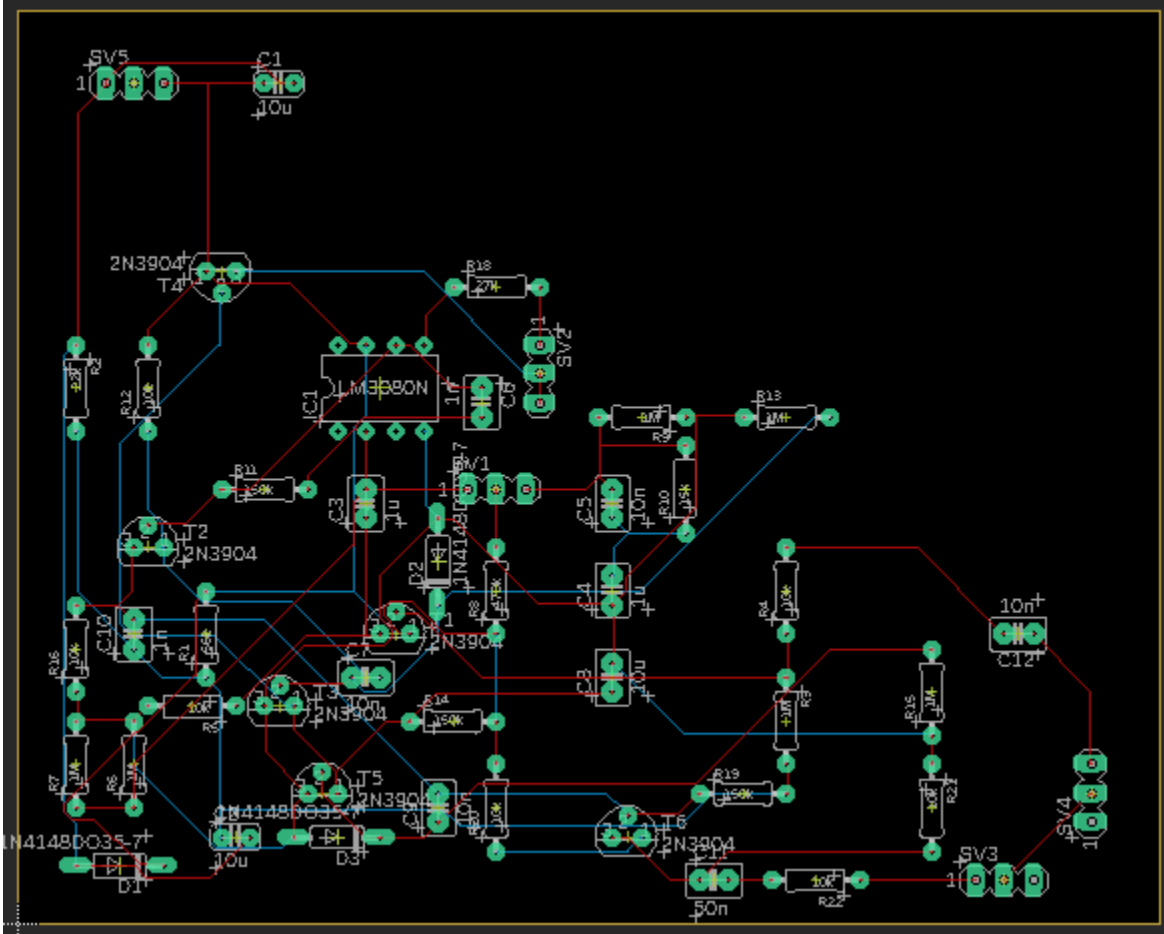
Figure 6.4.1.b - Compressor PCB

Throughout the winter break and the next semester we will likely change our components because of price or perhaps because of the housing that we use. These components may have different footprints or perhaps need to be placed differently from other design concerns. For now however, this is the PCB design for the compressor.

## 6.4.2 Fuzz Box PCB
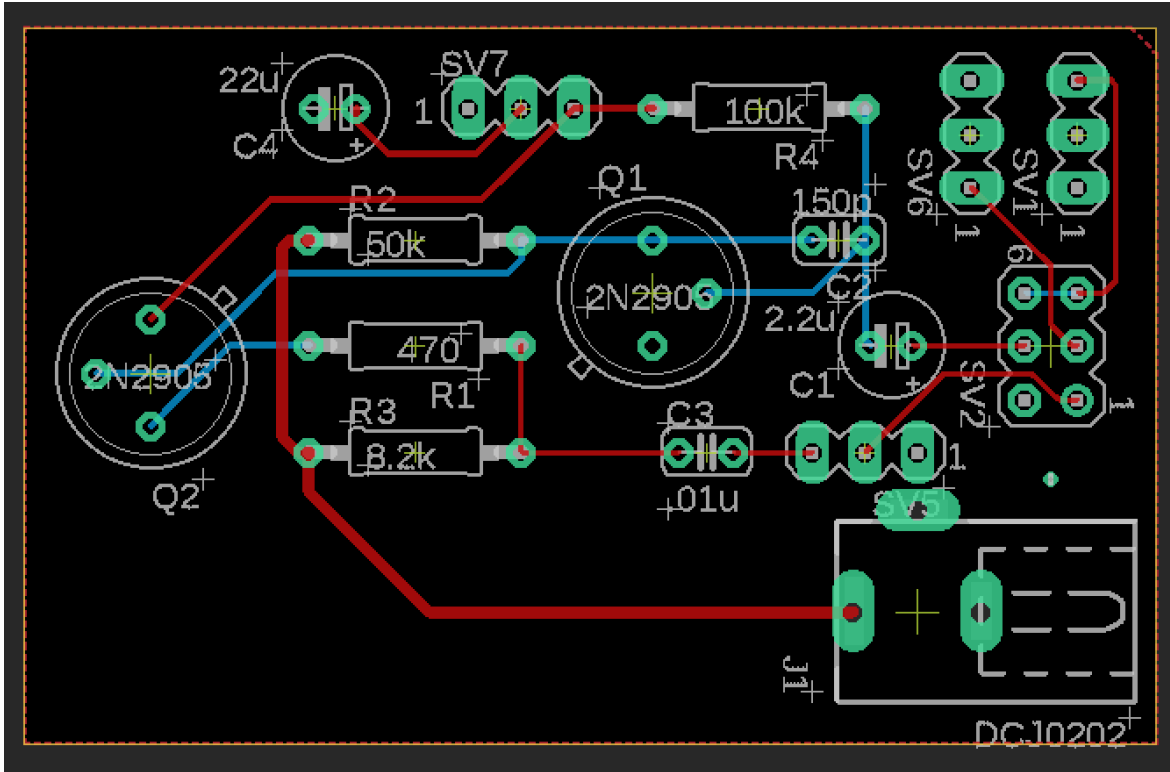


Figure 6.4.2.a - Fuzz Box PCB

| Qty | Value | Parts | Description | MPN |
|---|---|---|---|---|
| 1 | .01u | C3 | CAPACITOR, American symbol | MPMPP250S10J0F200 |
| 1 | 100k | R4 | RESISTOR, American symbol | MFR4-100KFI |
| 1 | 8.2k | R3 | RESISTOR, American symbol | MF25 8K2 |
| 1 | 2.2u | C1 | POLARIZED CAPACITOR, American symbol | MCKSK450M2R2F12S |
| 1 | 470 | R1 | RESISTOR, American symbol | MCF 0.5W 470R |
| 1 | 22u | C4 | POLARIZED CAPACITOR, American symbol | MAL211671229E3 |
| 1 | 50k | R2 | RESISTOR, American symbol | H850KBCA |
| 1 | 150p | C2 | CAPACITOR, American symbol | F151K25S3NN63J5R |
| 2 | AC128 | Q1, Q2 | PNP Germanium Transistor | AC128 |
| 4 | | SV1, SV5, SV6, SV7 | PIN HEADER | |
| 1 | | SV2 | PIN HEADER | |
| 1 | DCJ0202 | J1 | DC POWER JACK | |

Figure 6.4.2.b - Fuzz Box BOM

## 6.4.3 Distortion PCB

Shown above is the Eagle PCB schematic of the Distortion Pedal. It resembles the Multisim schematic with the important difference that all ground, 4.5V and 9V connections are connected as one node. This is of course for the purpose of creating a ground plane in the Eagle software. Another difference between these schematics is the correct

footprints applied such that the size of the components matches their real-world counterparts.
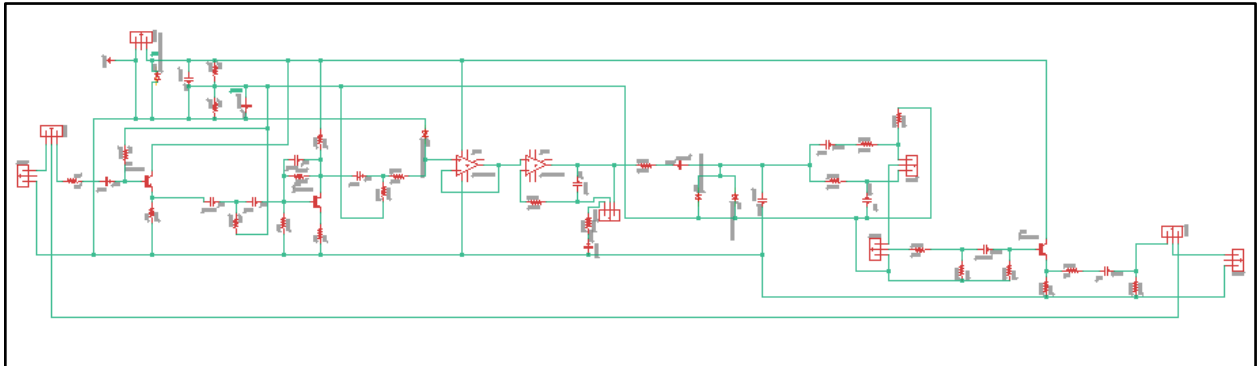


Figure 6.4.3.a - Distortion Schematic
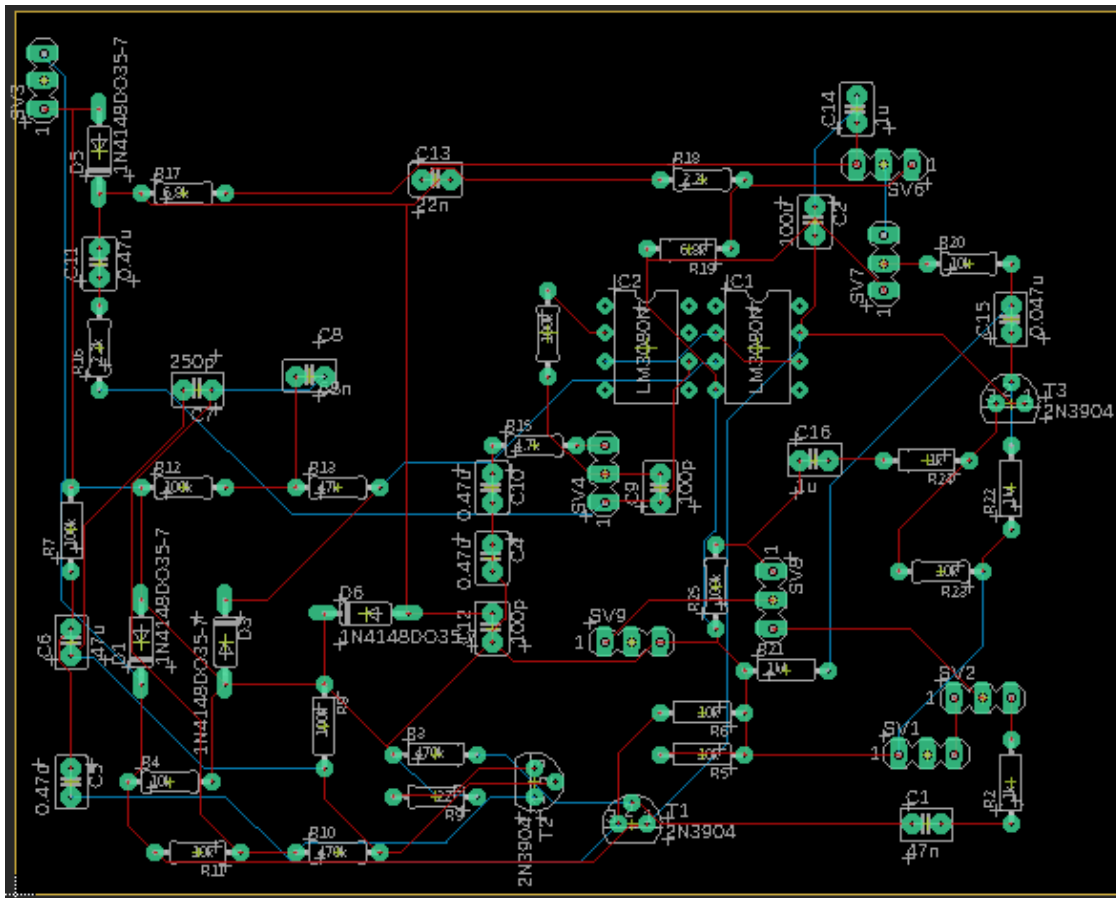


Figure 6.4.3.b - Distortion PCB

## 6.4.4 Phase 90 PCB
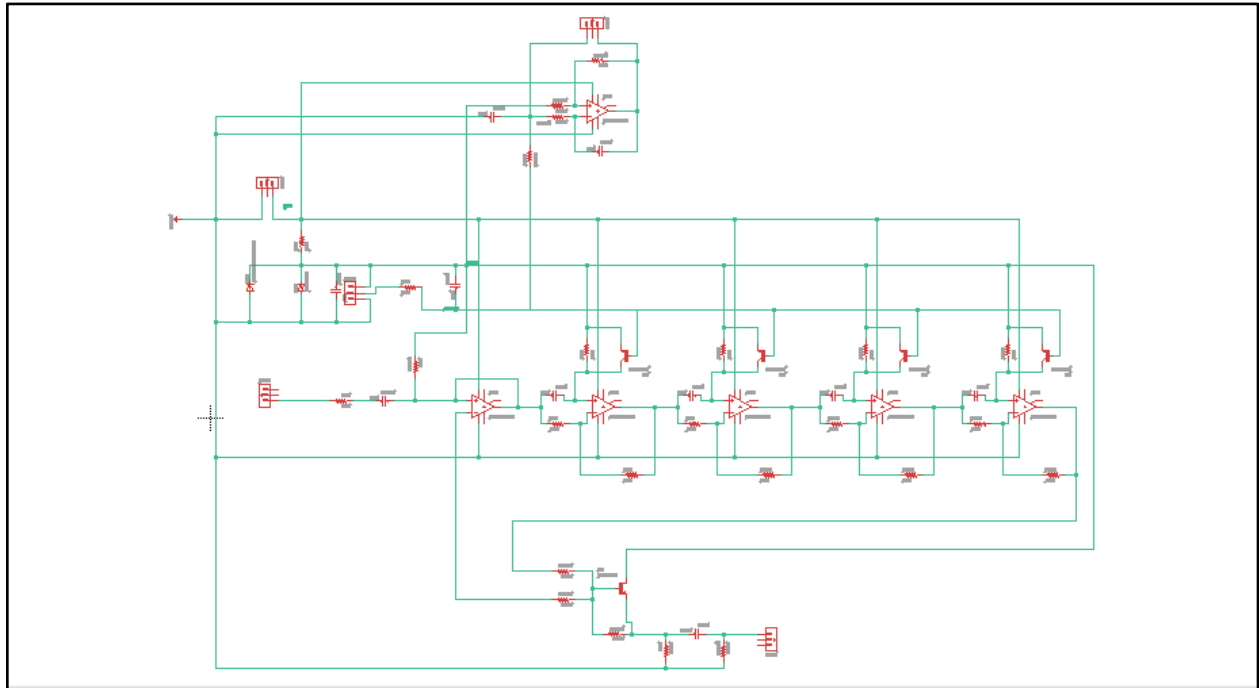


Figure 6.4.4.a - Phase 90 Schematic

Shown above is the Eagle Schematic of the MXR Phase 90. It is structurally the same as the multisim schematic but it has correct footprints for the components we hope to implement in Senior Design 2.
Shown above is the work in progress PCB design of the MXRPhase90 effect. As the following semester progresses and even throughout the upcoming holiday break this design will likely be altered a bit in order to accommodate real-world design concerns.
One of the notable challenges in designing this particular PCB is the 4-stage amplifier portion. Especially on the free Eagle PCB design software it was challenging to fit everything within the required bounds and to connect the components attached to the op-amps in a sensible way that could be iterated for all five amplifiers.
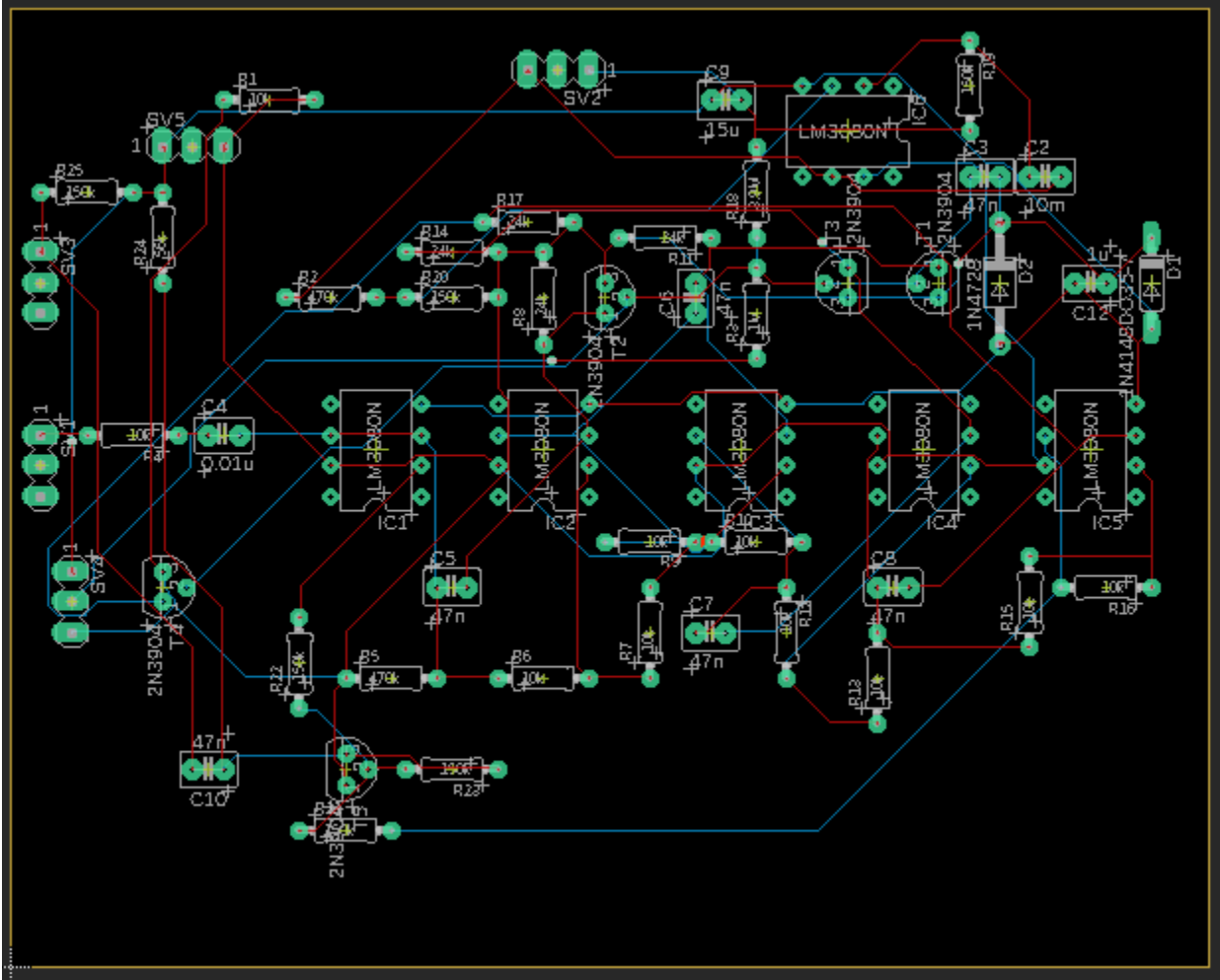
Figure 6.4.4.b - Phase 90 PCB

## 6.5 Housing
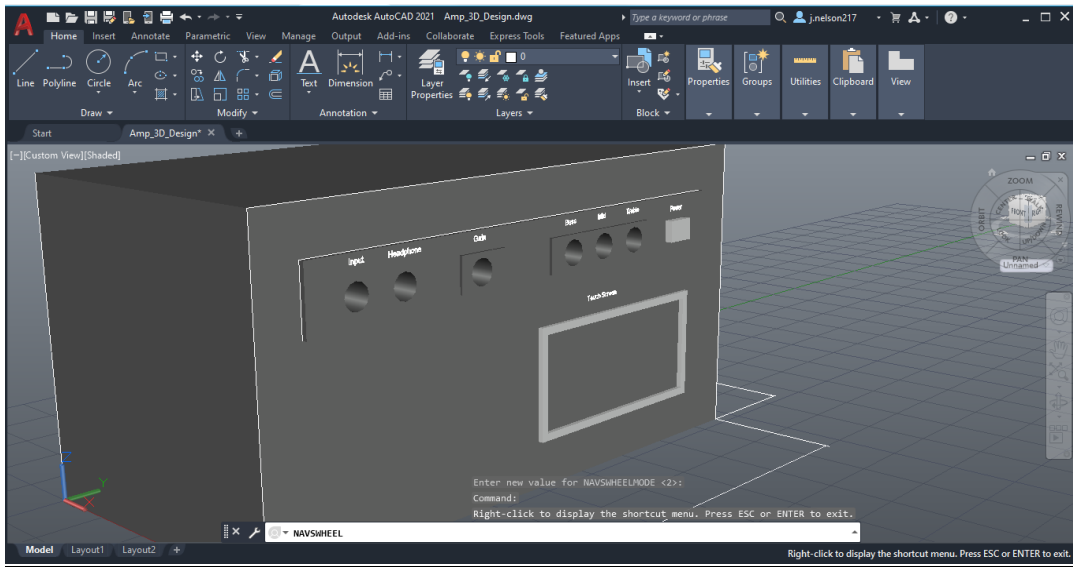


Figure 6.5.a - Prototype Illustration Front
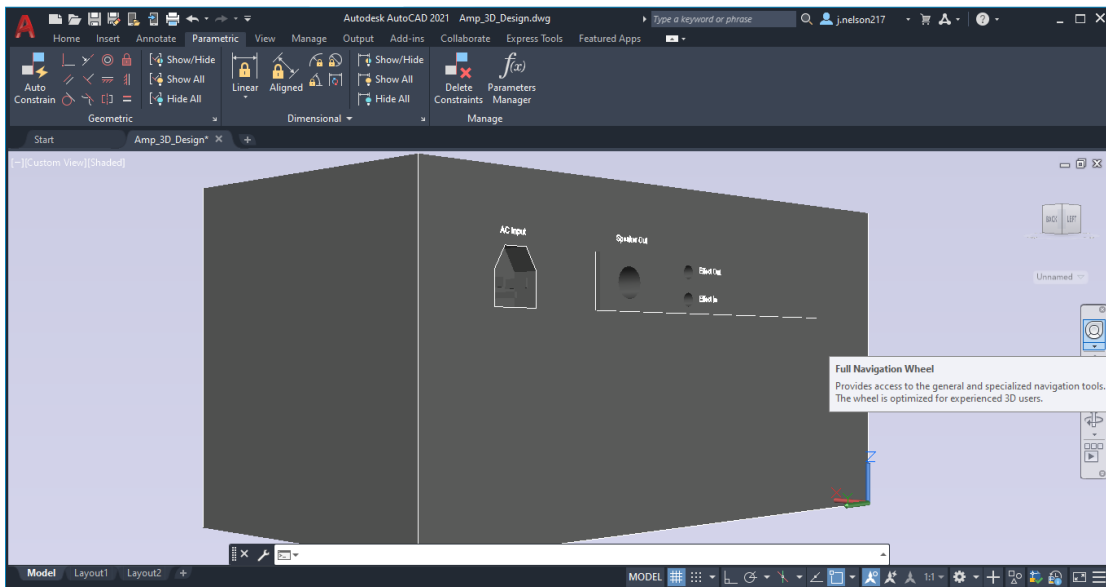


Figure 6.5.b - Prototype Illustration Back

Figure 6.5.a illustrates the front of our prototype, which has a touchscreen and knobs for control. Figure 6.5.b shows the back, including the different connectors the header will require.

# 7.0 Test Plan

## 7.1 Test Equipment

The table below, 7.1.a shows all of the equipment necessary for the testing of the unit. It is a combination of equipment found in the on campus lab as well as personal equipment of the team members.

| Equipment | Description | Location |
|---|---|---|
| Tektronix DPO 4034B | Oscilloscope | Senior Design Lab ENG1 - 456 |
| Tektronix DMM 4050 | Digital Multimeter | Senior Design Lab ENG1 - 456 |
| Tektronix Dual Arbitrary Function Generators | Simulation of input waveforms | Senior Design Lab ENG1 - 456 |
| Keithley 2230-30-1 | Power Supply | Senior Design Lab ENG1 - 456 |
| Rohde & Scharz FPC1000 | Spectrum Analyzer | Communications Lab ENG1 - 461 |
| Soldering Station | Soldering components to PCB | Senior Design Lab ENG1 - 456 James' Personal Supply Armon's Personal Supply Kristofer's Personal Supply |
| B&K Precision 1476 | Oscilloscope | James' Personal Supply |
| B&K Precision 2831C | Multimeter | James' Personal Supply |
| Audio Codec | Responsible for encoding and decoding incoming audio signals | To be purchased |
| DSP Development Board | Responsible for communicating with audio codec to perform digital signal processing on audio signal | To be purchased |

| Bass Guitar | Provide realistic audio signal to DSP for final testing | James's personal instrument Kristofer's personal instrument |
| --- | --- | --- |
| MCU Development Board | Responsible for controlling the screen and sending the user's inputs to the DSP unit | To be purchased |
| MSP4022 | Screen with touch overlay | To be purchased |

Table 7.1.a - Test Plan Equipment

## 7.2 Software Testing

### 7.2.1 User Interface Testing

To test the software for the user interface, the touch screen and microcontroller will be connected based on Figure 5.8.2.b using jumper cables. Once connected the microcontroller will have the software for the UI loaded onto it and run. It will stay connected to the computer so the terminal can see whether button presses are being accurately read. Once the button presses are positively identified across each section of the screen, the UART communication between the MCU and DSP will be tested. Test messages will be sent back and forth between the units to verify the UART communications are working as expected.

### 7.2.2 Digital Effects Testing

The audio codec will need to be soldered to the DSP board, this can be done either in the senior design lab or using Armon's personal equipment. After the code is written for the digital effects, it will need to be tested to see if the signal processing is doing what it is designed to do.

Firstly, some sort of input signal is needed. There are two options for using a sample audio input. The first one would be utilizing the actual bass guitar. The bass guitar can be connected using the corresponding jack. Although this method would be good for more final testing of the effects, it is impractical as only one person in the group owns a bass guitar. For the early and intermediate stages of testing, a waveform generator in the senior design lab will be utilized to provide an input signal to the DSP. Various test cases can be used, such as a sine wave, square wave, and triangle wave.

Since the audio codec contains the ADC and DAC, the audio signal will need to pass through there before it is filtered, and after. One channel of the oscilloscope can be connected to the input channel of the DSP development board to see the original signal coming in. The other channel of the oscilloscope can be connected to the output terminals or pins of the audio codec to see if the effect is being properly applied.
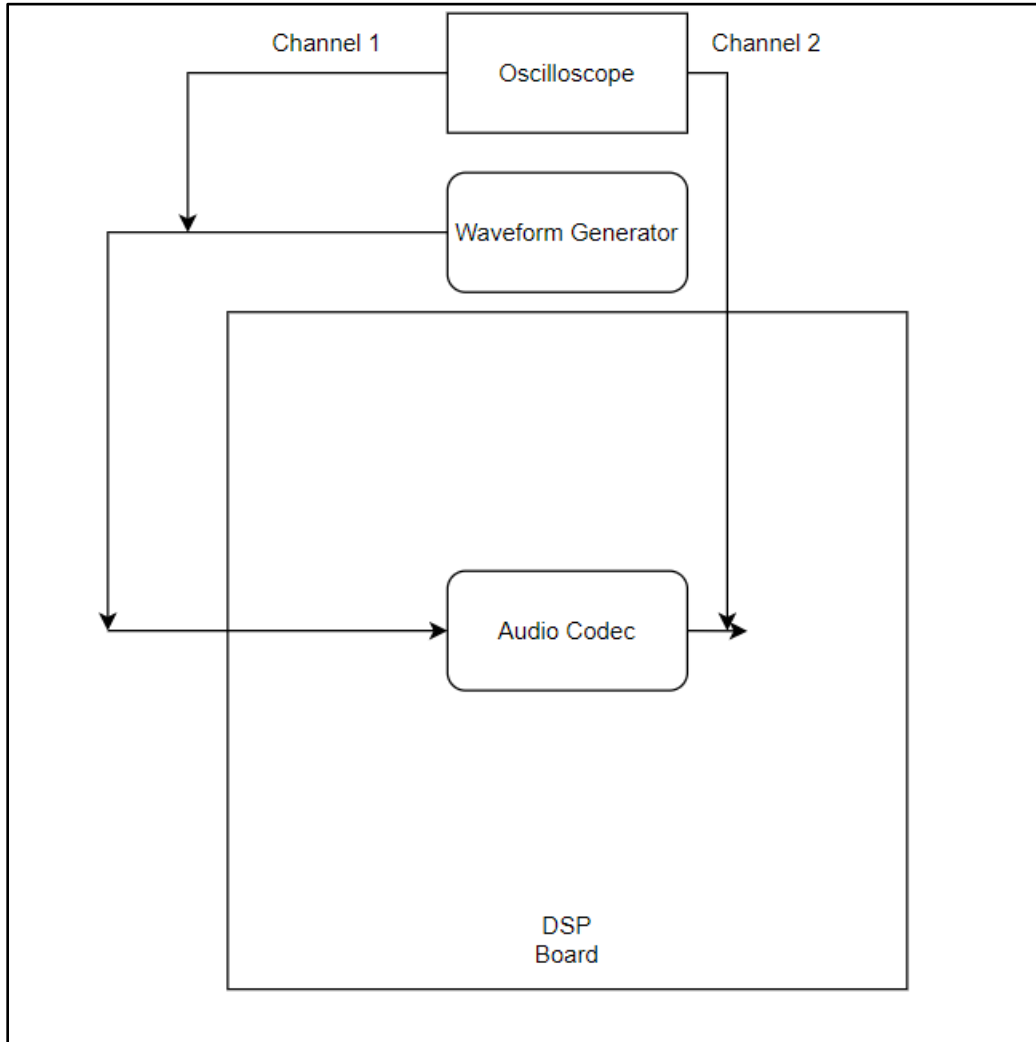
Figure 7.2.2.a - Block Diagram of Test Plan

## 7.3 Hardware Testing

### 7.3.1 Pre-Amplifier/Overload/Equalizer Testing

Testing the preamp/overload/equalizer consists of using a function generator to send an AC signal through the circuit and using an oscilloscope to measure the output. The simulated results were calculated for each discrete component of this circuit, being the preamp, overload detector, and equalizer. The oscilloscope will measure the input and output at each section of the circuit and determine whether the measured results align with the simulated results found in sections 5.1.2 for the preamp, 5.2.2 for the overload detector, and 5.3.4 for the equalizer.

### 7.3.2 Power Amplifier

Testing the power amp will work similarly to the preamp/overload/equalizer circuit mentioned previously. A function generator will be used to send an AC signal through

the circuit, and an oscilloscope will be used to measure the output. This output should align with the expected and simulated results for the power amplifier.

### 7.3.3 Power Supply

The power supply can be tested by taking a signal from a wall outlet to regulate. If it operates as designed, it will provide +35V, -35V, +15V, -15V, and +5V rails. A multimeter can be used to measure the voltage at each rail and compare it to the expected value, any deviations resulting in a failed test result.

### 7.3.4 Analog Effects

The analog effects will be tested similarly to the poweramp and preamp mentioned previously. Each of the effect pedals will have a function generator apply an AC input signal, while an oscilloscope measures the output of the circuit. Each output will be compared to the expected results found in the simulation, which can be found in 5.6.2 for the Compressor's expected results, 5.6.4 for the Fuzz Box's simulated performance, 5.6.6 for the Distortion pedals expected performance, and 5.6.8 for the expected performance of the Phase 90 pedal.

## 7.4 Requirements Testing

To ensure the product meets the requirements set forth at the beginning of the project, the table below lists the test plan for each of the requirements.

| Requirement | Target Value | Test Case |
|---|---|---|
| Size | 24" x 12" x 12" | Measure the dimensions of the final product and determine whether it meets the requirement. |
| Weight | 12lbs | Weigh the final product and determine whether it meets the 12lb requirement. |
| Output Power Rating | 100W @ 8Ω | Send an AC signal through the system and use a multimeter to measure the wattage at the output to determine if it meets the requirement. |
| Input Power Rating | 120VAC 60Hz | Use a wall outlet to power the unit, and use a multimeter to measure the voltage at each rail to determine if it is regulating the voltage correctly. |
| Frequency Response | 31Hz - 4.5kHz | Use a function generator to send varied AC signals through the unit, and use an oscilloscope to measure frequency and determine if it meets the requirement. |
| THD | 0.5% | Measure THD by inputting a sine wave into the |

| | | input and measure the distortion with a Spectrum Analyzer |
|---|---|---|
| Input Impedance | 1MΩ | Measure the resistance of the first filter stage of the Pre Amplifier |
| Output Impedance | 8Ω | Measure the output resistance of the speaker connected. |
| SNR | 80dB | Use the Spectrum Analyzer to measure the SNR |
| Analog Effects | 4 | Test each analog effect pedal and ensure there are at least 4 pedals applying the desired effect to the signal. |
| Digital Effects | 4 | Test each digital effect and ensure there are at least 4 effects applying the desired effect to the signal. |
| Inputs | 2 ¼ inch jacks | Verify there are two quarter inch inputs to receive the bass guitar's signal as well as the effect pedal's signal. |
| Output | Neutrik Audio Connector ⅛'' Headphone jack Bluetooth 5.0 | Verify there is a neutrik audio connector to send the signal to a speaker, as well as a headphone jack and a bluetooth output that can be received by bluetooth headphones or speakers. |

Figure 7.4.a - Requirement Testing Breakdown

# 8.0 Administration

## 8.1 Milestones

### 8.1.1 Senior Design I

The table below, 8.1.1.a, shows the outline for the milestones over the course of the first semester, Senior Design 1.  Each Milestone shows its expected completion date, as well as the current status and member or members responsible for completion.

| Milestone | Task | Due | Status | Responsible |
|---|---|---|---|---|
| 1 | Divide and Conquer V1 | Sep 16, 2022 | Complete | Group 26 |
| 2 | Technology Investigation | Oct 7, 2022 | Complete | Group 26 |
| 3 | Divide and Conquer V2 | Oct 7, 2022 | Complete | Group 26 |
| 4 | Pre Amp/Power Amp Research/Design | Oct 14, 2022 | Complete | Kristofer Edstrom/ James Howell |
| 5 | Analog Effects Research/Design | Oct 21, 2022 | Complete | James Howell |
| 6 | Digital Effects Research/Design | Oct 21, 2022 | Complete | Armon Eghbali |
| 7 | User Interface Research Design | Oct 21, 2022 | Complete | Kristofer Edstrom |
| 8 | Signal Flow Research/Design | Oct 21, 2022 | Complete | James Howell |
| 9 | 60 Page Draft | Nov 4, 2022 | Complete | Group 26 |
| 10 | Pre Amp/Power Amp PCB Design | Nov 11, 2022 | Complete | Kristofer Edstrom/ James Howell |
| 11 | Analog Effects PCB Design | Nov 11, 2022 | Complete | James Howell |

| | | | | |
|---|---|---|---|---|
| **12** | DSP/ Microprocessor inclusion to circuit | Nov 11, 2022 | In Progress | Armon Eghbali/ Kristofer Edstrom |
| **13** | GUI Development/ Implementation | Nov 11, 2022 | In Progress | Kristofer Edstrom |
| **14** | Power Delivery PCB Design | Nov 11, 2022 | Complete | James Howell |
| **15** | 100 Page Draft | Nov 18, 2022 | Completed | Group 26 |
| **16** | Finish PCB Design | Nov 18, 2022 | Completed | Group 26 |
| **17** | Final Document | Dec 6, 2022 | Completed | Group 26 |

Table 8.1.1.a - Senior Design I Milestones

**8.1.2 Senior Design II**

Table 8.1.1.b outlines the expected schedule for milestones in Senior Design II. Each milestone has an expected completion date, as well as a current status and the members responsible. This schedule is subject to some change as problems are encountered and changes are made to the project scope. A conservative early March completion date has been decided for the completion of a final product, so when problems are encountered, there will still be time to fix and construct a new product by the end of the semester.

| **Milestone** | **Task** | **Due** | **Status** | **Responsible** |
|---|---|---|---|---|
| **1** | Order PCB | Jan 9, 2023 | Not Started | Group 26 |
| **2** | Order MCU, DSP, MISC components | Jan 9, 2023 | In progress | Group 26 |
| **3** | Software Testing User Interface | Jan 23, 2023 | Not Started | Group 26 |
| **4** | Software Testing Digital Effects | Jan 23, 2023 | Non Started | Group 26 |
| **5** | Hardware Testing Preamp/Overload/ Equalizer | Jan 30, 2023 | Not Started | Group 26 |
| **6** | Hardware Testing Power Amplifier | Feb 3, 2023 | Not Started | Group 26 |

| 7 | Hardware Testing Power Supply | Feb 3, 2023 | Not Started | Group 26 |
|---|---|---|---|---|
| 8 | Hardware Testing Analog Effects | Feb 10, 2023 | Not Started | Group 26 |
| 9 | Prototype Completed | Feb 16, 2023 | Not Started | Group 26 |
| 10 | Test and Revise Integrated Product | Feb 30, 2023 | Not Started | Group 26 |
| 11 | Final Product Completed | Mar 3, 2023 | Not Started | Group 26 |
| 12 | Senior Design 2 Draft | TBD | Not Started | Group 26 |
| 13 | Senior Design 2 Final Document | TBD | Not Started | Group 26 |
| 14 | Conference Paper | TBD | Not Started | Group 26 |
| 15 | Final Presentation | TBD | Not Started | Group 26 |

Table 8.1.1.b - Senior Design II Milestones

## 8.2 Budget

Table 8.2.a below shows the estimated budget for the design and build of the project, each cost being an approximated value based on average costs for similar devices.

| Item | Quantity | Cost |
|---|---|---|
| DSP Processor Dev Board | 1 | $25 |
| Microprocessor Dev Board | 1 | $50 |
| Housing | 1 | $50 |
| Analog Effects PCB (with components) | 1 | $25 |
| Signal Flow PCB (with components) | 1 | $35 |

| | | |
|---|---|---|
| Pre Amp PCB (with components) | 1 | $25 |
| Power Amp PCB (with components) | 1 | $25 |
| Misc Components | X | $20 |
| Audio Connectors | X | $20 |
| Resistive Touchscreen | 1 | $40 |
| PC Fan | 1 | $5 |
| Cad Software License | 2 | Free (Education Version) |
| PCB Design Software | 4 | Free (Education Version) |
| Total | | $320 |

Table 8.2.a - Estimated Budget

## 8.3 Individual Contributions

Table 8.3.a below shows the contributions made by each team member, including what sections they made a contribution to, and an overall approximate count of how many pages are contributed to the document.

| Team Member | Section's Contributed | Page Count |
|---|---|---|
| Kristofer Edstrom | 1.0 Executive Summary<br>2.4 House of Quality<br>2.5 Project Block Diagram<br>3.4 Power Amplifier<br>3.8 Interface<br>3.9 Housing<br>4.1 Constraints<br>5.6 Analog Effects<br>5.8 Interface<br>7.3 Hardware Testing<br>7.4 Requirements Testing<br>8.1 Milestones<br>8.2 Budget | ~32 |

| Armon Eghbali | 2.1 Objective<br>2.2 Motivation<br>2.4 House of Quality<br>3.7 Digital Effects Research<br>5.7 Digital Effects Design<br>7.7 Digital Effect Test Plan | ~29 |
|---|---|---|
| James Howell | 2.3 Project Specifications<br>2.4 House of Quality<br>2.5 Project Block Diagram<br>3.1 Pre-Amplifier<br>3.2 Overload Detector<br>3.3 Equalization<br>3.6 Analog Effects<br>5.1 Pre-Amplifier<br>5.2 Overload Detector<br>5.3 Equalization<br>5.4 Power Amplifier<br>5.5 Power Supply<br>6.1 Pre-Amplifier<br>6.6 Analog Effects<br>7.1 Test Equipment<br>8.3 Individual Contributions | ~41 |
| Jeremy Nelson | 3.5 Power Supply Research<br>3.9 Housing<br>6.4 Analog Effects<br>6.7 Housing<br>3.7.8 FPGA and Digital Effects | ~21 |

Table 8.3.a - Individual Contributions

## **9.0 Conclusion**

Overall, the first semester of Senior Design was deemed a success. The challenges presented by the course were met with open arms and successfully completed one by one. The steps outlined DVC1, DV2, 60 Page Draft, and the 100 Page Draft allowed each team member to complete their component of the project with ease. The Milestone table, shown in Figure 8.1.1.a, were also pertinent to the success of the project. The goals outlined in the Executive Summary and Product Description were all met in the designed elements of the project. The simulated results found in Section 5, details this. The constraints outlined in Section 4, were also considered in all aspects of the design. The multiple pcbs required for the design of the project were successfully designed. Putting all of the designed components together, will yield a completed, working Bass Guitar Amplifier.

# 10.0 References

- https://sound-au.com/project152-1.htm

- https://www.ti.com/product/OPA1642

- https://admetro.com/news/the-basic-difference-between-4-wire-and-5-wire-resistive-touchscreens/#:~:text=The%20voltage%20read%20at%20the,and%205-wire%20resistive%20touchscreens.

- https://tru-vumonitors.com/touch-screen-basics/#:~:text=The%20five%20most%20common%20types,%2C%20and%20IR%20(Infrared).

- https://www.geeksforgeeks.org/difference-between-mcu-and-soc/

- https://www.digikey.com/en/articles/microcontrollers-with-a-touch-sensing-interface

- https://www.electronicdesign.com/technologies/analog/article/21801654/understanding-amplifier-operating-classes

- https://picture.iczhiku.com/resource/eetop/sykfGloQfYJRZBcv.pdf

- https://ses.library.usyd.edu.au/bitstream/handle/2123/10578/Marion%2C%20Bruno%20-%20Wah%20Wah.pdf?sequence=2&isAllowed=y

- https://benchmarkmedia.com/blogs/application_notes/152143111-audio-myth-switching-power-supplies-are-noisy

- http://www.lcdwiki.com/4.0inch_SPI_Module_ILI9486

- https://www.etechnophiles.com/esp32-dev-board-pinout-specifications-datasheet-and-schematic/

- https://www.electrosmash.com/