SAXOPHONE EFFECTS PEDAL

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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 the majority of 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 or not 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. The vast majority of 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 Visual Aid

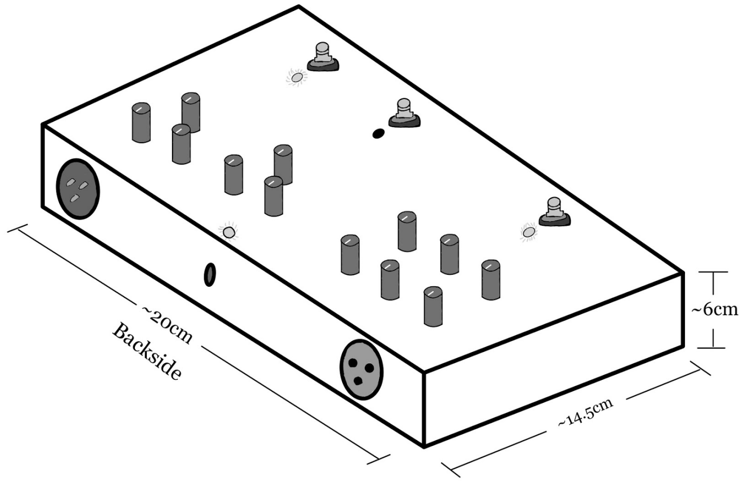


Fig. 1. Physical model of proposed effects pedal.

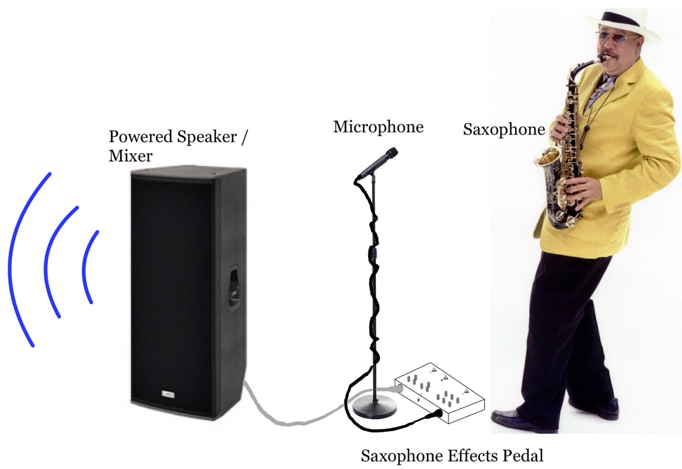


Fig. 2. Possible performance setup.

1.4 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

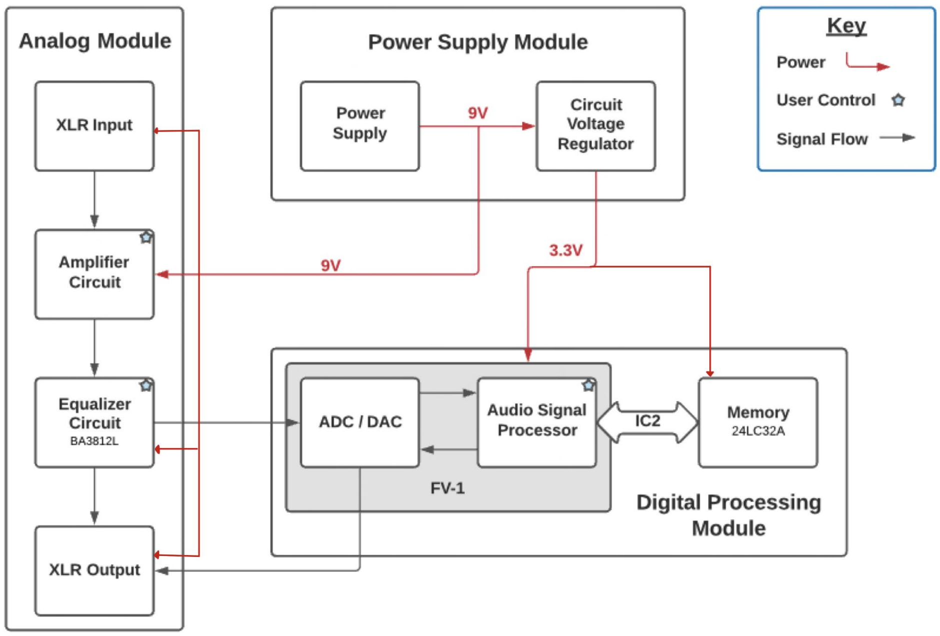


Fig. 3. High-level block diagram.

The signal from the microphone comes into the pedal through the XLR input circuit. This circuit takes sound from the microphone and gets rid of the noise before amplification in the amplifier circuit. The amplifier circuit will take this weak signal and bring it up to audible levels by increasing the amplitude to at least 20dB by a potentiometer. 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 reverb 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 analog signal from the EQ circuit (controlled via potentiometer). The resulting output is then sent through the XLR output circuit which packages the signal back up and sends it out of the pedal.

2.2 Power Supply

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.

2.2.1 9V Power Supply

We will be using a 9V DC power supply to power our system. We get the power from the outlet and this allows us to then step down the voltage to whatever is necessary for the components on the board.

Requirements	Verification
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.

2.2.2 Circuit Voltage Regulator

The voltage regulator will serve to step down the voltage from a 9V DC power supply to the voltage required by the op-amps in the equalizer circuit and the audio processor.

Requirements	Verification
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. 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. 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.

2.3 Digital Processing Module

The digital signal processing module, as seen 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 will then be converted back into an analog signal by the digital-to-analog converter and sent to the XLR output.

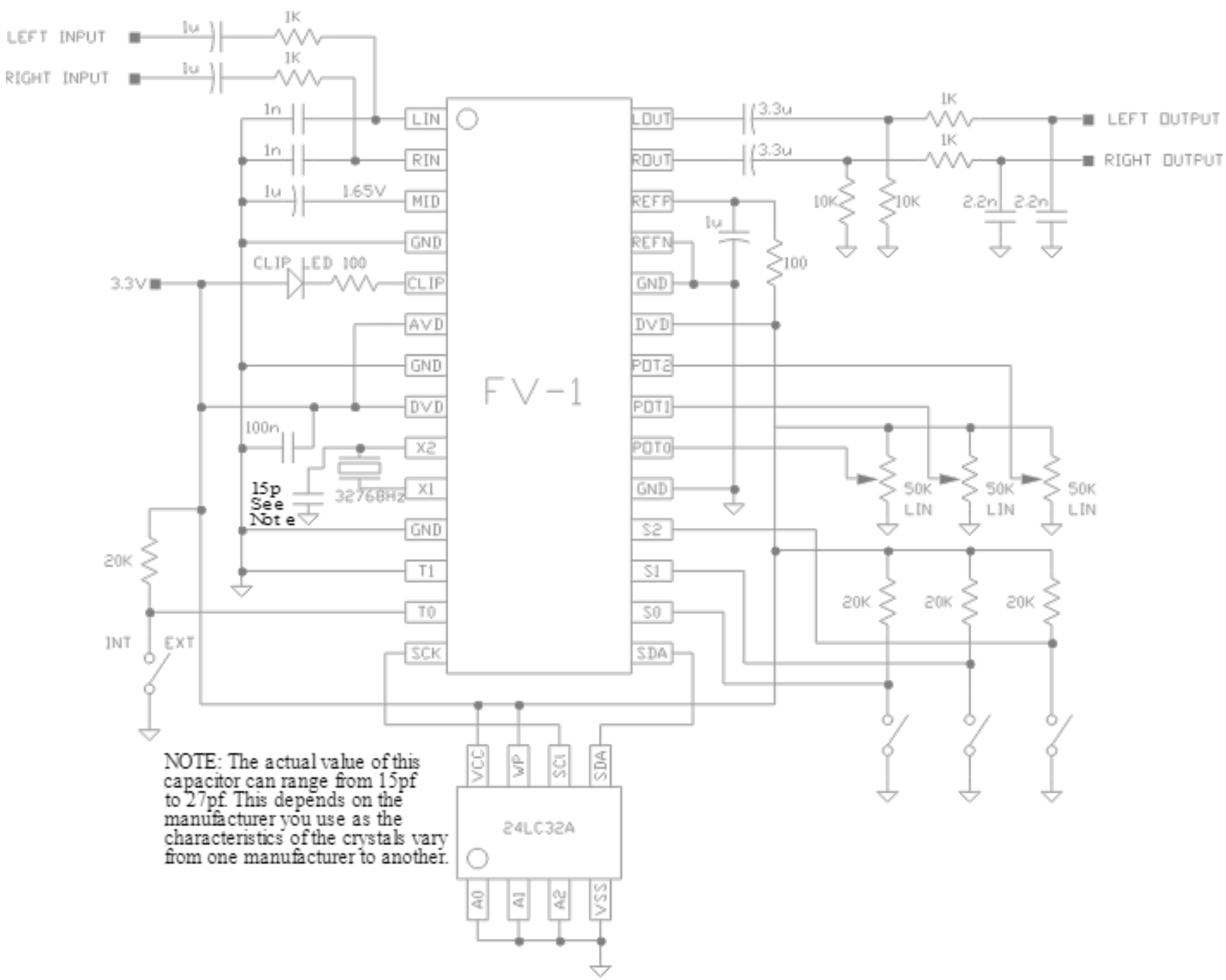


Fig. 4. Schematic of Audio Signal Processor (FV-1) connected to the Memory (24LC32A) [2].

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

The analog-to-digital converter is used to turn the analog signal coming from the equalizer into a digital signal that the audio processor can then perform the computations for the desired effect. The digital-to-analog converter is used to convert the output of the DSP from a digital signal to an analog signal, which then goes to the XLR output. The FV-1 chip has both the analog-to-digital and digital-to-analog converter built into its architecture.

Requirement	Verification
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.

2.3.2 Audio Signal Processor

The audio signal processor takes a digital signal and performs any necessary computations to apply the desired effects using DSP. It loads the programs that run the effects from the memory using I2C. It needs to be able to apply both reverb and delay effects. It will have an internal memory that stores the audio signal to perform the delay effect. The signal processor will receive and output digital signals.

There will be two potentiometers and two switches connected directly to the audio signal processor. The switches will be used to turn the reverb and delay effects on and off. The potentiometers will be used to vary the length of time for the delay effect and to vary the frequency of reverberations. We will also have a crystal oscillator attached to the audio signal processor that acts as a clock. The crystal oscillates at 32768 Hz. We choose the FV-1 chip since it is designed to be an effects pedal chip. It is designed to create both delay and reverb effects. The FV-1 chip is also able to store one second of delay memory at 32kHz.

Requirement	Verification
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 is played to ensure there is at least 0.5 seconds 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.

2.3.3 Memory

The memory is an EEPROM chip that will store the code for our delay and reverb effects. It will communicate with the audio signal processor with I2C. To encode the memory, we will use an IC memory programmer.

We chose the 24LC32A chip because our audio signal processor was designed to work with this EEPROM chip. It is able to store up to eight different programs at once, which is more than enough since we only need two, delay and reverb. The connection of the memory to the audio signal processor is shown in Fig. 4.

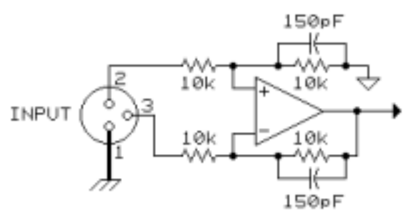
Requirement	Verification
1. Will have at least 256b of writable storage.	1. a. Load a 256b test half second delay effect program onto the memory with an IC memory programmer. b. Wire the memory to the audio signal processor and wire the audio signal processor output to an oscilloscope. c. Send a sample 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.

2.4 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.

2.4.1 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.



Typical Balanced Input

Fig. 5. Typical Balanced Input for XLR [3].

Requirement	Verification
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.*

2.4.2 Amplifier Circuit

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. The amplifier circuit is used to increase the amplitude of the signal to ranges processable by the DSP. Implementation of the transformer design will contribute to frequency shifts based upon the core material [4]. The basic design of our amplifier will be modeled after a Class-A amplifier as shown in Fig. 6.

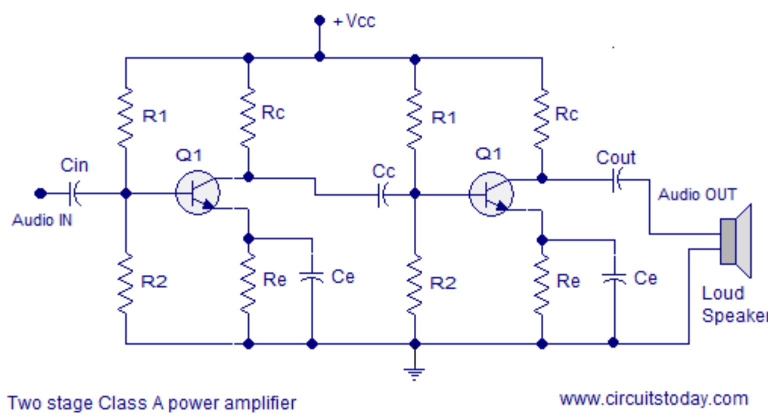


Fig. 6. Class-A amplifier circuit schematic [5].

Requirements	Verification
1. Gain of at least 20dB. 2. Output must not exceed the voltage range of -500mV to 500mV.	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. 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.

2.4.3 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 use 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 will have 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 will have a corresponding potentiometer knob to adjust it. The modified signal will then be sent to the FV-1 chip.

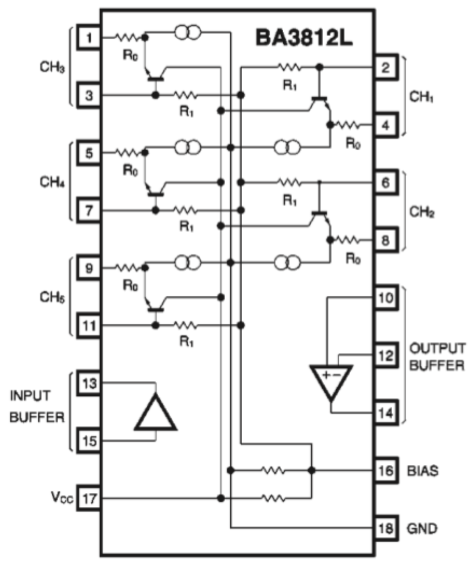
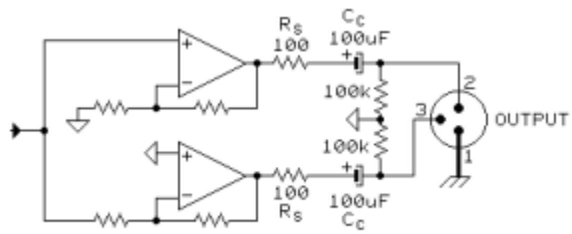


Fig. 7. Circuit Schematic Equalizer Amplification Circuit [6].

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

2.4.4 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 will have an XLR input to receive the audio. Fig. 8 shows a typical design for an XLR balanced output circuit.



Typical Balanced Output

Fig. 8. Typical Balanced Output for XLR [3].

Requirements	Verification
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.*

2.5 Software

The FV-1 uses its own type of assembly language. Since the FV-1 uses digital signal processing techniques, the chip will run all lines of code on every clock cycle. In order to create any type of effect, the DSP performs different algebraic equations on the input frequency. An example of how the assembly code alters the input frequency is shown below in Fig. 9.

For the delay effect, the FV-1 first stores the desired input frequency in its internal memory. The chip then uses counters to read the memory after the desired delay time has elapsed. It will then modulate the frequency with certain arithmetics to produce the desired delay effect. As for the reverb effect, the FV-1 will also store some of the input frequency in its internal memory. It will use this saved memory right after storing it. This memory will be modulated with different arithmetic than the delay for desired effects. The big difference is this one will be using low-frequency oscillation with sin and cos to get the spacious variation that reverb is known for.

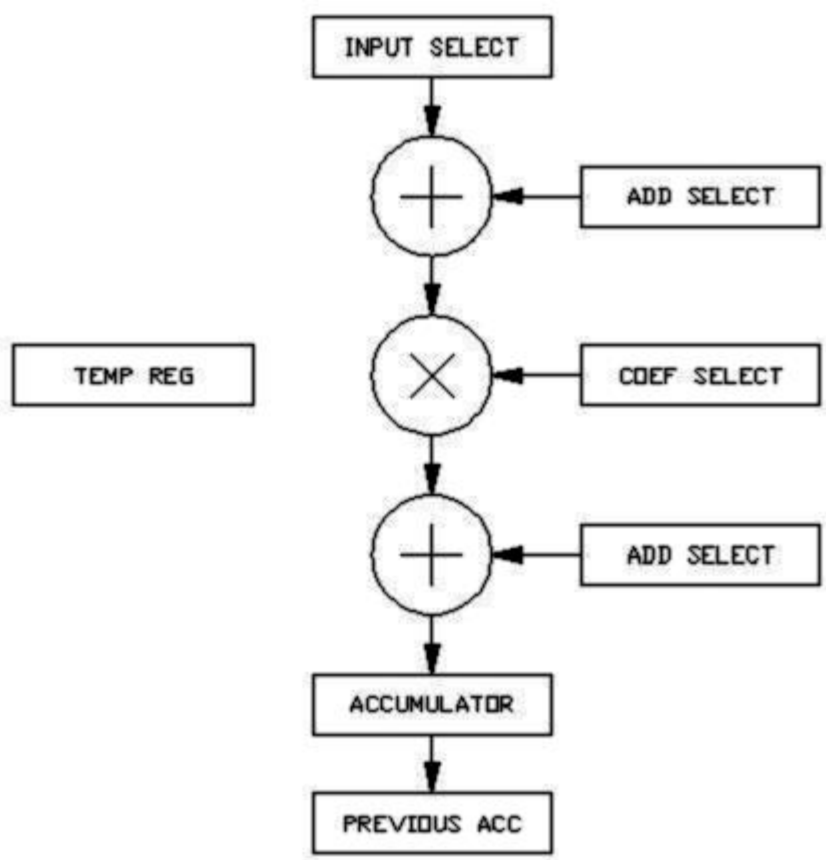


Fig. 9. Example of a mock arithmetic process block of FV-1 code [7].

2.6 Tolerance Analysis

The primary concern for this project is the introduction of various levels of noise from the different components in the circuit. Special consideration must be given in choosing the components for the voltage regulator and the op-amps used in the project. These components pose a special risk to adding distortion to the audio signal as it travels through the different parts of the circuit.

There are also additional sources of noise that are harder to control. Johnson noise occurs regardless of the material or electronic circuit in which it is observed. This noise is the result of thermal fluctuations of stationary charge carriers [8]. This puts a limit on the noise performance.

$$V_n = \sqrt{4kTRB} \quad \text{Eq. 1 [9]}$$

Use Eq. 1 from the textbook [9] to calculate noise. V_n is the RMS noise voltage, T is in kelvin, B is bandwidth in Hz, k is Boltzmann's constant, and R is resistance in ohms. Fig. 10 shows sample calculations of noise voltage given specific resistances.

TABLE 1.2 Resistances and their Johnson noise

Resistance (Ω)	Noise voltage (μV)	Noise voltage (dBu)	Application
1	0.018	−152.2	Moving-coil cartridge impedance (low output)
3.3	0.035	−147.0	Moving-coil cartridge impedance (medium output)
10	0.060	−142.2	Moving-coil cartridge impedance (high output)
47	0.13	−135.5	Line output isolation resistor
100	0.19	−132.2	Output isolation or feedback network
150	0.23	−130.4	Dynamic microphone source impedance
200	0.27	−129.2	Dynamic microphone source impedance (older)
600	0.47	−124.4	The ancient matched-line impedance
1000	0.60	−122.2	A nice round number
2500	0.95	−118.2	Worst-case output impedance of 10 kΩ pot
5000	1.35	−115.2	Worst-case output impedance of 20 kΩ pot
12,500	2.13	−111.2	Worst-case output impedance of 50 kΩ pot
25,000	3.01	−100.2	Worst-case output impedance of 100 kΩ pot
1 meg (10^6)	19.0	−92.2	Another nice round number
1 giga (10^9)	190	−62.2	As used in capacitor microphone amplifiers
1 tera (10^{12})	1900	−32.2	Insulation testers read in tera-ohms
1 peta (10^{15})	19,000	−2.2	OK, it's getting silly now

Fig. 10. Table of Johnson noise voltage given a resistance [9].

Similarly, the quantization of charge carried by electrons in a circuit will contribute a small amount of what is called "shot noise" [8]. Fig. 11 is a table that shows sample calculations of noise voltage given certain bursts of current.

TABLE 1.3 How shot noise varies with current

Current (DC)	Current noise (nA_{rms})	Fluctuation (%)	R (Ω)	Voltage noise (μV)	Voltage noise (dBu)
1 pA	0.000084	8.4	100	8.4×10^{-6}	−219.3
1 nA	0.0026	0.27	100	0.000265	−189.3
1 μA	0.084	0.0084	100	0.0084	−159.3
1 mA	2.65	0.00027	100	0.265	−129.3
1 A	84	0.000008	100	8.39	−99.3

Fig. 11. Table of shot noise given a current [9].

There is also a phenoma called popcorn noise which is a burst of low frequency noise that appears primarily in integrated circuits. If viewed on an oscilloscope you would see a change in output voltage from two discrete levels. The amplitude stays level up to a corner frequency, at which point it falls at a rate of $1/f^2$. Within the same device you may get differing burst-noise mechanisms which have varying corner frequencies. The source of these burst noise phenomena is not well understood or preventable [9].

If you look at the values provided by the table, you will notice that the scale of the noise voltage is small compared to the signal that is generated by the microphone. The microphone will generate voltages around 1mV and the voltage range of the noise is 1-10uV. Despite the large gap between the two, noise in the internal circuitry is additive and the values provided in the table are for a single component. The biggest risk to successful completion of the project is the noise that is introduced to the audio signal in the XLR input, the EQ, and the amplifier parts of the project. Even if we choose parts that are low noise and low drift, there is still a risk our circuit is noisy enough such that artifacts and distortion are introduced into the audio signal. For our project it is ideal that we maintain a SNR (Signal-to-Noise Ratio) of 30dB.

$$SNR = \frac{Audio\ Signal\ Voltage}{Noise\ Voltage} \quad \text{Eq. 2}$$

A SNR of 30dB equates to our audio signal being 31 times larger than any of the background noise that is generated from the circuit. We will continue to modify resistor values in our circuit to try and minimize the amount of Johnson noise generated while maintaining safe operation of all of the integrated circuits.

3. Cost and Schedule

3.1 Cost Analysis

Components	Quantity	Cost per unit
FV-1 (Effects Processor)	2	\$14.00
24LC32A (EEPROM)	2	\$0.44
BA3812L (Equalizer)	1	\$8.66
OP113FSZ (Op Amp)	2	\$5.62
TPS7A24 (Linear Voltage Regulator)	2	\$0.60
1N5226B (3.3V Zener Diode)	3	\$0.14
General resistors and capacitors	N/A	\$15.00
Film capacitors	10	\$2.00
AB38T (Crystal Oscillator)	1	\$0.21
JLM14 (Mic Transformer)	1	\$32.09
2N3403 (Amplifier BJT)	3	\$6.36
Hammond 1590BB (Enclosure)	1	\$14.85
Materials Total	28 Components	\$151.63

$$3 Engineers \times \frac{\$45}{hour} \times \frac{10hrs}{week} \times 16 weeks = \$21{,}600 \quad \text{Eq. 3}$$

Using Eq. 3, the projected labor cost for this project is \$21,600 for three people working across a sixteen-week period at a rate of forty-five dollars per hour. The total cost of the project is \$21,751.63 when accounting for the cost of materials.

3.2 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 presentation and demonstration.	Practice presentation and demonstration.	Practice presentation and demonstration.
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.

4. Safety and Ethics

There are a number of safety hazards that could potentially arise from the use of our proposed saxophone effects pedal. Since safety is a top priority, we are following IEEE code of ethics in that we “hold paramount the safety, health, and welfare of the public” [10]. We will ensure that all electrical components will be safely enclosed so they cannot cause harm to the user or any unsuspecting bystander. We will also look into the use of various insulators so current can not escape and cause bodily harm to the user in the event of a catastrophic failure. Since our device will be used with a speaker, the user could damage his or her hearing if the output audio is played at too high of a dB level. To help mitigate this occurrence, we will include a manual with the device that will include the chart of OSHA acceptable dB levels of sound [11].

As this is a device that is intended for both indoor and outdoor use, excessive moisture could potentially damage the device by causing a short circuit. Since the moisture could come from rain or even sweat from the user, we will follow IP67 guidelines [12] to prevent any short circuiting from liquid entering the device. This includes protection against water splashed from all directions. The effects pedal will be safe enough to handle if the user has excessive perspiration.

Our team strives to, “improve the understanding by individuals and society… of emerging technologies” [10]. As per the IEEE code of ethics, we want everyone to be able to use our device. We will include a manual on how to properly use our device along with the OSHA dB chart. We plan to rigorously test our design to make certain that no harm can come to the user. As a group we will hold 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” [10].

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