SAXOPHONE EFFECTS PEDAL

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Final Report for ECE 445, Senior Design, Spring 2021

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May 5, 2021

Team 28

ABSTRACT

The saxophone effects pedal is a portable audio processing unit used by saxophonists to both enhance the quality of sound in live performance and selectively modulate the signal with a variety of effects. The effects pedal is turned on by simply plugging in a standard 9V DC to the DC power jack on the rear plate. When powered, the saxophonist can elect which effects are utilized by clicking a button on the faceplate of the pedal. The effect operating status is determined by a white LED near the respective activator button. The effects are amplification, frequency equalizer, delay, and reverberation. Standard microphone XLR input and output cables are used when integrating this to an audio system for performance.

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1. Introduction

1.1 Objective

Modern music on the top charts does not come close in resembling the music from the early 1950s. While instrumentation and genre affinities contribute to this contrast, a primary difference in why these two eras of music sound unlike one another is due to processing audio with a variety of effects. Musicians often use these effects to change or embellish their sound for the benefit of adding tonal character or atmosphere to the musical piece. Nowadays, modern effects units used in performance are user-friendly, easily adjustable, and compact.

Quality effect pedals on the market specifically designed for saxophone players do not exist. The portable effects market only caters toward electric guitar and bass players. Circuit designs in guitar pedals do not match the appropriate type of input necessary for microphones needed by saxophonists. Not only does inconvenience occur in trying to find a matching microphone or adapter, but the signal quality will decrease. The bigger problem with utilizing these pedals is they do not emphasize the correct frequencies needed to make a saxophone sit well in a performance. Oftentimes, these pedals make the saxophone sound “tinny” or thin.

We propose building a multi effects pedal designed specifically for saxophone. Since most microphones use a balanced output, we will implement circuit designs for that type of input. Inside our pedal, we will implement a preamp and equalizer that will solve the aforementioned “tinniness” by emphasizing the important frequencies of a saxophone. The user can elect whether to utilize the practical performance effects, delay and reverb. As a result of the time, cost, and complexity to model distortion in the digital domain, we will design the EQ and preamp in analog circuitry while using digital signal processing for the delay and reverb effects.

1.2 Background

There are currently no effect pedals dedicated to saxophones despite a strong desire on the market. Well-known YouTube saxophonists such as Mark Maxwell, commonly referred to as Dr.Saxlove, and Chez Taylor both affirmed the desire for a saxophone pedal. Taylor told us “It would be so great to have a decent pedal which caters specifically to the Sax" while Maxwell commented “I, too, have tried many guitar pedals over the years and have found them lacking”. The critical acclaim does not end there as seasoned professional Dick Oatts said, “I do feel this would give young saxophonists an edge in more electric bands” and this product has the potential to make a saxophonist “more versatile on the job market”.

Using guitar-oriented pedals in place of a specifically tailored effects unit means using adapters or having a limited microphone choice. This “workaround” does not bridge the gap as will be discussed. Most microphones use a three-prong connection called XLR while guitars use a single quarter-inch input jack. Not only is this an inconvenience to find a suitable microphone or an adapter for the pedals, but there is also a quality drop in the signal. Unbalanced cables, such as the standard quarter-inch TS input jack used in guitars, do not cancel out unwanted noise like balanced XLR cables [1]. Furthermore, performers using a XLR to quarter-inch converter will experience a drastic tone change due to improperly matched impedance

Now, there are pedals on the market with microphone inputs made for vocals, but the problem is those circuits are designed for emphasizing typical vocal frequencies, not saxophone. Running a saxophone or other instrument through these often results in a comedic sound unfit for live performance. This is mostly because the upper harmonic frequency range of the saxophone is not accounted for in these devices as well as not properly sculpting the lower end. We are convinced from these observations there is both a need and a want in the market for a product that solves this problem.

1.3 High-Level Requirements

- The pedal will have at least 0.5 seconds max delay spacing and reverberation effects of up to 3 seconds of reflections.
- The equalizer circuit will be able to change the magnitude of frequency bands located at 36Hz, 160Hz, 720Hz, 3.2kHz, and 14kHz.
- The amplifier circuit will amplify the incoming signal by at least 20 dB.

2. Design

2.1 Block Diagram

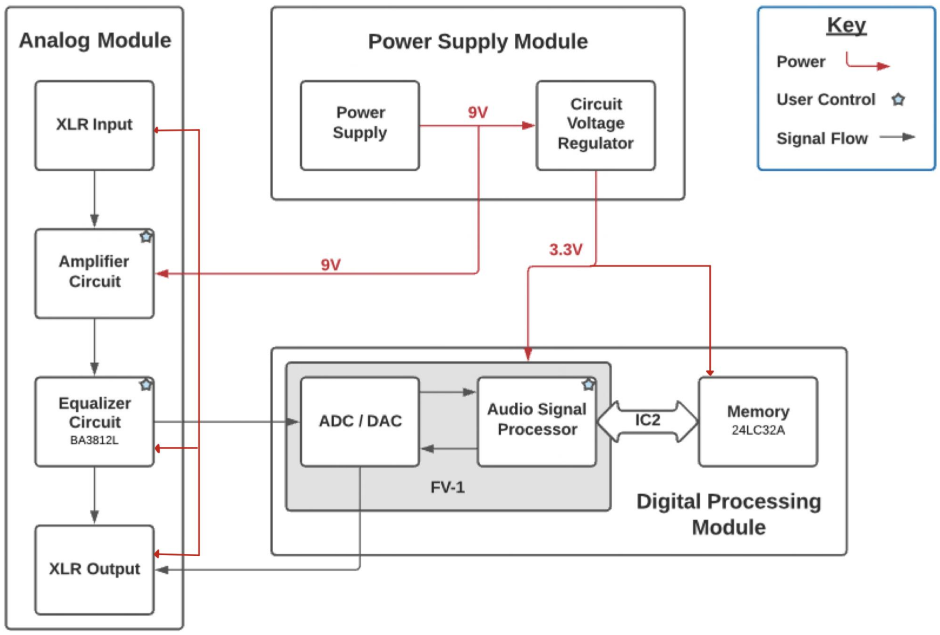
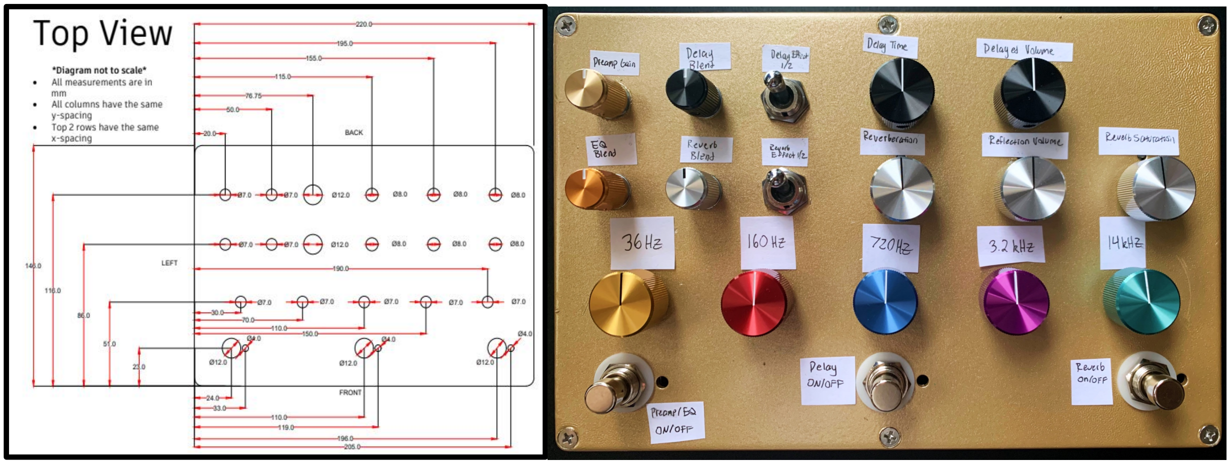


Fig. 1. High-level block diagram.

The signal from the microphone enters the pedal through the XLR input circuit. This circuit takes sound from the microphone and eliminates the noise before amplification in the amplifier circuit. The amplifier circuit will take the weak signal and raise the voltage to audible levels by increasing the amplitude by at least 20dB. From there, the amplified signal will be sculpted in the frequency domain by the equalizer (EQ) circuit. The EQ circuit has its five frequency bands centered around 36Hz, 160Hz, 720Hz, 3.2kHz, and 14kHz to best accommodate a saxophone's harmonics. These frequency amplitudes will be controlled by potentiometers.

Next, the newly shaped signal will be converted from analog to digital to enable manipulation by the effects processor which runs user created programs from the memory chip. There will first be a delay effect that will have adjustable duration, repeats, and blend. The second effect will be a reverberation effect that will not only have adjustable duration and blend, but also adjustable brightness to the reverberations. After the effects have been processed, they are then converted back into analog and mixed with a portion of the analog signal from the EQ circuit. This mixing is controlled by a potentiometer. The resulting output is then sent through the XLR output circuit which packages the signal and sends it out of the pedal.

2.2 Physical Design



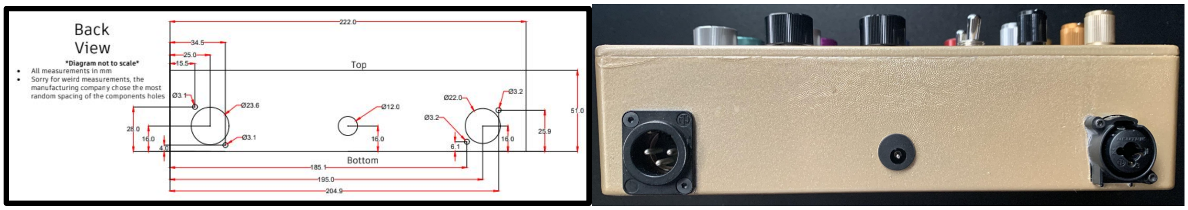
(a) Faceplate Schematic (b) Faceplate Implementation

Fig. 2. Faceplate physical design of Saxophone Effects Pedal

Our design allows the user to conveniently adjust and elect effects both before and during a performance. We precisely measured and modeled the layout of the faceplate (Fig. 2a) to best maximize total area between the knobs and switches. This was done to improve the ease of turning the knobs without accidentally moving the position of another one, thus unintendedly changing the effect.

As seen on the implementation of the design (Fig. 2b), the saxophone effects pedal was color coordinated and partitioned to group effects together. The top row of black knobs represents the delay, the middle row of silver knobs represents the reverb, and the bottom row of multi-colored knobs represents the equalizer. On the top left of the design there are the blend knobs whose colors correspond to their respective effects. This was all done to help the performer group together the signal chain, making it easier to understand the pedal.

We chose a metal casing for the saxophone effects pedal as it is extremely durable to accommodate for the intended, rugged stage use. The metal casing also provides added support when using the stomping switches on the bottom of the design. These switches allow the user to turn off and on the respective effects by use of their foot. This is advantageous as it allows the user to be able to switch up effects in the middle of a performance.



(a) Backplate Schematic (b) Backplate Implementation

Fig. 3. Backplate physical design of Saxophone Effects Pedal

As for the back of the saxophone effects pedal, great care was taken to ensure that the XLR input and output jacks were not colliding with the potentiometers up on the face plate. We modeled out the dimensions of the backplate (Fig. 3a) to ensure no boundary conflicts arose. In the implementation of the device (Fig. 3b), we decided to have the flow from input to output go from left to right to allow for standard integration with other devices such as a mixer or powered speaker.

2.3 Block Descriptions

Subsystem 1: Power Supply Module

The power supply module takes in power from a regular household outlet and supplies a 9V DC output voltage to the voltage regulator, which is then converted to the appropriate voltage to power each of the chips and transistors in the rest of the circuit.

9V Power Adapter

We used a 9V DC power adapter to power our system. The adapter gets power directly from the outlet and steps down the voltage from the 120 V AC to 9V DC. We chose to power the project this way because the use of the pedal requires other equipment like microphones and speakers which means that an outlet should always be available if the pedal is going to be used. Powering the project directly from the outlet eliminates the concern of worsening performance as a battery drains.

Linear Voltage Regulator

The voltage regulator stepped down the voltage from 9V DC to 3.3 V DC which is required by the signal processing chip as well as the memory chips. We chose a linear voltage regulator over a buck converter because a buck converter has a switching frequency that would introduce a lot of noise in an audio circuit. The TPS7A24 voltage regulator is designed to be low noise and low drift. The voltage regulator can also take up to 18 V DC at the input and step it down to 3.3 V DC, which allows us to use it in a safe region of operation.

Subsystem 2: Digital Processing Module

The digital signal processing module, as seen below in Fig. 4, alters the audio input to create the delay and reverb effects. The audio signal is delivered by the equalizer circuit as an analog signal to the analog-to-digital converter, which converts the analog signal into a digital signal. This signal is then sent to the audio signal processor which alters the signal to produce the desired delay and reverb effects. The code that is used to produce the delay and reverb effects are stored in the memory. The memory loads the program into the audio signal processor using I2C. The output of the audio signal processor is then converted back into an analog signal by the digital-to-analog converter and sent to the XLR output.

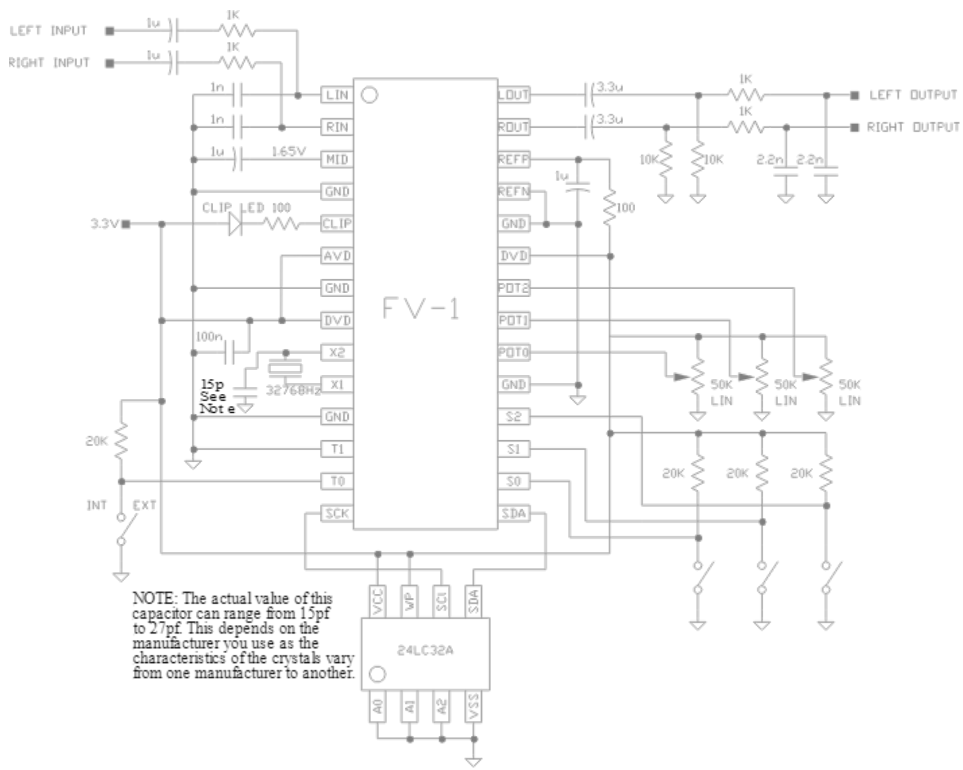


Fig. 4. Schematic of Audio Signal Processor (FV-1) connected to the memory (24LC32A) [2].

Analog-to-Digital & Digital-to-Analog Converter

The analog-to-digital converter was used to turn the analog signal coming from the equalizer into a digital signal which the audio processor can then manipulate to achieve the desired effect. The digital-to-analog converter was used to convert the output of the DSP from a digital to an analog signal, which then went to the XLR output. The FV-1 chip has both the analog-to-digital and digital-to-analog converter built into its architecture. We used a crystal that oscillates at 32kHz for our clock. This sets the sample rate that we used for the conversion.

Audio Signal Processor

We chose the FV-1 chip as our audio signal processor due to previous documentation and convenient mounting compared to other digital signal processors on the market. This chip does not only process the signal with effects but is also able to perform both the analog-to-digital conversion and digital-to-analog conversion mentioned in the prior section. By using an integrated system such as the FV-1 chip, this allows us to focus more on the effect algorithm quality. Upon our research, we concluded it would cost too much money and time to create a system from the ground up.

The audio signal processor takes the converted signal and performs the necessary computations to apply the desired effects. The audio processor loads the programs that run the effects from the memory using the I2C communication protocol. Since we designed the FV-1 chip to use a 32kHz crystal oscillator as a clock, the code will execute 32768 times per second.

There are two audio signal processors, one for the reverb and one for the delay. We added switches to the faceplate of the saxophone effects pedal to let the user switch between two reverberation effects and two delay effects. These effects can be turned on and off by use of the footswitch on the bottom of the faceplate. The delay effects have two potentiometers connected to its respective processor. One of the potentiometers is used for time of the delay while the other one is for volume. The reverb effects have three potentiometers. They are used for the amount of reverberations, pre delay, and volume of delay.

Memory

The memory used in the digital effects system is an EEPROM chip that stored the code for both our delay and reverb effects. To retrieve our algorithm designs for application use, the memory communicates with the audio signal processor using the I2C communication protocol. We encoded the memory by using an integrated circuit memory programmer. We chose the 24LC32A chip because our audio signal processor was designed to work specifically with this chip. The memory can store up to eight different programs at once, which is more than enough since we only need two, delay and reverb. We implemented the connection of the memory to the audio signal processor as shown in Fig. 4.

Software

The FV-1 uses its own type of assembly language. Since the FV-1 uses digital signal processing techniques, the chip runs every line of code on each clock cycle. To create any type of effect, the processor performs different algebraic equations on the input frequency. For the delay effect, some of the sound partials are allocated to memory which the FV-1 chip then uses later for delay and reverberations.

There are two types of delay that we implemented, single delay and wet delay. The single delay stored the audio signal in the delay memory. It then recalls it from 0s to 1s later depending on the current state of the potentiometer. The FV-1 chip then combines this delayed audio with the current audio to create the single delay. The wet delay works very similar to the single delay. The wet delay stores the audio signal in the delay memory. Then it pulls from multiple different points in the delay memory to create a smoother type of delay. The potentiometer also chooses how long the delay should be from 0 to 1 second.

As for the reverb effect, the two types of reverb that we implemented are hall reverb and room reverb. First, the audio is stored in the delay memory. Then, different points in the delay memory are taken out and are added and multiplied with the current signal before mixing them together. This is then modulated by sinusoidal waves to make a spacey, echoey sound. These waves are played with the current audio and put back into the delay memory. The potentiometers choose how loud the delayed memory should be, thus making the reverberations longer or shorter. The main difference between hall and room reverb is the number of sinusoidal waves the audio is modulated by. The hall reverb goes through more sinusoidal wave modulation to give it a more spacious sound to the point where it sounds like the signal is being played in a concert hall. The room reverb goes through less sinusoidal wave modulation to make it sound like it is being played in a relatively small room.

Subsystem 3: Analog Module

The analog module is responsible for taking the input from the microphone and using the XLR audio to eliminate the noise from the microphone. This analog signal is then put through an amplifier circuit to boost the signal voltage. After amplification, the signal passes through an equalizer circuit which has five frequency bands that the user will be able to adjust. This will emphasize key frequencies that sound best on a saxophone. The five bands will be adjusted by potentiometers that the user can control manually. After the analog signal has been modified, the signal will be passed to the analog-to-digital converters so that the signal processors can then apply the effects.

XLR Input

The XLR input is the part of the circuit that first receives the signal from the microphone wire. XLR wires have positive, negative, and ground signals. The positive and negative versions of the signal are used in a differential op-amp circuit to cancel the induced noise from the microphone wire. Fig. 5 shows a typical XLR balanced input circuit. We chose the XLR connection over a tip sleeve type connection because the XLR connection creates less distortion over longer distances. XLR input also makes the pedal more compatible with different types of microphones on the market.

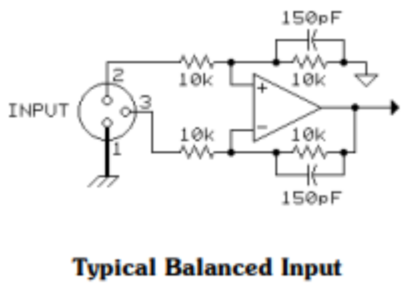


Fig. 5. Typical balanced input for XLR [3].

Amplifier Circuit

In the signal chain, the amplifier circuit comes after the XLR input and before the equalizer circuit. It takes the analog signal that comes from the microphone and increases the voltage amplitude. Microphone transducers typically produce low voltages around 1mV to 10mV, so the amplifier circuit was used to increase the amplitude of the signal to ranges processable by the digital signal processor.

Originally, one aspect that we wanted to incorporate into this design was a transformer coupled input to give the amplifier a nice saturation component often associated as a “warmer” sound. The reason we did not follow through with this design choice because transformers introduce a significant amount of noise to the system if not handled properly. Upon looking into amplifier designs with transformer coupling, we quickly realized we did not have a strong enough background nor the time to successfully create an amplifier that would address this issue. Instead, we turned to using a BJT base Class-A amplifier for our design. The basic design of our amplifier will be modeled after a Class-A amplifier as shown in Fig. 6. This design was chosen over other topologies like the Class-AB or Class-B due to the Class-A amplifier offering less noise contribution. This protects the fidelity of the signal.

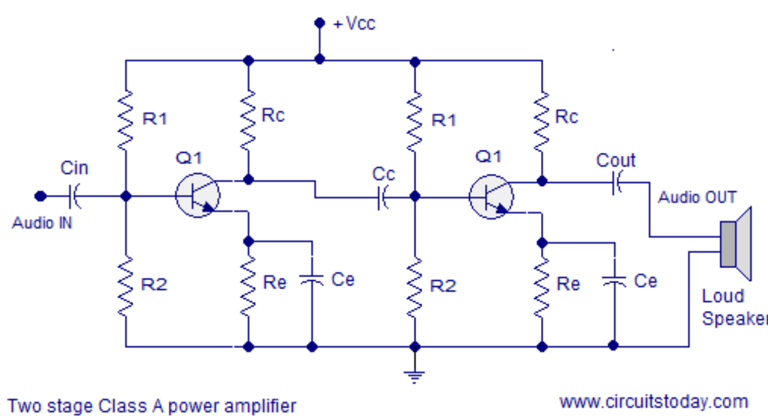


Fig. 6. Class-A amplifier circuit schematic [4].

Equalizer Circuit

The equalizer circuit modifies the analog signal after it has been amplified by the amplifier circuit. The equalizer is an essential component of the project because we used it to optimize the frequencies that are characteristic of the saxophone. The equalizer circuit uses the BA3812L integrated circuit shown in Fig. 7. The equalizer circuit has five frequency bands that will help modify and shape the sound as the user pleases. The five frequency bands allow selections of frequencies in low, low-mid, mid, mid-high, and high ranges. Each frequency band has a corresponding potentiometer knob to adjust it. The modified signal is then sent to the FV-1 chip. We chose this chip because it helped to substantially minimize the area we needed on a PCB. We chose to use film capacitors to center the frequency bands because they introduce less noise and distortion compared to electrolytic capacitors.

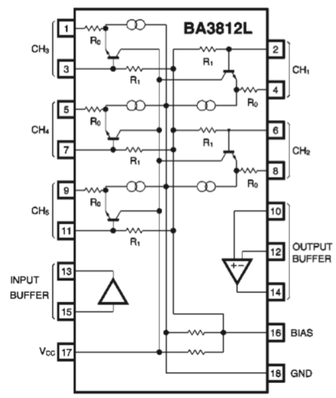


Fig. 7. Circuit schematic equalizer amplification circuit [5].

XLR Output

The XLR Output is the part of the circuit that receives the analog signal after it has been converted by the DAC (post signal processing). The XLR is necessary to transmit a low-noise copy of the processed audio to the speaker. The speaker has an XLR input to receive the audio. Fig. 8 shows a typical design for an XLR balanced output circuit. We used an XLR output type connection because powered speakers generally have XLR input connections. This makes the pedal compatible with speakers that are used for live performance.

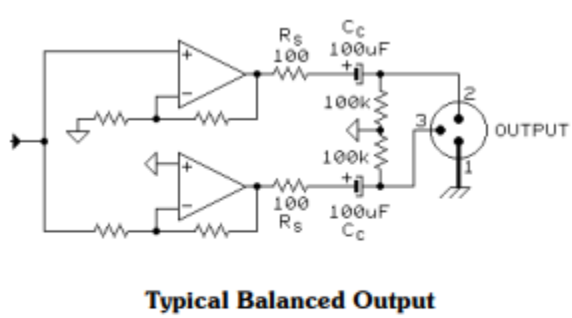
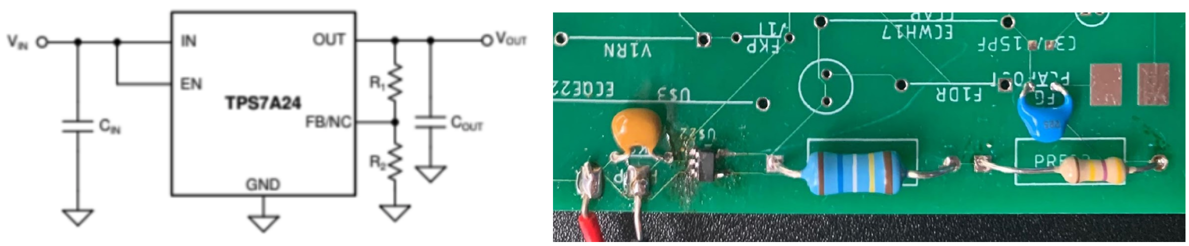


Fig. 8. Typical balanced output for XLR [3].

3. Verification

Subsystem 1: Power Supply Module

The power supply has two main components to it which include the power adapter and a linear voltage regulator. We tested the power adapter by connecting it directly to the wall socket in the lab. Although the power adapter is designed to provide 9V DC, we measured the power adapter with the oscilloscope and found that it was supplying 9.43 V DC. Then we tested the linear voltage regulator by providing the chip with a 9V DC input signal and measuring the output with an oscilloscope. The voltage regulator worked ideally when it was isolated from the rest of the components on the board. The regulator delivered 3.29 V DC at the output. We isolated the voltage regulator circuit by soldering it on the PCB first, along with the accompanying resistors and capacitors shown in Fig. 9 below.



(a) Schematic of voltage regulator [6]. (b) PCB with voltage regulator isolated.

Fig. 9. Schematic and PCB of the power supply module.

Unfortunately, when we finally integrated the voltage regulator circuit with the rest of the digital processing board, the voltage regulator blew. We tested the digital processing board by supplying 9 V DC from the power supply in the lab, but when we connected everything, the voltage regulator stopped working. While the power was live, we measured with the oscilloscope and saw that the voltage regulator was measuring 7.24 V DC at the output. After unplugging and retesting the power circuit, on a separate PCB, we confirmed that the chip was no longer functioning. We had spare voltage regulators, but we ran into the same issue.

One possible cause for the failure was that there was a thermal overload of the voltage regulator which caused the internal circuitry to fry. This is unlikely because the voltage regulator is designed to take up to 18 V DC. Another possible cause of failure could be found in the remaining components that are a part of the digital processing boards. There are 14 potentiometers and several capacitors. If one of those components happened to be faulty, it would have resulted in a short to ground. This would have exceeded the power requirements of the chip and resulted in a blowout. If we had further time to test our components in the laboratory, we would continue to test each one individually and integrate them more slowly.

Subsystem 2: Digital Processing Module

The three main components to the digital processing module are the analog-to-digital and digital-to-analog converters (ADC-DAC) converters, the audio signal processor, and the memory. We were unfortunately unable to verify the ADC-DAC converter's requirements since the system was fully dependent upon the power supply functioning correctly. If the power system were to have operated correctly, we believe the ADC-DAC would have passed the requirements if we were able to test them. We believe this as we designed the sampling rate to be around 16kHz using a 32kHz crystal oscillator.

We verified the audio signal processor effect capabilities with the FV-1 simulator called SpinCAD. An input audio file of a 1.5s beep was used to test the hall reverb and single delay. The delay and reverb times were set to maximum, and the results are shown below in figure 10 and 11 respectively. In both figures, the bottom wave is the original audio, and the top is the effect wave. In the delay effect, the delay audio starts exactly one second after the original. This verifies the requirement of at least half a second of delay. As for the reverb effect, the reverberated wave still has a tangible amplitude after ten seconds of propagation. This verifies the requirement that there needs to be at least three seconds of reverberations.

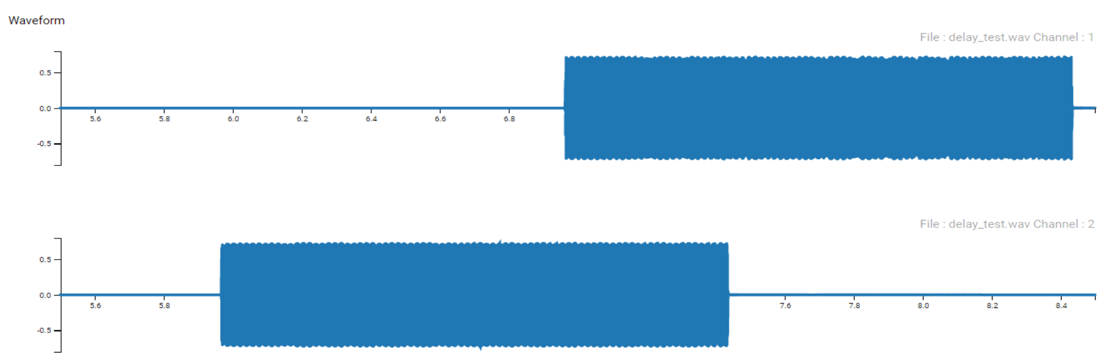


Fig. 10. Simple delay audio wave.

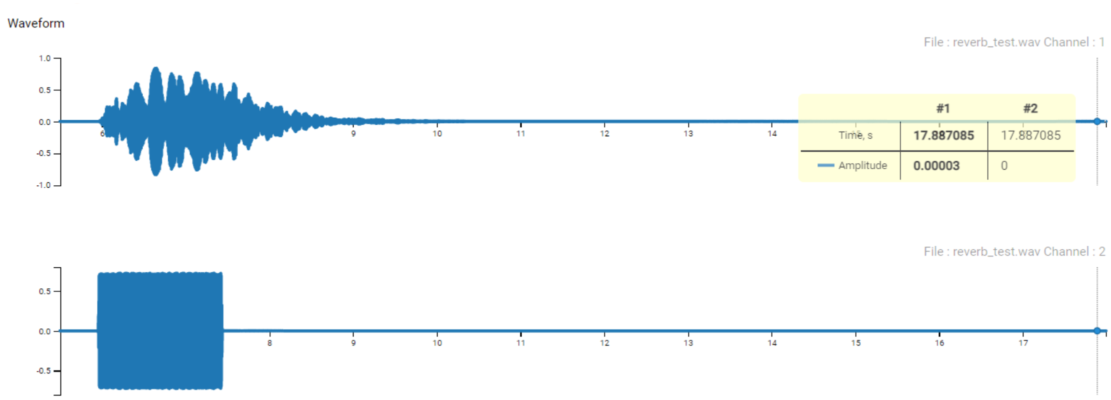


Fig. 11. Hall reverb audio wave.

The memory has a requirement of being able to have at least 256 bits of writable memory. Each instruction is two hex characters meaning each instruction is eight bits each. The code we wrote was thirty-two instructions since which translates to 256 bits of memory. We used the chip programmer to write our code to the memory chip. After clearing the simulator, we executed a command to read what was on the chip and store it on the computer's memory. Using this method, we were able to verify a successfully read algorithm from the memory chip.

Subsystem 3: Analog Module

The preamplifier, equalizer, XLR input, and XLR output are the main systems within the analog module. Since we were not able to fully integrate the system together, testing was done on a per system basis. For the preamplifier shown below in Fig. 12, we modeled a typical microphone input by sending a sinusoidal wave with an amplitude of 10mV to the input of the preamplifier.

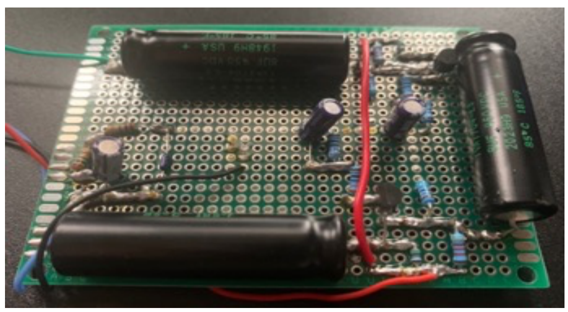


Fig. 12. Preamplifier implemented on a prototype board

Then, we probed both the input and output of the preamplifier at maximum gain conditions. Upon taking down the values for the input and output, we then calculated the gain of the system. For our implementation of the preamplifier, we had a gain of 21.8 dB, which exceeds our high-level requirement of a maximum gain of at least 20 dB. With this amount of gain, our preamp will never leave our predetermined voltage threshold range of -500mV to 500mV.

The equalizer system shown below in Fig. 13 was verified through analyzing the amplitude changes in the Fourier transform on an oscilloscope.

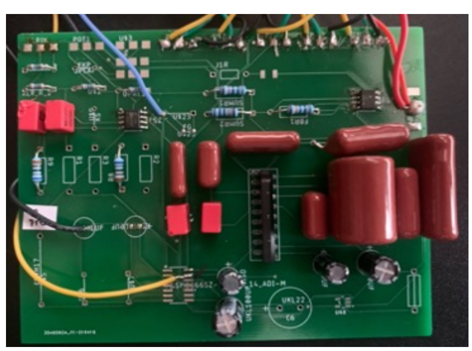


Fig. 13. Equalizer implementation on a PCB

To analyze the Fourier transform, a sinusoidal signal of an amplitude around 100mV was sent to the input of the equalizer. To test each of the high-level requirement frequencies, the sinusoid was set to the respective frequency being tested. From there, on the oscilloscope the output is probed, and a mathematical function was performed to calculate the Fourier transform. Once that was achieved, we varied the value of the potentiometer to see the range of the amplification or attenuation. From our results, we found that we had a 12.5 dB range of both amplification and attenuation at each of our high-level requirement frequencies. This exceeds the minimum amplification of at least 12 dB.

We could not fully verify the functionality of the XLR input and output circuits because we ran out of time in the lab. The main cause of this is because we were still waiting for the 3rd round order PCB which never got delivered to us. It would have been difficult to test the circuits on a breadboard because the main components were surface mount devices. If we had wired connections to each of the pins, there would have been too much noise introduced to establish correct functionality. If we had received the 3rd round PCB in time, we would have soldered the components on board and tested the circuit with a microphone input in the lab.

4. Cost Analysis and Schedule

The total cost of our project is shown below:

Total Cost = Cost of Parts + Cost of Labor = \$287 + \$33,750 = \$34,037 (4.1)

4.1 Final Costs of Parts

The final cost of all the necessary components used to complete this project can be found in Appendix D. In total, the cost to produce the saxophone effects pedal is approximately \$287. However, it should be noted that this number does not include shipping costs as well as duplicate orders for backup copies

4.2 Final Costs of Labor

$$3\,Engineers \times \frac{\$45}{hour} \times \frac{10hrs}{week} \times 10weeks \times 2.5 = \$33{,}750 \tag{4.2}$$

Using equation (4.2), the projected labor cost for this project is \$33,750 for three people working across a 10-week working period at a rate of forty-five dollars per hour. The 2.5 multiplier in the equation is used to account for project overhead.

4.3 Schedule

Week	Eliseo Navarrete	Peter Hevrdejs	Sean McGee
3/8	Design XLR input/output.	Design equalizer and pre-amplifier.	Create initial delay and reverberation effect algorithms.
3/15	Assist in prototype and debug circuit on breadboard.	Prototype and debug circuit on breadboard.	Program effects onto EEPROM and prototype effects circuit.

3/22	Design and order 1st PCB layout.	Assist in PCB layout design.	Create enhanced delay effect algorithms.
3/29	Assist in testing and verification.	Test and analyze circuit requirements. Make revisions as necessary.	Create enhanced reverberation effect algorithms.
4/5	Design metal enclosure and integration methods.	Design and order 2nd PCB layout.	Analyze and verify effects unit requirements.
4/12	Analyze and format data for presentation.	Assist in project finalization.	Solder components onto PCB and install into the enclosure.
4/19	Practice demo and presentation.	Practice demo and presentation.	Practice demo and presentation.
4/26	Prepare for final demonstration.	Prepare for final demonstration.	Prepare for final demonstration.
5/3	Write the final paper.	Write the final paper.	Write the final paper.

5. Conclusions

5.1 Achievements

While we were not able to meet all our goals given the time constraints and shipping delays, several key systems in our design were still successfully implemented. For instance, our device met all our high-level requirements, containing the core functionality of the project, when tested in their respective modules. This included the preamplifier having a maximum gain that exceeded 20 dB while also not being too high to amplify the signal voltage past the threshold range. The equalizer subsystem was successfully able to attenuate or amplify the predetermined frequencies by a factor of 12.5 dB. As for the digital effects system, while we were not able to implement the effect on the board, the signal could still be processed through a computer simulation to demonstrate the effects meeting their respective requirements. All the circuit boards can be successfully housed within our fabricated enclosure that strategically laid out the signal chain.

5.2 Uncertainties

The system integration aspect of our design was not tested due to the late arrival of key components. There was not enough time to implement all of our designs onto the PCB let alone verify the correct functionality before the demonstration. Since some key components of the system could not be verified, we decided to demonstrate each subsystem separately to avoid potential catastrophic failure. This leaves the uncertainty on whether the combined systems would work together as a whole.

The analog and digital systems were ordered on several different sets of PCBs. While having multiple PCBs in one box was fine for our testing purposes, we are unsure of the long-term durability when considering rugged use. The largest concern to this approach is interconnecting wires between boards being severed. If one board becomes unmounted from the fasteners, it could move around in the box and accidentally rip off a wire from a pad.

5.3 Future Improvements

There are still a few improvements that need to be made to make the saxophone effects pedal a more marketable device. The first improvement to be made is to implement the XLR input and output subsystems onto the board. As of right now, the only signal the board can modulate is from a waveform generator. Since this is not the intended input, not having the fully working subsystems hinders the intended utility of the product. The second improvement is to find a more reliable power scaler for the digital effects circuit. Improving this will ensure the power overloading problem we encountered will not occur again.

As for improvements to enhance the product from a manufacturing, moving the components all to one board will reduce the size of the overall layout as well as increase reliability. In addition to moving to one board, the other benefit of this is a smaller enclosure can be used to improve the portability aspects of the design. Another improvement to allow for higher throughput of devices is to replace the wire pads with through holes to allow for faster soldering and reliable solder joints.

As for future features for the convenience of the saxophonist, one attribute that could be added to this design is phantom power. Phantom power allows for a broader range of microphones to be able to use the pedal. This is achieved by sending 48V back up the XLR cable to power the microphone so that it may transmit a signal back down the other two lines.

5.4 Ethical Considerations

Our team strives to, "improve the understanding by individuals and society… of emerging technologies" [7]. As per the IEEE code of ethics, we want everyone to be able to use our device. The core purpose of the saxophone effects pedal is to increase creative avenues and inspire new art while demonstrating what modern technology allows us to do. We will include a manual on how to properly use our device along with the OSHA dB chart. As a group we have held each member accountable to maintain the IEEE code of ethics. It is our responsibility, "to support colleagues and co-workers in following this code of ethics" [7].

References

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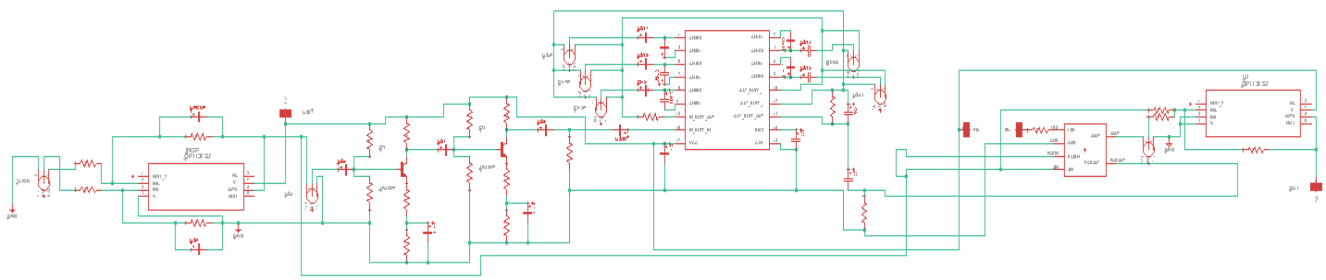
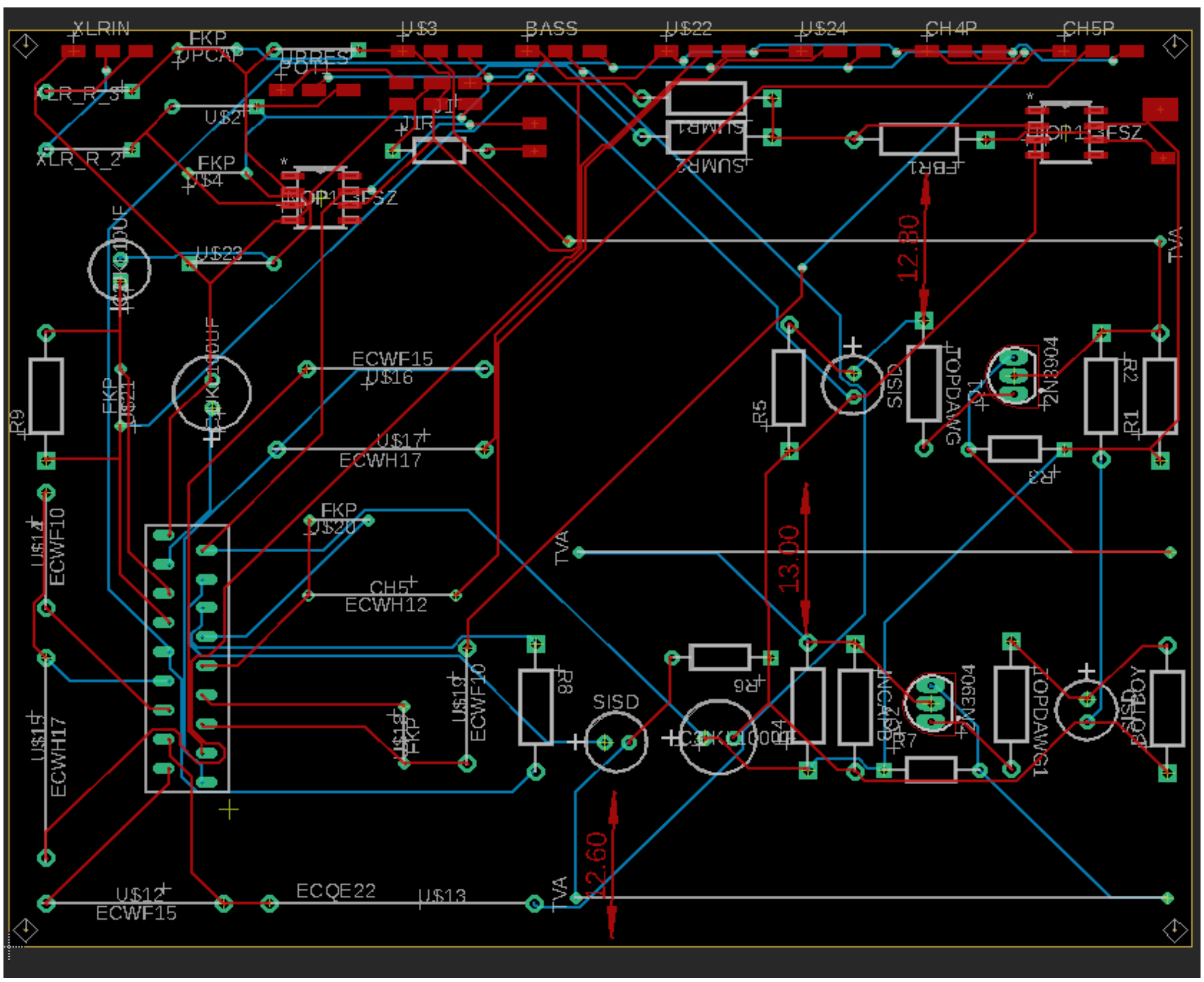
Appendix A: Analog Board Schematic & Layout

A.1. Analog Board Schematic

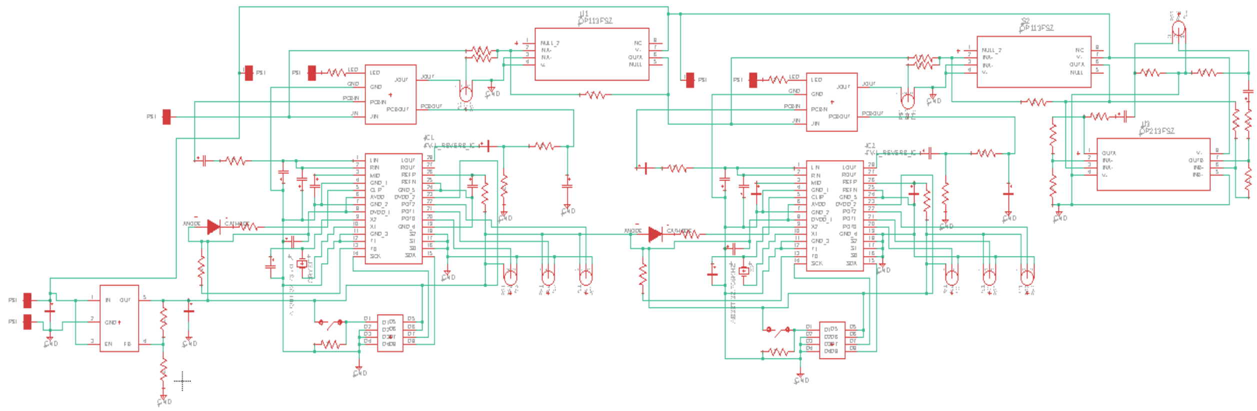


A.2. Analog Board Layout

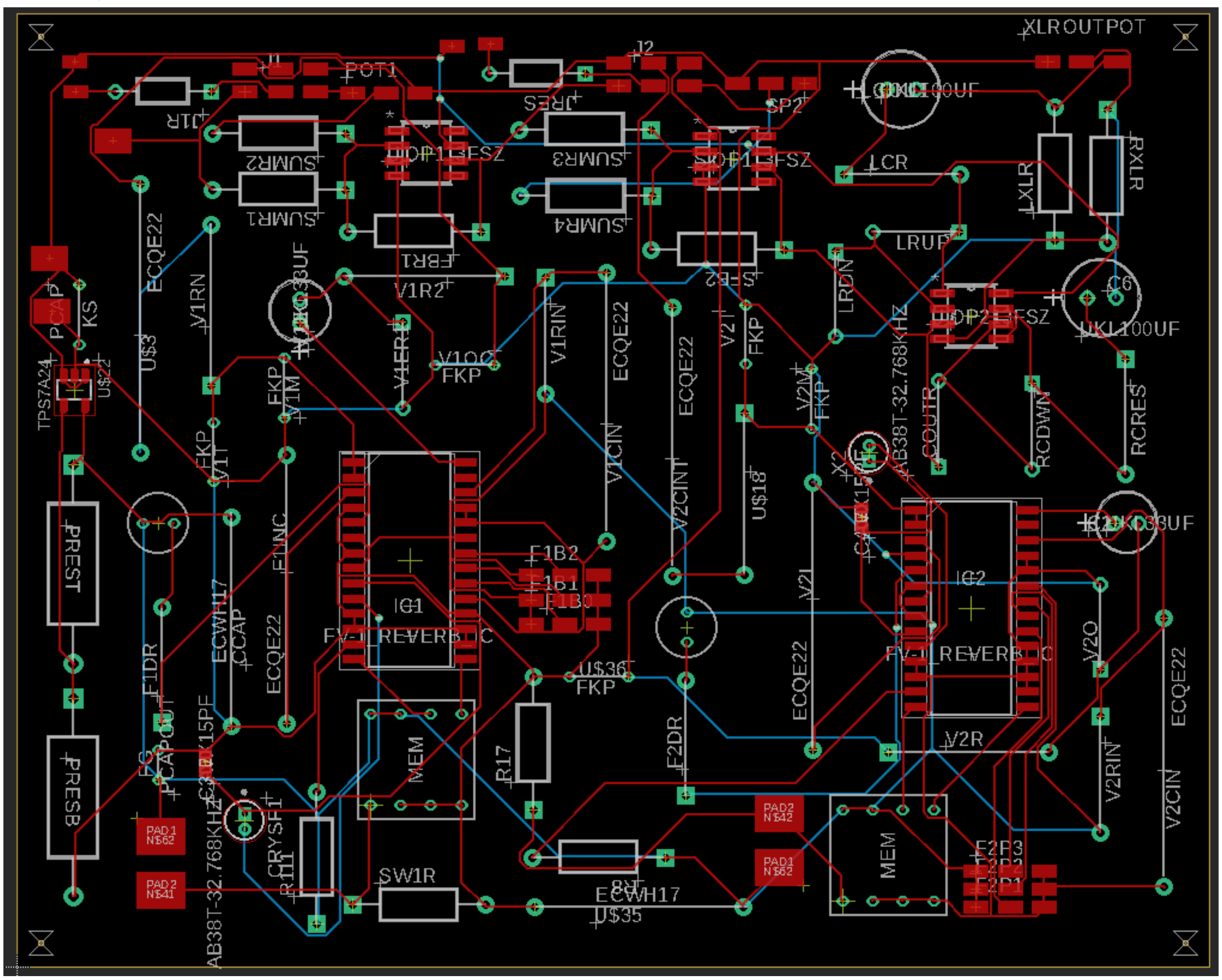


Appendix B: Digital Board Schematic & Layout

B.1. Digital Board Schematic



B.2. Digital Board Layout



Appendix C: Requirement and Verification Tables

C.1. 9V Power Supply

Requirements	Verification	Verified?
1. Power supply must convert 120V AC at 60 Hz to 9V DC continuously. 2. Power supply must be able to supply at least 1A of current. 3. Output voltage has less than 0.5% ripple voltage.	1. a. Plug power supply into the 120V 60 Hz AC wall socket and use an oscilloscope to measure the steady DC output voltage. 2. a. Create a simple circuit that has the power supply in series with parallel resistors that would generate at least 1A of current. b. Use a digital multimeter to measure the output current. 3. a. Using an oscilloscope, measure the output ripple voltage and calculate the percentage ripple by comparing the ripple voltage amplitude to the voltage amplitude found in step 1.a. If the ripple is less than 45mV then it is less than 0.5% of the output voltage.	Y Y N

C.2. Circuit Voltage Regulator

Requirements	Verification	Verified?
1. Voltage regulator must step down the voltage from 9V DC to 3.3V DC. 2. Voltage regulator must be able to supply at least 50mA of current. 3. Output voltage must have less than 0.5% ripple voltage.	1. a. Connect 9V DC power supply to the input of the voltage regulator. b. Use a digital multimeter to measure the steady DC output voltage to be 3.3V DC.	Y
	2. a. Supply 9V DC to input of voltage regulator and create a simple circuit where the voltage regulator is in series with a resistor that would generate at least 50mA of current. b. Use a multimeter to measure the current across the resistor.	Y
	3. a. Connect 9V DC power supply to the input of the voltage regulator. b. Use the oscilloscope to measure the ripple voltage and calculate the percentage ripple by comparing the ripple voltage amplitude to the measurement found in step 1.b. If the ripple voltage is less than 16.5mV then it is less than 0.5% of the output voltage.	N

C.3. Analog-to-Digital & Digital-to-Analog Converter

Requirement	Verification	Verified?
1. The ADC and DAC must be able to sample frequencies between 20Hz-15kHz 2. The ADC and DAC must keep the dB level the same when it enters the FV-1 as it leaves the chip.	1. a. Pass in an audio signal that is an audio sweep from 20Hz to 15kHz into the audio to digital converter. b. Have the signal go through the audio signal processor without having any effects alter the audio signal. c. Wire the digital to audio converter directly to an oscilloscope. d. Make sure the measured frequencies are the same as the input frequencies on the oscilloscope to ensure the frequency range is the same as the input. 2. a. Wire the FV-1 chip without any effects and wire the input and output to different multimeters. b. Send an audio file of a ten second impulse with constant voltage. c. Ensure the input voltage is the same as the output voltage.	N N

C.4. Audio Signal Processor

Requirement	Verification	Verified?
1. Will have up to 0.5 second max delay. 2. Will have up to 3 seconds of reflections.	1. a. Wire the signal processor with the factory settings for the max delay time. b. Send an audio file of a quarter second impulse and save the output file to a computer. c. Measure the difference in time from when the first impulse ends vs the end of the second time it plays to ensure there is at least half a second in between. 2. a. Wire the audio signal processor with the factory settings for the max reflections time. b. Send an audio file of a half second impulse and save the output file to a computer. c. Ensure the difference in time from when the audio starts vs when the audio ends minus half a second is over three seconds long.	Y Y

C.5. Memory

Requirement	Verification	Verified?
1. Will have at least 256b of writable storage.	1. a. Load a 256b test half second second delay effect program onto the memory with an ic memory programmer. b. Wire the memory to the audio signal processor and wire the audio signal processor output to an oscilloscope. c. Send a sample audio file through the audio signal processor. d. Measure the output frequency of the audio signal processor to ensure the output had half second of delay.	Y

C.6. XLR Input

Requirement	Verification	Verified?
1. XLR input circuit must be able to take voltages in the range of -150mV to 150mV. 2. XLR input circuit must not change the voltage from input to output.	1. a. Provide the inputs of the differential op-amp a voltage within the required range using a power supply. b. Use an oscilloscope to measure the output voltage to be unchanged from the input. 2. a. *Repeat steps 1.a. and 1.b. for requirement 2.*	N N

C.7. Amplifier Circuit

Requirements	Verification	Verified?
1. Gain of at least 20dB.	1. a. Use supply voltage to provide 100mV to the input of the amplifier. b. Use an oscilloscope to measure the output to be at least 1V.	Y
2. Output must not exceed the voltage range of -500mV to 500mV.	2. a. Connect the microphone, XLR input, EQ, and amplifier circuits. b. Provide a 100dB audio signal to excite a signal in the microphone. c. Use an oscilloscope to measure the input voltage of the amplifier. d. Use an oscilloscope to measure the output voltage.	Y

C.8. Equalizer Circuit

Requirements	Verification	Verified?
1. Peak frequency deviation must be no larger than 5% at 36Hz, 160Hz, 720Hz, 3.2kHz, and 14kHz frequencies.	1. a. Use an oscilloscope to analyze frequency output by filtering input test signals from a waveform generator.	Y
2. Must have at least +/- 12dB of control at aforementioned frequencies.	2. a. Use a waveform generator and sweep magnitude potentiometer to view decibel change on an oscilloscope.	Y

C.9. XLR Output

Requirements	Verification	Verified?
1. XLR output circuit must be able to take voltages in the range of -500mV to 500mV. 2. XLR output circuit must not change the voltage from input to output.	1. a. Provide the input of the differential op-amp a voltage within the required range using a power supply. b. Use an oscilloscope to measure the output voltage to be unchanged from the input. 2. a. *Repeat steps 1.a. and 1.b. for requirement 2.*	N N

Appendix D: Parts Cost Table

Part	**Quantity**	**Unit Price ($)**
Metal Enclosure	1	15.00
DC Jack	1	0.80
LED Bezel	3	0.27
White LED	3	0.33
Red LED	2	0.18
Toggle Switch	2	4.41
Footswitch	3	6.95
Potentiometer Caps	14	0.14
100kΩ Linear Potentiometer	10	0.66
50kΩ Linear Potentiometer	7	2.95
XLR Output	1	3.22
XLR Input	1	1.87
BA3812L EQ Chip	2	9.43
FV-1 Chip	2	21.00
IC Socket	2	1.54
Linear Voltage Regulator	2	0.60
Single Op Amp	5	5.62
Dual Op Amp	1	6.10
NPN BJT	2	0.50
32.768 kHz Crystal	2	0.20
100Ω Resistor	10	0.80
270Ω Resistor	5	0.26
1kΩ Resistor	10	0.79
1.5kΩ Resistor	3	0.19
4.22kΩ Resistor	1	0.70
5.1kΩ Resistor	3	0.26
6.8kΩ Resistor	5	0.26

10kΩ Resistor	16	0.18
20kΩ Resistor	10	0.18
42.2kΩ Resistor	3	0.26
100kΩ Resistor	20	0.28
1.02MΩ Resistor	3	0.49
1.69MΩ Resistor	3	0.49
15pF Capacitor	4	0.41
150pF Capacitor	2	0.43
1000pF Capacitor	10	0.29
1000pF (Film) Capacitor	2	0.41
2200pF Capacitor	6	0.43
6200pF Capacitor	2	0.67
10000pF Capacitor	2	0.69
0.03uF Capacitor	2	1.48
0.036uF Capacitor	2	1.55
0.1uF (Film) Capacitor	2	1.79
0.1uF Capacitor	7	0.67
0.15uF Capacitor	3	0.35
0.24uF Capacitor	2	1.87
0.33uF Capacitor	2	2.13
1uF Capacitor	15	0.38
1uF (Film) Capacitor	2	1.22
2.2uF Capacitor	2	0.56
3.3uF Capacitor	6	0.44
8uF Capacitor	3	5.57
10uF Capacitor	7	0.50
100uF Capacitor	6	0.37
Total		**286.57**

Appendix E: Digital Effect Algorithms

E.1. Room Reverb Algorithm

```
; Final_Room_Reverb.spcd
; Patch saved from SpinCAD Designer version 1027
; Pot 0: delay time
; Pot 1: pre-delay
; Pot 2: volume of delay
;
;------ Scale/Offset
RDAX POT0,1.0000000000
SOF 0.2900000000,0.0000000000
WRAX REG0,0.0000000000
;------ Scale/Offset
RDAX POT1,1.0000000000
SOF 0.5100000000,0.0000000000
WRAX REG1,0.0000000000
;------ Reverb_Room
SKP RUN ,6
WRAX REG8,0.0000000000
WRAX REG9,0.0000000000
WRAX REG12,0.0000000000
WRAX REG13,0.0000000000
WRAX REG14,0.0000000000
WLDS 0,20,100
RDAX REG0,0.1000000000
WRAX ADDR_PTR,0.0000000000
RDAX ADCL,0.5000000000
WRA 0,0.0
RMPA 1.0
WRA 3277,1.0
RDA 7751,0.5
WRAP 7278,-0.5
RDA 8288,0.5
WRAP 7752,-0.5
RDA 8956,0.5
WRAP 8289,-0.5
RDA 9748,0.5
WRAP 8957,-0.5
WRAX REG3,0.0000000000
```

```
RDA 26685,0.2
MULX REG1
RDAX REG3,1.0000000000
RDA 11642,0.5
WRAP 10764,-0.5
RDA 12930,-0.5
WRAP 11643,0.5
RDFX REG12,0.0200000000
WRHX REG12,-0.5000000000
WRAX REG4,-1.0000000000
RDFX REG8,0.5000000000
WRHX REG8,-1.0000000000
RDAX REG4,1.0000000000
WRA 12931,0.0
RDA 14467,-0.2
MULX REG1
RDAX REG3,1.0000000000
RDA 15436,0.5
WRAP 14468,-0.5
RDA 16804,0.5
WRAP 15437,-0.5
RDFX REG13,0.0200000000
WRHX REG13,-0.5000000000
WRAX REG4,-1.0000000000
RDFX REG9,0.5000000000
WRHX REG9,-1.0000000000
RDAX REG4,1.0000000000
WRA 16805,0.0
RDA 18696,-1.0
MULX REG1
RDAX REG3,1.0000000000
RDA 19375,0.5
WRAP 18697,-0.5
RDA 20503,0.5
WRAP 19376,-0.5
RDFX REG13,0.0500000000
WRHX REG13,-0.5000000000
WRAX REG4,-1.0000000000
RDFX REG10,0.5000000000
WRHX REG10,-1.0000000000
```

```
MULX REG7
RDAX REG4,1.0000000000
WRA 20504,0.0
RDA 22440,-1.0
MULX REG1
RDAX REG3,1.0000000000
RDA 23704,0.5
WRAP 22441,-0.5
RDA 24903,0.5
WRAP 23705,-0.5
WRAX REG4,-1.0000000000
RDFX REG11,0.5000000000
WRHX REG11,-1.0000000000
MULX REG7
RDAX REG4,1.0000000000
WRA 24904,0.0
RDA 3377,1.0
RDA 10201,0.5
WRAP 9749,-0.5
WRAX REG4,1.0000000000
RDFX REG14,0.1000000000
WRHX REG14,-1.0000000000
RDAX REG4,1.0000000000
WRA 3378,0.0
RDA 4277,1.0
RDA 10763,0.5
WRAP 10202,-0.5
WRA 4278,0.0
RDA 3978,0.7
RDA 4233,0.6
RDA 3686,0.5
RDA 4600,0.4
RDA 14467,0.7
RDA 18696,0.8
WRAX REG5,0.0000000000
CHO RDA,0,REG | COMPC,11743
CHO RDA,0,0,11744
WRA 11843,0.0
CHO RDA,0,COS  | REG | COMPC,19476
CHO RDA,0,COS ,19477
```

```
WRA 19576,0.0
;------ Output
RDAX REG5,1.0000000000
RDAX ADCR,1.0000000000
WRAX DACL,1.0000000000
WRAX DACR,0.0000000000
```

E.2. Hall Reverb Algorithm

```
; Final_Hall.spcd
; Patch saved from SpinCAD Designer version 1027
; Pot 0: delay time
; Pot 1:
; Pot 2: volume of delay
;
;
; ----------------------------
;------ Input
;------ Pot 0
;------ Scale pot 0
RDAX POT0,1.0000000000
SOF 0.7000000000,0.0000000000
WRAX REG0,0.0000000000
;------ Constant for filter
SOF 0.0000000000,0.0000000000
WRAX REG1,0.0000000000
;------ Reverb
RDAX REG0,1.0000000000
SOF 0.5500000000,0.3000000000
WRAX REG5,0.0000000000
RDAX ADCL,0.2511886432
RDA 7260,0.5
WRAP 7104,-0.5
RDA 7484,0.5
WRAP 7261,-0.5
RDA 7817,0.5
WRAP 7485,-0.5
RDA 8266,0.5
WRAP 7818,-0.5
WRAX REG6,0.0000000000
```

```
RDA 7103,1.0
MULX REG5
RDAX REG6,1.0000000000
RDA 13078,0.6
WRAP 11827,-0.6
RDA 14830,0.6
WRAP 13079,-0.6
WRAX REG4,1.0000000000
RDFX REG9,0.4000000000
WRLX REG9,-1.0000000000
RDFX REG8,0.0100000000
WRHX REG8,-1.0000000000
RDAX REG4,-1.0000000000
MULX REG1
RDAX REG4,1.0000000000
WRA 8267,0.0
RDA 11826,1.0
MULX REG5
RDAX REG6,1.0000000000
RDA 19220,0.6
WRAP 17777,-0.6
RDA 20564,0.6
WRAP 19221,-0.6
WRAX REG4,1.0000000000
RDFX REG11,0.4000000000
WRLX REG11,-1.0000000000
RDFX REG10,0.0100000000
WRHX REG10,-1.0000000000
RDAX REG4,-1.0000000000
MULX REG1
RDAX REG4,1.0000000000
WRA 14831,0.0
RDA 17776,1.0
MULX REG5
RDAX REG6,1.0000000000
RDA 26124,0.6
WRAP 24542,-0.6
RDA 28106,0.6
WRAP 26125,-0.6
WRAX REG4,1.0000000000
```

```
RDFX REG13,0.4000000000
WRLX REG13,-1.0000000000
RDFX REG12,0.0100000000
WRHX REG12,-1.0000000000
RDAX REG4,-1.0000000000
MULX REG1
RDAX REG4,1.0000000000
WRA 20565,0.0
RDA 24541,1.0
MULX REG5
RDAX REG6,1.0000000000
RDA 1274,0.6
WRAP 0,-0.6
RDA 2657,0.6
WRAP 1275,-0.6
WRAX REG4,1.0000000000
RDFX REG3,0.4000000000
WRLX REG3,-1.0000000000
RDFX REG2,0.0100000000
WRHX REG2,-1.0000000000
RDAX REG4,-1.0000000000
MULX REG1
RDAX REG4,1.0000000000
WRA 2658,0.0
RDA 8267,0.8
RDA 16707,1.5
RDA 22658,1.1
RDA 5451,1.0
WRAX REG7,0.0000000000
SKP RUN ,1
WLDS 0,20,50
CHO RDA,0,REG | COMPC,11877
CHO RDA,0,0,11878
WRA 11927,0.0
CHO RDA,0,COS  | COMPC,17827
CHO RDA,0,COS ,17828
WRA 17877,0.0
CHO RDA,0,REG | COMPC,24592
CHO RDA,0,0,24593
WRA 24642,0.0
```

```
CHO RDA,0,COS  | COMPC,50
CHO RDA,0,COS ,51
WRA 100,0.0
;------ Mixer 2:1
RDAX REG7,1.0000000000
RDAX ADCR,1.0000000000
WRAX REG14,0.0000000000
;------ Output
RDAX REG14,1.0000000000
WRAX DACL,0.0000000000
RDAX REG14,1.0000000000
WRAX DACR,0.0000000000
```

E.3. Simple Delay Algorithm

```
; Final_Delay_Simple.spcd
; Patch saved from SpinCAD Designer version 1027
; Pot 0: delay time
; Pot 1:
; Pot 2: volume of delay
;
;
; ----------------------------
;------ Delay
RDAX ADCL,1.0000000000
WRA 0,0.0
CLR
OR $007FFF00
MULX POT0
SOF 0.9972534180,0.0000305176
WRAX ADDR_PTR,0.0000000000
RMPA 1.0
WRAX REG0,0.0000000000
;------ Volume
RDAX REG0,1.0000000000
MULX POT2
WRAX REG1,0.0000000000
;------ Mixer 2:1
RDAX REG1,1.0000000000
RDAX ADCR,1.0000000000
```

```
WRAX REG2,0.0000000000
;------ Output
RDAX REG2,1.0000000000
WRAX DACL,0.0000000000
RDAX REG2,1.0000000000
WRAX DACR,0.0000000000
```

E.3. Wet Delay Algorithm

```
; Final_Delay_Wet.spcd
; Patch saved from SpinCAD Designer version 1027
; Pot 0: delay time
; Pot 1: no knob
; Pot 2: amount of feedback
;
;
; ----------------------------
;------ Delay
WRAX REG1,0.0000000000
RDAX POT2,0.4500000000
WRAX REG1,0.0000000000
RDAX POT0,0.9990000000
SOF 0.2500000000,0.0000305176
WRAX ADDR_PTR,0.0000000000
RMPA 1.0
WRAX REG0,1.0000000000
MULX REG1
RDAX ADCL,0.5000000000
WRA 0,0.0
RDAX REG0,1.0000000000
WRAX REG0,0.0000000000
;------ Mixer 2:1
RDAX REG0,1.0000000000
RDAX ADCR,1.0000000000
WRAX REG2,0.0000000000
;------ Output
RDAX REG2,1.0000000000
WRAX DACL,0.0000000000
RDAX REG2,1.0000000000
WRAX DACR,0.0000000000
```