Bass Guitar Amplifier

Initial Project and Group Identification Document
Divide and Conquer, Version 2.0

60-page draft should be "dominated by design" -Richie

Group 26 Fall 2022
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Table of Contents (need to further breakdown further, write table of contents for 120 pages, plan what each person should write)

1.0 Executive Summary

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.

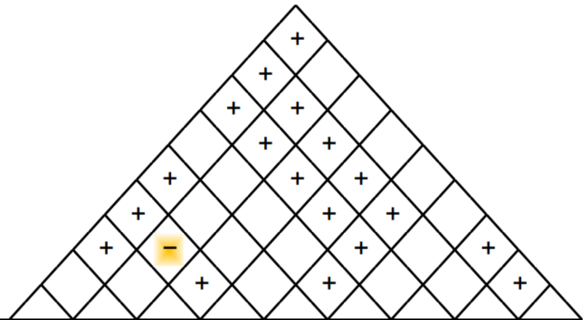
- Touch screen interface capable of controlling digital effects
- Analog effects such as fuzz, distortion, overdrive
- Digital effects such as reverb, delay, and other custom effects implemented with digital circuitry.
- Ability to output to a speaker or headphones

2.3 Project Specifications

Attribute	Description
Size	24” x 12” x 12” (W x H x D)
Weight	12lbs
Output Power Rating	50W @ 8Ω
Input Power Rating	120VAC 60Hz
Frequency Response	40Hz - 3.5kHz
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 ¼” (6.35mm) Effect Input Jack
Outputs	¼” (6.35mm) Speaker Pass Through ⅛” (3.5mm) Aux Output Bluetooth 5.0

Table 2.3.1 - Project Specifications

2.4 House of Quality Analysis



Weight	Customer Requirements (Explicit and Implicit) / 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
	Column #	1	2	3	4	5	6	7	8	9
	Direction of Improvement	▲	◇	▼	▲	◇	◇	▲	▲	▼
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	▼

Figure 2.4.1 - House of Quality

Figure 1 shows the House of Quality, which illustrates our customer and engineering requirements.

2.5 Project Block Diagram Overview

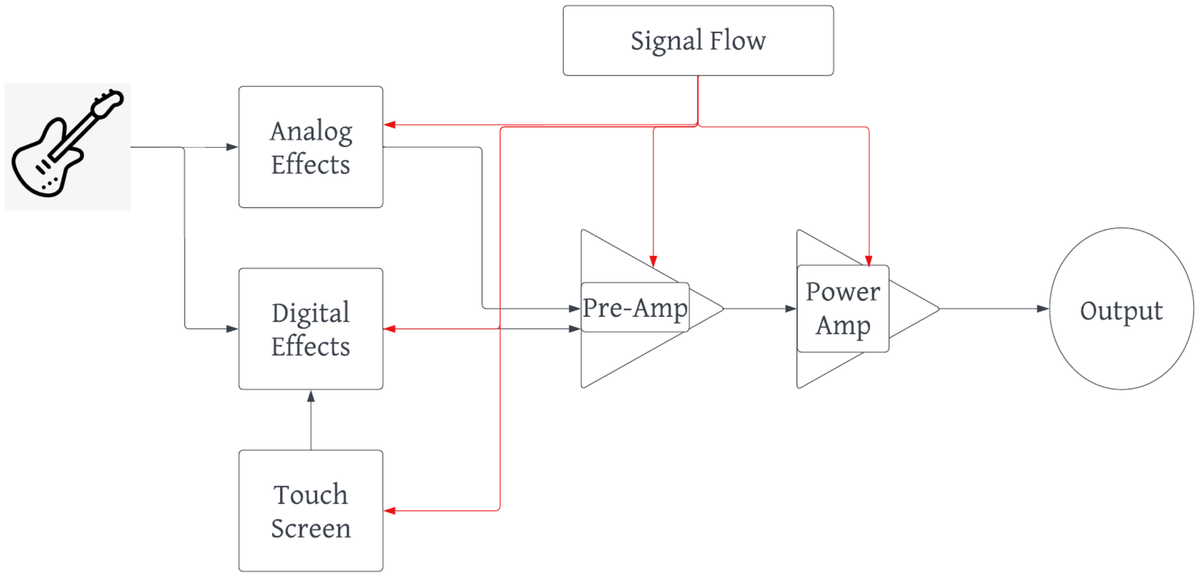


Figure 2.5.1 - Project Block Diagram
(MORE Detail, between touch screen and digital effects)

Analog Effects:

1. James Howell
2. Apply Analog Effects to the signal.
3. Status: Research
4. Inputs: Bass Guitar
   Outputs: Pre-Amp

Digital Effects:

1. Armon Eghbali
2. Apply Digital Effects to the signal.
3. Status: Research
4. Inputs:Bass Guitar
   Outputs: Pre-Amp

Touch Screen:

1. Kristofer Edstrom
2. Provide a touch screen and knobs to allow the user to control the various effects.
3. Status: Research
4. Inputs: User input
   Outputs: Digital Effects

Signal Flow:

1. Jeremy Nelson
2. Supply power to the rest of the amplifier.
3. Status: Research
4. Inputs: None

Outputs: All

Pre Amp:

1. James Howell
2. Bring the output of the Bass Guitar/Effects up to line level.
3. Status: Research
4. Inputs: Analog Effects, Digital Effects, Bass Guitar
   Output: Power Amp

Power Amp:

1. Kristofer Edstrom
2. Takes the line level input and applies a gain to it in order to drive the output.
3. Status: Research
4. Inputs: Pre Amp
   Output: Output

Output:

1. Jeremy Nelson
2. Interface the device with a variety of outputs.
3. Status: Research
4. Inputs: Power Amp
   Output: External Device

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.

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.

3.1.3 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.

3.1.4 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.

3.1.5 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:

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/)	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.

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 a dedicated IC can be used in this situation. For simplicity, it was decided to use a LM397 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

3.4 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.

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 Topology

3.5 Power Supply

3.6 Analog Effects

3.6.1 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 5 and 6 show, graphically, what these compression ratios would look like.

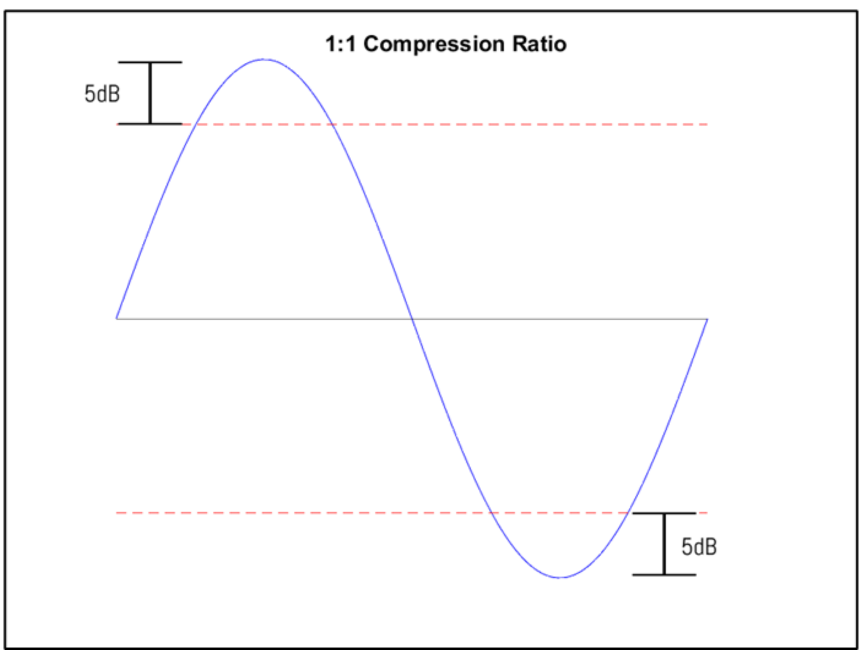


Figure 3.6.1.a – 1:1 Compression Ratio

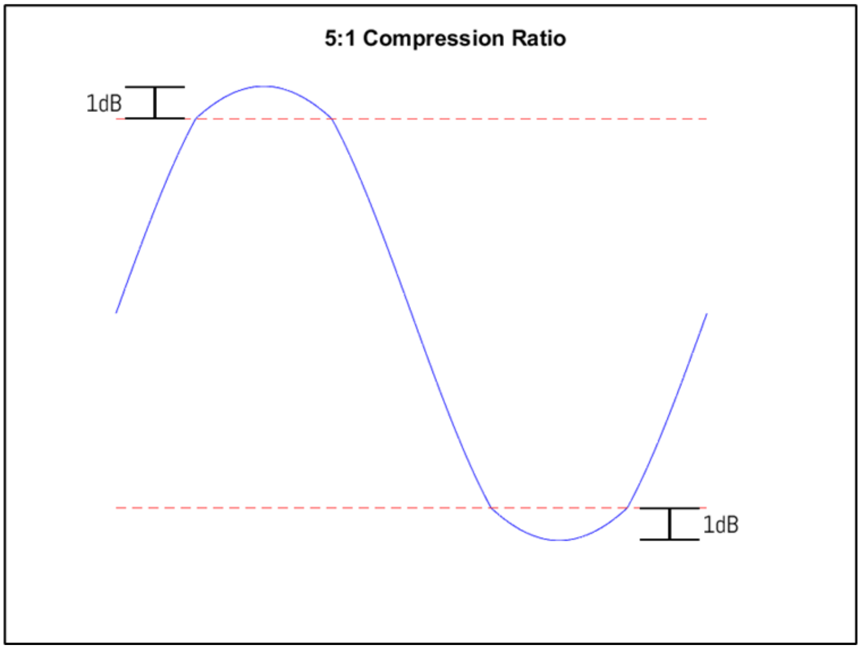


Figure 3.6.1.b – 5:1 Compression Ratio

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.

3.6.2 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 7 shows types of clipping found in Overdrive, Distortion, and Fuzz Effects.

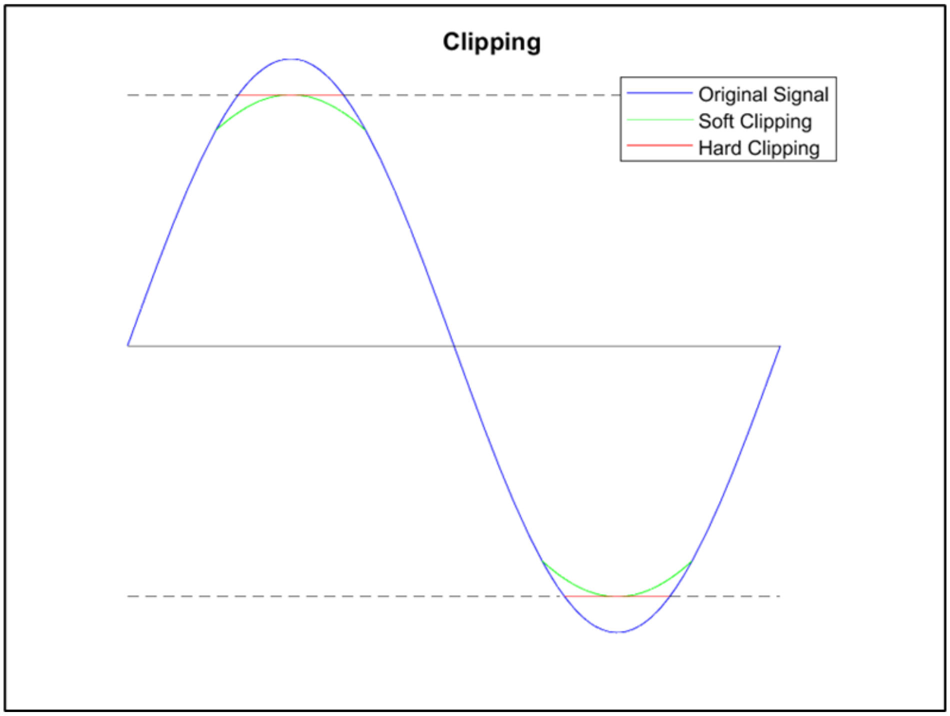


Figure 3.6.2.a - Clipping

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.

3.6.3 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.

3.6.4 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.

3.6.5 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.

3.6.6 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.

3.6.7 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 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.

3.7.2 Echo

Echo is part of a group of effects produced by time delay. A delay of 50 ms or greater causes the echo effect.

3.7.3 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.

3.7.4 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 implemented 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.

3.8 Interface

4.0 Constraints and Standards

4.1 Constraints

Constraint	Description
Cost	Maximum limit of $400 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.

4.1.1 Economic and Time

4.1.2 Environmental, Social, and Political

4.1.3 Ethical, Health, Safety

4.1.4 Manufacturability and Sustainability

4.2 Standards

Standard	Description

Bluetooth	Would comply with Bluetooth 5.0 Standards
RoHS Compliant	All components will be compliant with the RoHS standard for hazardous materials

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.a below..

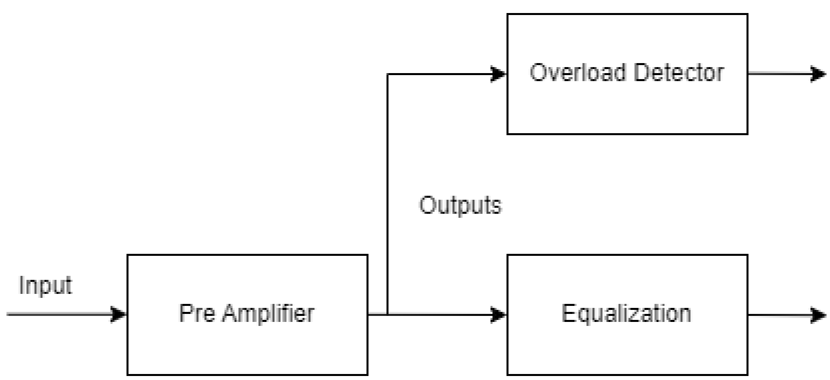


Figure 5.1.1.a - Signal Flow Diagram of Pre-Amplifier

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 thing.thing.1 or Formula thing.thing.2

Formula 1 applies when the switch S1 is open. This is the low gain mode of the amplifier. Formula 2 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 3.

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.

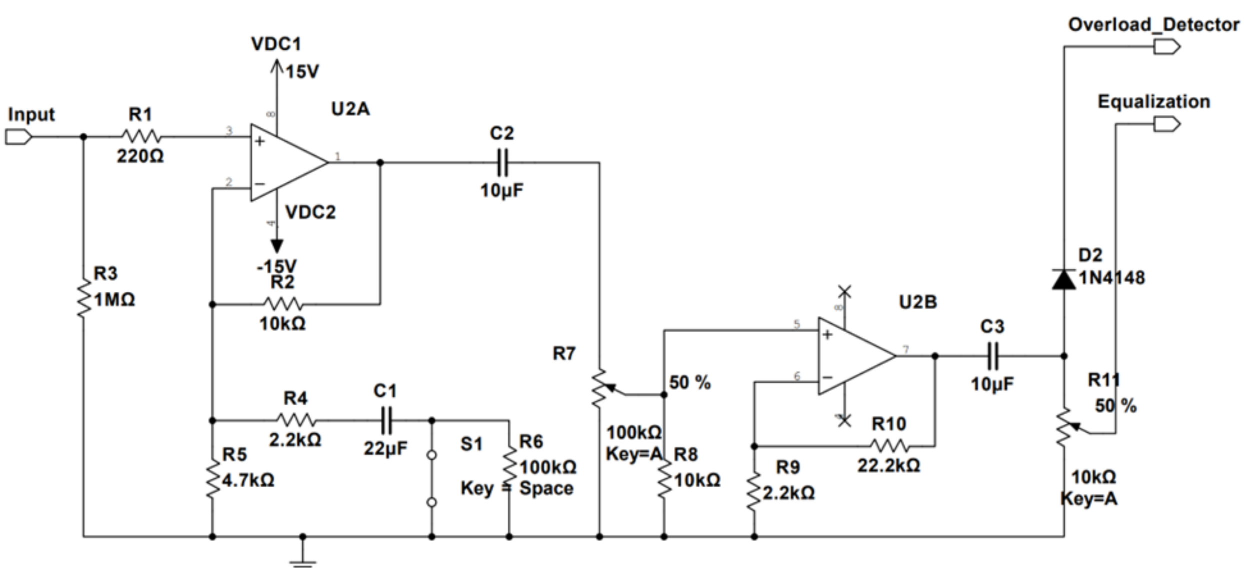


Figure 5.1.1.b - Pre-Amplifier Design Schematic

Qty	Parts	Value	Description	Price
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	
1	R5	4.7k	RESISTOR	
1	R6	100k	RESISTOR	
2	R9, R10	22.2k	RESISTOR	
1	S1		DIP SWITCH	

Table 5.1.1.c - 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.

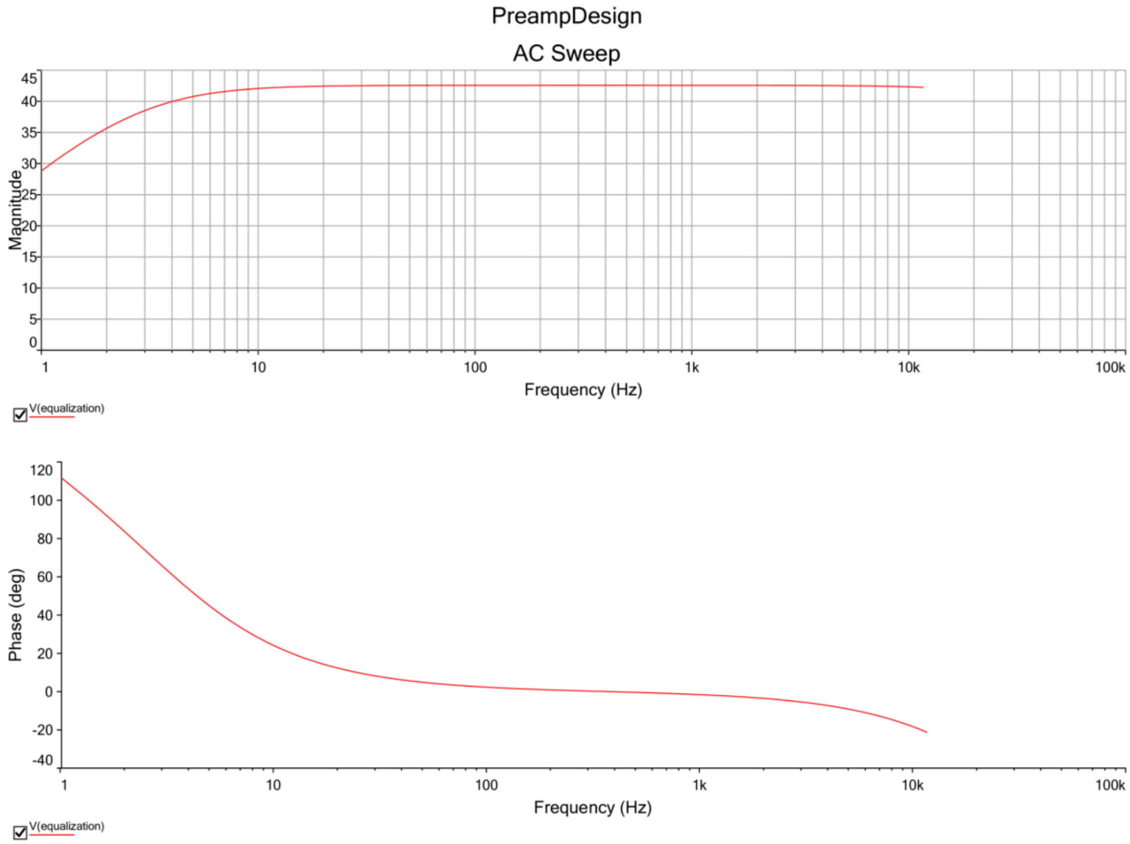


Figure 5.1.2.a - Overall Frequency Response

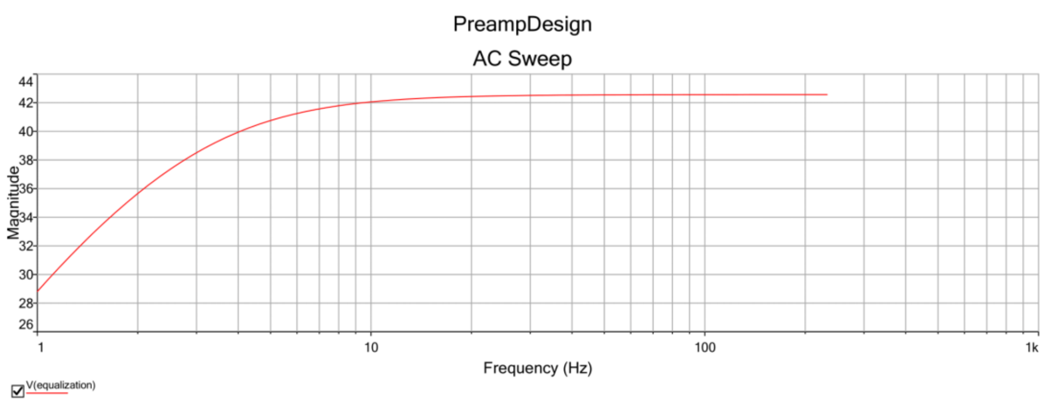


Figure 5.1.2.b - Bass Response

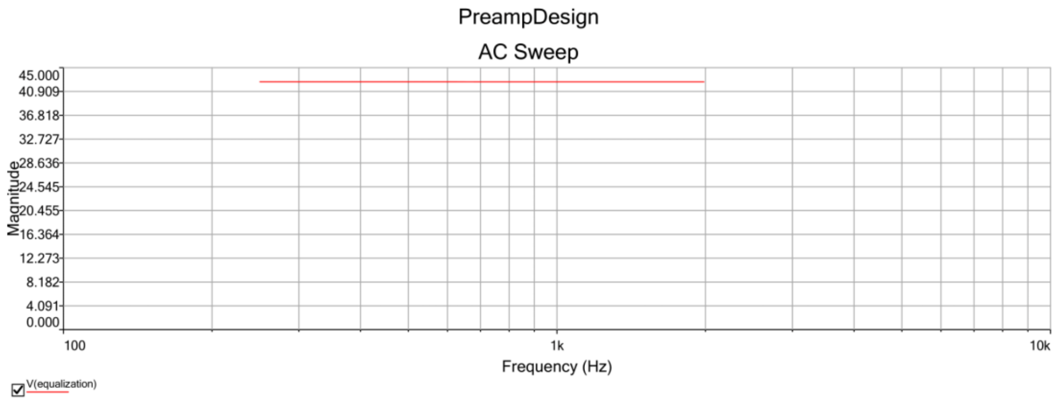


Figure 5.1.2.c - Mid Reponse

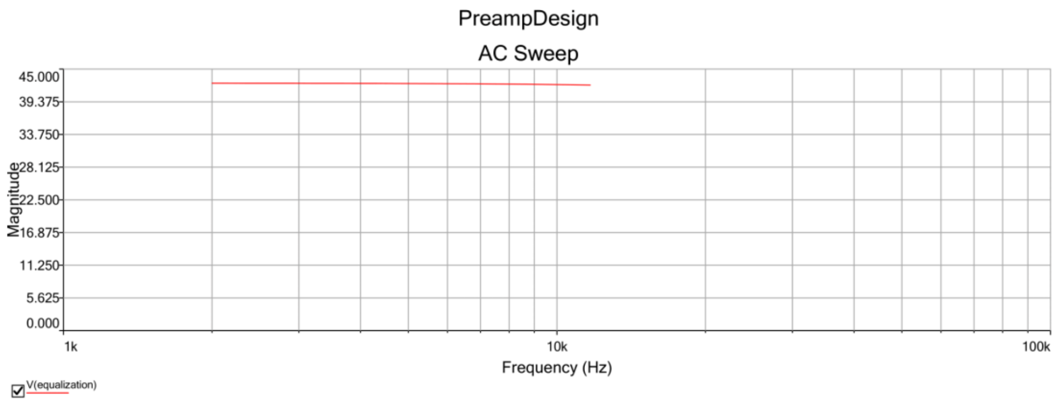


Figure 5.1.2.d - Treble Response

5.2 Overload Detector

5.3 Equalization

5.4 Power Amplifier

5.5 Power Supply

5.6 Analog Effects

5.7 Digital Effects

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.

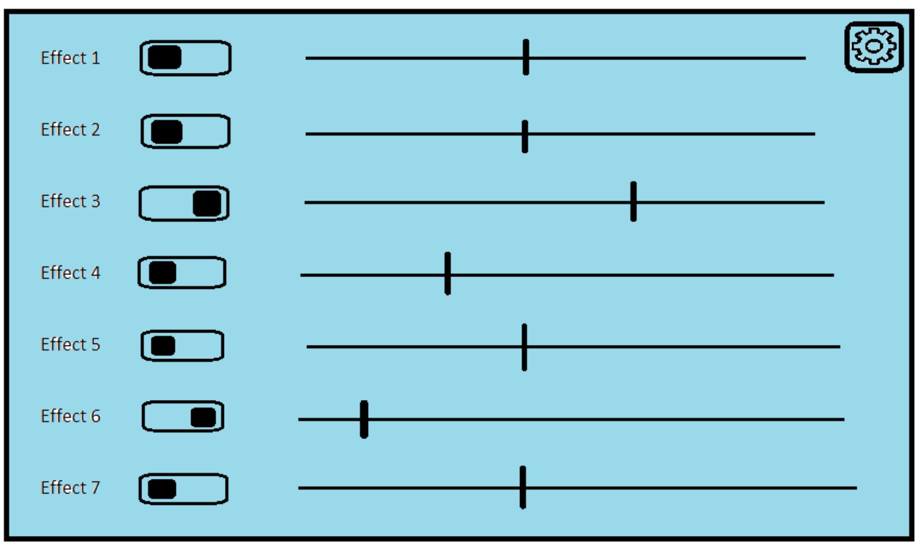


Figure 5.8.2.a - Touchscreen Interface

<u>6.0 Prototyping</u>

6.1 Pre-Amplifier

6.2 Overload Detector

6.3 Equalization

6.4 Power Amplifier

6.5 Power Supply

6.6 Analog Effects

6.7 Digital Effects

6.8 Interface

6.9 Housing

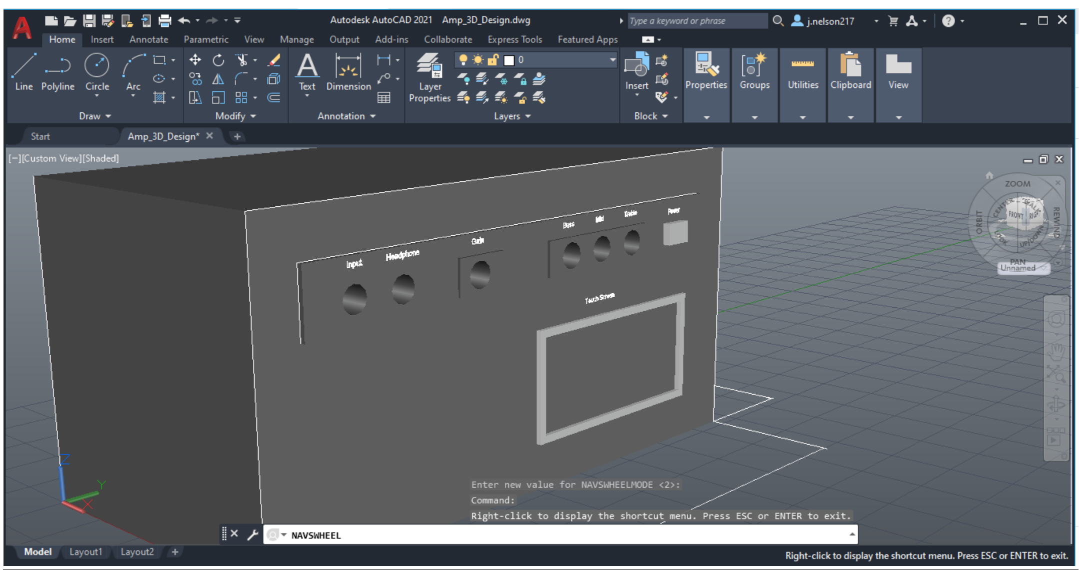


Figure 6.9.a - Prototype Illustration Front

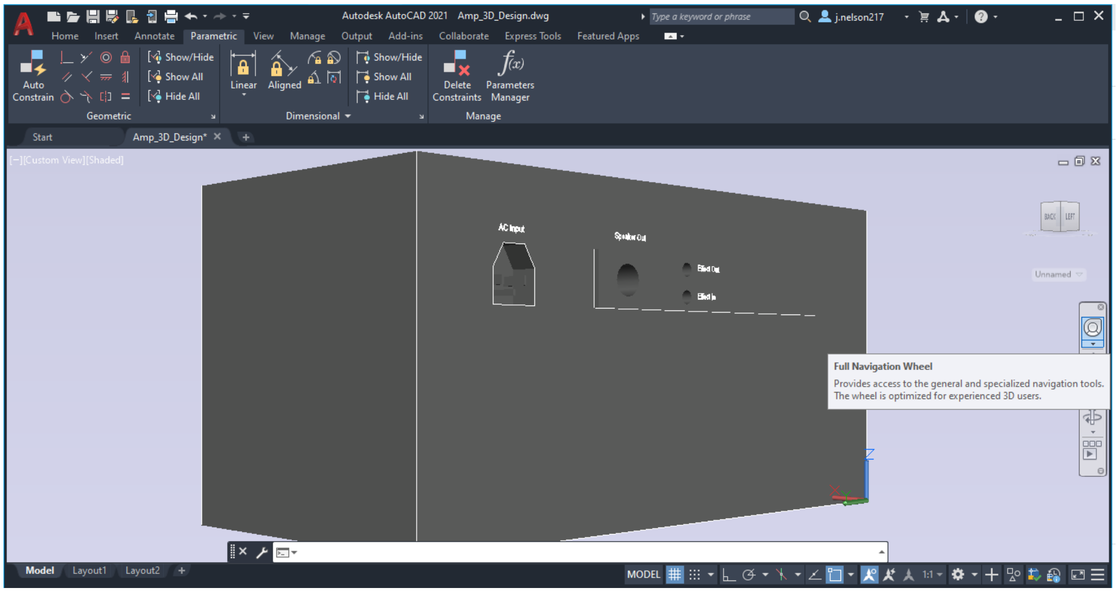


Figure 6.9.b - Prototype Illustration Back

Figure 6.9.b illustrates the front of our prototype, which has a touchscreen and knobs for control. Figure 4 shows the back, including the different connectors the header will require.

7.0 Test Plan

7.1 Pre-Amplifier

7.2 Overload Detector

7.3 Equalization

7.4 Power Amplifier

7.5 Power Supply

7.6 Analog Effects

7.7 Digital Effects

7.8 Interface

8.0 Administration

8.1 Milestones

Senior Design I:

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	In Progress	Kristofer Edstrom/ James Howell
5	Analog Effects Research/Design	Oct 21, 2022	In Progress	James Howell
6	Digital Effects Research/Design	Oct 21, 2022	In Progress	Armon Eghbali
7	User Interface Research Design	Oct 21, 2022	In Progress	Kristofer Edstrom
8	Signal Flow Research/Design	Oct 21, 2022	In Progress	Jeremy Nelson
9	60 Page Draft	Nov 4, 2022	Pending	Group 26
10	Pre Amp/Power Amp PCB Design	Nov 11, 2022	Pending	Kristofer Edstrom/ James Howell
11	Analog Effects PCB Design	Nov 11, 2022	Pending	James Howell
12	DSP/ Microprocessor inclusion to circuit	Nov 11, 2022	Pending	Armon Eghbali/ Kristofer Edstrom
13	GUI Development/ Implementation	Nov 11, 2022	Pending	Kristofer Edstrom

14	Power Delivery PCB Design	Nov 11, 2022	Pending	Jeremy Nelson
15	100 Page Draft	Nov 18, 2022	Pending	Group 26
16	Finish PCB Design	Nov 18, 2022	Pending	Group 26
17	Final Document	Dec 6, 2022	Pending	Group 26

Table 4 - Senior Design I Milestones

Senior Design II:

Milestone	Task	Due	Status	Responsible
1	Order PCB	TBD	Not Started	Group 26
2	Prototype Completed	TBD	Not Started	Group 26
3	Test and Revise	TBD	Not Started	Group 26
4	Senior Design 2 Draft	TBD	Not Started	Group 26
5	Senior Design 2 Final Document	TBD	Not Started	Group 26

Table 5 - Senior Design II Milestones

8.2 Budget

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	1	$35

components)		
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 3 - Estimated Project Budget ($500 dollars expected - Richie)

9.0 References

https://sound-au.com/project152-1.htm
https://www.ti.com/product/OPA1642