



**ASHESI UNIVERSITY**

**AFFORDABLE DIGITAL GUITAR EFFECTS PEDAL FOR LOCAL GUITARISTS**

**CAPSTONE PROJECT**

B.Sc. Electrical & Electronic Engineering

**Victor Hazel**

**2019**

**ASHESI UNIVERSITY**

# **AFFORDABLE DIGITAL GUITAR EFFECTS PEDAL FOR LOCAL GUITARISTS**

## **CAPSTONE PROJECT**

Capstone Project submitted to the Department of Engineering, Ashesi

University in partial fulfillment of the requirements for the award of

Bachelor of Science degree in Electrical & Electronic Engineering.

**Victor Hazel**

**2019**

# DECLARATION

I hereby declare that this capstone is the result of my own original work and that no part of it has been presented for another degree in this university or elsewhere.

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I hereby declare that preparation and presentation of this capstone were supervised in accordance with the guidelines on supervision of capstone laid down by Ashesi University College.

Supervisor's Signature:

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Date:

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## **Abstract**

Music has been a mode of communication and a source of identity for many cultures, especially in Africa. The sound from the guitar is one of the many components that make up music, The electric guitar is a musical instrument, usually made up of six strings which is connected to amplifiers to produce sound. Technological advancement has led to the creation of Guitar effects pedals.

However, manufacturers keep making new models of guitar effects pedals with a continuous increase in cost. It is therefore difficult for low-income young guitarists to have access to guitar effects pedals. This project seeks to use a general purpose microcontroller to create a very affordable guitar effects pedal.

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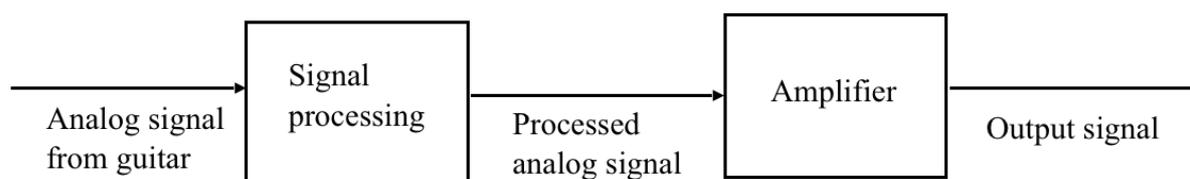
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## Chapter 1: Introduction

There are many stringed instruments used in African music, and the guitar is a classic example. It is a string instrument which has been used across many different genres of music. As music evolved, guitarists began using new technology to enhance their play. In 1948, the first commercially standalone effects unit was made by DeArmond Trem Trol [1]. A guitar effect unit is a device that modifies the sound of the guitar to produce new tones. Such tones include distortion, chorus, delay, and flanger. Several other units were produced as the years went by. In the 1980s, the first multi-effects unit was produced. A guitar multi-effects pedal takes in an analog signal from the guitar, performs signal processing on the signals and passes the resulting analog signal output through an amplifier as shown in **Error! Reference source not found.** as shown in Guitar effect pedals were



*Figure 1 Functional block diagram of an effects box*

made in single units called stomp boxes. Every stomp box was created with a circuit for a one guitar effect such as distortion stomp boxes. A guitarist will, therefore, have to buy several stomp boxes, one for each effect tone he desired to have. As such, guitarists had the liberty to mix several tones (effects) during live play [2]. Guitarists had to arrange many stomp boxes on a pedal board and interconnect them with a patch cable. Finding the source of feedback when many stomp boxes are connected then became a problem. The inception of the multi-effects pedal allowed guitarists to carry all these effects in a single unit. The guitarist will not have to take a long time to connect several stomp boxes and spend money on buying patch cables. A downside of the multi-effects pedal is that it is relatively expensive as compared to the stomp boxes. Also, when a part of it gets damaged, the whole unit has to be replaced, unlike the stomp boxes which a unit can easily be replaced. On the hand, new multi-effects give guitarists the chance to customize their effects and save it on their pedal.

## 1.1 Background

The processing of guitar signals with Digital Signal Processes requires a DSP chip. Digital signal processing involves converting continuous varying quantities such as voltages and temperature into discrete values for processing. Previously, signal processing was done through the building of electrical circuits. However, signal processing can now be done digitally, thanks to the invention of the DSP chip, a high-speed integrated circuit optimized for high-speed huge mathematical computations [3]. Digital processing involves manipulations of many sampled units of a signal. Currently, technological advancements have made it possible for the creation of cheap micro-controllers which can be used to perform digital signal processes. However, the quality of processed signals by micro-controllers may not be as high as that of a dedicated DSP chip but will be of acceptable quality. Acceptable quality is the audio quality that has low or little noise compared to the signal of interest.

Guitar multi-effects are very expensive in Ghana. As a result, most Ghanaian guitarists cannot afford them. Most guitarists who have an effect pedal find it difficult to buy a newer version due to the cost involved. Also, the guitar multi-effects are limited to sound from the guitar alone, which has processed to give a new tone. This project aims at creating an acceptable quality digital effects board for a guitar which is less than 300.00 GHC. A search on some of Ghana's retail online shops, OLX and Tonaton showed that the lowest price of guitar multi-effects is GH¢300.00 and that is for a used effects pedal. The table below shows the prices of new effects boards.

### The prices of new effects boards

Brand	Cost (GH¢)	Site
Roland GR-55s Guitar Effect Pedal	3,000.00	www.ghanabuysell.com
Boss GT 10 Guitar Multi-Effects Processor	2,000.00	www.tonaton.com
RP90 DigiTech Guitar Effect	1,900.00	www.ghanabuysell.com
Boss GT 10 Guitar Multi-Effects Processor	2,000.00	www.tonaton.com

This project will employ the use of a general-purpose microchip for the processing of signals: ATmega2560. The price of an ATmega2560 chip on an Arduino costs not more than GH¢60.00. The other components required for building the circuits and developing the whole product will not cost more than a GH¢100.00. It is, therefore, possible to produce on a large scale at a price less than GH¢300.00. A digital effect pedal that costs GH¢300 will be the same price of an average used effects board 50% less than the price of some second-hand effects pedals. This price will be competitive on the market and will be preferred for the large middle-income guitarists in Ghana.

### 1.2 Scope of Work/Research

- 1) Building Circuits
- 2) Signal processing of analog signals to produce different tones
- 3) Testing the entire project

The final product will be a standalone guitar multi-effects board. A standalone guitar multi-guitar effects pedal is a separate unit. It will take an input signal from the guitar and perform some algorithms on the signal to produce a new effect.

### **1.3 Significance And Contribution/ Outcome**

As a guitarist myself, I understand the influence of having an effects pedal and how it changes the way I play. It revolutionizes my playing. I do not want other guitarists to be unable to experience this revolution, just because it is too expensive. My project will make this experience more affordable for guitarists, and in doing so, tap into deeper potentials previously unknown.

## Chapter 2: Related Work

Digital signal processing has been done in several ways over the years. Options include software digital signal processing and hardware signal processing. Software has made it possible to process digital systems with mathematical filters to perform signal processes. MATLAB and Simulink provide a platform for analysis and processing of signals. The Data Acquisition toolbox in Matlab helps to make this real time. The hardware necessary for this is the NI 9201 board, NI 9263 board and an audio amplifier. MATLAB also provides a complete workflow for fixed-point design and C

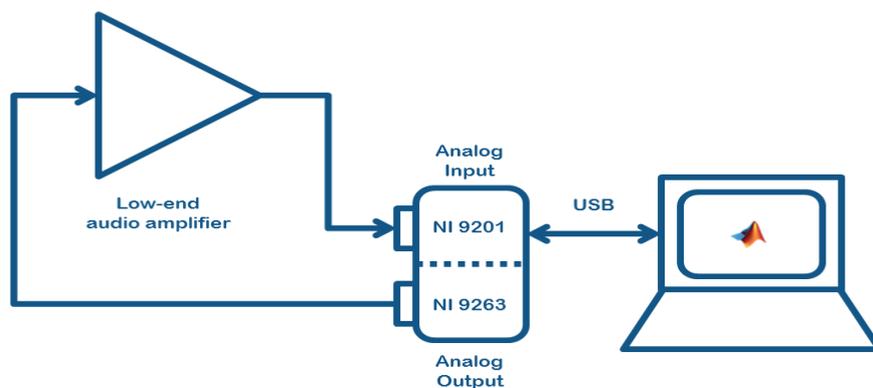


Figure 2 A block diagram of NI9201 block interfacing with Matlab

and HDL code generation. The benefit of using MATLAB is that it requires very little hardware and that Simulink provides a graphical interface for processes [4]. **Error! Reference source not found.** illustrates how the filter, DSP card, and Matlab are used for signal processing.

A project by Czubak et al. in making a real-time digital effects system used MATLAB Simulink to design the filters [5]. According to them, MATLAB Simulink allows for fast calculation and representation of filter designs. MATLAB Simulink provides building blocks which contains the necessary mathematical calculations for creating filters. A DSP board together with Code Composer 3.1 made it possible for real-time demonstration.

Another project carried out by Kevin Salvador employed a TI/Spectrum digital DSK6713 for digital signal processing on guitar signals [5]. Again, MATLAB was used to perform signal processing on the signals.

## Chapter 3: Requirements and Design

This project focused on enhancing customer usage in the local industry and the ability of customers to purchase guitar effects pedals. As a result, an analysis of customer needs for the product. The requirements of this project consist of the customer requirements and design requirements. The requirements will be updated as the project progresses.

### 3.1 Design requirements and specifications with justification

Table 1 Table of requirement and Specifications

Requirement	Specification	Justification
Light in weight	Less than 4.5Kg	The product must be light enough to be carried around
Audio output must be clear	The output must be of high quality.	This is to prevent distorted audio signals from the amplifier
Portable	Product size must be no more than $(20 \times 20 \times 5) \text{ cm}^3$	It should be small enough to be carried in a guitar bag
Cost	The product must not be more than 300.00 cedis	Most guitarists wish to spend less than 400 cedis on an effects board of this caliber

User friendliness	Controls should be simple and should be easy to connect and play	Guitarists like to connect and play on the spot.
Power Supply	5V and -5V output	The different blocks require 5V or -5V for operation

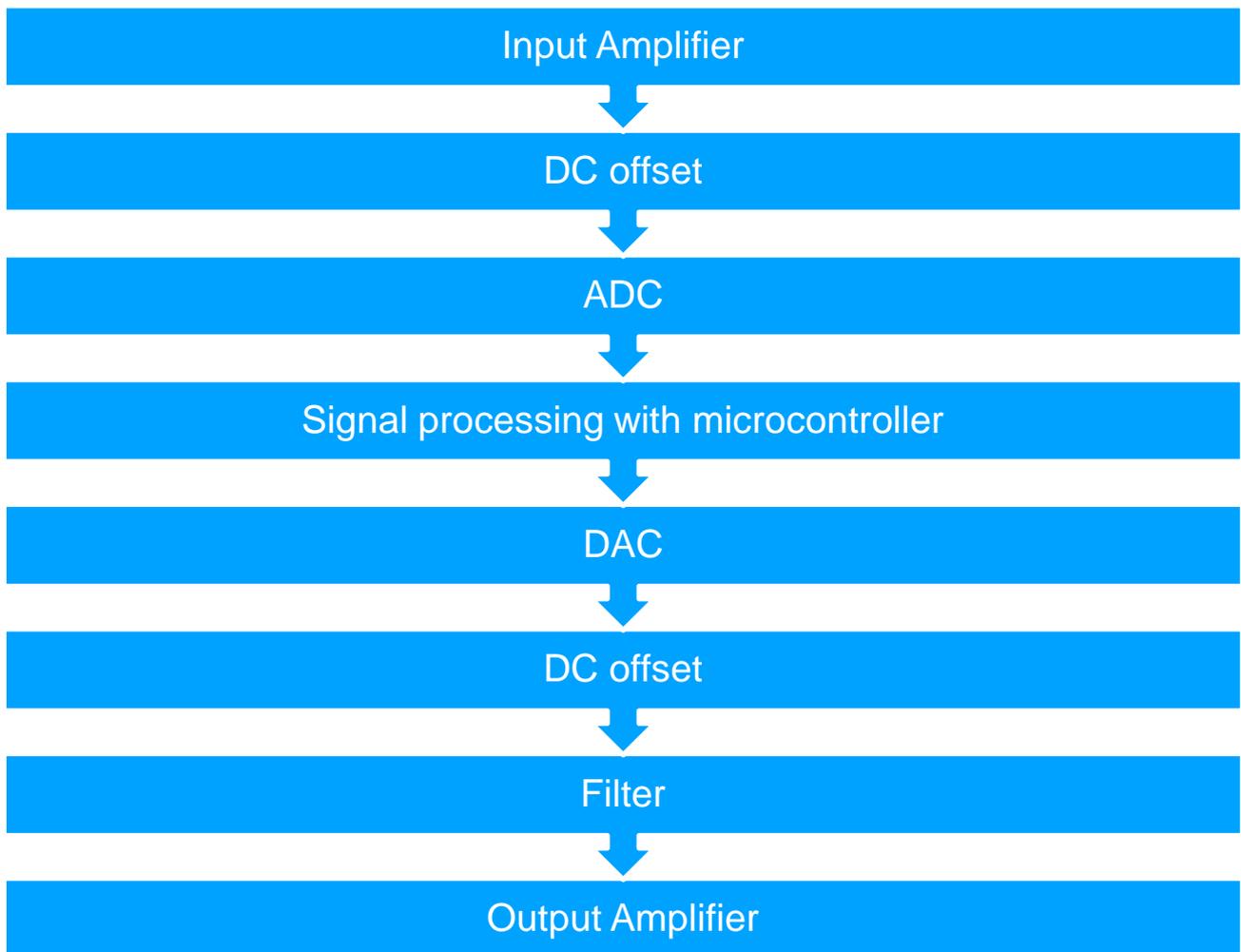
## Inputs and outputs

*Table 2 Table of inputs and outputs*

Inputs	Outputs
Guitar Analog signal	Analog signal with effects
Power supply	Different power levels for various components
Volume control /Gain Amplification	Varying volume levels from no volume to highest volume
Function Switches	Cause the microcontroller to switch between different effects
Equalizer	Controls the bass, mids, and highs of the tone.

### 3.2 Description of Block Diagram

This project is made up of different blocks. A low-level block diagram of the various blocks reveals that the project is made up of the following blocks. These blocks are explained together with their design implementation.



The various blocks are further explained:

1) Input Amplifier: The amplitude of the input signal of the guitar is very small. This is about 220mV peak-to-peak. It, therefore, has to be amplified for other processes to be performed. The alternatives for the construction of the input amplifier are the push-pull amplifier and the non-inverting operating amplifier. The Push-Pull Amplifier is used to supply high power and low distortion signals to the

load. It consists of two transistors which push and pull the signal on the output signal on the positive and the negative half cycle respectively [6]. The advantages of this amplifier are low distortion, absence of magnetic saturation and cancellation of power supply ripples [7]. The push-pull amplifier needs to have identical transistors and requirement coupling transformers. The output of a non-inverting amplifier is in phase with the input signal.

The alternative is the non-inverting amplifier using an operational amplifier. The non-inverting amplifier has a high input impedance and low output impedance. This makes it a good circuit buffer to protect the source of the audio signal from overloading by the rest of the circuit.

The non-inverting Operational Amplifier was selected. This is due to its ability to protect the incoming signal source from the rest of the circuit. The operational amplifier is an all one built device with transistors which is performs can be used as an amplifier or a switch. It is light in weight, small in size and more cost-effective as compared to the push-pull amplifier. The non-inverting operational amplifier circuit is shown in Figure 3 below.

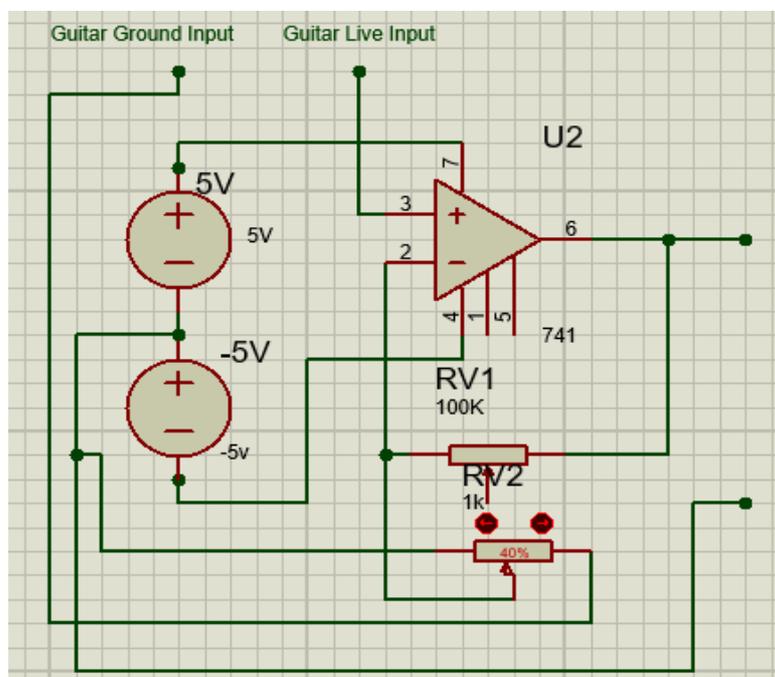


Figure 3 Op-amp based amplifier for input signal

The circuit in Figure 3 shows the implementation of the non-inverting amplifier. The potentiometer (RV2) which is connected to the ground input from the guitar gives the user the flexibility to adjust the gain of the signal.

2) DC-OFFSET: This block sets the offset voltage for oscillation of incoming signals. By default, audio signals from guitars oscillate about 0V. If this wishes to be changed, a DC offset is built. The incoming signal is capacitor coupled to a potential divider whose voltage is at the desired offset voltage. The addition of the two signals enables an oscillated at the DC voltage. From **Error! Reference source not found.**, the capacitor receives the input signal, and the output is signal goes through the intersection of the capacitors and resistors. The circuit of a DC offset is shown in Figure 4 below.

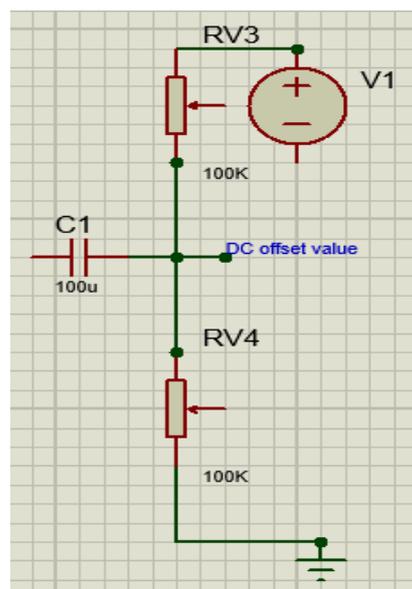


Figure 4 A DC offset circuit

3) ADC – The Analog to digital converters transforms continuous varying signals to discrete signals to enable the microcontroller to process it. The design options considered for this block are the flash ADC

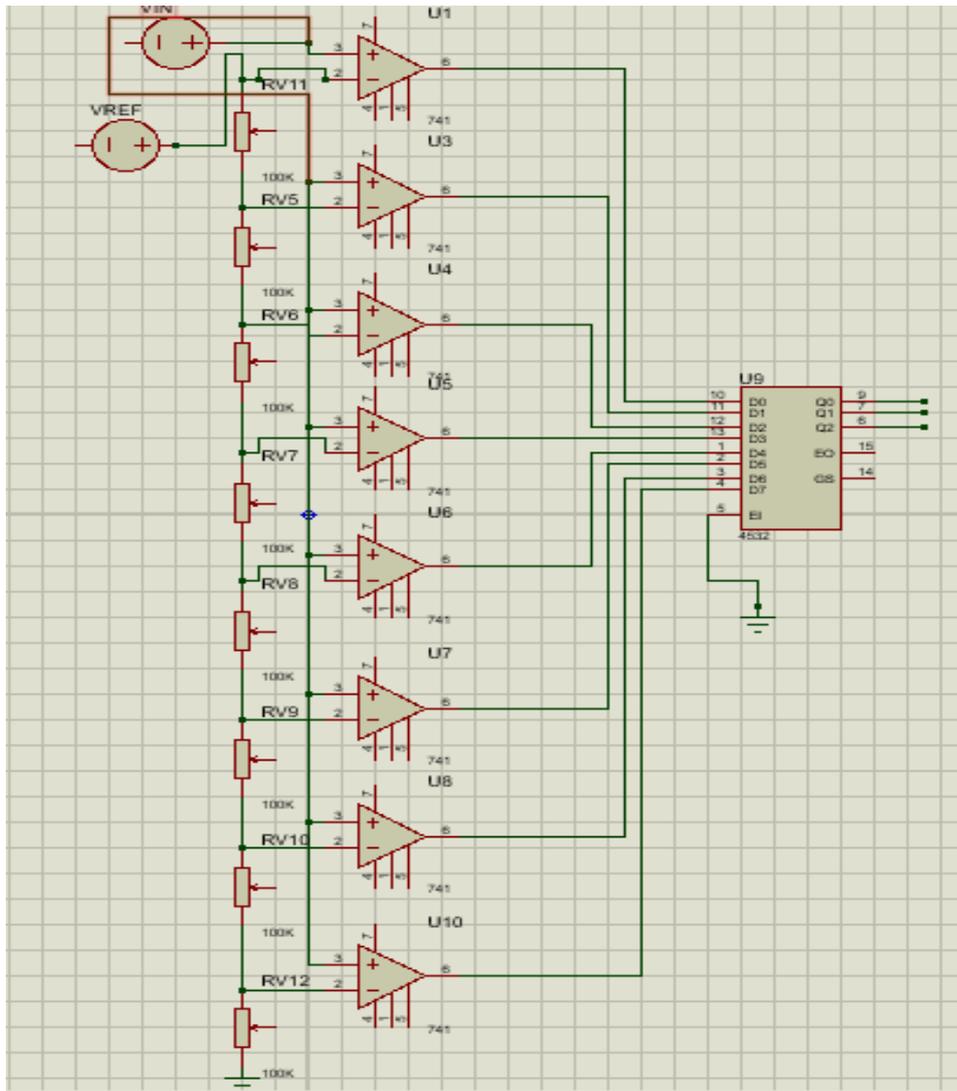


Figure 5 A 3-bit flash comparator for ADC

Flash ADC is the fastest type available. It is made up of comparators and a string of resistors. A flash analog to digital converter uses  $2^{n-1}$  comparators where n is the number of bits of the ADC. This means, for an 8 bit ADC which this project seeks to employ will require 255 comparators. This circuit compares the input signal to a unique reference voltage at each point. The output from each comparator is an input to a priority encoder circuit. The encoder then outputs the binary equivalent. [8]. An example of a 3-bits flash comparator is shown in Figure 5 as an illustration.

Regardless of the speed of flash ADC's, they tend to draw significant current due to the number of comparators required. The Flash ADC is not economically suitable for this project due to the high cost in component required [9].

Another design alternative is to use the in-built Arduino commands for DSP. The Arduino Mega has pins that can read analog signals. The command used is ‘AnalogRead()’. This command accepts analog signals and assigns them a value between 0 and 1023. This suggests that it is a 10 bit ADC. It does so at a frequency of 8000Hz. The frequency required for this project is 40,000Hz. The Arduino This means that the sampling frequency of the ADC is not high enough to reproduce the signal to the quality expected.

The final design option is the to use of bitwise operations to directly program the registers of the ATMEGA chip. This procedure bypasses the ‘AnalogRead’ command and gives the user the flexibility to change the sampling frequency by programming the ADC registers and other registers. It, however, requires more programming skill due to the complex code writing involved.

The latter design option (Bitwise operation) was implemented. This is due to the low cost, and no extra components required. This project prioritizes low cost. Therefore, a faster process was traded for a cheaper process.

4) DAC – The DAC block is responsible for converting the processed digital signal to analog signals. In other words, it converts binary data to voltage or current. In this project, the Digital audio data would have to be converted to voltage to be heard by the ear. The design options being considered are the binary-weighted-input DAC and the R-2R binary ladder.

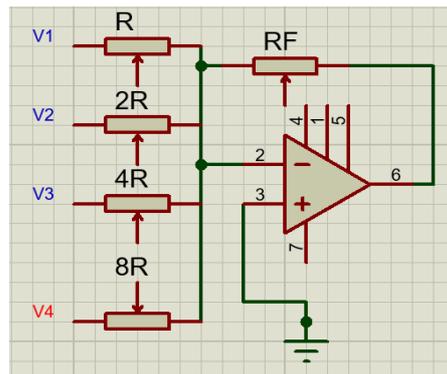
The binary-weighted-input DAC works on the principle of the inverting summing opamp. For three different voltage input with voltages V1, V2 and V3 into the opamp with resistors R1, R2 and R3 respectively and feedback resistor Rf, the output voltage can be calculated using the equation of the inverting amplifier. Equation of the inverting amplifier is

$$-V_{out} = \left( \frac{R_f}{R_{in}} \right) * V_{in}$$

Therefore, the output voltage of the summing amplifier is

$$-V_{out} = \left( \frac{R_f}{R_{in}} V_1 + \frac{R_f}{R_{in}} V_2 + \frac{R_f}{R_{in}} V_3 \right)$$

The binary weighted ADC uses a doubled value of each input resistor  $R_{in}$  as shown in Figure 6. That is,  $R_1=1K\Omega$ ,  $R_2=2K\Omega$ , and  $R_3=4K\Omega$ . This suggests that, for an 8bit, resistors will be required in  $\frac{1}{2}$  ratio of each pair up to the resistor value  $2^{(n-1)}$  which is  $2^7=128K\Omega$ . This can be limiting if all the values of resistors are not available [10].

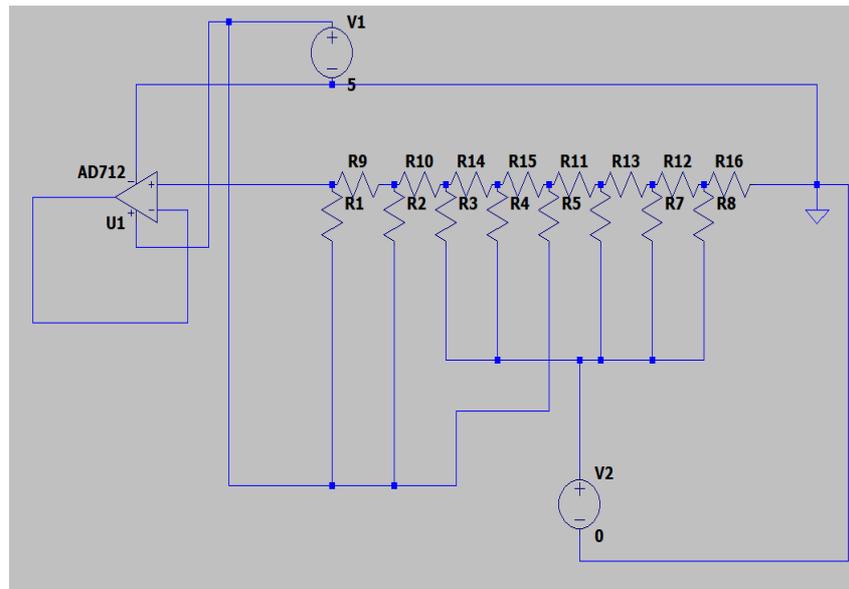


*Figure 6 Binary Weighted DAC*

The second design alternative for the DAC block is the R-2R ladder. It uses only two values of resistors with values  $R$  and  $2R$  no matter the number of bits of DAC being constructed. It is fairly simple to use since the output impedance of the ladder is equal to  $R$ . The R-2R ladder circuit is analyzed by Thevenin's theory.  $D_0$  is voltage-divided. The voltage is simply  $D_0/2$ . The Thevenin equivalent resistance (when the source is removed) at that point is two  $R$ s in series  $2R$ . It is realized that there is another voltage divider circuit showing up. There is a repetitive occurrence of this shape. Continuing the calculation reveals that the Thevenin equivalent resistance is always  $R$ . To know the weight of voltage each input voltage we use Thevenin's open voltage. In the first stage,  $D_0$  (LSB) was reduced by half; It keeps reducing by a ratio of  $\frac{1}{2}$  throughout the circuit. Hence for a 4 bit DAC, the LSB will contribute  $D_0/16$ .

The total output voltage is the weighted average of all the input voltages:

$$A_{out} = \frac{D_0}{16} + \frac{D_1}{8} + \frac{D_2}{4} + \frac{D_3}{2}$$



*Figure 7 8-bits R2R DAC*

Figure 7 is an LTSpice simulation of an 8bit R-2R DAC. The digital value written to it is 00010011. The output voltage was passed through a non- inverting amplifier which serves as a buffer to the circuit as well as amplifying the output.

Micro-controller –The microcontroller is responsible for the processing of digital signals to produce the desired output signals. The microcontroller has General Input/Output pins (GPIO) which are necessary to interact with signals in the real world. This project will consider two microcontrollers from the same family: The Arduino Uno and the Arduino Mega. Figure 8 shows the general architecture of the Arduino.

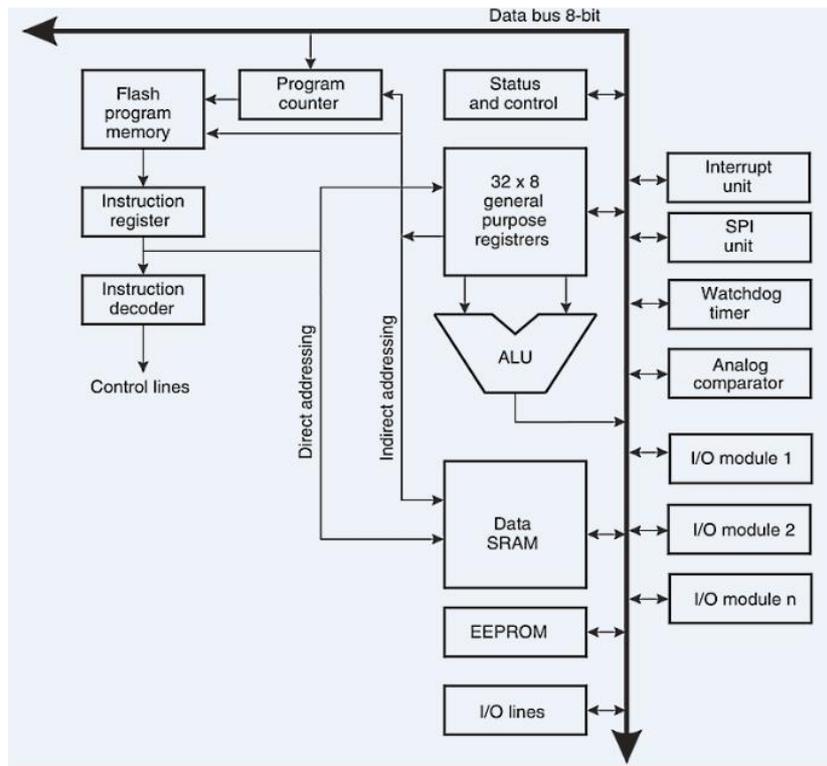
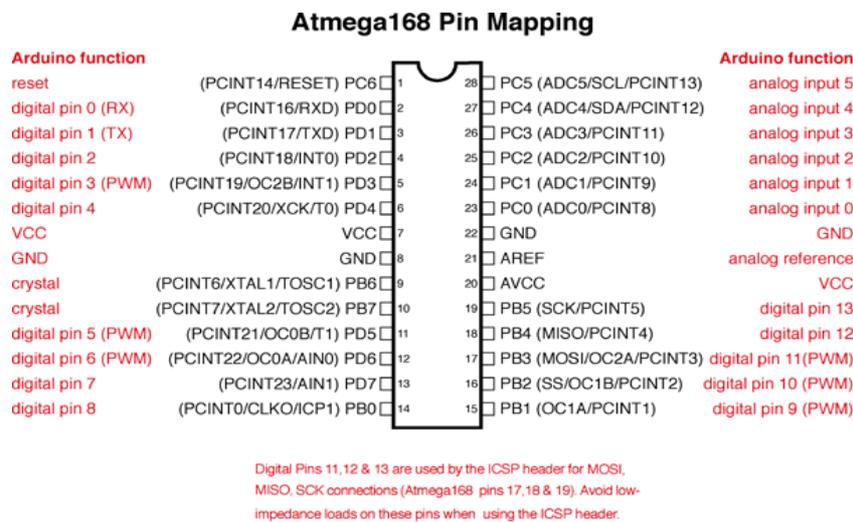


Figure 8 Architecture of Arduino

[11]

Arduino is an open-source easy-to-use platform for building electronic projects. It has a programmable controller and an Integrated Development Environment which is used to write and upload instructions unto the controller [12]. The Arduino Uno controller has two memories: program memory which stores code and data memory which stores data. The main IC is ATmega328P. The ATmega328P has 32KB of flash memory for storing code, 2KB of SRAM and 1KB of EEPROM and operates with a clock speed of 16MHz [13]. Arduino Uno has 14 GPIOs (general input-output pins) which allow the connection of sensors, actuators, and other electrical components.



*Figure 9 Pin mapping for Arduino Uno*

[14]

The Arduino Mega is an advanced model of the Arduino Uno microcontroller. The microchip of the Arduino Mega is ATmega2560 ( Pin mapping shown in Figure 10). The Arduino Mega has 54 input/output pins and has 256 kB of flash memory. It also has 16 analog pins and 14 pulse width modulation (PWM) pins [15].

The Arduino Mega was selected for this project due to its large flash memory as compared to the Uno. This decision was taken because some signal processing requires the use of buffers which may take up quite a large amount of memory.

Both the Arduino Uno and Mega have the ability to sample analog signals with the command 'AnalogRead,' which samples analog signals. It, however, does so at a rate of 8kHz. The goal of the project is to sample at about 40kHz. For this reason, bitwise operations had to be made to registers directly and interrupts used to run commands.

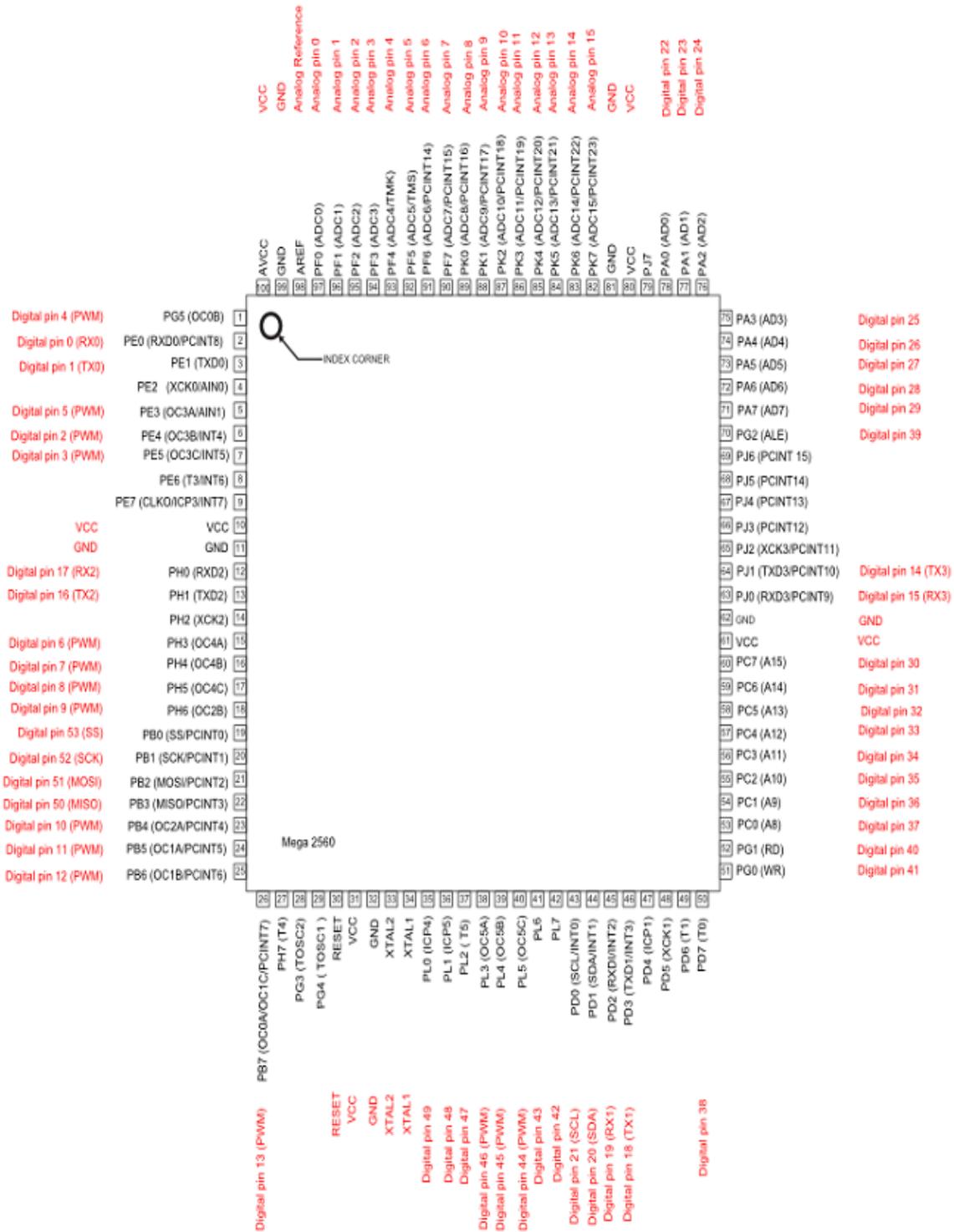
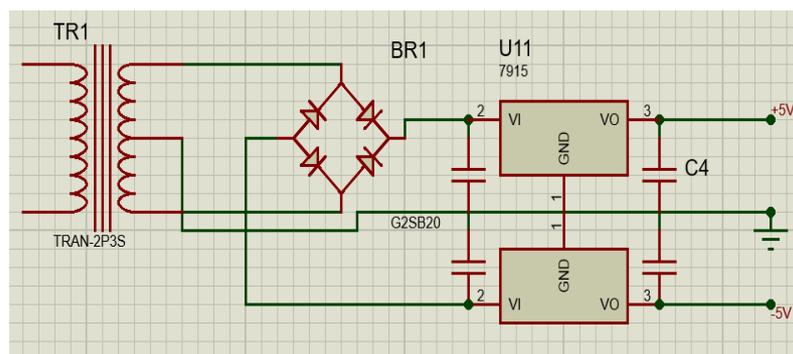


Figure 10 Pin Mapping for ATmega 2560

[16]

a. Power supply: The power supply will provide the right power to the various components of the product. This includes the micro-controller and the amplifier and the DC buffer. This project seeks to reduce the cost guitarists spend on their effects pedals. Hence, a power supply is designed which takes 230V AC and outputs a DC voltage for the operation of the device as opposed to the buying of batteries. This project needs a +5V and -5V supply. A full-wave bridge rectifier allows for DC generation on both the positive and negative half cycles. The average voltage output is, therefore, higher than the half wave rectifier's output. This circuit as shown in Figure 11, makes use of a center tapped transformer, and four power diodes arranged such a way that, for every half cycle, a pair of diodes allow current to be rectified to produce a positive voltage on the positive cycle and a negative voltage on the negative cycle. A smoothing capacitor is connected across the load to smoothen ripples in the rectified current.



*Figure 11 Full-wave bridge rectifier*

For this project, the full wave bridge rectifier was chosen because it provides both positive and negative voltages which are required for the functioning of some blocks.

## **Chapter 4: Implementation, Tests, and Results**

### **Implementation**

This project consists of two main parts: the hardware component and the software component (firmware). The hardware component consists of the physical and tangible parts of the project. This is mainly composed of the electrical circuit component. The main electrical components used are resistors, capacitors, and op-amps. This project avoided the use of pre-made modules to allow for more flexibility and to keep the cost as low as possible. The software consists of computer programs and the firmware of the micro-controller. Software used include LTSpice, MATLAB, Arduino IDE. Proteus was used to design the schematic for this project. In this section, the circuitry and software are explained.

#### **4.1 Hardware**

There are four main circuits in this project: The input circuit, the Digital-to-Analog converter, the output circuit and the power supply circuit.

##### **4.1.1 Input Circuit**

This circuit is responsible for receiving the guitar signal and get it ready for the micro-controller to sample and perform signal processes. This circuit consists of a ¼ guitar input jack which receives the cable from the guitar. Since the guitar produces a low voltage signal, about 1VRMS max, it needs to be amplified. The amplifier built is a non-inverting amplifier. The potentiometer is connected to the ground of the input jack which is connected to the feedback resistor. The purpose of the potentiometer is to get to be able to control the gain of the signal such that no signals are cut off. The inability to get a fixed resistor of the exact value required to keep the signal at an amplitude of 2.5V also influenced the use of the potentiometer. The signal is then passed through a non-inverting operational amplifier. The amplifier used is the UA741CP general purpose operational amplifier. The amplified signal is then passed through the DC offset which holds the signal oscillating about 2.5V. This is

important because the micro-controller reads a maximum voltage of 5V. If the signal oscillates about 0V, the lowest amplitude will be  $0V - 2.5V = -2.5V$  and which will not be read by the controller, thereby leading to aliasing. However, when the DC offset is set to 2.5V, the lowest amplitude will be  $2.5V - 2.5V = 0V$  and the highest amplitude will be  $2.5V + 2.5V = 5V$ . The micro-controller will be now be able to read all values of the incoming signal. From the DC offset, the signal goes to pin A0 of the Arduino Mega.

#### 4.1.2 Output Circuit

After the processing has been done on the Arduino, the output is passed to the R-2R ladder DAC. The R-2R method was chosen because of the readily availability of the components required. This is an arrangement of resistors which takes digital bits, weights them and sums them to produce a voltage between 0 and 5V (We are using 5V because the  $V_{max}$  for Arduino is 5V). The DAC built is an 8-bit meaning it has  $2^8 = 256$  levels between 0V and 5V. This means the digital signal coming is between 0 and 256 and will be converted into a voltage between 0 and 5V.

All the digital pins were connected to each section of the resistor ladder. Since this is a DAC, the digital signal was manually supplied (ie. between 0 and 255); Where 0 = 0V and 255 = 5V. Every pin on PORT D (Digital Input and Output Port) contributes to the digital value. The weighted binary sum of the pins on the port make up the digital value being supplied to the DAC.

The voltage output is given by  $\frac{digitalvalue}{255} * 5V$ . The period of a signal is the duration of each step \* the number of steps. For example, if I sample and hold a signal for 5 $\mu$ s, (still using an 8-bit DAC), then the total number of steps is 256. Then the period is

$$(50 * 10^{-6})s * 256 \text{ steps} = 0.0128s. \text{ Frequency is the inverse} = 78\text{Hz}.$$

The value of  $R=10K\Omega$   $2R=20K\Omega$ . The DAC is connected to the Arduino Mega by connecting the DAC inputs to Port C. The

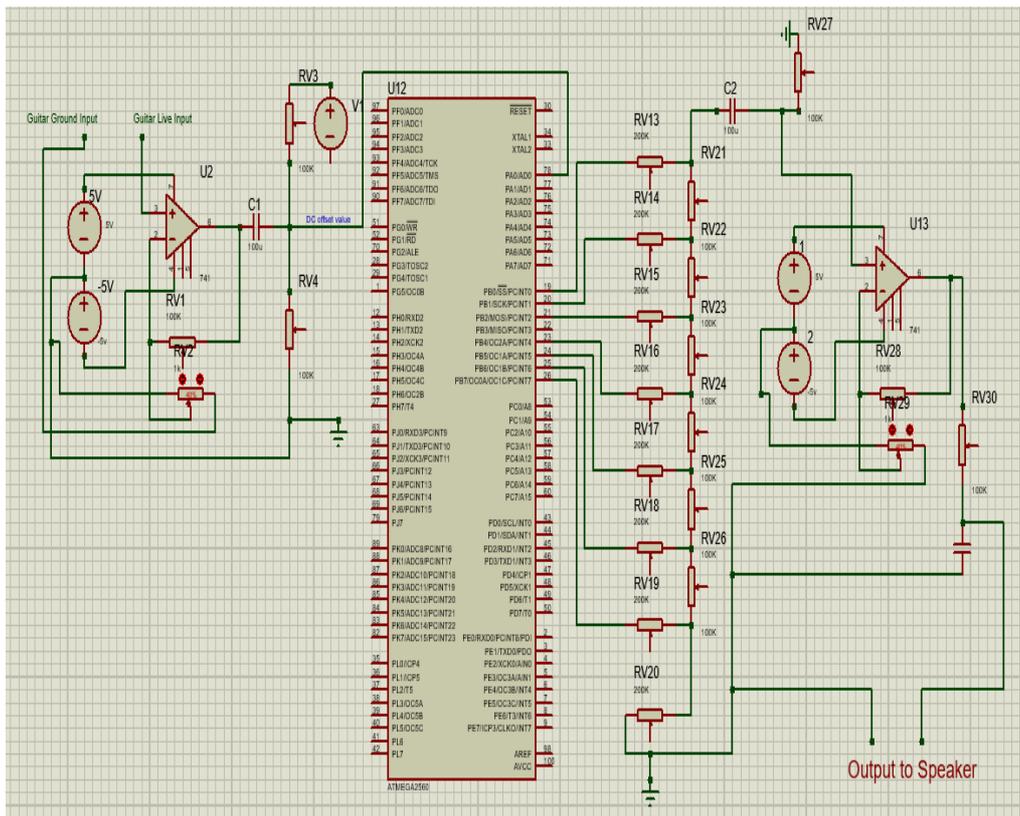
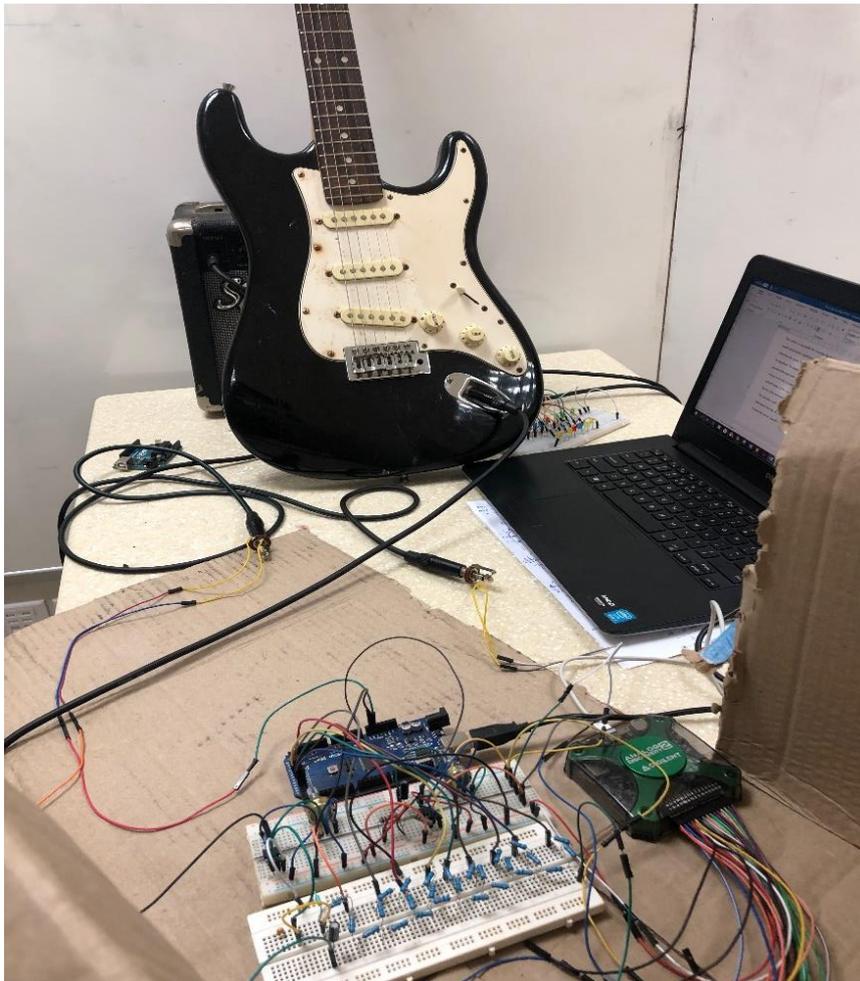


Figure 12 Complete Circuit Schematic



*Figure 13 Implemented Bread Board Circuit*

## **4.2 FIRMWARE**

The code was written in C/C++ using the Arduino IDE. It is code modified from a project by Amanda Ghassaei in 2012. Bitwise operations were performed on the ADC registers to configure them for operation. The code has an integer variable; ‘effects’ which determines the effects that will be operating. Delay, Tremolo, Reverb. Clean Boost.

- 1) Delay: A delay buffer was created to store 7,200 samples. These samples are stored into three different buffers named `buffer_write_index1`, `buffer_write_index2`, `buffer_write_index3`. Each buffer stores 1/3 of the 7,200 samples. At any time  $t$ , the new sample is added to the value at `buffer_write_index1`, `buffer_write_index2`, `buffer_write_index3`. The samples in

each buffer are played differently with different magnitude scales to determine the amplitude of the delay.

Pseudo-code for delay:

- The processor reads the next sample,  $S_n$ , from the ADC
- It reads the value,  $Br$ , at the current read index in the buffer.
- It adds the  $S_n$  to  $Br$  and writes the output to the DAC
- It writes the value  $S_n$  to the buffer at indices,  $Bw1$ ,  $Bw2$  and  $Bw3$ , where:
  - $Bw1 = Br + (7200/3)$
  - $Bw2 = Br + 2*(7200/3)$
  - $Bw3 = Br + 3*(7200/3)$
- It then increases  $Br$  by one
- Definitions:
  - $S_n$ : Current sample value from ADC
  - $Br$ : Current read index of delay buffer
  - $Bw1$ : Current write index for first delay in buffer
  - $Bw2$ : current write index for second delay in buffer
  - $Bw3$ : current write index for third delay in buffer

2) Clean boost: This effect type boosts the sound of the clean guitar. Only the amplitude of the signal is affected. It is achieved by the circuit amplification of the signal. No manipulation was made in the software to increase the amplitude because the circuit amplification gets the signal close enough to the clipping amplitude.

- 3) Tremolo: This is a modulation effect that rhythmically changes the amplitude of the signal. A low-frequency oscillator is convolved with signal which results in the signal oscillating. [17]

Pseudo code for tremolo

- The user inputs the tremolo depth and time, which will be used to determine the parameters,  $f$  and  $C$  in the equation:  $V(t) = A\sin(2\pi f t) + C$ .
- The processor reads the next sample,  $S_n$ , from the ADC
- The volume coefficient,  $V_c$ , is determined from the equation for  $V(t)$
- The output,  $V_c * S_n$ , is written to the DAC

- 4) Reverberation:

Pseudo code:

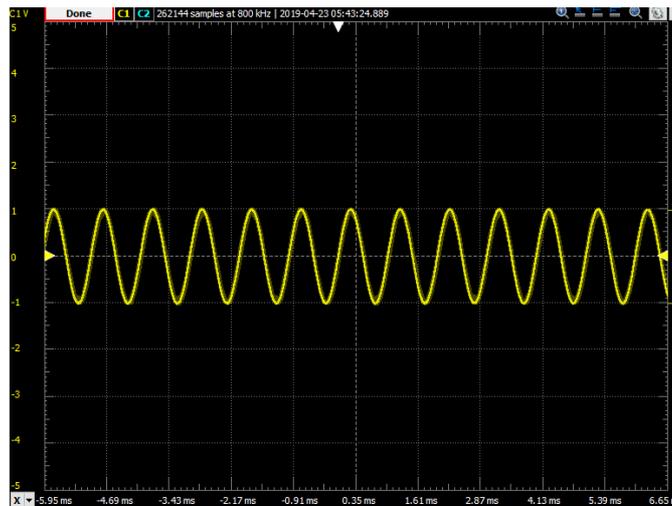
- The impulse response for a reverb room was recorded and saved in an array on the processor.
- The processor then reads values from the ADC into a buffer and convolves impulse response of the reverb, with the values stored in the buffer. It then writes this output to the DAC

## Tests and Results

To test the circuit, A sinusoidal signal of 1Hz was passed into the input of the circuit by the wave generator of analog discovery 2. The output was recorded when the signal passed through each block.

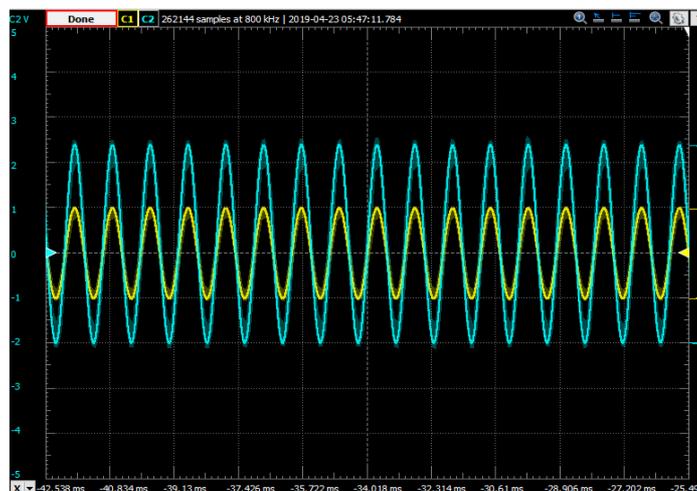
All graphs are plotted together with the original input signal (yellow graph).

Input signal:



*Figure 14 Input Signal: 1Hz, 1V peak-to-peak*

The input signal has an amplitude of 1V and a frequency of 1Hz. The scope of the signal is taken at the input terminal of the circuit. Signal after passing through amplifier



*Figure 15 Amplified Signal vrs Original Signal*

The signal passes through the amplifier. The amplified signal (blue) still oscillates about 0V and is in phase with the input signal.

After 2.5V DC offset

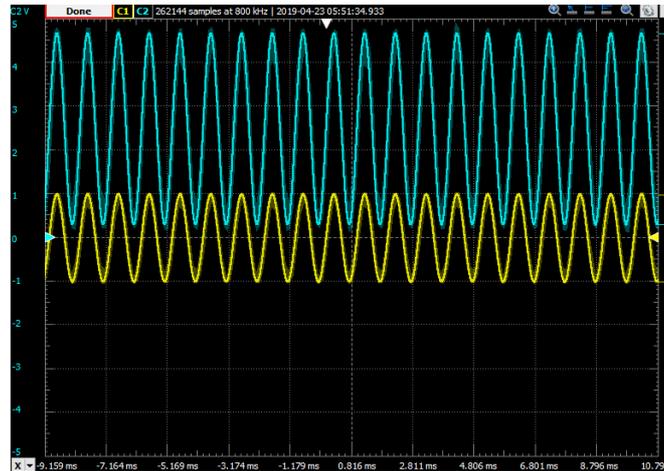


Figure 16 Amplified Signal, offset at 2.5V vs Original Signal

The signal then passes through a 2.5 DC offset. This causes the signal to oscillate about 2.5V. This is necessary for the Arduino Mega to be able to sample the signal since the signals above 5V and below 0V cannot be read.

After DAC (Notice vivid steps in signal)

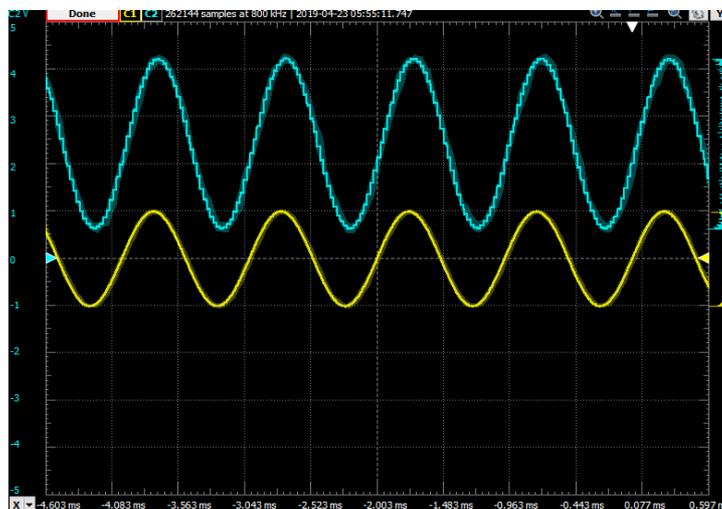
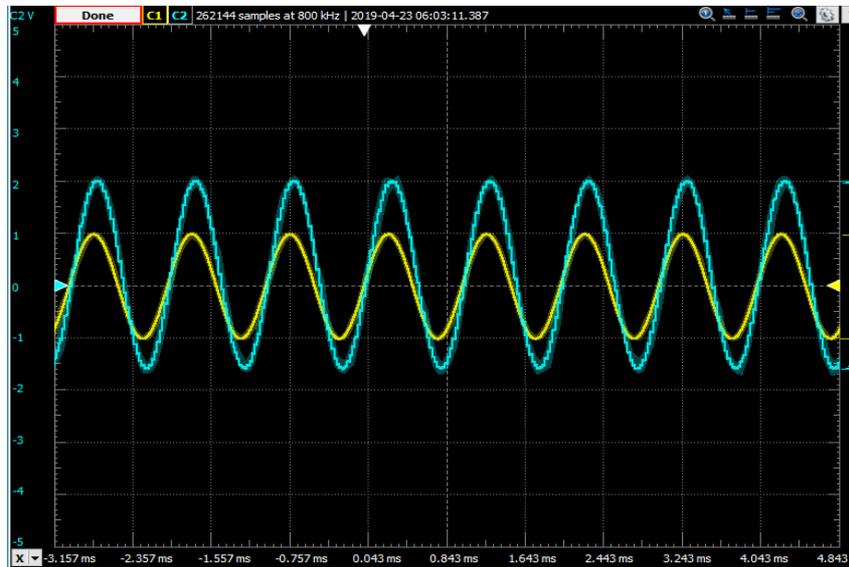


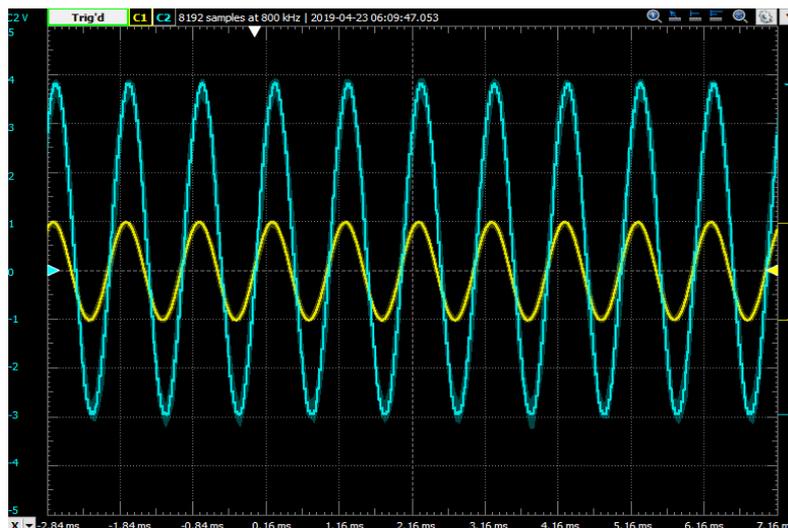
Figure 17 Digital-Analog Converted Signal vs Original Signal

After the signal is processed by the microcontroller, it passes through a DAC to convert the signal back to an analog signal. This is evident in the visible ripples in the signal (amplified signal oscillating about 2.5V). The ripples are as a result of sampling of the input signal phase.



*Figure 18 0V offset vrs Original Signal*

The analog signal passes through a 0V DC buffer. This causes the signal to oscillate about 0V. This is necessary to reduce the noise in the output. The signal is still in phase with the original input signal.



*Figure 19 Amplified output signal*

The signal is finally amplified through an output amplifier. The signal final signal is an amplified version of the input signal and is still in phase.

The effects were tested in MATLAB before their implementation in the arduino environment. Due to memory issues, some could not be implemented. This signal was used for the testing of the effects that were generated.

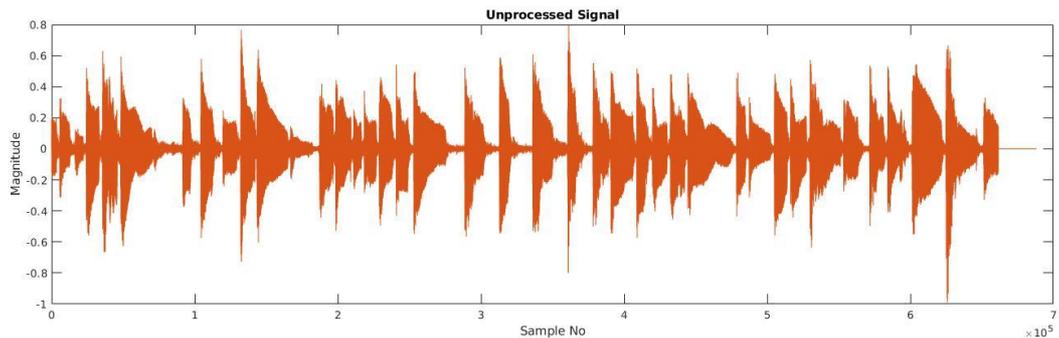


Figure 20 Unprocessed Signal

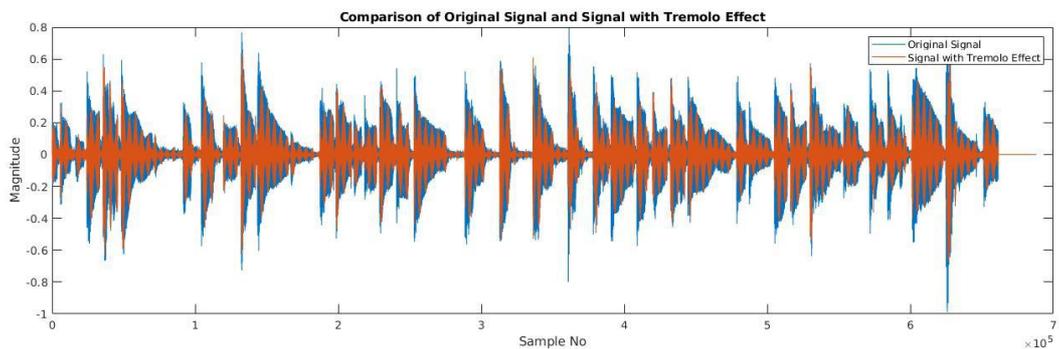


Figure 21 Original Signal(orange) vrs signal with Tremolo effect(blue)

For the implementation and testing of the reverberation effect, an impulse response was generated for an impulse input in a room with reverb. The impulse input used was a snare drum.

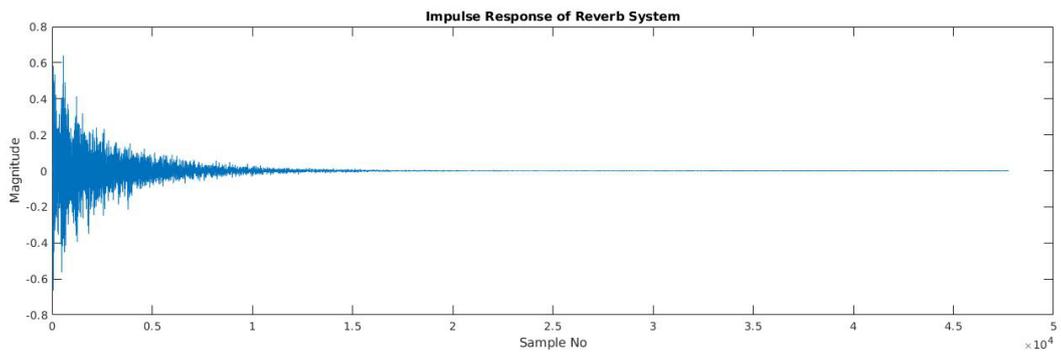


Figure 22 Impulse response of a reverb system

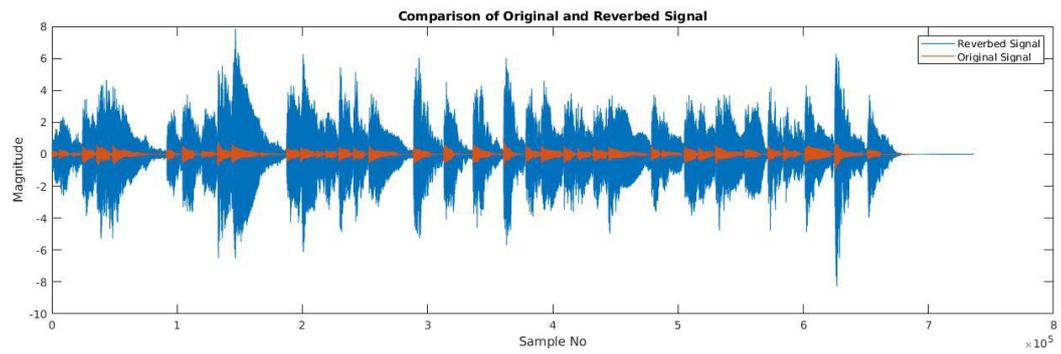


Figure 23 Original Signal vrs Reverbed Signal

## **Chapter 4: Future improvement, limitations, conclusion**

This project can be extended to have more effects like distortion, flanger, fuzz, etc. A voice box effect can be incorporated to create a new sound effect. For this to be achieved, there is a need for a second analog to digital converter for the microphone. There are two separate ADCs for the microphone and guitar signal because they will have to be working simultaneously. After these signals are processed, they will be an input into the micro-controller where processing will be on the digital signals. At this block, the equalizer, volume and function switches can be used as to affect the output sound. System memory stores the functions and commands for operation by the microprocessor. The final digital signal produced by the microcontroller is converted to an analog signal and sent to an amplifier. Wireless transfer of signals is also a feature that can be explored. Bluetooth communication might be an option to achieve.

Some limitations faced in this project was lack of sufficient memory for the Arduino. Again, noise in the output signal was expected to be generated from loose breadboard connections.

This project has been a learning experience and has been a successful proof of concept. This project can be pursued to create a marketable product.

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## APPENDIX

Code

```
//Half Frequency Playback
```

```
//by Amanda Ghassaei
```

```
//https://www.instructables.com/id/Arduino-Vocal-Effects-Box/
```

```
//July 2012
```

```
/*
```

```
* This program is free software; you can redistribute it and/or modify  
* it under the terms of the GNU General Public License as published by  
* the Free Software Foundation; either version 3 of the License, or  
* (at your option) any later version.
```

```
*
```

```
*/
```

```
int effect = 2;
```

```
// Parameters for delay
```

```
// Parameters for tremolo effects
```

```
// User-defined parameters
```

```
float tremolo_time = 0.1; // s
```

```
float tremolo_depth = 0.8;
```

```
// Calculated parameters
```

```
float tremolo_frequency = 1 / tremolo_time;
```

```
float tremolo_peak = 3;
```

```
float tremolo_amplitude = tremolo_peak / 2;
```

```

float tremolo_offset = 2*tremolo_amplitude - tremolo_amplitude*tremolo_depth;

float tremolo_multiplier = 1;

float pi = 3.14159265358979323846;

//audio storage

const unsigned long sample_rate = 9000; // Hz

const unsigned long buffer_size = 0.8 * sample_rate;

byte delay_buffer[buffer_size];

byte incomingAudio;//storage variable

unsigned long buffer_read_index = 0;

unsigned long buffer_write_index1 = buffer_read_index + buffer_size / 3 - 1;

unsigned long buffer_write_index2 = buffer_read_index + (buffer_size / 3) * 2 - 1;

unsigned long buffer_write_index3 = buffer_read_index + (buffer_size / 3) * 3 - 1;

float magScale1 = 0.8;

float magScale2 = 0.8;

float magScale3 = 0.8;

byte get_from_buffer(unsigned long index){

    return delay_buffer[index];

}

void append_to_buffer(unsigned long index, byte new_value){

    delay_buffer[index] = delay_buffer[index] + new_value;

```

```
}
```

```
void remove_from_buffer(unsigned long index){
```

```
    delay_buffer[index] = 0;
```

```
}
```

```
void setup(){
```

```
    DDRC = B11111111;
```

```
    DDRF = B00000000;
```

```
    cli();//diabile interrupts
```

```
    //set up continuous sampling of analog pin 0
```

```
    //clear ADCSRA and ADCSRB registers
```

```
    ADCSRA=0;
```

```
    ADCSRB = 0;
```

```
    ADMUX |= (1 << REFS0); //set reference voltage
```

```
    ADMUX |= (1 << ADLAR); //left align the ADC value- so we can read highest 8 bits from ADCH  
register only
```

```
    ADCSRA |= (1 << ADPS2) | (1 << ADPS0); //set ADC clock with 32 prescaler- 16mHz/32=500kHz
```

```
    ADCSRA |= (1 << ADATE); //enable auto trigger
```

```
    ADCSRA |= (1 << ADIE); //enable interrupts when measurement complete
```

```
    ADCSRA |= (1 << ADEN); //enable ADC
```

```

ADCSRA |= (1 << ADSC); //start ADC measurements

sei();//enable interrupts

Serial.begin(9600);

}

ISR(ADC_vect) { //when new ADC value ready

incomingAudio = ADCH;

// This is the delay effect

if (effect == 0){

// Play incoming signal plus delay at that point

PORTC = get_from_buffer(buffer_read_index) + incomingAudio;

remove_from_buffer(buffer_read_index);

// Store delayed signals

append_to_buffer(buffer_write_index2, incomingAudio * magScale2);

append_to_buffer(buffer_write_index3, incomingAudio * magScale3);

// Increment buffer indexes

buffer_read_index = (buffer_read_index + 1) % buffer_size;

buffer_write_index1 = (buffer_write_index1 + 1) % buffer_size;

buffer_write_index2 = (buffer_write_index2 + 1) % buffer_size;

buffer_write_index3 = (buffer_write_index3 + 1) % buffer_size;

}

```

```

// This is the tremolo effect

if (effect == 1){

    tremolo_multiplier = tremolo_amplitude * sin(2*pi*tremolo_frequency*micros()*pow(10,-6)) +
tremolo_offset;

    PORTC = tremolo_multiplier * incomingAudio;

    Serial.println(tremolo_multiplier);

}

// This is the clean guitar(no effect)

if (effect == 2){

    PORTC =incomingAudio;

}

}

void loop(){

}

```