Bass Guitar Amplifier with Analog and Digital Effects

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***Abstract* — This paper intends to showcase the design of a cost-effective, yet powerful, bass guitar amplifier designed as a Senior Design Project. Using the TDA7294, the amplifier can deliver 70 watts into an 8Ω load. The Pre-Amplifier and Equalizer stage can adjust the signal to line level and allow for 6 bands of equalization. In addition to this, the design includes 4 Analog Effects and 4 Digital Effects. The showcased design is intended as a lower cost, yet similarly powerful, alternative to other options available on the market.**

***Index Terms* — Analog Design, Class-AB, Digital Design, Equalization, and Power Amplifier**

I. INTRODUCTION

Delving into the world of music can be expensive and daunting due to all the various options on the market. On top of the base product that one will purchase, there are many accessories that can be bought to enhance the instrument playing experience. This overload of information and technology can cause newcomers to be hesitant in trying an instrument out. Our project, an all-in-one bass guitar amplifier, aims to mitigate this fear of committing substantial amounts of money for less of a product, and assists the customer in preventing them from needlessly having to research and purchase separate effect pedals. Moreover, the amplifier will be offered at a more affordable price compared to other products on the market.

II. PROJECT SPECIFICATIONS

This design project will satisfy the following specifications:

- Output Power Rating: 70W @ 8Ω
- Frequency Response: 31 Hz – 4.5kHz
- Total Harmonic Distortion: 2%
- Input Impedance: 1MΩ
- Output Impedance: 8Ω
- Signal to Noise Ratio: 80dB
- Four analog effects
- Four digital effects

III. DESIGN OVERVIEW

The Block Diagram, shown in Figure 1, details the signal path as it progresses through the system. There are four core stages: the Bass Guitar Input, the Pre-Amplifier, the Equalization, and the Power amplifier. Additionally, there are four substages: the Analog Effects, the Digital Effects, the Digital Signal Processor, and the User Interface. The first core stage of Fig. 1 is the bass guitar input. The output of a standard four-string bass guitar is in the range of 10mV to 100mV with a frequency response of 41Hz to 5kHz. The output of a five string would be much the same except for a lower fundamental frequency of 31Hz. The second core component of Fig. 1 is the Pre-Amplifier. The Pre-Amplifier will take the output of the bass guitar and increase the voltage to a range of approximately 1V to 2V. The next stage is the Equalizer. The Equalizer will allow the user to enhance the signal at six specific frequencies by applying a boost or cut. The last core component of Fig. 1 is the Power Amplifier. The Power Amplifier will apply a large current gain so that the signal will be able to drive an 8Ω bass speaker.

The first substage is the Analog Effects. The Analog Effects will consist of four analog filters that will modify the input signal and output to the next stage. These four filters will include: the MXR Dyna Comp, a Fuzz Face, Boss DS1, and Phase 90. The second substage is the Digital Effects. Much like the Analog Effects, this stage will use Digital Filters to modify the input. The digital filters will be generated with the third substage. The third substage is the Digital Signal Processor. Finally, the last substage is the User Interface. The User Interface consists of a ESP32 Microcontroller along with a touch screen used to interface between the various effects.

The last component, not shown in Figure 1, is the Power Supply. The Power Supply will supply each core and substage with the voltages and currents necessary for operation.

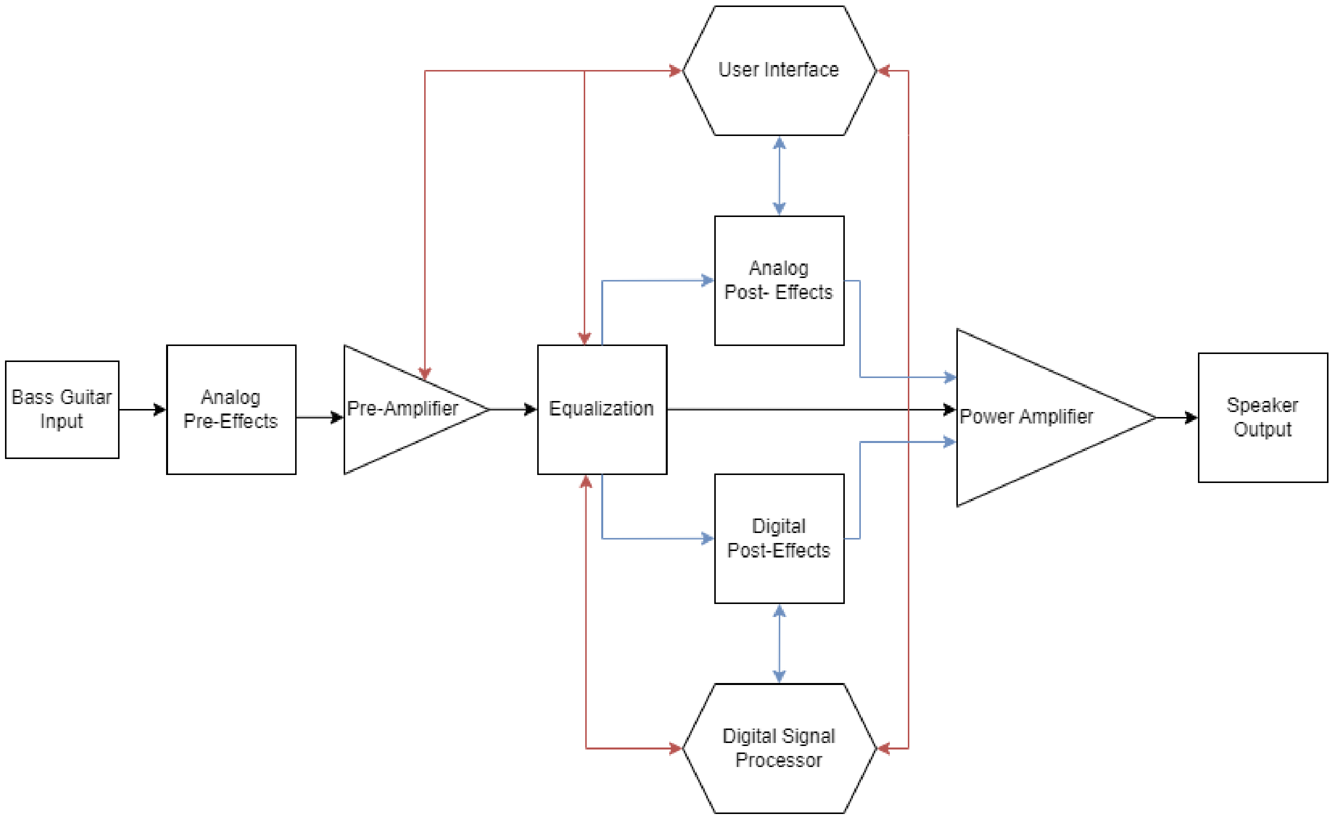

Fig.1: System Block Diagram

IV. SUBSYSTEM ANALYSIS

A. Pre-Amplifier

The Pre-Amplifier design features two cascaded op amp stages. Both stages are primarily focused on applying a voltage gain to the input signal. It is very important for a preamp stage to contribute the least amount of noise to the system as possible because the noise added in this stage will be further amplified in future stages. Texas Instrument's OPA1642 JFET audio op amp met all the necessary conditions for the design and therefore was chosen.

The second op amp stage is another voltage gain stage, but with the ability to adjust the gain. Potentiometer R7 creates a voltage divider between output of the first stage and the non-inverting terminal of the second. Resistors R9 and R10 set the overall voltage gain to 10 V/V. Capacitor C3 is another decoupling capacitor between the output and Potentiometer R11, connecting to the Equalizer Stage. Potentiometers R7 and R11 have been combined into a single, dual gang potentiometer that controls the output gain of the system. Figure 2 below shows the completed schematic for the Pre-Amplifier stage.

The first op amp stage of the Pre-Amplifier is arranged in a non-inverting configuration so that the output can never be less than the input. The very large input impedance by this configuration, as well as the reverse-biased input junction of an JFET helps limit the loading effect on the input signal. In addition to this, Resistor R1 is a current limiting resistor used so that the small amount of current flowing through op amp inputs does not place a load on the signal. Resistors R2, R4, R5, R6, and Capacitor C1 control the voltage gain of this stage. With Switch S1, the gain mode can be swapped between low and high. Capacitor C2 is a decoupling capacitor between the first and second stage.

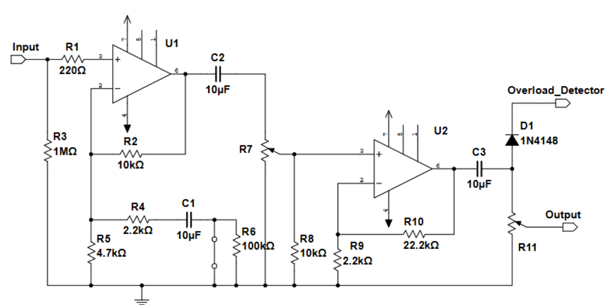

Fig. 2: Pre-Amplifier Schematic

The last component of the Pre-Amplifier is Diode D1. D1 connections the decoupled output of the second stage to the Overload Dectector. The Overload Dectector uses a LM393 Voltage Comparator to detect if the output voltage is larger than 5V. If this is true, a red LED is turned on. Voltages above this 5V threshold could have clipping or distortion and is above the voltage limit for the input of the Power Amplifier. Figure 3 below shows the completed schematic for the Overload Detector.

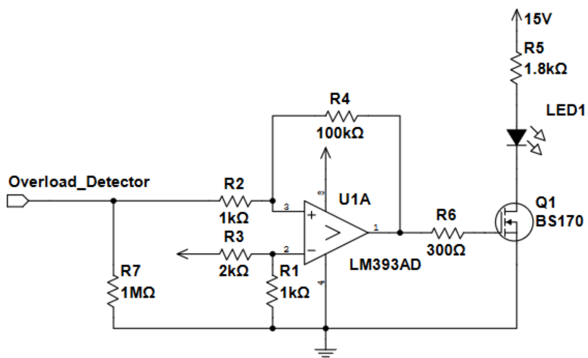

Fig. 3: Overload Detector Schematic

B. Equalizer

The designed Equalizer can be divided into two separate stages. The first being a High Pass Filter and the second being a six band Graphic Equalizer.

The High Pass Filters is a basic second order active filter with a corner frequency targeting 25Hz. The lowest fundamental frequency of 4-string bass guitar is 40Hz and 31Hz for a 5-string. Any frequency under this 31Hz mark is unnecessary for the design and may cause unwanted sound if left unfiltered. The op-amp used in this design was once again the NE5532. Figure 4 below shows the designed High Pass Filter.

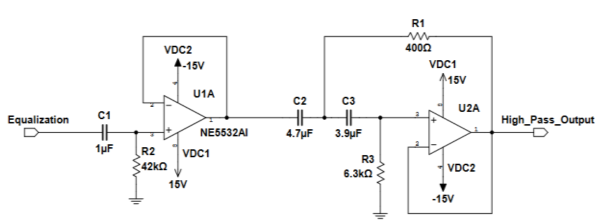

Fig. 4: High Pass Filter

The second stage of the Equalizer is a six band Graphic Equalizer. Each band uses a gyrator notch filter designed to target its specific band frequency. The bands are then added together through a summing op amp to yield the completed equalizer. The targeted frequencies for the six bands are as follows: 80Hz, 250Hz, 500Hz, 1.5kHz, 3kHz, and 5kHz. The formula for the gyrator filter resonance frequency is shown in Formula 1.

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

The Bass band gyrator filter was designed to target a frequency of 80Hz. R1 was fixed to a value of 560Ω. R2 was fixed to a value of 75kΩ. Using (Formula) to calculate the values for the capacitors, the commonly used values for capacitance were tested for what best fit the design. C1 was calculated to be 94nF and C2 was calculated to be 1uF. The design for the Bass band gyrator notch filter is shown in Figure 5 below.

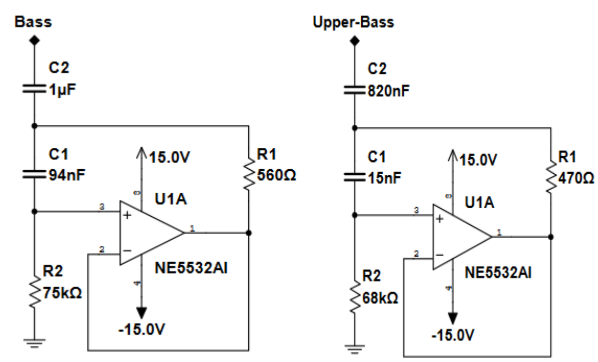

Fig. 5: Gyrator Notch Filter Design Part 1

The Upper-Bass band gyrator filter was designed to target a frequency of 250Hz. R1 was fixed to a value of 470Ω. R2 was fixed to a value of 68kΩ. C1 was calculated to be 15nF and C2 was calculated to be 820nF. The design for the Bass band gyrator notch filter is shown in Figure X above.

The Low-Mid band gyrator filter was designed to target a frequency of 500Hz. R1 was fixed to a value of 470Ω. R2 was fixed to a value of 68kΩ. C1 was calculated to be 8.2nF and C2 was calculated to be 400nF. The Mid band gyrator filter was designed to target a frequency of 1.5kHz. R1 was fixed to a value of 470Ω. R2 was fixed to a value of 68kΩ. C1 was calculated to be 3.5nF and C2 was calculated to be 100nF. The Upper-Mid band gyrator filter was designed to target a frequency of 3kHz. R1 was fixed to a value of 470Ω. R2 was fixed to a value of 62kΩ. C1 was calculated to be 1nF and C2 was calculated to be 100nF. The Mid band gyrator filter was designed to target a frequency of 5kHz. R1 was fixed to a value of 510Ω. R2 was fixed to a value of 68kΩ. C1 was calculated to be 750pF and C2 was calculated to be 39nF. The designs for these bands of gyrator notch filters are shown in Figure 6 below.

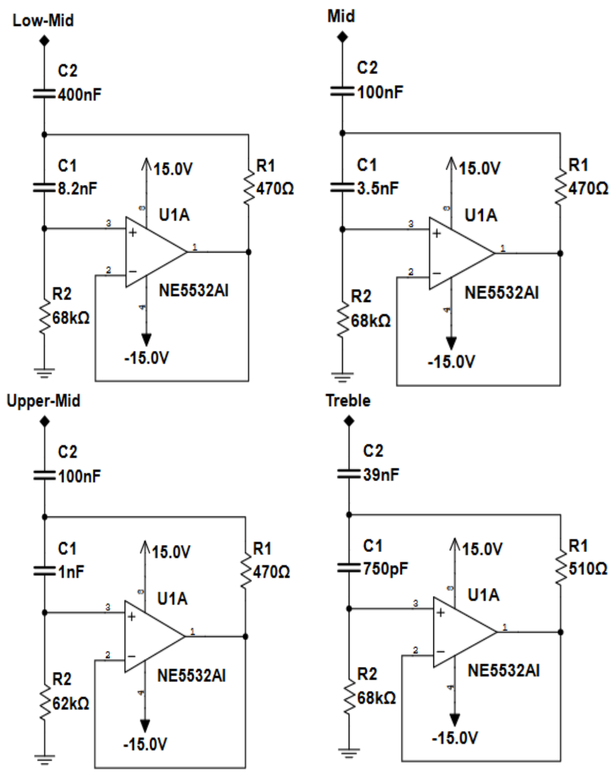

Fig. 6: Gyrator Notch Filter Design Part 2

Each of the six bands are then summed together using a summing op amp with an inverting topology. Due to the nature of notch filters and inverting op amps, the final output waveform is correctly oriented for minimal distortion. To control the boost and cut of each band, the input to the summing stage is controlled by a 10kΩ potentiometer for each band. To prevent the filters from shorting out when a potentiometer is turned completely to the left, a small resistor of 100Ω is inserted in series with each potentiometer. The resistor R9 was chosen to allow for a maximum boost of +12dB and a maximum cut of -12dB. Figure 7 details the complete design for the Input and Summing Stage of the Graphic Equalizer.

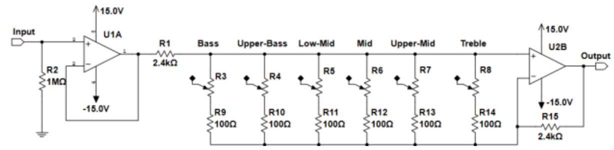

Fig. 7: Input and Summing Stage of Equalizer

C. Analog Effects

The four analog effects used in this project are the compressor effect, phase90 effect, fuzz face and distortion effect. Each of these are staples of guitar effects and each provides a unique tone and opportunities for the bassist.

The Compressor effect attenuates the signal received by reducing the volume of the loud sounds and giving more body to the quiet sound to provide the output with a more uniform volume. The amount the signal is adjusted is called the compression ratio, measured in decibels. 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 be adjusted to be 1dB above the threshold. The compressor is used by adjusting a potentiometer on the pedal.

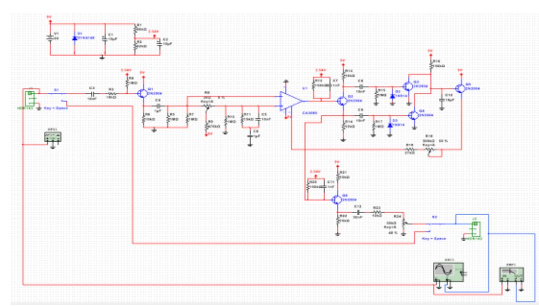

Fig. 8: Compressor Schematic

The distortion effect applies heavy clipping, phase shifts, noise, changes in the frequency response, and other non-linear transformations of the input signal. This clipping effect is achieved by amplifying the input signal well above its maximum gain. Uniquely to the distortion effect, the amount of distortion remains constant to any input signal regardless of the loudness or softness.

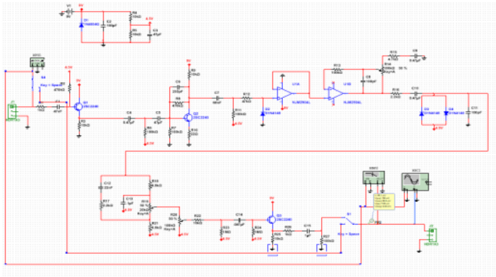

Fig. 9: Distortion Pedal Schematic

Phase shift effects split the input waveform into two copies, one that is identical to the original signal, and the other that is phase shifted 180 degrees. These signals are then recombined and create a ”notch” in the frequency response (a frequency where the signals cancel each other). The MXR Phase 90 has two of these notches that can be swept up and down the frequency domain by varying the potentiometer in the LFO stage of the effect.

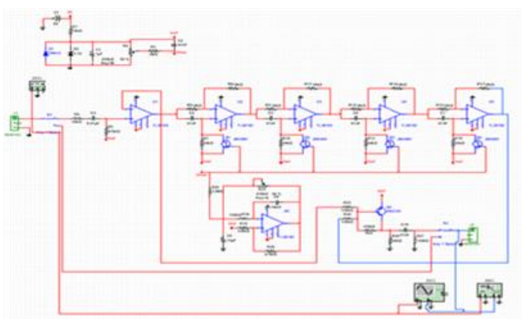

Fig. 10: MXR Phase90 Schematic

The fuzz face effect is similar to the distortion effect, in that it adds clipping, phase shift and wave modulation, but the fuzz face takes this a step further and transforms the input into an imperfect square wave. The sound produced has a lot more buzz than the other effects and is harsher even than the distortion pedal.

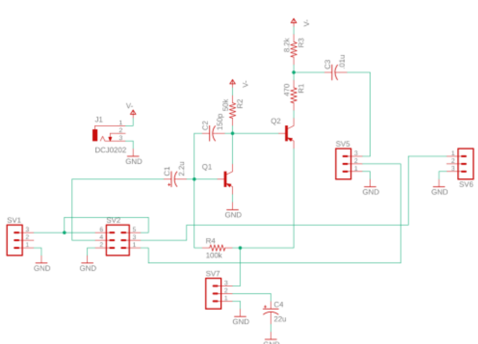

Figure 11: Fuzz Face Schematic

D. Digital Effects

An understanding of required hardware and hardware setup is crucial for proper digital signal processing. The STM32F446RE was chosen for its built-in ADC and DAC along with its use of Direct Memory Access, which frees up the CPU and allows for better performance. There are also enough GPIO pins, which will be used to communicate with the touch screen. The user will be able to toggle the desired effect, which includes echo, flanger, reverberation, and wah-wah.

A consideration to take is the negative values of the incoming audio signal. ADCs cannot deal with negative values. Thus, a DC offset needs to be applied to the input of the ADC and reversed at the DAC output to be sent to the speaker. To achieve this, an inverting op-amp configuration circuit with unity gain can be used to add a DC offset to the incoming signal. The MCU will be powered by a 5V regulated power supply. Meaning that the dynamic range for the ADC will be between 0V-5V. Since there is a 5V supply readily available, it can be used at the non-inverting terminal of the op-amp. This would create an offset of 10V, which is above the ADC” s reference voltage. The 5V should pass through a voltage divider, to provide something closer to 2V, leading to a DC offset of 4. The values of R1 and R2 can be adjusted based on availability of resistor values

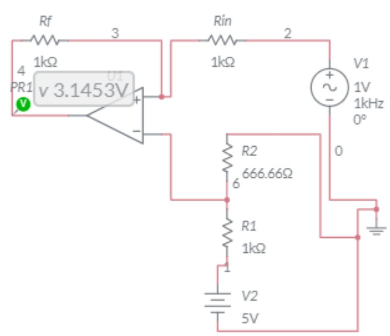

Fig. 12: Input DC bias Circuit

The reverse of this process must take place after the DAC so we can have an accurate reconstruction of the input signal. The same circuit configuration can be used, however the 5V must pass through a unity gain inverting op-amp, yielding –5V. The inverting terminal will see a –2V and the DAC output will be connected to Rin. This results in an output voltage that is V_{DAC} -4V.

The sampling rate of the ADC and DAC conversion is set to 44.1kHz. The ADC and DAC are both triggered by the Timer 2 event, which happens at the overflow at its auto-reload register. Timer 2 is running on the APB1 timer clock which is default at 84MHz. Dividing the default timer clock by the desired sampling speed gives a value of 1904, which is the value placed inside the autoreload register.

A circular buffering scheme is used for the ADC and DAC memory. Circular buffering uses FIFO logic, which is widely used in audio processing applications. Old information is overwritten, and current information is kept in its current location, as opposed to non-circular buffering where elements need to be shifted after a buffer element has been utilized. The buffer size was determined based on the maximum delay needed to implement certain effects. The echo sound effect requires the most samples out of any effect to produce the desired delay, which is a delay of 50ms. At the set sample rate, this would require 2205 samples. In the event a greater delay is needed, more samples are needed. Thus, as safe buffer size is 5000. A half-buffer complete callback and full-buffer complete callback are used to change input and output buffer pointer positions. Meaning only half the buffer is being written to while the other half is used for the data processing.

The Echo effect is a time-based delay that can be realized by the following difference equation, where y(n) is the DAC output, x(n) is the ADC input, alpha is some gain around 0.5, and N being the delay in samples.

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

Fig. 14: Echo Difference Equation

Reverberation is a natural phenomenon that occurs due to reflections of sound waves off different surfaces. Reverberation can be realized by the following difference equation

$$y(n) = (1 - \alpha)x(n) + \alpha x(n - N)$$

Fig. 15: Reverberation Difference Equation

Wah-wah is produced by creating a bandpass filter with a varying center frequency. It can be implemented by the following difference equation. The bandpass filter can be created by cascading a lowpass filter and highpass filter. Where Q1 is the Q-factor and F1 is an oscillating wave of center frequencies.

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

Fig. 14: Highpass filter Equation

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

Fig. 15: Bandpass filter Equation

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

Fig. 16: Lowpass filter Equation

Flanger is created by having a sinusoidal varying delay. It utilizes the same difference equation as reverberation, but N varies sinusoidally. The wave is dependent on the ratio between the LFO frequency and sampling rate.

E. Power Amplifier

The power amplification stage of the design is based on the TDA7294 from STMicroelectronics. This power amplifier can output a maximum of 100W @ 8Ω under the most ideal conditions. It has built in thermal shutdown protection and a mute/standby option. The choice to base the design on this power amplifier came down to how well it compared to other options available.

The supplied voltage to the TDA7294 can be up to a maximum of ± 40V. With the condition that the main rails of our power supply will output ± 35V, we chose to use ± 35V in our design. Based on our supply voltage, the design of the power amplifier will be able to output 70W of continuous RMS power into a load of 8Ω. Given these conditions, the Total Harmonic Distortion of this amplifier is calculated to be 0.5%.

One of the major design elements of any power amplifier is power dissipation. Based on testing and calculations, it was determined that the maximum power that needs to be dissipated would be 40W. Attaching a large, extruded aluminum heatsink to the integrated heat spreader is enough to satisfy these conditions. In addition, the airflow provided by the case fans will improve the heat dissipation.

The designed Power Amplifier circuit is shown below in Figure 17.

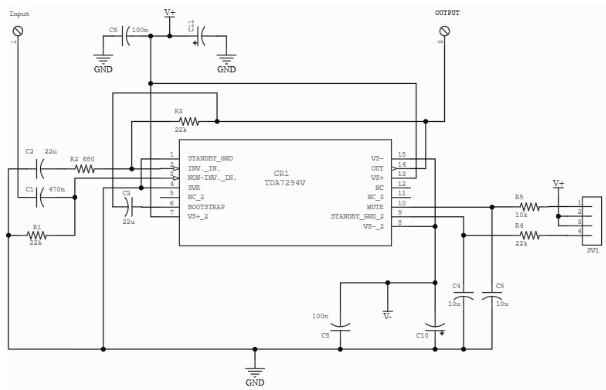

Fig. 17: TDA7294 Power Amplifier Design

F. User Interface

The analog effects will be controlled by potentiometers integrated into each of their respective circuits, such as the bass, mid, and treble of the equalization stage, or the effect strength in one of the analog effect pedals. The digital effects are controlled by inputs from the user to a MSP4022 touchscreen, which is processed by an ESP32 microcontroller. The logic for this is shown in Figure 18, so when a touch input is detected, it will check where the input happened, and will execute certain logic if one of the toggles was selected or if one of the sliders was moved. When the microcontroller processes the input, it will send the data to the DSP unit, which will apply the digital effects to the audio signal.

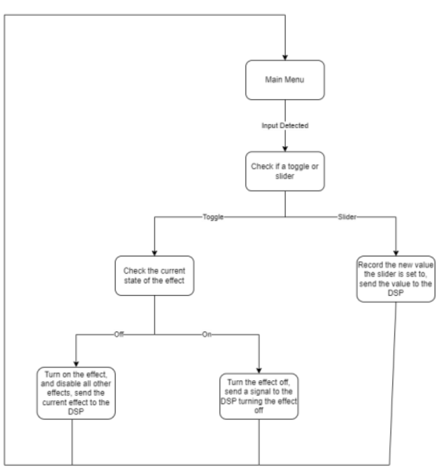

Fig. 18: User Interface Flowchart

G. Power Supply

The design of the power supply was made to be simple, yet effective at supplying power for the rest of the design. The primary winding of the transformer is connected to 115 VAC standard North American residential power. There is a DPST switch in-line to connect and disconnect the circuit from the outlet as necessary. The transformer used was the Hammond 187F36. The 187F36 has a secondary winding of 36 VAC CT with a current rating of 2.8A. The total power rating of this transformer is 100VA. The secondary winding is connected to a full wave bridge rectifier. Capacitors C1 and C2 are the large filter capacitors. These are rated at 1F and a maximum voltage of 50V. Capacitors C3 and C4 are smaller filter capacitors rated at 470uF and a maximum voltage of 50V. C5 and C6 are bypass capacitors and are rated at 0.1uF. The Resistor R1 is a bleeding resistor and is rated at 10kΩ. The main positive rail of the power supply was designed to have a DC voltage of +35V and the main negative rail was designed to have a DC voltage of -35V.

The voltage regulation stage of the power supply uses LM317 and LM337 IC's to supply DC voltages of +15V, -15V, +12V, and +5V. The first LM317, IC1, used 11kΩ and 1kΩ resistors in a voltage divider to bias the adjustment pin to output a voltage of +15V. IC2, an LM337 negative voltage regulator, also used 11kΩ and 1kΩ resistors in a voltage divider to output -15V. IC3, an LM317, used 8.6kΩ and 1kΩ resistors for an output voltage of +12V. IC4, an LM317, used resistors of 3kΩ and 1kΩ for an output voltage of +5V. Each of the LM317 IC's had 0.1uF bypass capacitor at their inputs and a 1uF bypass capacitor at their outputs. The LM337 had an 1uF bypass capacitor at its input and output. Figures 19 and 20 below shows the entire designed circuit for the power supply.

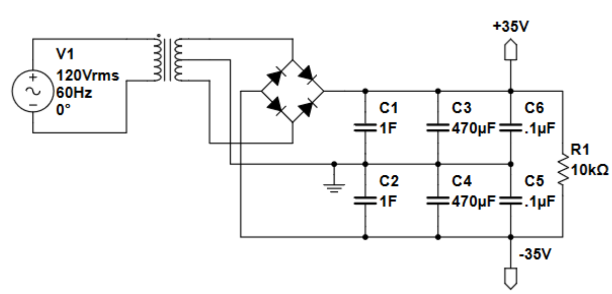

Fig. 19: Power Supply Schematic Part 1

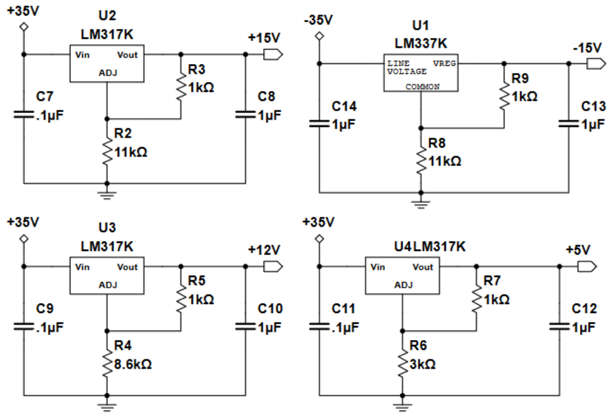

Fig. 20: Power Supply Schematic Part 2

VII. Conclusion

As earlier discussed, the world of music can be expensive and daunting due to all the various options on the market for musical equipment. On top of the base product that one will purchase, the many accessories that can be bought can cause newcomers to be hesitant in trying a new instrument out. The all-in-one bass guitar amplifier lowers the barrier of entry and provides the consumer an avenue to not have to needlessly research and purchase separate effect pedals. The accessibility of this product will allow many newcomers to experience the world of music where they would have otherwise been excluded.

Biography

James Howell is graduating, with honors, from UCF with a Bachelor of Science in Electrical Engineering in May 2023. He has currently accepted a Hardware Design Engineering position at Eizo Rugged Solutions. His hobbies include bowling and fishing.

Kristofer Edstrom is graduating from UCF with a Bachelor of Science in Computer Engineering in Spring of 2023. He is currently working at Eizo Rugged Solutions as a Hardware Design Engineer and will continue his employment with Eizo post-graduation. His interests include retro console repair/modification and reading.

Armon Eghbali is graduating, with honors, from UCF with a Bachelor of Science in Electrical Engineering in May of 2023. He is currently an intern Test Engineer at Lockheed Martin with a full-time offer as a Digital Engineer at Northrup Grumman. Interests of his include sports, weightlifting, music, and being in nature.

Jeremy Nelson is graduating from UCF with a Bachelor of Science in Computer Engineering in Spring of 2023. He is currently looking forward to a summer of peace and quiet. His interests include video games, music, and reading.

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