# SAXOPHONE EFFECTS PEDAL


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# 1. Introduction

### 1.1 Objective

If you listen to any song on the top charts today, odds are it does not come close in resembling the music from the early 1950's. While instrumentation and genre affinities contribute to this contrast, a primary difference in why these two eras of music sound unlike one another comes down to processing the 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. Unlike in recent past, modern effects units used in performances are made to be user friendly, easily adjustable, and compact - sometimes encapsulated in little modules for stage use called pedals.

Currently, there are no quality effect pedals on the market specifically designed for saxophone players. The portable effects market is currently dominated in its catering towards electric guitar and bass players. One problem in utilizing these guitar pedals for a saxophone is the input does not match the output of the majority of microphones. 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 that 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.

From this, we propose building a multi effects pedal designed specifically for saxophone. Since the majority of microphones use a balanced output as opposed to the unbalanced outputs used in guitars, we will need to implement circuit designs not only to that type of input, but also the frequency range of a saxophone. In our pedal, we will implement a preamp and equalizer that will solve the aforementioned "tinniness". The user can elect whether or not to utilize practical performance effects - delay and reverb. The delay and reverb will be implemented on a digital signal processor while the EQ and preamp will be based in analog due to the time, cost and complexity to model distortion.

### 1.2 Background

As mentioned in the previous section, there currently are no effect pedals on the market dedicated to saxophonists let alone brass or woodwind players. Using guitar pedals by means adapters or microphone choice do not bridge this 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 as one needs to find a suitable microphone or an adapter to use these 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]. Not to mention, if one was to use a XLR to quarter-inch converter they will experience a drastically different tonal response due to the impedance not being properly matched.

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 believe from these observations there is both a need and a want in the market for a product that solves this problem.

### 1.3 Physical Design

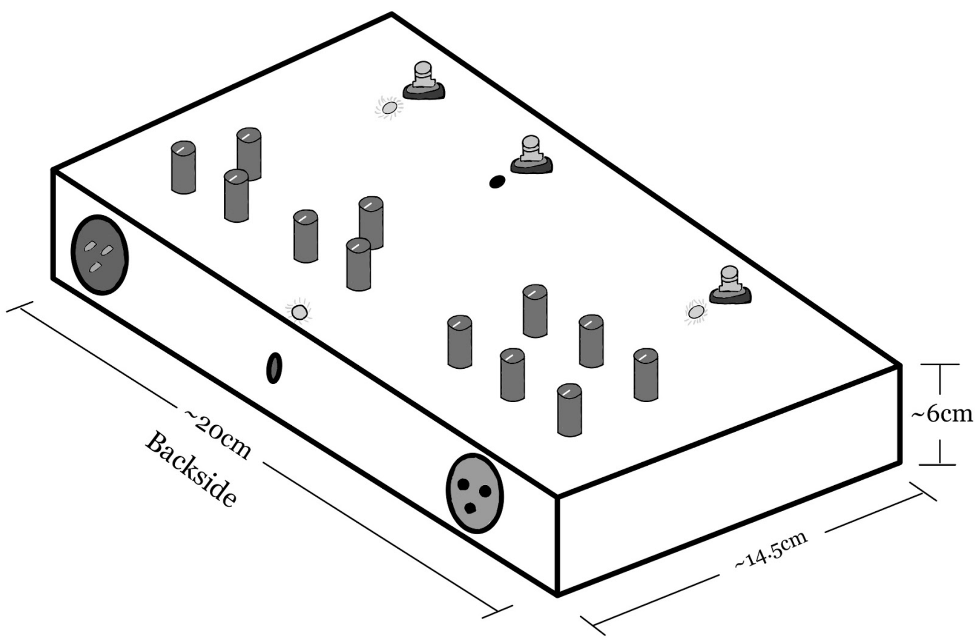


Figure 1. Physical model of proposed effects pedal

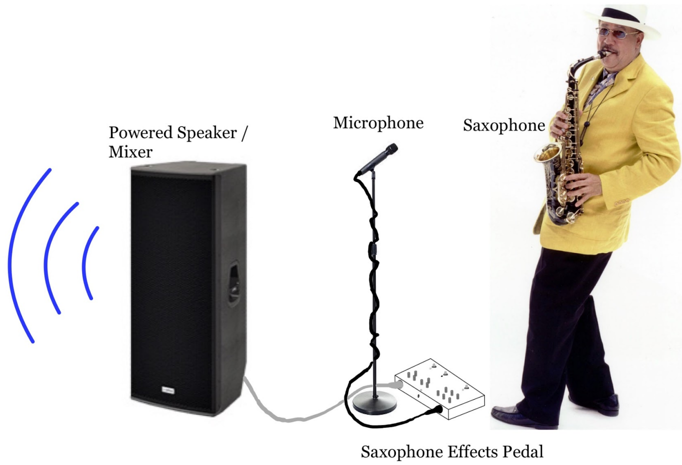


Figure 2. Possible performance setup

### 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.

- Users will be able to adjust the magnitude of the 5 frequency bands centered at 36Hz, 160Hz, 720Hz, 3.2kHz, and 14kHz for the EQ.

- The amplifier circuit will amplify the incoming signal by at least 20 dB

# 2. Design

## 2.1 Block Diagram

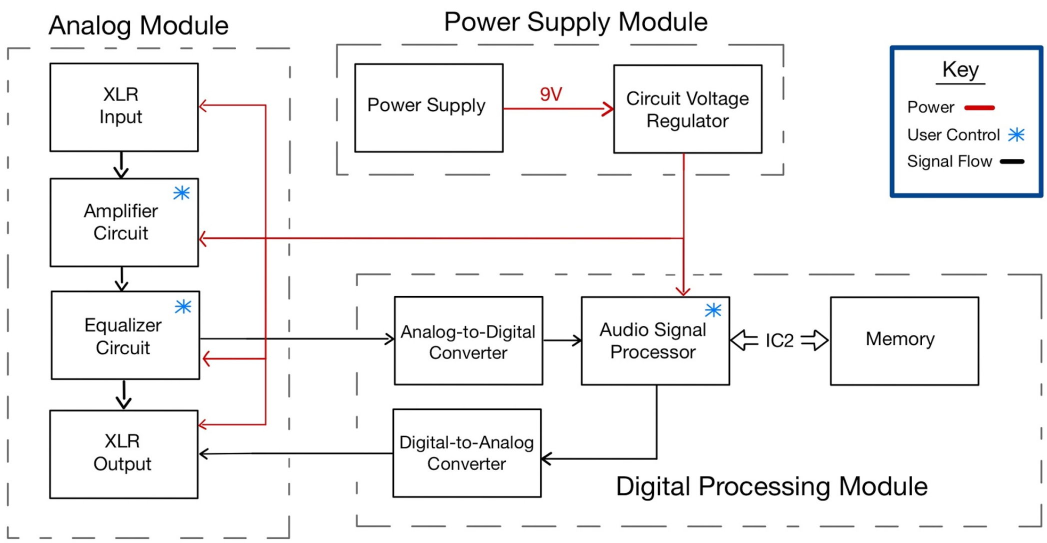


Figure 3. High-level block diagram

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 5-bands centered around the frequencies 36Hz, 160Hz, 720Hz, 3.2kHz, and 14kHz to best accommodate a saxophone's needs. These frequency amplitudes will be controlled by potentiometers.

From there, the newly shaped signal will be converted from analog to digital to be manipulated 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.

*Requirement 1: Power supply must convert 120V AC at 60 Hz to 9V DC continuously and be able to supply at least 1A of current.*

*Bonus Requirement: The adapter should have at least a 5ft cable.*

### 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.

*Requirement: The regulator must step down the 9V DC to 5V DC for the op-amps and 3.3V for the DSP integrated circuit.*

## 2.3 Digital Processing Module

The digital signal processing module is the part of the PCB that will handle the conversion of the analog signal from the microphone to a digital signal. This signal is then sent to different chips which process the signal to produce the desired effects. The output of that chip will then be converted back into an analog signal and sent as an XLR output to a speaker.

### 2.3.1 Analog-to-Digital Converter

The ADC 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.

*Requirement 1: The ADC must be able to convert frequencies between 0-15kHz*

*Requirement 2: The voltage to power the ADC must be between 3.3-5V*

### 2.3.2 Audio Signal Processor

The audio signal processor should be able to take a digital signal and perform any necessary computations to apply the desired effects. We need to be able to apply both reverb and delay effects. The signal processor will receive and output digital signals.

*Requirement 1: Will have up to 0.5 second max delay.*

*Requirement 2: The voltage to power the processor must be between 3.3-5V*

### 2.3.3 Memory

The memory is used to store and replay the sounds for the delay effect. It will communicate with the audio signal processor with I2C.

*Requirement: Will have a minimum clock speed of 32.768KHz and at least 32Kb of memory.*

### 2.3.4 Digital-to-Analog Converter

The DAC is used to convert the output of the DSP to an analog signal, which then goes to the XLR output.

*Requirement 1: The DAC must be able to convert frequencies between 0-15kHz*

*Requirement 2: The voltage to power the DAC must be between 3.3-5V*

## 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 being amplified, the signal passes through an equalizer circuit which has 5 frequency bands that the user will be able to adjust. This will emphasize key frequencies that sound best on a saxophone. The 5 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.

*Requirement 1: The op-amp circuit must be able to take two 150mV input voltages from the microphone.*

### 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 that the DSP would be able to process. Transformer design will be implemented which will attribute to frequency shifts based upon the type of transformer used [2].

*Requirement 1: The amplifier must not boost amplitude beyond +/- .5V DC*

### 2.4.3 Equalizer Circuit

The equalizer circuit is used to modify the analog signal after it has been amplified by the amplifier circuit. The equalizer is an essential component of the project as it is where we can optimize the frequencies that are characteristic of the saxophone. The equalizer circuit will have 5 frequency bands that will help modify and shape the sound as the user pleases. The 5 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 the frequency band. The modified signal will then be sent to the ADC and the DSP thereafter.

### 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 in order to transmit a low-noise copy of the processed audio to the speaker. The speaker will have an XLR input to receive the audio.

## 2.5 Risk Analysis

Despite requiring relatively simple components, the part of the project that has the highest risk of failure is between the amplifier circuit and the equalizer. This section of the project is where noise and distortions are most likely to be added to the incoming audio signal. If we don't correctly center the frequency bands, then we could potentially attenuate parts of the frequency response that are crucial to good sounding saxophone audio. Every other part of the project requires simple connections, therefore the amplifier/equalizer poses the biggest risk to successful completion of the project.

# 3. 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 of ours, we are following IEEE code of ethics in that we “hold paramount the safety, health, and welfare of the public” [3]. 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 cannot 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 their 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 [4].

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 to prevent any short circuiting from liquid entering the device. This includes protection against water splashed from all directions. It is our intention that a user may also be able to handle the device safely with slightly damp hands.

Our team strives to, “improve the understanding by individuals and society… of emerging technologies” [3]. 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 on rigorously testing 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” [3].

# 4. References

[1] Aviom Blog. 2021. *What's the Difference Between Balanced and Unbalanced?*. [Online] Available: <https://www.aviom.com/blog/balanced-vs-unbalanced/> [Accessed 18 February 2021].

[2] Robjohns, H., 2021. *Analogue Warmth*. [Online] Soundonsound.com. Available: <https://www.soundonsound.com/techniques/analogue-warmth> [Accessed 18 February 2021].

[3] "IEEE Code of Ethics," *IEEE*. [Online]. Available: https://www.ieee.org/about/corporate/governance/p7-8.html. [Accessed: 18-Feb-2021].

[4] "Department of Labor logo UNITED STATESDEPARTMENT OF LABOR," *1910.95 - Occupational noise exposure. | Occupational Safety and Health Administration*. [Online]. Available: https://www.osha.gov/laws-regs/regulations/standardnumber/1910/1910.95. [Accessed: 18-Feb-2021].