

# **G12 PedalVision**

*User Programmable Instrument Multi Effects Pedal and Light Interface*



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### **Biographies**

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**Brian Boga** is an Electrical Engineering student here at the University of Central Florida. He started school initially after a degree in Mechanical Engineering, but quickly changed his path to Electrical Engineering after some careful evaluation of potential jobs that he desired to have in the future, and the best way of obtaining those positions was with a degree in Electrical Engineering. Brian is also a current member in the United States Air Force Reserves where he gets to both load munitions onto fighter jets, but he also gets to troubleshoot Electrical issues within the weapons release systems which has helped him reinforce some electrical and computer information that he had learned while in school. He has not dealt with many of this projects problems, such as digital signal processing, frequency calculations, or power supply design. That in mind he is thrilled to obtain this knowledge through research as well as through trial and error.

**Kevin Leone** is a Computer Engineering student here at the University of Central Florida. He started school at UCF in 2013 without initially knowing what he wanted to do. After two years, and building his own computer later, he decided to pursue a Computer Engineering degree. He also has an interest in mathematics, giving him the motivation to complete a mathematics minor. He is currently working as a Software Engineering Intern through UCF's CWEP, where he can work with embedded and high level programming. This project was a new topic for him, since he was never introduced to digital signal processing in the courses that were required to be taken. Also, he did not have any previous experience with guitars or effects pedals. Although this may have held him back initially, the research was found to be very interesting. He primarily shows interest in software design, but is very eager to learn and understand the hardware applications of any system.

**Jose Ramirez** is an Electrical Engineering student at the University of Central Florida. He started his at UCF in 2013 without initially which discipline to pursue. He knew he wanted something that he could incorporate into his interest in audio recording and live performance, so after having conversations with professors upon completion of his first semester he decided to pursue an Electrical Engineering degree. Due to the artistic nature of this project, he was thrilled to be able to work on a product that he has used before. Having seen disadvantages of current products, he wanted to design a system without the current disadvantages. He is currently working as a Reliability Engineering Intern through the College Work Experience Program, where he is helping track failures and implement corrective actions to ensure a high-quality product is delivered to the customer. He has recently accepted a job offer from Texas Instruments for an Applications Engineer position, which he will start upon graduation from UCF.

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## 1. Executive Summary

When it comes to music an individual's sound is everything. Musicians are constantly searching for the "best" sound they can find, whether that be with regards to a specific tone quality or cool sounding effect to enhance what is being played. Sometimes people want something to enhance a show's visual aspect and want the visual part of their performance to match their audio portion. With this project idea we aim to kill two birds with one stone by allowing the user an assortment of sounds, both digital and analog, and a light display that will react to the electrical signal created by the instrument playing.

As it stands currently, many musicians crave the "warm" sound of analog circuitry but must purchase each analog effect unit separately, thus creating a large foot print that the musician has to worry about with regards to mobility. On top of having to purchase each effect separately the price of each box is high in comparison to the cost of the components that make up the effect unit. The only alternative to paying these high prices is to switch to digital effects. On top of the reduced price of the digital effects, they offer more customizability in that they can be programmed to recall settings instantly. Although the cost of a digital effects unit with multiple sounds is relatively cheap as compared with the analog unit, many musicians feel that the digital effects are not as "organic" sounding or have a "fake" sound to them; therefore, the digital effects are seen as inferior.

With this project we aim to combine the best of both worlds by including both the "warm" tones of analog effects and the versatility of digital effects into one unit. This combination will help reduce the overall footprint of the multi effects unit and save the user money since they will only need to buy one unit as opposed to a different unit for each effect they want.

In addition to the audio effects, this project aims to enhance the user's visual performance as well. The multi effects board will be able to interface with an LED display, allowing the user multiple modes of operation for the display. The user could use the LED display to determine if the current pitch being played matches the musical pitch standard A440 (440 Hz), the musical note A above middle C used as a general standard for musical pitch, in order to ensure pleasant listening experience. The LED display could also strobe or change color depending on what the output of the user's instrument indicates.

Ultimately, the group's objective is to create a product that will have the capability to satisfy the modern musician's desire for analog effects, programmability and versatility of digital effects, and the visual appeal of the attachable LED display in order to create the ultimate performance tool.

## **2. Description**

### **2.1. Project Motivation**

For many guitar players finding the perfect sound is an uphill battle, requiring many hours of trying different stomp boxes and dialing in the best settings for that musician's taste. Then once the musician is happy with one of their stomp boxes and they connect it to a second box they come to find out that they need to change their settings again in order to compensate for the effects of the second box on their desired sound. On top of this already frustrating process the price of each box can be very high, making the process of building a specific sound very costly with respect to time and money. The purpose of this project is to create an all-in-one unit that will have all of the common effects musicians are looking at an affordable price point and portable size. On top of the benefits of the all-in-one unit, the project will have a visual display made of LEDs to serve as a tuning device and as a fun visual display that changes color and pattern depending on what the musician is playing. This will allow the performer to play whatever he or she wishes and not have to worry about following a strict tempo in order to have their visual effects match their performance. With this product we hope to give the musician everything they need to have a great performance both aurally and visually.

### **2.2. Project Goals and Objectives**

With this project being an all-in-one unit we want to meet the needs of many different musicians and their playing styles. Many musicians want the classic sound quality of analog effects. Many players believe that the analog effects produce a warmer more "organic" tone. But since these are analog effects they are not easy to change. With this in mind some musicians instead want the versatility of the digital effects. Although they do not produce the exact sound of the input, with today's technology the reproduced signal is getting closer to being reconstructed more accurately, making the digital effects sound closer to the analog effects. Due to this divide the project will have the best of both worlds by combining the analog and digital effects. The analog effects will be have effects such as overdrive and distortion which do not need to be changed frequently as they play more of a tone conditioning role in a musician's desired sound. While the digital effects, such as delay and reverberation, will be constantly changed depending on the tempo of the song and how much of the effects the musician wants to hear.

While performing the musician needs to put on a great show, with this the visual aspect comes into play. By incorporating a multi-functional LED display the performer can tune their instrument exactly by using the LED to indicate if the pitch of the instrument matches tuning standard of 440Hz. The Display can also be used as a fun display to help add dimension to the performance. A system block diagram is shown in Figure 1 below. With this product the musician will have the ability to produce classic sounds with modern technology, excellent versatility due to the digital effects, and an eye catching display to enhance their performances.

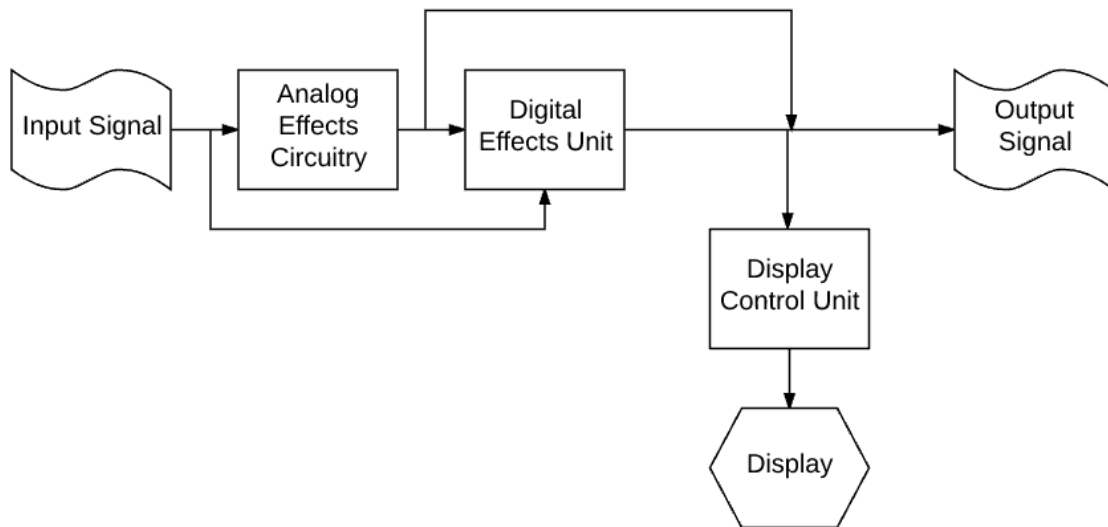


Figure 1: System Block Diagram

## 2.3. Requirement Specifications

The G12 PedalVision will have the capability to create soft overdrive clipping, hard distortion clipping, many equalization filtering options and a compression/limiting circuit all through analog circuitry. The unit will also have the capability to create delay/echo effect, reverberation, and flange through the digital controller of the effects unit. A user programmable option will be available in order to allow for more flexibility and for possible expansions of the system. The addition of an LED display will also be incorporated which will be able to read the input provided by the effects output of from the output of the instrument itself in order to determine what action it will take with regards to what to display. All effects will be able to be toggled on and off through foot switches. Settings for the digital and analog effects can be adjusted by hand through knobs, while the digital effects and LED control will also be able to be adjusted through a connection through USB.

- Analog effects
  - Input impedance of at least 500K
  - Output impedance of no more than 10K
  - Bypass full frequency response from 20 Hz-20 KHz
  - Knobs to adjust volume, drive, and tone
  - Foot controls to toggle effect on and off
- Digital effects
  - DSP chips will be used to create the multiple digital effects
  - LCD interface for changing settings of DSP chip.
  - Knobs for adjusting digital values such as speed, depth, and wet signal
  - Foot control to toggle effect on and off
- LED display
  - A microcontroller will be used to control the LED operation
  - LED display will have multiple modes of operation

- Size and Weight
  - No more than 30 lbs
  - No larger than 15  $in^3$

## 2.4. House Of Quality Analysis

In order to ensure that we have a successful product we need to take into account the different factors that will affect the functionality and appeal of the product. Understanding the relationship between function and market factors is key as they strongly affect each other. The engineering and marketing tradeoffs are shown in Table 1 below to better illustrate the main points that will affect the success of our product.

- Positive Correlation =  $\uparrow$
- Strong Positive Correlation =  $\uparrow\uparrow$
- Negative Correlation =  $\downarrow$
- String Negative Correlation =  $\downarrow\downarrow$
- Increase Requirement = +
- Decrease Requirement = -

			Engineering						
			Cost	Size	Weight	Input Impedance	Output Impedance	Power Consumption Audio Effects	Number of Effects
			-	-	-	+	-	-	+
Market	Sound Quality	+	$\downarrow\downarrow$	N/A	N/A	$\uparrow$	$\uparrow$	N/A	N/A
	Versatility	+	$\downarrow$	$\downarrow$	$\downarrow$	$\uparrow$	$\uparrow$	$\uparrow$	$\uparrow\uparrow$
	Ease of Use	+	N/A	N/A	N/A	N/A	N/A	N/A	$\downarrow$
	Portable	+	$\uparrow$	$\uparrow\uparrow$	$\uparrow\uparrow$	N/A	N/A	$\uparrow$	$\downarrow$
	Cost	-	$\uparrow\uparrow$	$\uparrow$	$\uparrow$	$\downarrow$	$\downarrow$	N/A	$\downarrow$
			Less than \$500	Smaller than 15 $in^3$	Under 30 lbs	Min 500k	Max 10K	Max 1.5W	5 minimum

**Table 1: G12 PedalVision Tradeoff Matrix**

The cost analysis can be broken into two parts; audio, and LED system. The estimated cost for the audio portion is less than \$300. The estimated cost for the LED system is less than \$200.

## 3. Project Related Research

### 3.1. Analog Effects

For this project we will have multiple analog audio effects. The effects that will be included in this design are overdrive, distortion, compression, and equalization filter network. On top of the aforementioned effects, an adjustable input stage will be implemented as well. A flow chart of the analog effects is depicted in Figure 2 below.



Although we could have implemented many different types of effects, we decided to include these four effects since they are the most common effects used by the modern musician. Another reason we decided to implement these effects using analog circuitry is that they are more of a “set and forget” effect. The reason for this is that these effects deal more with the frequency aspect of the musician’s sound which is not changed frequently between songs unlike other effects like delay and reverb. With this in mind we do not need to implement these digitally to allow for quick modification. Another reason to implement these effects using analog circuitry is that the output waveform will be exact as opposed to a digital implementation which will not have the same signal reconstructed exactly due to the effect of sampling and digital level approximations.

In order to activate each effect, there will be a toggle system for each effect that the user will be able to manipulate. This system will allow the user to have a single or multiple effects active at the same time through the use of multiple footswitches, one per effect to be exact. The foot switching system is crucial to this project due to the fact that the user may need to change effects quickly while playing his/her instrument and cannot use their hand to toggle the desired effect. The values of the many parameters such as gain, drive, tone, threshold, and volume will be controlled through many knobs the user will be able to adjust by hand on the effects unit. Having the ability to change the parameter values and toggling ability will allow for better user control of each of the analog effect.

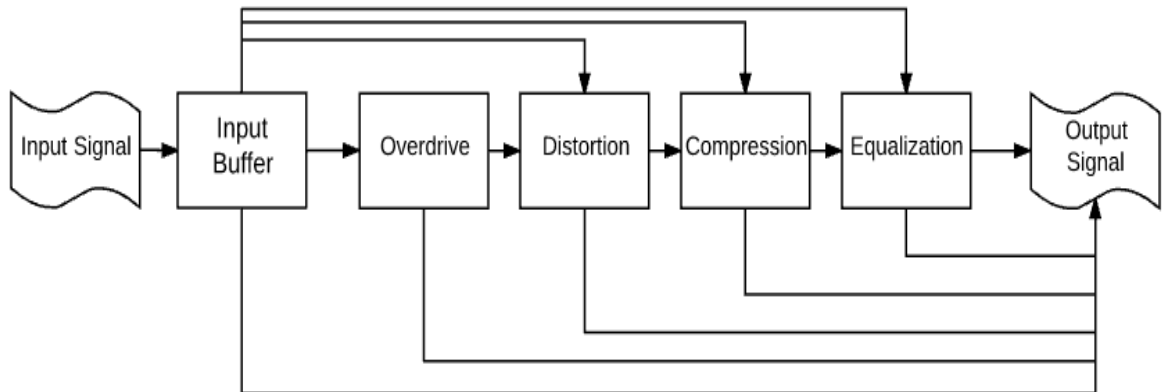


Figure 2: Analog Effects Flowchart

The following sections will cover explanations of how each effect will work individually. An explanation of common implementations of each analog effect will be discussed in detail as well. Finally a breakdown of the chosen parts and a comparison between components will be included as well in the sections to come.

### 3.1.1. Necessary Features

These are key features that the analog effects unit must have, and some features that may be added to the product in its later life due to time constraints.

- Must haves
  - Must be able to take pass original signal unaffected when in bypass mode
  - Overdrive circuit must be able to adjust overall gain
  - Distortion circuit must be able to adjust overall gain
  - Compression circuit must be able to amplify small signals, and attenuate large signals quickly without unwanted modulations being added
  - Equalization network cut off frequencies must be able to be adjusted by the user
  - Equalization network filters must be able to be changed
  - All analog effects must have an LED light indicator to show if the desired effect is toggled on or not
  - Analog effects must have knobs and switches in order to allow the user to easily adjust the unit to their desired settings.
  
- Expandability/Wants
  - May have the ability to link third party effects units after the analog effects but before the digital effects unit
  - May have various extra capacitors to control some of the lower frequencies that are passed in the overdrive and distortion circuits

### **3.1.2. Input Buffer**

Although the input buffer may not seem like a must have effect in a signal chain it plays a very vital role in keeping your signal true to what you are playing. The purpose of the input buffer is to replicate what it sees at the input at the output of the circuit. This basic function allows the more accurately replicated signal to pass through the rest of the circuits in the signal processing chain. When implemented correctly, the input buffer will increase the high frequency response of the signal chain due to the capacitance of instrument cables, long lengths of the instrument cable, and other effects that may be present further down the signal chain. The key to a good input buffer is a high input impedance and a low output impedance.

#### **3.1.2.1. Input Buffer Implementation**

In order to create an effective input buffer pedal you need and a high input impedance and a relatively low output impedance. To do this we can use either transistors or an op amp. Some of the transistors that could be used to create an input buffer are bipolar junction transistors, mosfets, and jfets. For all three of these transistors you connect your input signal to the base or gate of the device while the output can be connected to the source or emitter of the device.

For comparison sake, the first device for the input buffer we will look at is the mosfet. The mosfet in theory has a very high input impedance due to the oxide separating the gate from the substrate of the device. But since this gate is very sensitive to certain charges extra protection is needed. Also since mosfets have a relatively high capacitance they may remove some of the higher frequency components that we are trying to preserve.

The next device we will look at is the jfet. Like the mosfet, the jfet has a very high input impedance. Unlike the mosfet, the jfet does not have the problem of gate sensitivity so it is better to this application due to its ruggedness as well. The only down side is that it does not provide the lowest output impedance. Many input buffer circuits use jfets though due to more linear properties of this device.

Bipolar junction transistors are the last transistor we will look at for the input buffer options. The bipolar junction transistor is a reliable part that is not as sensitive to base charge like the mosfet. Although it does not have the highest input impedance it does allow for a very low output impedance which is another property we are looking for.

Another option for an input buffer is an op amp. An op amp allows for the fewest number of parts for this input buffer circuit and has true unity gain of 1. The input impedance of an op amp is very large like that of a jfet and a mosfet and has a low output impedance like that of a bipolar junction transistor.

A summary of the above information is illustrated in Table 2 below.

Device	Input Impedance	Output Impedance	Number of Components
<b>MOSFET</b>	High	Low	High
<b>JFET</b>	High	Medium	Medium
<b>BJT</b>	Medium	Low	Medium
<b>Op Amp</b>	High	Low	Low

Table 2: Input Buffer Summary

### 3.1.2.2. Input Buffer Part Selection

For the selection of the input buffer we need to consider which has the highest input impedance and the lowest output impedance. With only looking at those criteria the op amp is the ideal choice for the input buffer circuit. Many argue though that the sound of an op amp is cold and sterile and that it does not color the sound. But since the purpose of an input buffer is to replicate the input signal exactly at the output, the circuit should not color the tone of the signal.

The op amps that we are looking at for this function are:

- TL07xx Low-Noise JFET-Input Operational Amplifiers
- OPA827 Low-Noise, High-Precision, JFET-Input Operational Amplifier
- OPA164x SoundPlus™ High-Performance, JFET-Input Audio Operational Amplifiers

For this input buffer circuit we want to have the most transparent design as possible. Meaning we want, once again, a high input impedance, low output impedance and large unity gain frequency response in order to allow for all possible audio frequencies to pass unaltered. Other parameters we want to look at are the signal to noise ratio and the distortion of the op amp as well since we do not want the op amp to create a noisy signal or color the tone due to internal distortion.

The first parameters we will look at for these op amps are the input and output impedance. For a good input buffer we need the input resistance to be as high as possible while having a low output impedance as well. Upon inspection of the data sheet of each op amp the following input and output impedances were found. The value of each input and output impedance is shown in Table 3 below.

Op Amp Model	Input Impedance	Output Impedance @1k
TL07xx	$10^{12} \Omega$	Not in data sheet
OPA827	$10^{13} \Omega$	20 $\Omega$
OPA164x	$10^{13} \Omega$	10 $\Omega$

Table 3: Input Impedance Comparison (Courtesy of Texas Instruments)

We can see from the table that the input impedance of the OPA827 and OPA164x are one order of magnitude higher than the TL07xx. We also see that at 1 KHz the output impedance of the OPA827 and OPA164x are in the tens of ohms. With these parameters the OPA827 and OPA164x are the better candidates if we look at the input I and output impedances.

Now that we know the input and output impedance of the three potential op amps we now need to look at their characteristic with regards to their frequency response, signal to noise ratio and internal harmonic distortion. It is important to look at these parameters of the op amps because we want the signal passing through the circuit to be as clean as possible. Table 4 below illustrates the parameter values of each of the three op amps.

Op Amp Model	Gain-Bandwidth Product	Input Voltage Noise @ 1KHz	Total Harmonic Distortion
TL07xx	3 MHz	18 nV/ $\sqrt{\text{Hz}}$	0.003%
OPA827	22 MHz	4 nV/ $\sqrt{\text{Hz}}$	0.00004%
OPA164x	11 MHz	5.1 nV/ $\sqrt{\text{Hz}}$	0.00005%

Table 4: Frequency and Noise Comparison (Courtesy of Texas Instruments)

With these parameters now investigated the obvious choices would be either the OPA827 or the OPA164x op amps. The OPA827 is only available in 1 channel packaging while the OPA164x is available in 1, 2, or 4 channel packages. This variability packaging allows for more flexibility when designing the circuit while still retaining the same

properties we are after. The price of the two op amps will also play a big roll in this decision as well. As you can see the price of the OPA827 is very high in comparison to the OPA164 for the same number of channels. The price of the different op amps and their configurations are in Table 5 below.

Op Amp Model	Number of Channels	Price
OPA827	1	\$10.13
OPA164x	1	\$2.88
OPA164x	2	\$4.20
OPA164x	4	\$5.29

Table 5: Cost Comparison (Courtesy of Texas Instruments)

### 3.1.3. Overdrive and Distortion

Since their creation, overdrive and distortion effects can be heard all over contemporary music. Overdrive is created when a signal begins to clip, thus flattening the peaks of the incoming signal. In order to create this effect in the 1950's and 1960's when guitar amplifiers were starting to become popular in popular music, the musician had to turn up the volume of their amplifier to very high levels in order to create this clipping effect. Although this did work, it was not pleasing to the ear to play at such loud volumes. Due to this problem, different ways of clipping one's signal were created later on. Although the two effects function very similarly, for the sake of this product overdrive will be considered as soft clipping while distortion will be considered as hard clipping. Examples of simple overdrive and distortion circuits and their respective output waveforms are shown in Figure 3 and Figure 4 below. Soft vs Hard clipping is a descriptive of how flat the signal peaks become when view at the output.

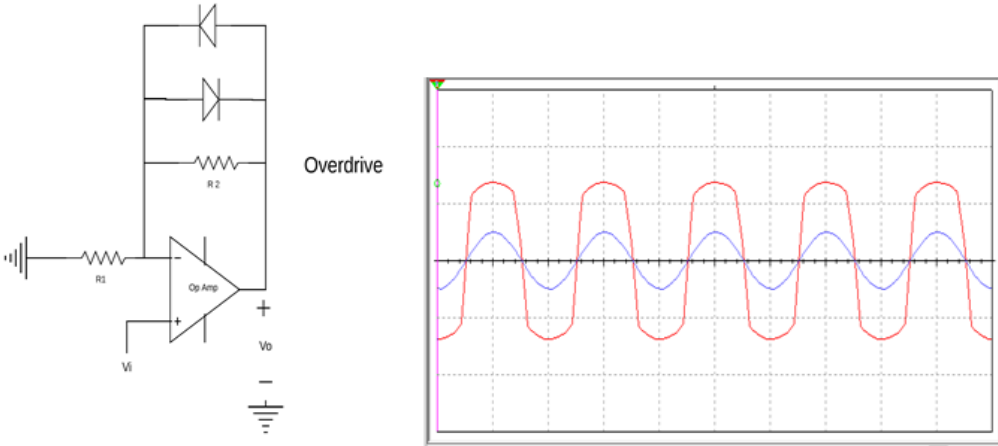


Figure 3: Simple Overdrive Circuit and Waveforms

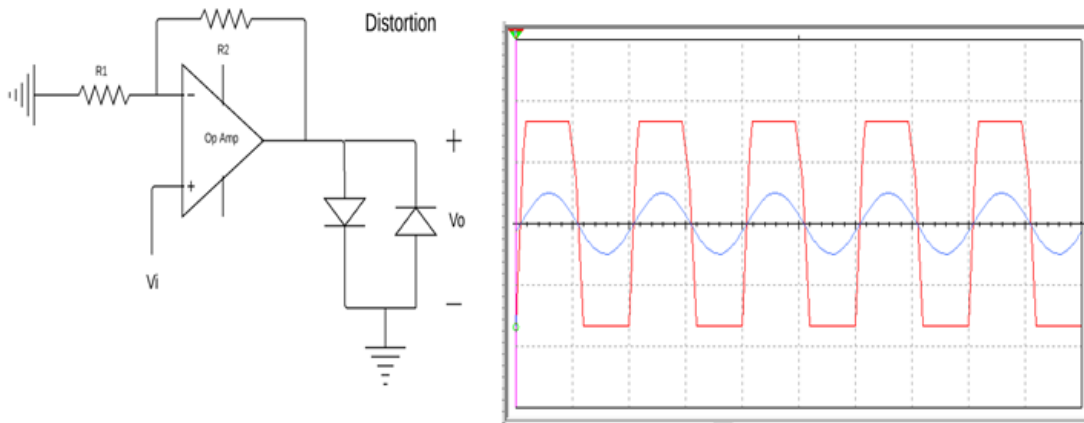


Figure 4: Simple Distortion Circuit and Waveforms

### 3.1.3.1. Overdrive and Distortion Implementations

In order to create an overdrive or distortion effect we need to clip the top and bottom of our waveform. Traditionally this has been done by amplifying the signal so much that it reaches the rails or maximum bias of the amplifier which will then truncate the signal passing through the amplifier. Today different circuits have been designed to achieve this clipping without amplifying the signal to extreme levels.

The first method is to link multiple gain stages together in order to amplify the input signal close to the rail or bias voltage of the devices, thus having the need to reduce the amplitude at the output with a voltage divider. A solution for this is through the implementation of diodes in the circuits. By implementing diodes in the circuits we can avoid having to greatly amplify the signal to the point of rail clipping.

One method of creating soft clipping is to create an amplifier with two back to back diodes connected from the collector to the base of the transistor. This creates a soft clipping effect but has a limited amount of gain. If this circuit is linked to a second identical circuit then this circuit combination creates more of a hard clipped sound. The output from the two circuits linked together creates a “fuzzy” sound.

If you do not want a “fuzzy” tone quality but still want soft clipping in order to create some harmonic distortion then placing two back to back diodes in the feedback loop of an amplifier circuit will do the job nicely. This soft clipping may add volume to the signal due to the need to amplify the signal enough to reach the forward turn on voltage of the diodes, but in many applications this is desirable to an extent.

If you are want a more clipped sound you would need to increase the gain of the previous designs. But if you want clipping without considerable increase on volume then placing two back to back diodes from the output to ground could create hard clip sound without the “fuzz” characteristic sound quality or increase in volume due to excess amplification. By placing the two diodes to ground the upper and lower peaks of the signal passing

through the amplifier are flattened greatly. This creates an almost square wave like signal that allows for harmonic distortion which creates a pleasing sound to the ear.

Another factor that has not been mentioned yet is symmetrical versus asymmetrical clipping. Although the two will clip the peaks of the signal passing through the circuit the symmetrical clipping circuit will clip the peaks of the top and bottom of the input signal evenly. On the other hand, asymmetrical clipping will clip either the upper or lower peak of the signal more than the other. There is an ongoing debate in the world of guitar effects pedal builders over which form of clipping is better. In general many people think that asymmetrical clipping sounds better than symmetrical clipping, but this is not to say that symmetrical clipping is undesirable. Many think asymmetrical clipping sounds better due to the more aggressive, raspy, and more “complex” sound this sort of clipping creates. This complex sound is generally associated with the sound you would get when you are driving your power tubes in an amplifier hard. While symmetrical clipping on the other hand has a smoother and mellower sound that is associated with driving the preamplifier portion of a traditional tube amp.

A summary of the above information is illustrated in Table 6 and Table 7 below.

<b>Level of Clipping</b>	<b>Quality</b>	<b>Volume</b>
<b>Soft</b>	Slight harmonic distortion	Large boost in volume
<b>Hard</b>	Lots of harmonic distortion	No boost in volume

Table 6: Clipping Level Summary

<b>Clipping Style</b>	<b>Quality</b>	<b>Volume</b>
<b>Symmetrical</b>	Smooth/Mellow	NA
<b>Asymmetrical</b>	Complex/Aggressive	Boost volume of half cycle

Table 7: Clipping Symmetry Summary

### 3.1.3.2. Overdrive and Distortion Part Selection

Much like the input buffer circuit we will be using op amps for the main blocks for our overdrive and distortion circuits. For the overdrive and distortion we still want a high input impedance and a low output impedance. We also still care about the large unity gain frequency response, as well as a small input voltage noise and minima harmonic distortion. It may seem weird that we want low distortion in an overdrive and distortion circuit, but we want to be able to control the distortion through the exterior components of the circuit.

The op amps that we are looking at for this function are:

- TL07xx Low-Noise JFET-Input Operational Amplifiers
- OPA827 Low-Noise, High-Precision, JFET-Input Operational Amplifier
- OPA164x SoundPlus™ High-Performance, JFET-Input Audio Operational Amplifiers

Like the input buffer we want the op amp to be transparent in tone in order to allow for the other components that are supposed to color the sound to serve their function accurately. In order to achieve this we still need a high input and low output impedance. As referenced, the value of each input and output impedance is shown in Table 3: Input Impedance Comparison. On top of the input and output impedance values, we still want to have low noise and harmonic distortion in order to allow for the exterior circuit components to manipulate the signal as desired. As mentioned before, Table 4: Frequency and Noise Comparison, shows the details of each op amps frequency response as well as the input noise and harmonic distortion.

With the above parameters analyzed we need to consider the different packages and price point of each op amp. As you can see the price of the OPA827 is very high in comparison to the OPA164 for the same number of channels when referencing Table 5: Cost Comparison.

With the comparison of the op amps complete we can begin to compare the different types of diodes that we may need to use.

When look for diodes for the purpose of creating an overdrive or distortion circuit you can use many different types of diodes such as silicon diodes, germanium diodes, and even LEDs. But we will only consider silicon and germanium diodes for this application.

The diodes that we will be looking at for this application are:

- 1N60
- 1N270
- 1N914
- 1N4148
- 1N34A

The different parameters that could affect the sound are the forward voltage, junction capacitance, and material. Table 8 shows the breakdown of these parameters for each diode.



Diode Model	Forward Voltage	Junction Capacitance	Material
1N60	0.32-0.65 V	2-6 pF	Ge
1N270	1V	0.8 pF	Ge
1N914	1 V	4 pF	Si
1N4148	1 V	4 pF	Si
1N34A	1 V	1 pF	Ge

Table 8: Diode Comparison

Since the data sheets only show the maximum forward voltage it is a little difficult to determine how this will affect the sound quality with the forward voltage alone. With regards to this though we will need to run experimental tests to determine which diode sounds better and in what configuration to place them. The capacitances for all of the diodes are fairly similar which makes it difficult to determine how big of an impact these very slight differences will make. Although we are unable to determine how this will affect our sound quality we can still see how the cost breakdown looks below in Table 9.

Diode Model	Cost
1N60	\$1.00
1N270	\$0.99
1N914	\$0.10
1N4148	\$0.10
1N34A	\$0.99

Table 9: Diode Cost Comparison

### 3.1.4. Compression

Much like overdrive and distortion, compression is used to condition a musician's sound in order to get a more desirable sound. But unlike the previous two effects, overdrive and distortion, compression in the traditional sense is supposed to be transparent. This effect detects the amplitude of the incoming signal and either reduces or increases the amplitude of the output signal in order to get more of a uniform amplitude for the output signal. One side effect of compression is the creation of more sustain due to the amplification of lower amplitude signals and the reduction of higher amplitude signals. Due to this change in amplitude of the input signal the effect creates the illusion of a very long steady pitch when in actuality you input signal is not very steady or as sustained.

#### 3.1.4.1. Compression Implementations

The way a compressor works is by amplifying small signals and attenuating large signals. This is done by having a system that can sample the output signal of the circuit and either amplify or attenuate the signal different means. A flow chart illustrating this concept is illustrated in Figure 5 below.

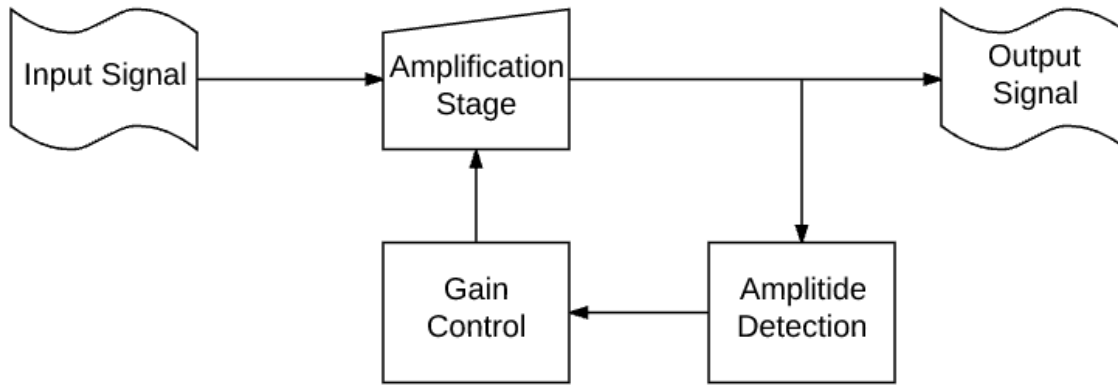


Figure 5: Compression Function Block Diagram

In order to achieve this function we can follow some common implementations for this which are all similar to two approaches. These approaches are to sense the output and turn the amplifier on and off depending on the signals amplitude, or sensing the output and either leaving the amplitude of the input the same or attenuating it before passing through an amplifier.

The first method used to create a compression effect is by turning the amplifier on and off. This is done through the use of an amplifier with an input bias current pin. The circuit amplifies the input signal and then through the use of a network of transistors creates an input bias that is fed to the input bias current of the amplifier. When the signal is small enough the transistors create the input bias current for the amplifier which in turn amplifies the small signal passing through the system. If the input signal is too large, then the transistors will not create an input bias current which will turn off the amplifier. With amplifier off, the input signal will pass through the signal without adding gain. This switching operation creates an output whose waveform looks fairly consistent in amplitude and aurally constant volume.

The second method used to create compression is to sense the output and attenuate the input signal before it gets amplified. To achieve this, the output signal is fed to the gate or base of a transistor which begins to turn on if the signal is large enough in amplitude. Once the transistor starts to turn on, the input signal starts to be attenuated depending on if the transistor is fully on or not. With the signal attenuated it will pass through the amplifier and have a steady amplitude. If the signal is small enough at the output of the amplifier then the transistors will not turn on so the input signal will not be attenuated at all. This allows for the small signal amplitude at the output to be similar to the amplitude of the large signals at the output.

### 3.1.4.2. Compression Part Selection

Like the previous effects we want a high input impedance and a low output impedance. We also still care about a large enough frequency response, low input noise and harmonic distortion, but in order to get a compression effect we need to either turn the amplifier on

and off by applying an input bias current through the implementation of a network of transistors, or by attenuating the input signal before it is amplified by a normal op amp.

The op amps that we are looking at for this function are:

- TL07xx Low-Noise JFET-Input Operational Amplifiers
- OPA827 Low-Noise, High-Precision, JFET-Input Operational Amplifier
- OPA164x SoundPlus™ High-Performance, JFET-Input Audio Operational Amplifiers

The input and output impedances of the following op amps are shown in Table 3: Input Impedance Comparison. Frequency response, input noise, and harmonic distortion are referenced in Table 4: Frequency and Noise Comparison and finally price comparison is shown in Table 5: Cost Comparison.

The op amps above would only be usable if we decide to design the circuit to attenuate the input before amplification. If we decide to turn out amplifier on and off depending on the output we will need an op amp with an input bias current that we can control.

Since many of these types of ICs are now obsolete, they are becoming hard to come by. Due to this we will look at the following amplifiers this function:

- BA6110
- LM13700 Dual Operational Transconductance Amplifiers With Linearizing Diodes and Buffer

Since these types of amplifiers are becoming harder to find we will need to acquire both amplifiers and test in order to determine the sound quality add to the signal at the output (Datasheet Reference).

With regards to price point the cost of the two amplifiers are as shown in Table 10 below.

<b>Amplifier Model</b>	<b>Cost</b>
<b>BA6110</b>	\$3.00
<b>LM13700</b>	\$1.36

**Table 10: Amplifier Cost Comparison (Courtesy of Texas Instruments)**

In order to create this bias current necessary for the amplifiers to work, we need some transistors to accomplish this. Since these transistors do not affect the quality of the signal passing through any general purpose bipolar junction transistor should work for this application, as the transistor will be functioning as a switch in order to allow or block current from entering the input bias of the operational amplifier.

### 3.1.5. Equalization

Equalization is an effect that can greatly shape an individual's sound due to the fact that it is a network of filters that the user can adjust to their liking. The equalization pedal works by boosting or cutting certain frequencies or a frequency ranges in order to further shape the tone of the output signal. This method of boosting and cutting is done by passing the musician's input signal through the desired filter; whether that is a low pass, high pass, bandpass, or band reject filters as illustrated in Figure 6 below.

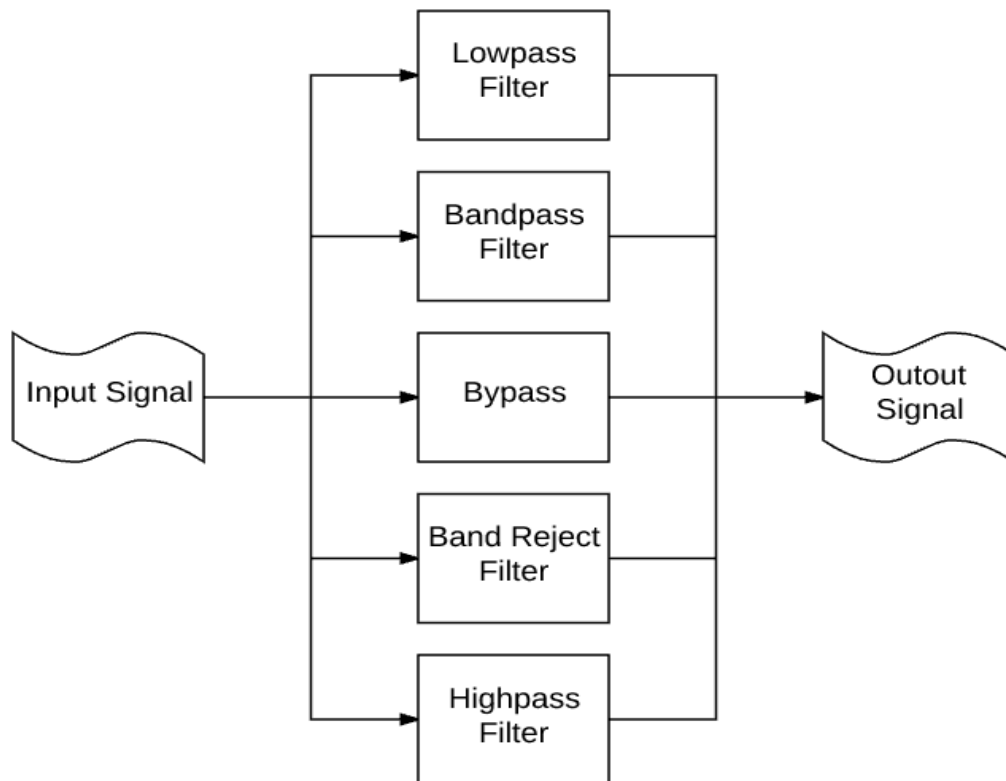


Figure 6: Equalization Network

#### 3.1.5.1. Equalization Implementations

The common implementations for this type of effect are all very different. To put it simply since equalization is just implementing different filters many different pedal builders design their own filter for the sound characteristic they deem to be best for the given effect. For many some effects the equalization option is a band reject filter created by having a high pass and low pass filter together with the ability to blend the output of the two filters in order to create a cut in the middle frequency range. Other effect pedals have only a high or a low pass filter which can be useful for some applications but can be frustrating due to their limited ability to shape the tone of the signal passing through the circuit. Another method of equalization for some effects is a band pass filter which allows the user to adjust the over lap of the low pass and high pass filter much like the band reject filter mentioned earlier. This manipulation of the two filters allows the user to create a bump or boost of the middle frequencies within a certain range.

### 3.1.5.2. Equalization Part Selection

Much like the input buffer circuit we will be using op amps for the main blocks for our overdrive and distortion circuits. For the overdrive and distortion we still want a high input impedance and a low output impedance. We also still care about the large unity gain frequency response, as well as a small input voltage noise and minima harmonic distortion. It may seem weird that we want low distortion in an overdrive and distortion circuit, but we want to be able to control the distortion through the exterior components of the circuit.

The op amps that we are looking at for this function are:

- TL07xx Low-Noise JFET-Input Operational Amplifiers
- OPA827 Low-Noise, High-Precision, JFET-Input Operational Amplifier
- OPA164x SoundPlus™ High-Performance, JFET-Input Audio Operational Amplifiers

Like the previous effects we want the op amp to be transparent in tone of the equalization filter network in order to allow for the other components that are supposed to color the sound to serve their function accurately. To do this we need a high input and low output impedance, which can be seen in Table 3: Input Impedance Comparison; as well as low noise and low harmonic distortion, as compared in Table 4: Frequency and Noise Comparison, to allow for excellent signal fidelity.

With the above parameters analyzed once again we can see that the price point of the OPA827 is much greater than the price of the OPA164x as shown in Table 5: Cost Comparison along with the package sizes.

## 3.2. Digital Effects

There are a few digital effects that our team is going to emulate for this project. The three effects include flanger, reverb, and delay. There are many effects that we could have implemented using digital signal processing (DSP), however we decided to stick to these three. The reason for this is because these effects are heavily dependent on time. Because of this key feature of these effects, it makes sense to implement them digitally as opposed to with analog. Using DSP, it will allow for easier manipulation by the user.

We also would like to create an interface that will be used to communicate with the microcontroller and allow the user to change the settings of the digital effects, as well as turn them off if desired. In order to toggle effects the user should be able to easily cycle to the effect they want to turn on or off, then simply use a foot switch in order to toggle on or off. The reason for the footswitch is because a user of this multi-effects pedal will need to be able to turn these effects on or off while on stage playing. Historically the way this is done is by having a footswitch on each separate pedal that can be pressed in order to enable or disable them. Since this product will include each effect in a single pedal enclosure, we will set up an interface to change the values of each effect, as well as a way to toggle each effect on or off.

Having the ability to change values and toggle will allow for better user control of each of the digital effects. On a normal effects pedal, the values for each parameter are set by turning a knob until the desired settings are made without knowing the exact values. With the ability to change settings, the user will be able to choose more exact values for each effect. An example of this would be where the user will be able to choose the exact time between each echo of a note with the delay effect. Having this detailed control of each effect will allow for the user to get the exact sound they are trying to create for their music using the multiple controls shown in Figure 7.

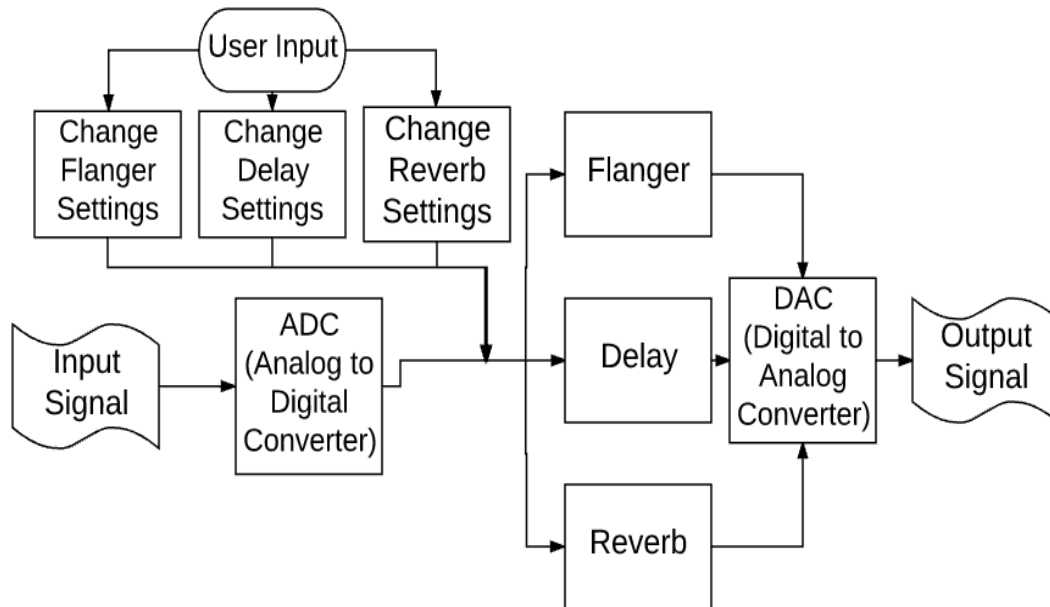


Figure 7: Digital Effects Flowchart

Digital effects have become much more popular in recent effects pedals. The output sound as a result is very similar to that of an analog effect pedal, but there are slight modifications to it before and after the DSP. In order to use DSP in signal modification, an analog to digital converter (ADC) must be used to take the analog input signal from the guitar, and turn it into something that a microcontroller or DSP chip can understand. This conversion is a vital part of digital effects pedals, and must be understood and done correctly before any code can be used effectively.

### 3.2.1. Necessary Features

These are key features that the digital effects unit must have, and some features that may be added to the product in its later life due to time constraints.

- Must haves
  - Easy to use interface.
  - Must be able to take pass original signal unaffected when both switches are off.
  - ADC/DAC must be high resolution and high precision for best sound.

- DSP chip must be at least 16 bit precision and must use floating point arithmetic.
- Each DSP chip must have an LED light indicator to show if the desired effect is toggled on or not.
- DSP chips must have knobs and switches in order to allow the user to easily adjust the unit to their desired settings.
- Expandability/Wants
  - To be able to flash the DSP chips to add more effects later on.
  - Low power consumption.

### 3.2.2. Delay/Echo

The delay effect, also known as echo, is a fairly simple effect. The best way to explain how it sounds is to compare it to an actual echo that might be produced by speaking in a large room, or cave. After the initial sound is produced there are recurring copies of the same sound that gradually get softer and softer. It also could seem like the sound is being repeated but every repetition it is further away from the point of origin. This effect is produced by the sound waves bouncing off of a wall or something similar and traveling back to the original location moments later. This produces more and more repetitions the more times it is able to bounce off of the walls. This effect can be illustrated as shown in Figure 8. This figure shows the sound waves being emitted originally by the source, then having them bounce back at a lesser intensity.

This is a natural phenomenon that occurs any time a sound wave hits an object. The same idea of the effect is used in echolocation, where a signal is sent out and the reflected wave is measured. This allows data to be gathered about the object that is reflecting the sound. This process can also allow us to understand how to modify the sound digitally to make it sound as if the repetitions of sound are occurring from an object close by or further away.

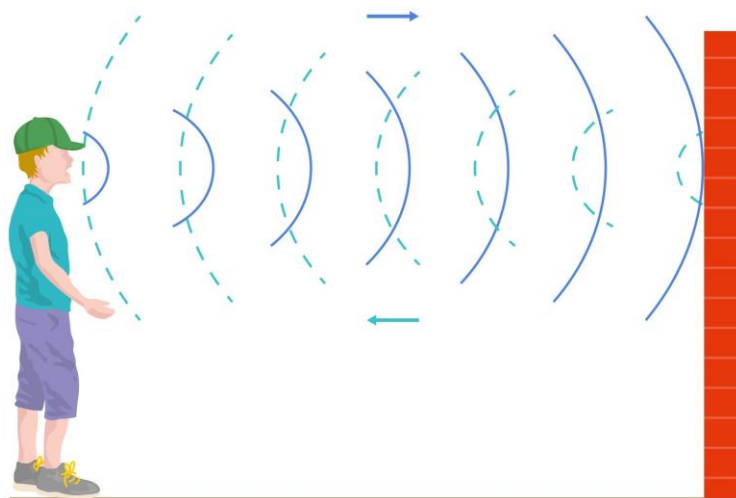


Figure 8: Delay/Echo Example [18]

Historically, the effect was first used by playing back the audio on a special machine that took the tape reel and delayed it for a certain amount of time. This process was used in the 1950's and did not allow for much modification of the sound. As time passed, the technology used to develop the effect advanced. The 60's introduced using magnetic disks to get better audio quality than the tape reel. Around the 70's the effect was implemented in an analog effect pedal. This created an affordable and efficient way for artists to use the effect in their recording. Digital delay pedals were introduced around the 80's and gave users the ability to have better user control of the effect, and to allow for increased delay times.

This effect is very simple and does not take very much computation. In order to create this effect a simple delay is needed. In order to create this delay the function  $y(n) = x(n) + d * x(n - M)$  can be used.

- $x(n)$  = Function of the original signal at the input value of the time step of the signal.
- $d$  = depth desired for output signal. This is a value between 0 and 1. The value 0 does not produce the effect, and 1 allows for the maximum effect.
- $n$  = time interval.
- $M$  = Delay time.

This is similar to the function used to create the flanger effect, described in section 3.2.3. This delay function is the basis to mostly every time based effect. With this we are able to generate the sound that we want, and by changing the values it allows us to personalize the sound even more. In order to create the effect of multiple delays one after the other decaying as time passes, this delay function can be modified. The modified function can be used for a finite number of echoes. For example, to modify the delay function for 3 echoes, it would change to  $y(n) = x(n) + d * x(n - M) + d^2 * x(n - 2M) + d^3 * x(n - 3M)$ . As the number of repetitions increases the  $d$  value for depth will decrease exponentially. This is how the signal appears to decay for later iterations. Without this the signal would simply stay at the same level for however many iterations. [14]

### 3.2.2.1. Coding Delay/Echo

In order to code the delay or echo effect, the main component includes using a delay line. A delay line is basically a buffer that takes samples of data and stores them to be accessed after a certain amount of time has passed. Figure 9 shows a block diagram of how a delay line works. Common notation for delay lines include the input function  $x(n)$ , the output function  $y(n)$ , and the delay given by  $z^{-M}$ .

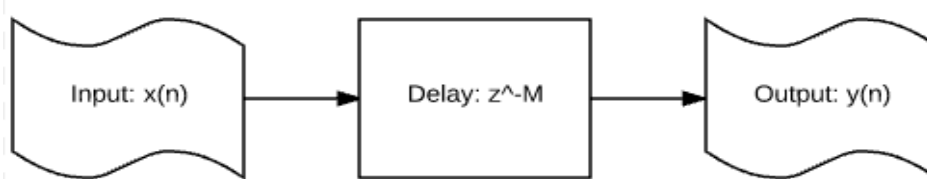


Figure 9: Delay Line



This basic idea allows for implementing time based effects fairly simply. In the case of the echo effect the delay line is used to create the echo portion of the signal, and is mixed with the original signal. The echo is repeated over a certain amount of time, and decays slightly after each repetition. These delay lines can be represented by the formula  $y(n) = x(n) + d * x(n - M)$ . In order to make the effect, there must be multiple delay lines in the same code. To code this the user must set the length that they want the echo to last. The longer the effect continues, more software delay lines are needed. This process is called a comb filter, where one line is passed to a delay of some sort, and the other line carries the original signal directly to the output. This comb filter is explained in more detail in section 3.2.4.1, but can be represented by the Figure 22 below. This figure is a more advanced version of the regular comb filter, but it has the same function, just with multiple delay lines.

The  $M$  value in the delay line is where the  $x(n - M)$  comes from. By setting the value of  $M$  for the delay it creates the time that ticks before each repetition. For the echo effect a delay of greater than 50 ms is normally used to achieve the desired output signal. Another key component to creating this effect is to make sure that the sampling rate is taken into account for the  $M$  value. [24]

When receiving samples from the ADC, the DSP chip will get the signals at a specific rate. If a delay needs to be of a certain length, it is necessary to know this sampling rate. By delaying a signal by  $M$  it is simply reading in the original value and waiting for  $M$  samples to be received before letting the delay line go through to the output. In Figure 10 the delay lines are shown, where there can be unknown amount at first. The value “ $i$ ” in the figure is determined by the amount of delays desired by the user. Each signal delay line is delayed by an incrementing multiple in order to make sure that the signals do not mix incorrectly. The length of each delay line is  $i * delayLength$ . By doing this calculation, no two repetitions of the sound will come through at the same time, always allowing the sound to create a uniform decay of repeated signals.

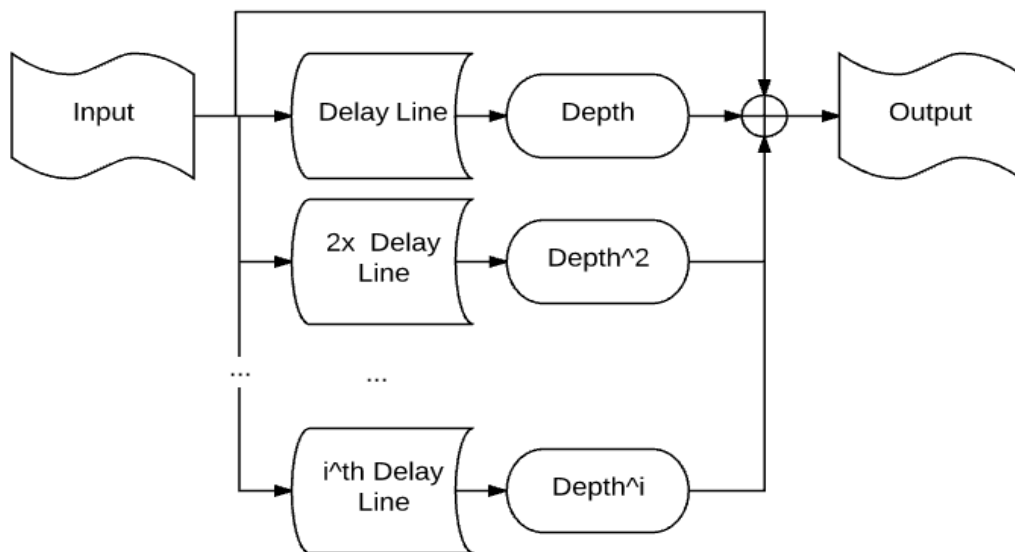


Figure 10: Iterative Delay/Echo Block Diagram

A loop may be used in order to create these delay lines dynamically so a user will be able to choose any amount of echoes desired. To perform this dynamically a loop can be run when creating the delay lines based on the value set for the length of the echo effect. This user configurability is a great feature and allows for easy manipulation of the effect to fit the desired needs. Some people might want a short echo that only lasts for a few iterations, while others want it to continue on for 15 seconds. This is all up to the user of the product.

Although the method described by figure 11 is a viable option to create the delay effect, there is a better way of doing it. The better option would be to use the process shown in Figure 11. This process allows the signal to only use a single delay line. After each delay the signal is send as the output as well as taken back to the beginning where it is able to go through the delay line again. This process does the exact same thing as above, but uses less memory when coding this algorithm.

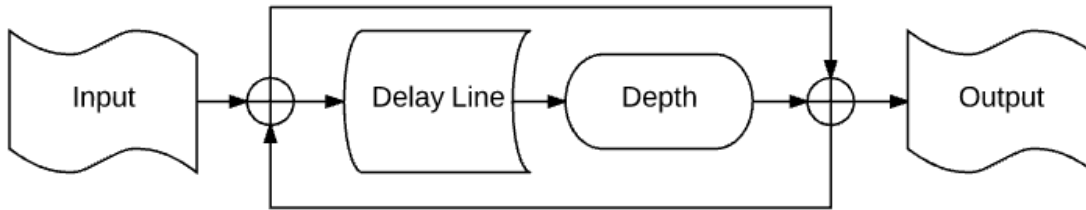


Figure 11: Recursive Delay/Echo Block Diagram

### 3.2.3. Reverb

Reverb is an effect used to create a sound that appears as if the recording is being done in a large room. This is a natural effect that occurs when listening to any audio in a room that does not absorb the sound waves. As the sound travels from the source it is able to bounce off walls in all directions before actually reaching the target. Figure 12 shows an example of how this works. Along with the direct sound that is coming from the source, the sound waves that bounce off of the walls ultimately make it to the target, but create a echo type effect. These bouncing signals are called early reflections. These reflections of the original sound occur somewhere between 5 and 20 milliseconds after the sound is produced.

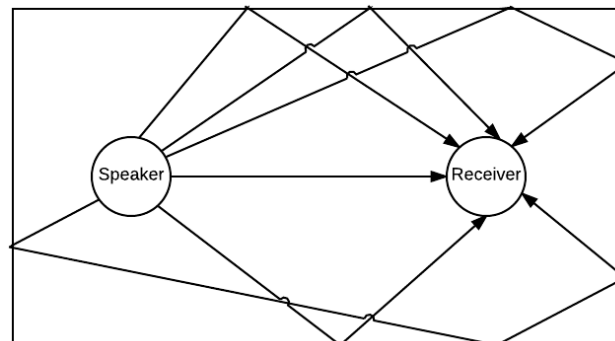


Figure 12: Natural Reverb Example

This effect has multiple parts to it which include the original signal, as well as a collection of modified signals in an attempt to create the illusion signals bouncing off of the walls. In around the 60's, before the use of effects pedals, this effect was produced by simply having the sound played on the opposite side of a large room and a recording device on the receiving end of the room. This method was called a chamber reverberator and mimicked the method in which natural reverberation is created.

Around the same time, another method called plate reverberators were also used to create the effect. The way these would work involved passing sound waves across a plate of metal and receive the sound on the other side of the plate. Depending on the type of material used, the receiving end of the plate gets the signals at different times and levels. The last method commonly used in creating the reverb effect before the use of digital signal processing was the spring method. This method was even implemented in an amplifier released in the Fender Twin Reverb Amplifier. The way this method works similar to the plate method where a signal is applied to one side of the spring, and the other side receives the sound. By using the spring tension the device is able to control the time and level of the receiving sound. [16]

Finally the reverberation method more commonly used today, especially in guitar pedals is the digital signal processing method. The way this process works is by taking a converted analog signal, then taking the samples and modifying them to be passed to the DAC. In order to modify the signals, multiple delay lines are used in order to create the early reflections. Several delay lines are used at the same time but at slightly different sample times to mimic multiple signals coming from different directions from the walls. This series of delays fade out over time, allowing the sound to appear as if they are decaying. The effect is shown simply in Figure 13 below. This shows the initial sound, then the first reflection that is heard, which is then followed by the early reflections which fade off over time. The time for it to fade is the Reverberation time. This is the time it takes for the sound to drop 60 dB below the original level. [20]

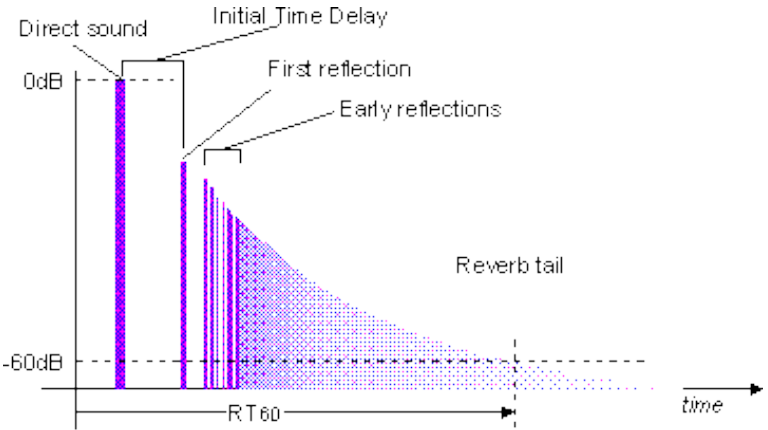


Figure 13: Reverb Explanation [1]

### 3.2.3.1. Coding Reverb

Since there are two ways in which reverb can be coded, it is necessary to make the classification of the method being used. The two methods that could be used to create this digital effect include delay lines or convolution. In section 3.2.2.1 delay lines are explained, with Figure 9 giving a basic flowchart of how the idea works. The other way of producing the effect is by using the convolution method. This method involves using the impulse response of a room. If this is known then the convolution method can create a very accurate portrayal of the real world reverberation effect. The impulse response is used to convolve with the original input signal. While this creates a very realistic effect, it is mainly used to modify recorded signals instead of a real time modification of it. For the purposes of a guitar pedal, it is clear that real time processing of the signal is necessary for the application. For example an artist would mainly use a pedal such as this when playing live in order to modify the output of the guitar during a song. For this reason, the product will include delay lines as the method for creating reverberation. [3]

There are also multiple real time processing methods to creating the effect. The first one is Schroeder reverb. This is created by implementing delay lines and filters. The process is similar to that of the echo effect, but instead of using multiple delay lines one after the other, they are used in unison to create the multiple echo signals that can be observed with the effect. The delay lines are used in a comb filter as illustrated in figure 14. After the signal goes through the comb filter, it is sent through an allpass filter. The filter used for reverb is set up slightly different. Figure 14 is able to show the setup of the Schroeder reverb structure. The way it works is by sending in the original signal to multiple comb filters then taking that signal through a series of allpass filters. The number of comb and allpass filters is determined by the programmer and cannot be changed by the user. Each delay line in the comb filters will have a different delay time in order to create a more complex structure of echo sounds. [19]

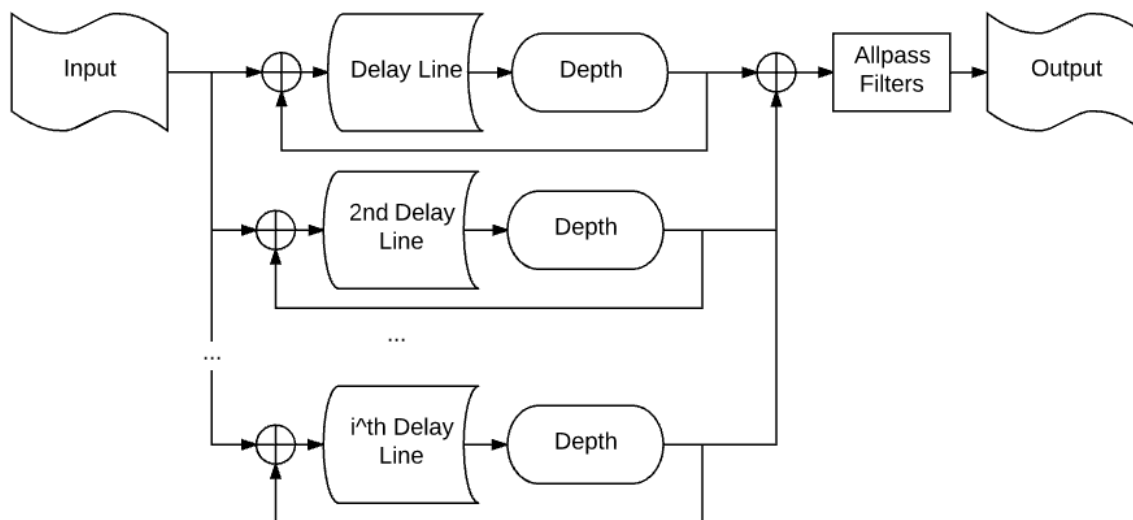


Figure 14: Schroeder Reverb Block Diagram

The other method of creating reverb is called the Moorer reverberation. This includes a similar structure to that of the Schroeder reverb, but also adds tapped delay lines in order

to create the early reflections. This signal is then passed to the output signal, as well as through a Schroeder reverb setup. Having this extra signal coming from the tapped delay lines adds the early reflections and makes the effect a little more realistic compared to the Schroeder reverb. Figure 15 illustrates the Moorer reverb as a modification to Schroeder's reverb. The tapped delay line is a normal delay line that allows an output periodically through the delay at each "tap."

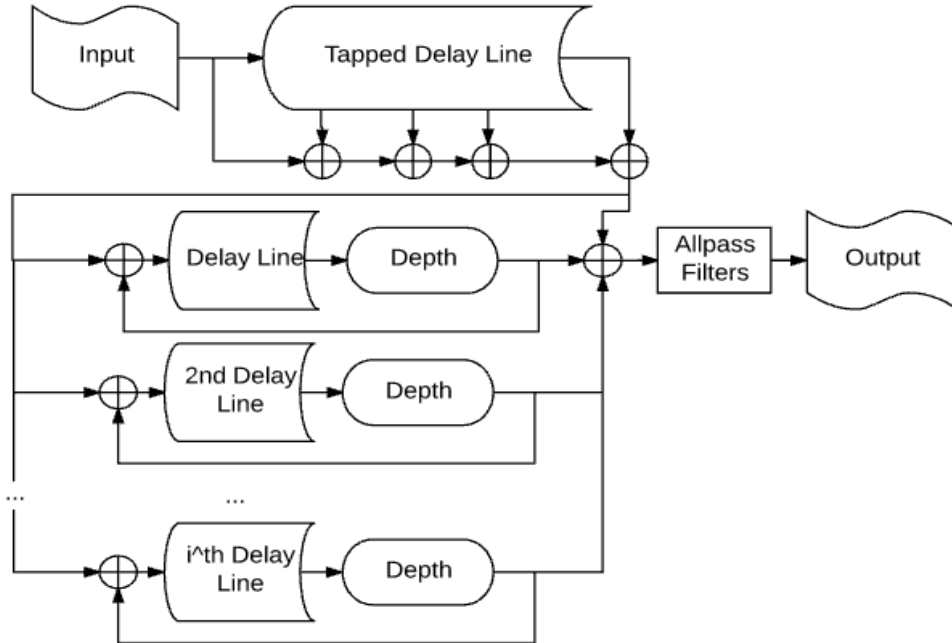


Figure 15: Moorer Reverb Block Diagram

### 3.2.4. Flanger

Flanger is an effect used in signal processing in order to create a unique output sound. The input sound is modified to create a sweeping effect. This effect has been used in signal processing since the mid 1900's and was originally discovered and used by Les Paul. Since computers were far from advanced at this time in history, the effect had to be created in a much different way than it is created today. Originally the effect was achieved by using two tape machines that would play the same audio. One of these tape machines would be physically touched to slightly delay its signal. These two outputs were then mixed together equally. Because of the small delay in one of the machines, the result of the output ends up with a sweeping sound. This process is depicted in Figure 16, where tape 1 and tape 2 initially have the same audio. In order to stop the flanger effect, the other tape would also be slowed down by the same amount, in order to match the signals once again.

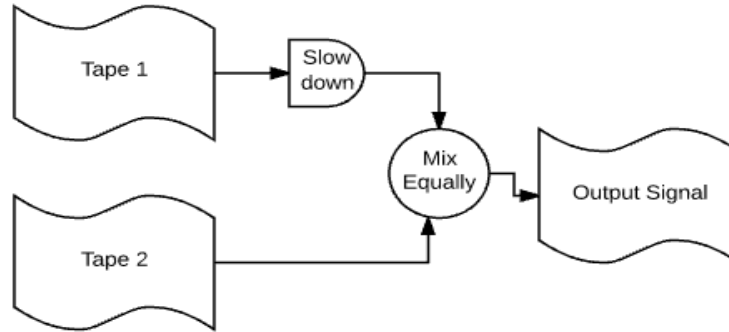


Figure 16: Flanger Flowchart

As technology advanced, so did the process of creating the flanger effect. This effect was implemented with analog before digital was used. In the analog circuit the signal is taken in like normal, then mixed with a delayed version of the signal. This method is still very prominent today in the market for effects pedals. However, for the purpose of this project, the digital implementation was chosen for this specific effect due to the fact that it is a time based effect.

DSP became more popular for newer applications of the flanger effect. In order to achieve this through DSP, a feedforward comb filter is used. This filter is able to perform the same operation as the original way of creating the flanger effect. In the comb filter the input signal is taken down two paths. One path takes the original signal directly through to the output, while the other path goes through a delay and a depth operation. The equation can be represented as  $y(n) = x(n) + d * x(n - M(x))$ .

- $M(n)$ = Length of delay-line at time interval  $n$ .
  - $M(n) = M_0 * (1 + A \sin(2\pi f n T))$
  - $A$  = Amplitude
  - $f$  = Frequency
  - $n$  = Sample time step
  - $T$  = Time step size

This function allows for the original signal to be added to the modified signal. The modified signal uses  $M(n)$ , which is a function that is used to shift the signal at different intervals over time. If the signal were shifted by a constant amount, then the effect would be called a phase shifter effect. This effect is not very different from the flanger effect, but both create a unique sound that an end user of the product would be able to notice. The flanger effect will have more of a sweeping and modulating sound. The reason for this is shown in the  $M(n)$  function above. If this function was a constant value, it would create the phase shifter effect. However, since it will be modulated based on multiple factors, it will be able to create the flanger effect that is desired.

There are some drawbacks to creating this digital effect. This is because the effect is created with a time variant function, meaning the modulation to the signal varies based on time. This is why the function uses  $M(n)$  which will change based on time, shown in the equation above. This type of function can make the calculations for the effect more difficult to understand and simulate properly.

### 3.2.4.1. Coding Flanger

Implementing the flanger effect will be the most complicated out of the three effects we are trying to implement. The reason for this is because of the slightly delayed signal that needs to be added to the original signal. This is not just a simple delay of the signal that is added to the original. Instead this delay is modulated with time, in order to create an inconsistent pattern of delays. These delays usually range between 1 and 3 milliseconds. This creates notches that are present in the frequency response of the signal. Using the frequency response graph is also a way to show the difference that is created when the delay is modulated as opposed to being a constant value. In Figure 17, the delayed signal that is added back to the original signal is always a fixed interval, creating an evenly spaced graph. The modulated signal added back to the original causes a more chaotic pattern of notches that are clearly apparent in Figure 18.

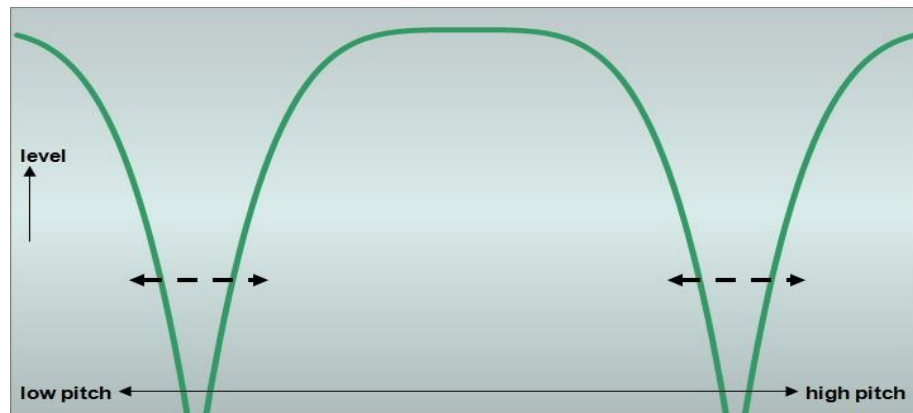


Figure 17: Phase Frequency Response [6]

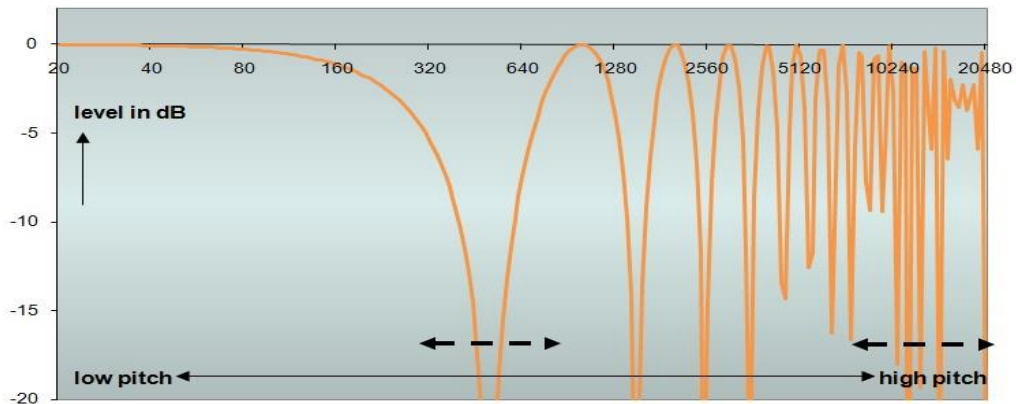


Figure 18: Flanger Frequency Response [6]

The frequency response of a simple guitar output is shown below in Figure 19. This frequency response can be observed to be fairly smooth. Due to the nature of an output wave of a guitar being very different from a single sine wave, the frequency response is not going to be a perfect curve.

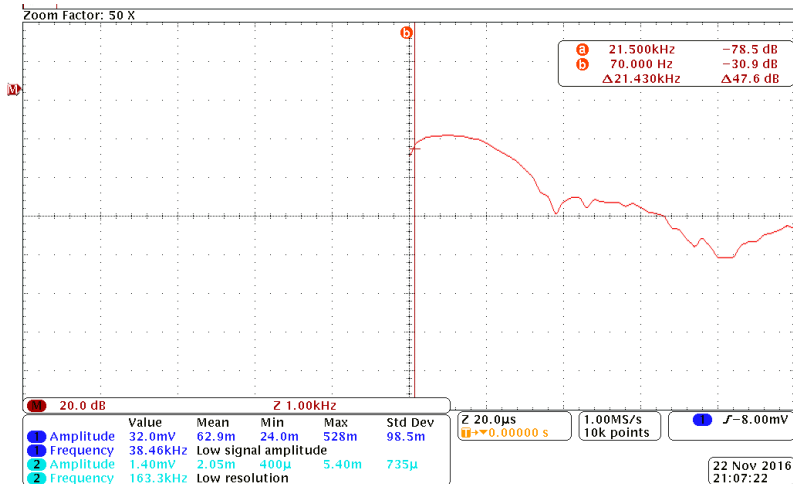


Figure 19: Frequency Response of Dry Signal

Figures 20 and 21 were created by probing a guitar using both effects. These effects were created by using the TC Electronic brand pedals for phaser and flanger. Although store purchased pedals were used to get the frequency responses, the purpose was to show the difference between a simple phase shifter and the flanger pedal. Once the G12 PedalVision is working correctly, these images will be re-created using it instead of the store bought pedals. The figures below can be seen fairly simply, in figure 20 showing the phaser effect pedal frequency response, there are dips in the graph periodically. Figure 21 has many more dips in the graph and are more unevenly spread out. This is because of the modulated time intervals for the delays.

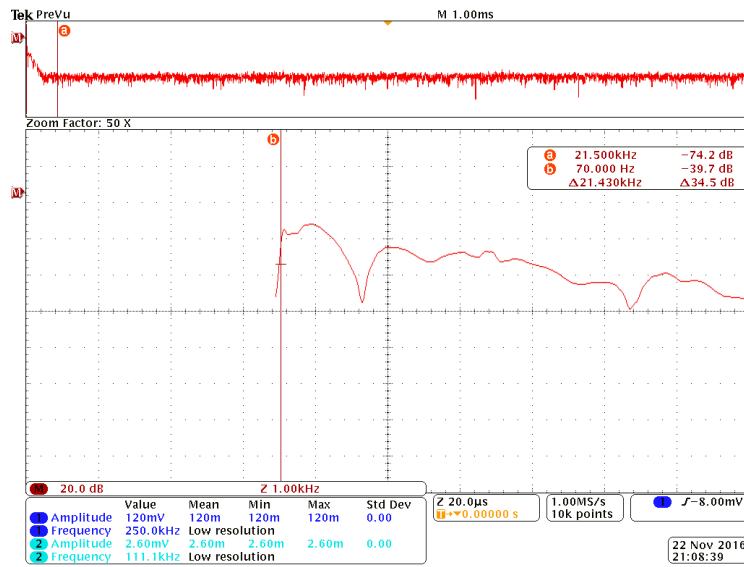


Figure 20: Probed Phase Frequency Response



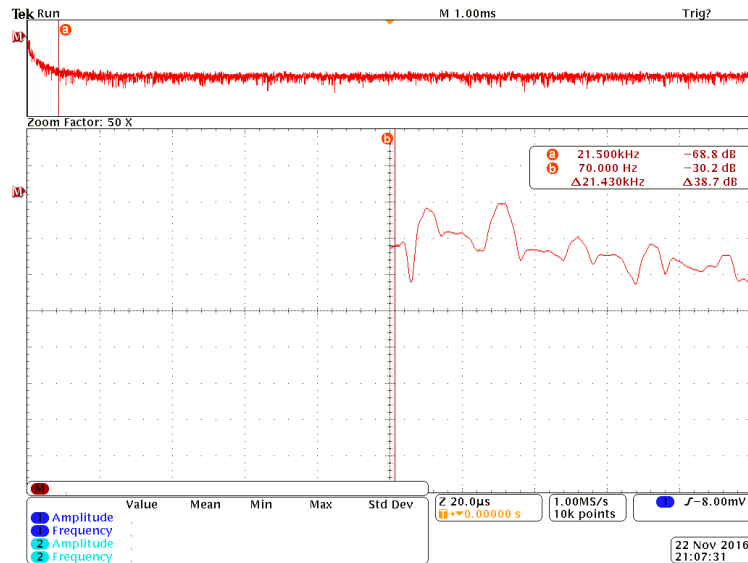


Figure 21: Probed Flanger Frequency Response

In order to create this effect the code must allow for the input values from the digital to analog converter to be added to values from a previous time in the signal. To achieve this delay of signal, a buffer can be used to hold values of the original signal that will be passed after a certain amount of time. This time interval that the values are held for is determined by the  $M(n)$  function. The way that this can be done is by using delay lines.

The delay buffer will be initialized in order to make sure that there will be no output until a value is passed in from the input function. As a simplified example to explain the delay line, we will look at a discrete time impulse signal. For each time interval  $n$  the value that is sampled at  $x(n)$  moves into the buffer of length  $M$ . As  $n$  increases, the buffer shifts in order to allow the new value from the input function. The value at the end of the buffer, which is at the  $M^{th}$  spot, is taken as the output value  $y(n)$ . By making the values pass through the buffer in order to become the output value, it creates a delay of  $M$  intervals.

In order to implement this for use with the flanger effect, it will take a more advanced understanding of the use of a delay line. In the example from the previous paragraph, the  $M$  value was fixed. Because of this, the delay that is achieved will always be the same. For the flanger effect, the delay is not a constant value. It is actually a function of time that fluctuates to allow the sound to be more varying and create a unique output signal. This type of delay line is called an interpolated delay line, and allows for smooth variations over time.

When creating delay lines, if the sample spot was just randomly moved around, it would cause the output to have a “zipper like” sound. This is because the transition between varying sample taps would be more jagged. To avoid this the interpolated delay line is used to create a smooth variation between sample speeds. The process of doing this does make it more complicated, but it offers much better quality in the product overall.

The delay line will be used in a feedforward comb filter. The way this filter works is exactly how the flanger effect is created. The input signal is taken down two paths. One

path goes directly to the output, while the other path takes the signal through the delay line. After this delay line the signal is modified by a depth value, which determines how much of an effect the signal modification will have on the output signal. This signal is added to the original input signal in order to create the output. This comb filter is illustrated in Figure 22.

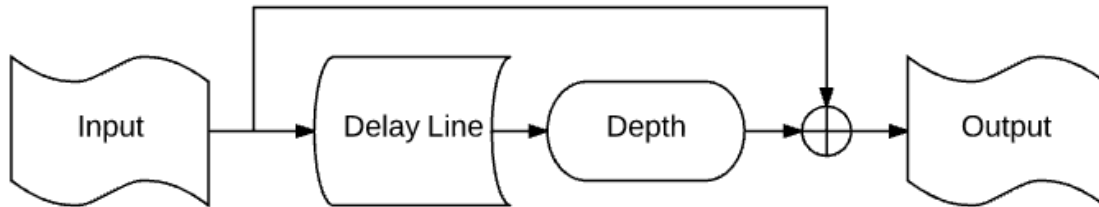


Figure 22: Feedforward Comb Filter

For this effect to be used in a multi-effects pedal, the user must be able to modify some values that allow for personalization of the sound. The values that should be able to be changed in the flanger effect are the depth and the speed of the delay. With these controls of the effect, the user will be able to mix the sound how they see fit.

### 3.2.5. Software Interface

There are many options for creating a user interface for the digital effects. In order to create a complete system, there must be an interface for controlling the settings of the effects, as well as turning them on or off.

#### 3.2.5.1. Effects Interface

In order to modify the values for each effect that is being implemented by the DSP chips, an interface is needed. There are many ways that this interface can be set up in order to effectively change settings. Some of the main concerns that must be considered when deciding the interface is usability. As an end user, there should be very little to no learning curve for using the system. It should include terms that are known in the industry for each of the effects. For example, a user might know what reverb is, but if we label it as “Schroeder Reverb” then the user might become confused if they do not know that this is just a way of implementing the digital effect. To avoid confusion, very basic terms will be used, as well as controls that are also user friendly. Most guitar pedals today contain dials without actually having values. For the digital effects on our product we feel that adding a number value to the controls will allow for better usability of the system. If a user is able to see that the best sound for the effect comes from using a specific value, then they can always change it back to that to get the exact sound every time. This avoids the situation where a dial gets turned slightly making the value different, and very difficult to get back to the same spot.

The first proposed method in order to change the values of the DSP chips would be to simply load a file from a storage device that includes configuration settings for each chip.

This method would be the cheapest and easiest way of interfacing the settings for each digital effect, however it is the least advanced and user friendly approach. This method would not add to any large components to the system and would be fairly easy to implement. However, the drawback to this method is that it is not user friendly and does not allow for quick modification of the chips. This is a big concern for users since it is possible that the pedal will be used while playing a song live where the settings of the effects might change for different parts of the song.

Another proposed interface would be to use an LCD screen on the actual pedal. This method would be very simple because it would allow any user to be able to change the settings without needing any external hardware. In order to change the settings, the interface could be controlled using a simple button and knobs. The button can be used to select the desired digital effect to be loaded onto each DSP chip, then the knobs can be used to begin changing the value that is for each setting. At most four knobs will be needed to changed values for the digital effects. The display will show which effect is currently being used, as well as the values of each setting for that effect. Once the user has changed the settings to the desired value, they may begin to play. This method would be very simple to implement and use, however it also has some drawbacks. The first drawback is that the interface needs hardware to be added to the pedal. One of the purposes of the G12 Pedalvision is that it is a low cost alternative to buying multiple effects pedals. By adding more hardware to the product, it also increases the price, especially if quality components are desired.

Because of the drawbacks of the LCD display interface, another proposed method could be to use a computer to change the values of each setting. To implement this method, a user interface would be created to run as a program on all platforms including Window, MacOSx, and Linux. The interface would create a visually pleasing user interface for changing values. A slider or dial will be able to be turned or moved while showing the value in a text box next to it. The value can be directly changed in the textbox as well to give the user the ability to set exact values to always get the same sound from their effect. In order to get the values to change on the actual DSP chip, the program must connect to the board in some way. This method would be most logically implemented using a usb cable that connects to the pedal. When connected, each DSP chip will be able to be configured based on what the user wants. When disconnected after the settings are loaded, each DSP chip will hold the values until they are changed again. This method does also have some drawbacks that come along with it. The main concern is that not every customer will have access to a computer at all times, which will make the product limited on where it can go.

A method that has a similar concern as the program to run on a computer is to use a smartphone to change the value. The concern that goes along with this is that not every has a smart phone. Although it is very likely that users will have a phone that could install an application to change the settings, it is still possible that some users will not be able to do this. Another concern would include the extra hardware needed to implement a wireless communication between the phone and the DSP chips. This can be done by adding a Bluetooth receiver in the board, but might increase the cost slightly. The

application that would be developed includes all of the same functionality as the computer program, except it is in the handheld environment. Each setting can be modified for either DSP chip, by using a slider or typing in the exact value. This is all done through wireless serial communication.

Below in Figure 23 is a general layout for how the program for the computer or the smartphone application will be created. There will be a section for each DSP chip that can be modified, both which include the settings for each effect to be changed. Since only one effect can be chosen at a time, a dropdown menu is used to change which effect is currently being used. For each dropdown choice, the settings on the right side will be changed based on what effect is being used.

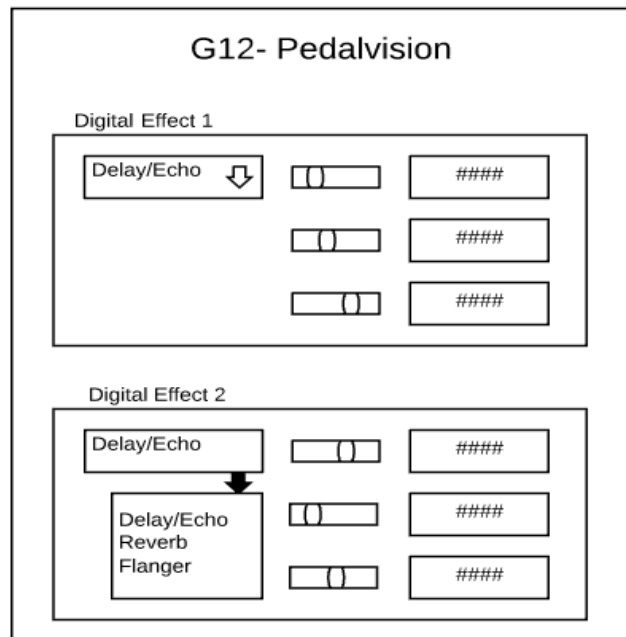


Figure 23: Proposed Interface Design

### 3.2.5.1.1. Effects Interface Selection

Due to the simplicity of the LCD screen on the pedal, we believe that this is the best option for the G12 PedalVision. This effect interface will allow for all users to be able to use the system without having to learn something different than they are used to. Since all effect pedals use knobs to change the settings, it will keep the product from having a learning curve. Using a simple button to cycle through effects will allow for the user to quickly change which digital effect they are currently using for each chip. The LCD screen will be helpful as well, since it will be able to display the current effect that is loaded while also giving exact values for each of the settings. With exact values for the settings, the user will be able to easily recreate the exact sound they had previously set. It has not been decided if there will be an LCD screen to interface each DSP chip, or if one interface will be used. If only one interface is used, the user will choose which chip they are programming.

This interface selection is also better for a user who would be purchasing the product. The reason for this is because there is not a need to connect anything to the pedal. Everything that is needed to change values and select effects will be included all in the same box.

### **3.2.5.2. Toggle Interface**

One idea for toggling on or off desired effects would be to use the same interface that is used for changing the values of each effect. In this interface the user would be able to choose the ones that are to be toggled on or off, or simply turn down all of the effect values to allow the original signal to go through as an output. This would be the simplest way of turning each effect on or off, but it would not be a very practical application for a guitar pedal. Having to go into the interface to turn down all values will take a considerable amount of time and would also cause the user to have to stop playing in order to turn them off. An end user of the product who is using the multi-effects pedal for live play of a song might need to turn an effect off very quickly in order to transition into the next part of a song. Having to stop playing in order to toggle certain digital effects would interrupt the flow of the song and make for an undesirable user experience. In order to get around this, it is common to use footswitches in order to allow for the user to tap the effect on or off while also playing at the same time.

The footswitch is an integral part in guitar pedals. It gives the user instant control over what effects the signal is being modified by. See section 5.1.7 for more details about the footswitch controls. There are multiple different ways to implement a footswitch, however we believe a hard switch is the best option because it keeps the original sound intact.

The footswitch interface for toggling the digital effects can be implemented in multiple ways. One of which is to set up a cycle switch, then an on/off switch. This implementation is a good way to use the footswitches for easy manipulation of the toggle signals. The cycle switch when pressed by the user will choose between which effect to turn on or off. The other switch will toggle the value of the current effect chosen by the cycle switch. By allowing the user to toggle values in this way it will give them control over which effect they have on at each time.

However, there are a few drawbacks to this approach. One of these drawbacks is that it would be hard to transition between effects in the middle of a song. In order to change from the flanger effect to a delay effect, the user would have to cycle to flanger, toggle it off, then cycle to echo to toggle it on. This is much more useful than having to go into the interface to turn the effects on or off, but it still is a hassle for a user of the pedal during a song. Another drawback would be that only one effect would be able to be on at a time since there would only be one chip running the digital effects. Because of this the switches to turn them on or off would only be able to interface a single chip. If another chip were added, another set of cycle and toggle switches would have to be added as well. This will possibly go against cost and size constraints that are set for the pedal. Also each one of these switches would need a small LCD display in order to allow the user to

see which effect they are cycling to. Possibilities of which display to use are shown in section 3.2.6.3.

The last idea for toggling effects on or off is to simply put a single footswitch for each DSP chip that is being used. This way the user would set which effect is to be used on the chip, then the switch will simply either let the signal go through the DSP chip or bypass it before the signal goes into the ADC. This switch will be easy to implement in hardware and does not call for any special software modifications. The drawback to this approach is that it requires more hardware to implement depending on how many DSP chips are used. For each chip, a separate ADC and DAC must be used in order to ensure if no digital effect is being used, the signal never has to be converted from analog to digital. This keeps the sound as accurate as possible.

### **3.2.5.2.1. Toggle Interface Selection**

The interface for toggling effects on or off was chosen to be done by using a single footswitch to allow the signal to go to the DSP chip or simply bypass it and go straight to the output. Doing it in this way will ensure the best sound possible when not using the digital effects. Because the switch will simply bypass the DSP chip altogether, the analog sound will not be converted, meaning there will be no slight signal modifications when going through the ADC and the DAC. This type of interface will also give the opportunity to add more functionality to the pedal. Because each DSP chip is only able to perform one digital effect modification of the signal at a time, it would make it more configurable if two chips are used. There is not much of a need for more than two chips since most users will not try to apply more than two digital effects at the same time. The more modifications of the signal can cause the output signal to be very “ugly” and unappealing to an audience.

In order to avoid undesirable output signals, most end users would only use at most two of these digital effects at a time. To meet the needs of possible customers, we plan on implementing two DSP chips and ADC chips. Each chip can be configured to perform one of the desired digital effects at a time. The toggle switches that are used route the signal past the desired chip. To ensure the best sound when using a single chip, both DSP chips have an identical setup. This way, if neither chip is on, the signal never gets converted to digital, and goes straight through to the output. If only one of the two effects wants to be used, the conversion is done for it separately. This avoids having to always take the digital signal through the first DSP chip. With this functionality, the user will be able to configure each chip before playing, then choose if both or just one of the chips are on. Figure 24 shows the block diagram of the how each module will be connected with the switches.

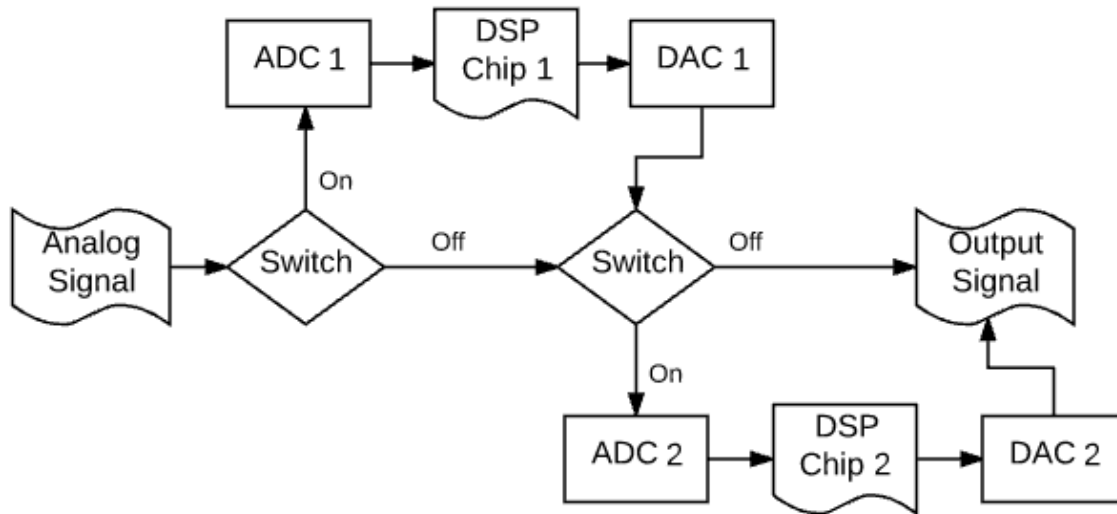


Figure 24: DSP Module Block Diagram

### 3.2.6. Digital Effects Hardware

The hardware used for creating the digital effects will include a DSP chip, which will be the brain of the system, and the interface in which the user will control the digital effects.

#### 3.2.6.1. DSP Chip

For digital signal processing applications, there are multiple options for which the processes can be completed. The two ways this can be done are by using a microcontroller or a dedicated DSP chip. Using a microcontroller would work, however it is not meant for high rate real time operations. Because of this, a dedicated DSP chip will be necessary to complete the digital effects. While a microcontroller may be used to interface with the DSP chip, all calculations will be done on the DSP chip.

In order to use this, the interface must be able to load the program onto the chip. With this ability, multiple different DSP chips can be programmed with different digital effects. By doing this the user will be able to set up their effects in the order they want, with their desired settings.

There are multiple options for choosing a DSP chip. In our application we are going to be using two separate digital signal processing setups. This will allow the user to have more flexibility on which effect to use, without sacrificing any quality in sound. As described in section 3.2.4.2.1, the setup for the G12 PedalVision must include a way to switch on or off each of the chips. Because of this the all in one chips that include multiple DSP and ADC modules would not be desired at this stage in the product. For the initial build of the product it would be best to use a combined ADC/DAC module, with a separate dedicated DSP chip. This way the quality is not traded for a slightly more compact size. One of the constraints that has been set on this project is to create a high-quality product, but also keep it as compact as possible. The way that is proposed to design this product will take up more space than if we used an all in one chip, but it will not make the product too

large for our constraints. For this reason, the separate DSP chip will keep the quality high, without making the product too large.

### 3.2.6.1.1. DSP Chip Selection

When choosing a DSP chip, it is necessary to have a setup in mind. The reason for this is because there are many kinds of DSP chips. Some of these chips are packaged together with a built-in ADC and DAC, or a small microcontroller. For our application, we will simply need a standalone DSP chip. This is because we will have a separate microcontroller already which is being used for controlling the LED matrix portion, see section 3.3.3.1.

There are multiple options when it comes to a standalone DSP chip. The two main options that need to be decided are the arithmetic format and data width. The selection is made based off of what the use of the product will be for. Two different types of arithmetic formats are available. These include fixed point and floating point. In fixed point arithmetic numbers are represented as integers or fractions between -1 to 1. In floating point arithmetic, the values are represented by what is called a mantissa and exponent, and is described by the function  $mantissa \times 2^{exponent}$ . Fixed point arithmetic does not allow for as large of a range as what floating point allows. Fixed point arithmetic can also be more difficult to program, but do come at a lower complexity of internal circuitry, and lower power consumption. Since the power consumption is not that large of a difference, for the application in a guitar pedal, the floating point arithmetic will be a better option [4].

Floating point arithmetic has a much larger, and allows for the same precision for numbers on both sides of the spectrum. If values in the fixed point arithmetic get too small, precision is lost, while values that are too large end up clipping. For this reason, we have decided to use the floating point arithmetic DSP chips for our system. [15]

The next option we have to choose for the DSP chip is the bit precision, which is also known as the data width. Most of the fixed point chips on the market use a 16 bit word size. This goes up for the floating point arithmetic, with most chips typically using a minimum of 32 bit word length. This allows for  $2^{32}$  different bit patterns vs the  $2^{16}$  patterns in the fixed point. Noise is also a deciding factor when choosing DSP chips. The noise associated with storing a number in these chips is because of the rounding of values. Since the gap between values in the floating point 32 bit precision is much smaller than the gap between values in the fixed point precision, it allow for much less noise to occur from storing the values. [25]

Because of the reasons discussed above, it has been decided that the G12 PedalVision will use a floating point DSP chip. There are still multiple options to choose from. The Table 11 below shows possible chips for the DSP application. It includes some fixed point arithmetic options as well for comparison.



DSP Chips	Manufacturer	Arithmetic	Precision	Cost
<b>C55x Series</b>	Texas Instruments	Fixed-Point	16 Bit	\$3-\$10
<b>C64x Series</b>	Texas Instruments	Fixed-Point	32 Bit	\$25
<b>C674x Series- TMS320C6745DPTP4</b>	Texas Instruments	Fixed/Floating-Point	32 Bit	\$25
<b>C67x Series - TMS320C6720</b>	Texas Instruments	Floating-Point	32 Bit	\$16
<b>ADAU144x</b>	SigmaDSP	n/a	28 Bit	\$10-\$15
<b>ADAU1701</b>	SigmaDSP	n/a	28 Bit	\$5-\$10

Table 11: DSP Chips (Courtesy of Texas Instruments)

The chip that was chosen for our application was the TMS320C6720. This DSP chip is commonly used in audio signal processing applications. It allows for a 32 bit single precision (64 bit double precision) floating point operations. We decided to go with the PowerPAD pin design over the Ball grid array, due to the extra flexibility available with the pins. We will be able to observe any failures much easier, since all pins are visible at once, whereas the ball grid is facing against the board, not allowing access to the joints.

### 3.2.6.2. ADC/DAC

An analog signal is actually just continuously changing energy levels that can be observed when a transducer converts the energy into the current and voltage signals we are accustomed to. These signals form a waveform that is able to be measured and sampled at certain time intervals. This process is what the ADC does in order to convert the analog signal to a digital one. As the analog signal is sampled, the values at each interval are measured and passed to the digital signal processing chip.

The sampling rate of an ADC is very important to ensure quality in the samples for the digital signal. If too little samples are taken in a certain amount of time, the digital signal will not accurately represent the real world analog signal. To avoid this completely, the sampling rate of the ADC must be at least twice as much as the highest frequency coming in through the analog signal. Since noise is a definite concern when dealing with analog signals from a guitar, it is necessary to filter out any outlier frequencies that may occur. In order to do this a frequency range must be set based on what the analog signal is capable of. Once the range is set, an anti-aliasing filter must be used in order to remove any samples that go above the range.

Once the filter is built, the signal is converted to a digital signal. There are two ways that this conversion can be made. One is by what is called batch processing. This takes a block of samples at a time and then converts it. The other way is to sample continuously. This process involves taking a sample at each time interval given by the sample rate, causing the digital signal to be produced continuously. The sampling type is given based on the ADC chip that is chosen. See section 3.2.5.2.1 for the chosen hardware for the

ADC chip. Below in Figure 25 is an example of how the samples are taken from the analog signal then converted to the digital signal at each sample interval. The initial signal comes in, then the discrete values are calculated. These discrete values are then given bit values to be put passed in as a digital signal.

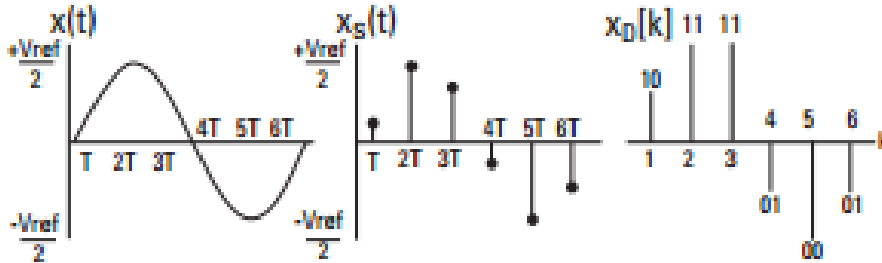


Figure 25: ADC Example Waveform (Courtesy of Texas Instruments) [27]

Each value is then passed to the DSP chip where the modification of the signal takes place. Each of the digital effects change the signal in different ways. Each use different processes in order to create the unique sounds. However, the three effects that have been chosen implement a time based modification, so delay lines are a part of each of them. These are explained in more detail for each application based on the effect in section 3.2.

Once the values are modified by the DSP chip, the digital signal must be converted back to an analog signal in order to be output. To do this, a digital to analog converter (DAC) is used. This is a chip that takes the modified digital signal from the DSP chip and converts it back to an analog signal. This process is not exact. The output from a DAC has an apparent waveform that is composed of steps at each sample time rather than a smooth analog signal. This stair appearance is able to be reduced partially by using a process called zero stuffing.

This signal comes into the DAC in samples as shown in figure 26. Each sample is padded with a zero afterward as shown in figure 27, which is then replaced with a value averaged from the two samples it is between as shown in figure 28. The reason the zero is to be added in order to oversample the signal which helps to make it a smooth output waveform. The samples come in at the sampling rate, while the zero stuffing allows the samples to be recorded at twice the original rate.

7

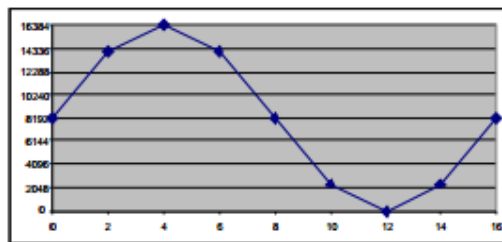


Figure 26: Initial Signal (Courtesy of Texas Instruments) [26]

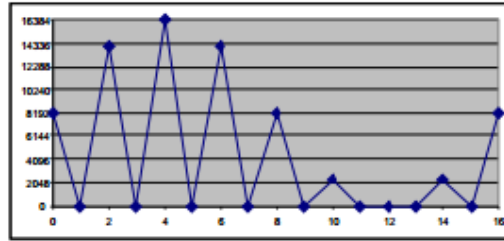


Figure 27: Zero Stuffed (Courtesy of Texas Instruments) [26]

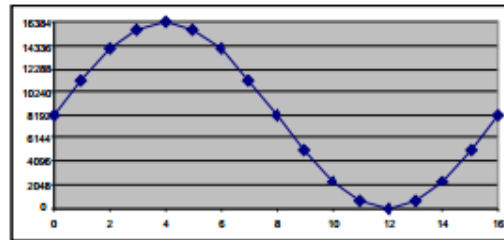


Figure 28: Zero Padded Double Sampled (Courtesy of Texas Instruments) [26]

### 3.2.6.2.1. ADC/DAC Selection

In order to get an effective signal from a digital conversion, filters must be used before and after the conversion. These are commonly known as the anti-aliasing filter for before the signal goes into the ADC, and an anti-imaging filter for when it comes out of the DAC. The anti-imaging filter is also known as a reconstruction filter. These filters are used to make sure that the input and output of the chip are as accurate as possible. The possibilities of how to implement the two filters include mainly using multiple low pass filters in series to increase the order of the filter to increase the roll off rate. Depending on the chip that is chosen for the project, these filters may already be implemented on the chip. If we choose one that does not have them, then the circuit must be created so the signal passes through it before going into the chip.

There are different types of ADC chips that can be chosen from. Each type uses a different method in which the conversion is made. These different types are Flash, Successive approximation, and Sigma-Delta. When choosing an ADC these types need to be considered based on the use of the chip in our system, as well as availability.

The flash ADC is the fastest of the three types, however it uses more power than its counterparts. This is because it uses  $2^{bit\ length}$  comparators in order to convert the signal. With this many comparators it is clear that the power consumption will be high. The next type is the successive approximation, which uses one comparator and logic to convert the signal. It starts by comparing the input to half of the reference voltage, setting the bit if it is greater. This continues, cutting the comparison to the reference voltage by another half. This method is simple in the hardware that it uses, but it takes many clock cycles to complete the conversion. The last type is the sigma-delta ADC, which uses filtering and oversampling to create a very high resolution conversion. Because of the oversampling the conversion may take extra clock cycles to complete, but the tradeoff for a higher

resolution compared to the successive approximation is worth it. The sigma-delta type is also very precise for use as a DAC. For that reason, if the ADC chosen is a sigma-delta, we will also use a sigma-delta DAC. [13]

Another consideration when choosing an ADC/DAC chip is to have separate chips for the ADC and DAC, or to have a combined chip. The combined chips can be used to save on the size of the product, which is a concern for the system we are trying to build. Having them combined does not mean there must be a sacrifice in resolution. Listed in table 12 below are different ADC/DAC chips, some of which are combined chips, also of varying bit resolution.

<b>Product</b>	<b>ADC/DAC</b>	<b>Filters Included</b>	<b>Bit Resolution</b>	<b>Cost</b>
<b>PCM3002</b>	Both	Yes	24 Bit	\$11
<b>PCM3060</b>	Both	Yes	24 Bit	\$6.15
<b>MCP3002</b>	ADC	No	10 Bit	\$2.30
<b>MCP4911</b>	DAC	No	10 Bit	\$3.50

**Table 12: ADC/DAC Comparison Table**

The ADC/DAC chip that we have decided to use was PCM3060. Because of its high bit resolution high precision due to it being a sigma-delta converter, this chip seems to be the best option for our system. The input to the DSP chips must be high resolution and as accurate as possible in order for the output sound to be as desired.

### **3.2.6.3. Interface Hardware**

The interface that is used to change the settings of each of the DSP chips on the board can be implemented in multiple ways. Section 3.2.4.1 goes into detail on the possibilities of the interface that could be used.

In order to implement the interface using a file loaded to a storage device, the hardware must allow for a storage device to be connected, either internal or external. In order to easily set this interface up, a USB flash drive can be used to load the file into the microcontroller where it will be parsed and sent to the correct DSP chips. This will require some hardware to be implemented including. One potential part to use to allow a USB device to be plugged in to the pedal would be a USB expansion board. This will allow for data to get passed from the device to the microcontroller which will then set the digital effects desired by the user.

If using the interface option of the LCD display on the product, there is definitely a need for extra hardware on the system. This hardware would include an LCD screen and a microcontroller capable of interfacing the screen. The LCD display would not have to be very advanced since only simple displays will be needed if this interface is used. A large screen is not necessary as well since only one setting needs to be displayed at a time.

Ideally the hardware would allow for seeing all settings at once, however due to size and cost constraints, smaller LCD screens such as the ones listed below will be effective for this implementation. The options for LCD screens are shown below in table 13, and are compared based on the size and cost.

<b>LCD Display</b>	<b>Size</b>	<b>Cost</b>
<b>Nokia 5110</b>	84x48 (pixels)	\$3.00
<b>GDM1602K</b>	16x2 (segments)	\$16
<b>RRLCD204WB</b>	20x4 (segments)	\$10.99
<b>GDM2004D</b>	20x4 (segments)	\$14.36

**Table 13: LCD Display Options**

The next interface option includes using a serial method to communicate with the DSP chips. This will allow for a computer to be used to change the values of the settings in each DSP chip. Having a computer to interface these chips is very helpful because it allows for a much nicer looking user friendly graphical user interface. Since the user interface is easier to understand the user will be able to change the values much easier. The hardware that is needed for this method is a serial peripheral interface or SPI. There are many options for hardware to implement this. Below in table 14 is a list of possible hardware options for an SPI.

<b>SPI</b>	<b>Cost</b>
<b>μUSB-PA5</b>	\$25
<b>FT232RL</b>	\$15

**Table 14: SPI Options**

For a mobile interface a Bluetooth or Wi-Fi module is needed in order to get values passed from the program to the DSP chips. Having a wireless connection to interface the chips will allow for users to have more flexibility when setting the values for the pedal. When playing they can easily press a few buttons on their phone to load a different effect almost instantly. Between songs would be the best time for this to be done in case it takes some time for the effect to be loaded. The program would use serial communication in order to send the values to the board. This communication will be done wirelessly using a chip connected to the board. A Bluetooth and wifi option are shown below in table 15. Each of these will allow for the same functionality.

Chip	Wireless Type	Rate	Interface	Cost
BC127	Bluetooth	3 Mbps	UART	\$27
RN-52	Bluetooth	3 Mbps	UART	\$25
XBee Module	Wifi	72 Mbps	UART	\$45

Table 15: Wireless Communication Options

### 3.2.6.3.1. Interface Hardware Selection

The interface that is most probable to be implemented for the G12 PedalVision is the LCD screen interface. Because of this, the selection of hardware is based off of this interface. The hardware chosen for the LCD display is going to depend on how much space we have when building the product.

For this purpose, the RRLCD204WB will be used if only one LCD display is used for all of the digital effect. This display has a high enough pixel density to be able to fit the messages we need to display easily. Also it includes and LED backlight. This is a plus because the product is likely to be used on stage where it may be low lighting. With the backlight on the screen, the user will be able to see the display at all times. Since this display is not very large or expensive, it will also be used if multiple displays are needed. Ideally, one display will be used in order to keep the size of the pedal as small as possible.

Along with the LCD screen, the interface will need a microcontroller in order to display to values to the screen, as well as to communicate with the DSP chip. At the moment, the microcontroller that is being used to control the LED matrix will also be used for this purpose. See section 3.3.3.1 for more details about this microcontroller. This is the current selection, however it is possible that when implementing the design, there may be a need for a separate microcontroller. In this case a low powered MSP430 may be used in its place since the logic needed is fairly simple and does not need a high processing speed.

In order for the microcontroller to communicate with the LCD screens as well as the DSP chip, I2C communication protocol will be used. This will allow for the microcontroller to be the master device, while the LCD display and each DSP chip are the slave devices. Using the simple two line

## 3.3. LED Matrix Feedback Display

### 3.3.1. Purpose

The LED matrix feedback display is an array of individually addressable LED lights that are toggled on and off to different colors and intensities dependent on the frequency and amplitude of the incoming signal. This matrix does not add any value to the actual guitar effects pedal section of the project. It does not offer any type of filtering or distortion to the input signal. The input signal is merely used as a reference to determine the LED settings to be displayed. What the display does offer is the ability for the end-user to have

a visual feedback of the notes that he or she is playing and their affected output after any type of effects are applied to the signal. This may not be important to all end-users due to the fact that it doesn't enhance the sound quality or playback effect, but to the sit at home and play in their room with the lights low user, it can add a feeling of being in sync with your music. Allowing them to turn their amplifier up loud to feel the music, of course hear the music, and now see the music like never before.

This is the effect that we are going for, adding that new dimension to music playing that the average user hasn't been able to experience. This being said, we realize that those other end-users that don't desire this encapsulation of the music need to get some usefulness out of the LED matrix feedback display. We have accomplished this by offering a tuning feature that is built into the unit. This feature will not only allow users to tune their instruments quickly and efficiently, but it will serve as a conversation piece to show other musicians. Having a unique design for the tuning feature gives it that one of a kind feel and look that will make it worthwhile to share with others.

### 3.3.2. Necessary Features

These are key features that the LED matrix feedback display must have, and some features that may be added to the product in its later life due to time constraints.

- Must haves
  - Must be able to set LED colors individually
  - Must be able to adjust overall brightness of each LED individually
    - Proportionally adjusting the current through red green and blue evenly
  - Must be able to set a minimum of 7 unique colors to the LEDs
    - This is due to the main harmonic frequencies
  - Must be able to take an input that is the instrument being played after passing through the G12 guitar pedal
  - Must be able to interpret the frequency of the input signal accurately
  - Must be able to interpret the amplitude of the input signal accurately
  - Must be able to change the color of each LED at a speed at which the normal human eye will not see a flicker
  - Must have a tuning feature that can tune the input instrument to at least the level of the standard tuner on the market
  - Must have buttons and switches installed on the board that allow the user to select various modes that they would like to use
  - The MCU must have a clock fast enough to read the input signal for its frequency and amplitude, as well as interpret this and serially communicate the appropriate settings to the LED driver
- Wants
  - May have the ability to link multiple units together to create a larger array of lights

- May have the ability to have modes selected wirelessly through a companion application
- May have a diffusion sheet placed over the top of the unit causing the lights to have a smoother transition between them
  - This would be purely aesthetic and a drawback of it would be that it would lower the perceived intensity of the lights

### **3.3.3. LED Module Hardware**

#### **3.3.3.1. MCU**

The ATmega2561 is a multi-purpose microcontroller that we have selected to use as the MCU that will do all of the high level thinking for the LED matrix display unit. The ATmega2561 was selected based on several key features that it offers that are of importance to the displays functionality.

256 KB Flash memory. This is a more than a comfortable amount of memory for programming space needed to accomplish what we have set out to accomplish, yet still leaves room for open ended growth. Leaving room for the potential for updates to be pushed out that would offer new display features, or better power consumption options. Whatever it may be to come, it leaves room for the product to grow.

16 MHz operating speeds. This MCU needs speeds of at the minimum above 40 KHz, but everything greater allows us to compute complex equations between samples granting us a better display product with more time for computation and information transmission between chips such as the TLC 5955 and the ADC chip that is reading in the values and sending those values over serial communication to the MCU.

54 Input/Output pins. Having this many pins available to us is a luxury that this chip brings us. For the first version of our system this chip may use only a fraction of these available input/output pins, but again this allows the project to stay very open ended allowing us to, in the future, include Bluetooth, Wi-Fi, touch-screen displays, switches, accelerometers, humidity sensors, temperature sensors, etc. The list can go on and on for add on pieces to improve the products usability.

16-bit resolution pulse-width-modulation. Having such a fine resolution of pulse-width-modulation allows one of two things. One, allows us to if the need arise, create a perceived voltage level between zero and five volts where we can step it around 76 micro-Volts, giving us great precision. More importantly, this allows us to recreate varying clock frequencies at the cost of resolution bits of course.

8 10-bit analog to digital converters. For this particular section of the project we will require an analog to digital converter, but for that we will not be using the analog to digital converter built into the ATmega2561 itself. This is due to the time required for this particular analog to digital converter to obtain its value and transfer that value out to the program. The built in one just isn't fast enough for the application of reading in audio signals. That being said, these can still easily be used and integrated into further add on pieces such as reading in a potentiometer for a knob that would be integrated onto the housing, these applications aren't as time sensitive as audio applications.



The Physical size of the chip. The ATmega2561 has a footprint of only 16 millimeters by 16 millimeters. This compact size, especially for the pin count was a large deciding factor as to going with it since size is fairly important for the end product since the LEDs and the LED drivers will be taking up majority of the space on the printed circuit board. Also the chip being a surface mounted device allows for us to place the chip on only one layer of the printed circuit board, and not have it cut through and potentially disturb other internal traces that may be routed below it if it were a through hold chip.

It can be seen that the previously laid out capabilities of the ATmega2561 can, and will easily tackle the large job that is set out for it within this project.

### **3.3.3.2. TLC 5955**

#### **3.3.3.2.1. Abilities of the TLC 5955**

The TLC 5955 has 48 channels in which each channel is able to have an individual PWM value set to it. This allows for 16 multi-color LED that have input values for red, green and blue to each be set individually allowing for a multitude of colors to be set.

The TLC 5955's outputs are constant current outputs. With 65,536 achievable steps per individual output through PWM, the total achievable colors for the chip per multi-color LED is 281 trillion colors.

The TLC 5955 has a 128-step, 7-bit, output setting that can set a "global brightness" per each set of 3 outputs, correlating to the red, green and blue legs of each LED which allows the overall brightness of the LED to be changed independently for each LED.

If a 64 LED arrangement is desired, only 4 chips will be required to achieve this.

The TLC 5955 works as a shift register, therefore stringing multiple TLC 5955 chips in series would work without issue allowing each one to be set and triggered simultaneously.

The main MCU can control the clock and the data being shifted in bit-by-bit and enable all of the subsequent chips simultaneously to have the settings take effect in unison.

The TLC 5955 can have an input clock as fast as 25MHz, allowing for a much faster reaction than the human eye can perceive. When talking about a system that is highly dependent on speed due to real time reaction speeds this chip allows for speeds that are greater than needed for the project. These features allow for a smooth transition for the viewer and for a better experience since a delay won't be noticeable.

With built in LED open detection the TLC 5955 can protect itself from any opens. The TLC 5955 has a register built in to set the maximum current allowable through an output. This eliminates the need for an external resistor.

The plan is to have the TLC 5955 work as a slave to the main MCU, where the main MCU will give commands about light settings to have which are dictated based on the mode that the display is currently set to, and based on that information it will receive the proper register values to be stored.

The colors will be chosen based on main harmonic frequencies. Where each of the main frequencies is assigned a particular color and when referencing a combination of frequencies, this is meant as if multiple notes are played simultaneously then each one of those colors are added to make a combination of the colors, the more colors are added to it, the closer to a white light the LED becomes.

The fundamental frequencies that we will be referencing are starting from the reference point of A4 being equal to 440 Hz, and using the logarithmic scale from there to reference all other notes about it.

The control signal will work as a digital signal processor signal that will be fed from the G12 pedal where within it, it is first converted from an analog signal, to a digital signal. This is at a 28-bit resolution. This signal will then feed through a series of digital filters to get it to the appropriate new values where it is then sent out through its output to a new MCU that interprets the data and processes it to a proper value to be fed out to the TLC 5955.

### **3.3.3.2.2. Various Display Modes**

#### **3.3.3.2.2.1. Tuning Mode**

This mode allows the user to get a visual feedback of their note that they are playing. It will display the letter of the note, along with that letter's color from red to yellow to then green where red indicates that you are far off from the center frequency, and as it transitions to green through the color yellow, it indicates that you are getting closer to the center frequency until you get a bright green color that then flashes three times indicating that you are within a margin to the center frequency.

#### **3.3.3.2.2.2. Unison Mode**

This display mode will use the entire 64 LED display to act as one single color transitioning between brightness and colors dependent on color and intensity of the notes being played, transitioning to different colors as frequency of the input signal changes to the main MCU, and the control signal changes to the TLC 5955. The lights will pulse in and out having their intensities set by the amplitude of the input signal to the main MCU.

#### **3.3.3.2.2.3. Segmented Mode**

This display mode will treat each column as a single node having them display the same intensity and color while transitioning in time to the beat from left to right. This will show eight different segments of beat colors simultaneously allowing the user to get an almost real time feel of the music for the last eight segments of time. This mode may also be shifted to transition from top to bottom rather than from left to right.

### **3.3.3.2.2.4. Continuous Mode**

This mode is similar to the segmented mode in that the previous frequency played is retained for some time. The difference will be that this mode travels in as one LED at a time, starting from top left, to top right and then traveling down to the next row and repeating across all lights until it reaches the end in which it then starts to write over itself from the top left again.

### **3.3.3.2.3. Various Storage Settings**

#### **3.3.3.2.3.1. Gray Scale**

This register is 768-bits wide and stores the data for the PWM controls for each individual LED color, each of which is 16-bits wide for all three colors, and then for all 16 sets of colors. This register can be written to through passing data serially through the SIN pin, and using the SCLK pin to shift the data over from the LSB to the MSB. The MSB must be set to the value 0 so that when the LAT pin is set to HIGH the following 768-bits are placed into the gray scale register, storing the PWM settings that are desired for each individual color in each LED. The gray scale register is the register that allows us to have access to over 281 trillion unique colors. This is handled from bit 0 to bit 768 leaving bit 769 as the control bit to decide whether or not this data is stored into the grayscale latch or the control data latch.

#### **3.3.3.2.3.2. Control Data Latch**

The control data latch is comprised of the dot correction data (DC), the maximum current data (MC), the Global Brightness control (BC), and finally the function control data.

- The dot correction data is 336 bits long, and controls the current output to each individual color in each LED. This is based on the limited value already, taking it from 100% to 26.2% based on each 7-bit value stored for all of the 48 outputs. This is handled from bit 0 to bit 335.
- Maximum current latch
- Global Brightness control
- Function Control data latch

### **3.3.3.3. LED**

When it came down to selecting LEDs for this project it was not an easy task. Not because it was abnormally difficult to find an LED capable of performing at the level that we would ask of it, but because there were countless LEDs that could take the job on. The requirements that we set out for the LEDs are as such.

Must be surface mount LEDs. This is to conserve usable space on our printed circuit board. A through hole design would take up potentially valuable real-estate on the top, bottom and everywhere in-between since it passes directly through the board.

Must be able to produce at the minimum seven unique colors. These seven unique colors are what will identify the seven fundamental frequencies of music. Through research and

## Group 12 G12 PedalVision

prior use of LEDs it was found that for this to be achievable and minimal on space we would need a four contact LED. One contact for each of the main light spectrum colors; red, green and blue. One contact as the common node for ground.

Must be small in size. Size of the LED is important for this project because the printed circuit board will need to hold all of the LEDs and we are limited by price for the size of the printed circuit board, as the price drastically goes up with larger printed circuit boards.

Must be top viewing with a greater than a 90 degree viewing angle. The LED must be top viewing do to the design of the board and the placement of the board relative to the user. The board will be placed on the floor in a housing that will protect it yet still let the light through from the top of the housing. As for the viewing angle, this was more of an arbitrary desire to allow for the lights to blend some and fill the room more.

Should be mid-level in power consumption. Low power consumption is not key with this project, but power consumption is still a factor. And with power consumed comes heat to dissipate, that being said we sought after an LED that would offer a mid-range brightness, with a mid-level power consumption.

Should be low cost, below one dollar per unit. The first design of the project calls for 64 of these LEDs to be used. To keep costs at a reasonable level, and to account for extra LEDs to be purchased, to replace broken ones, we would need the costs to be below one dollar per each LED.

With these requirements in mind we went out and searched for the LED that would be used on the LED Matrix Feedback Display. Through this we narrowed our search down to one disturber in particular, Kingbright. Kingbright is a company geared towards LED manufacturing, where you can even have custom LEDs made. Narrowing the search down to this manufacturer only made things slightly easier as they have countless types of LEDs, but following our set out guidelines for what we needed and wanted our LEDs to have helped narrow it down to a page of LEDs. It was at this point that we had one major difference between the LEDs, either 2 milliamp LEDs or 20 milliamp LEDs. These had different lumen levels, but nothing majorly different. We decided to go with the 20 milliamp LEDs since a brighter product would offer a better appeal, that and the fact that the TLC 5955 can only offer constant current as low as 3 milliamps. Now the last main difference between the remaining LEDs is where their red green and blue fell on the wavelength spectrum. This did not matter to us really, so we went with one close to a range that we were familiar with. Thus bringing us to our choice of the 3.5 x 2.8 millimeter RGB SMD LED, or more formally known as the part number AAA3528BGRS/129/C3.

### **3.3.3.4. Analog to Digital Converter**

Analog to digital converters do exactly as their name says, convert an electrical signal from analog, to a digital format. This is a necessity for this project, both in the digital signal processing of the input signal, and in the LED Matrix Feedback Display. This is a requirement for the LED Matrix Feedback Display due to the need to convert the input signal, which is the output signal of the G12 PedalVision, to a digital signal allowing the

ATmega2561 to interpret the value and take appropriate steps in order to obtain the desired color spectrum of the matrix.

As with all of the other subsystems, the analog to digital converter must meet a certain standard of wants and needs that we have set forward for the analog to digital converter to perform as required. These requirements are as follows.

Must have no less than a 10-bit resolution. The resolution is what gives an analog to digital converter its power, allowing it to differentiate from one level of a signal to another. Having too low of a resolution would equal to a loss of the signal, while there will always be loss of signal data when converting from an analog signal to a digital one, we want to minimize this as to capture the best signal that can be determined.

Must be able to sample at a rate of at least 40 Thousand samples per second. Humans can typically hear sound in a range of 20Hz - 20KHz. According to Nyquist sampling theorem, in order to capture a signal correctly you must sample that signal at a sample rate of at least twice the frequency of the input signal. Due to this theorem, sampling at 40 thousand samples per second will accurately capture all of the hearable frequencies for humans. While there will be a lot of wasted data when sampling lower frequency inputs at 40 thousand samples per second, this is not something that will negatively affect the product, but not sampling at a fast enough rate will definitely affect the output of the product.

Should be surface mounted. The analog to digital converter should be a surface mounted device to help keep things uniform, and to save available space for traces that can be routed below the chip. If it is through hole, then it is not going to be an absolute no on the product since we will only be using one of these on the LED Matrix Feedback Display board, but we would like to if at all possible keep it as a surface mounted device.

Should have serial communication. To help keep things simple and uniform, we would like it if the chip communicated back to its host using some form of serial communication protocol as to keep things easier on the programming end and uniform. This would also help reduce the necessary pin count on the chip itself if we are dealing with any resolution greater than or equal to 10-bit.

Some potential items to consider when discussing a system such as this one, while considering using an analog to digital converter would be the fact that the input signal is an AC signal. The AC input signal will be a signal that will vary about a center of zero volts. This can be an issue when looking at analog to digital converters as most cannot take input signals below the ground level. This can be handled in one of two ways.

One, purchase an analog to digital converter that can take in an alternating signal about ground level. This would be the simplest solution, as no modification to the input signal would need to be done to correct the alternating portion of the signal.

Two, adding a DC offset to the signal as to center it around the center voltage acceptable to the particular analog to digital converter being used. For example, if we are using an analog to digital converter that accepts a voltage between zero volts and five volts, we would add a DC offset of two and a half volts that would now center our alternating signal about two and a half volts. While this is a more complicated method compared to just purchasing a compatible analog to digital converter, it is still a viable option.

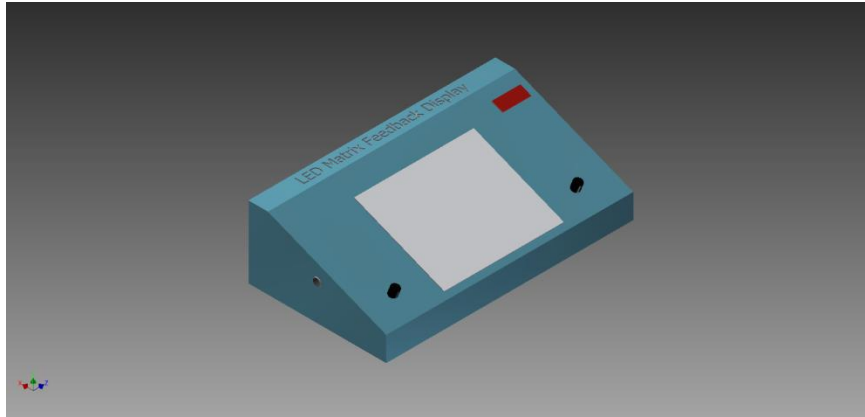
Another potential issue that we may encounter with our analog to digital converter is having too high, or too low of an input signal. This would be a signal outside of the limits of the analog to digital converter. To handle this on the high end of things, we will be incorporating a Zener diode with a level at the upper voltage level as to keep the voltage level from exceeding that upper limit. To protect the chip from the voltage reaching a level below its threshold, in the case that its threshold is zero volts, we will be adding a diode in line with the circuit to keep the potential from going negative, and to keep current from flowing backwards through the chip.

With all things considered we decided to go with the Texas Instruments PCM3060. This chip was selected because it meets all our criteria with a sampling speed of up to 192KHz. It also has a resolution of 24-bits. This particular chip has built in digital to analog converters, which may come into play for potential future projects such as an onboard speaker. Another large deciding point to use the PCM3060 was that it was already being integrated in the project, and by buying these items in bulk we are able to bring costs down a bit.

### **3.3.3.5. Housing**

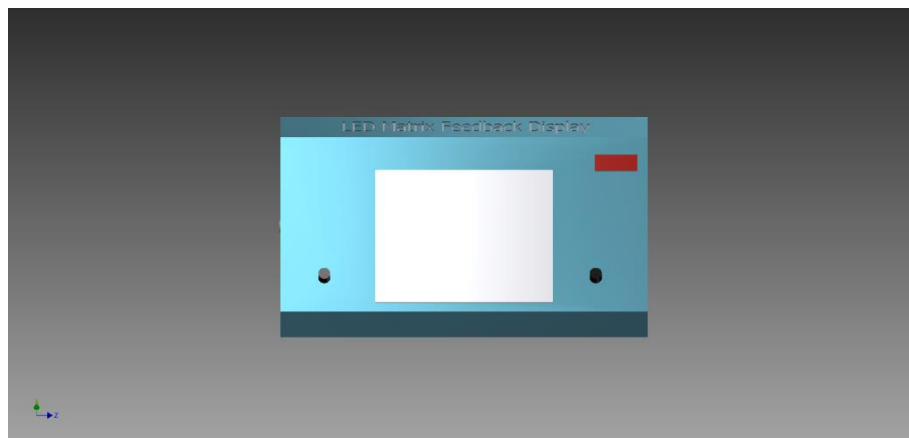
The Housing that will encase the LEDs and all of its components will serve as both the protector of the internal components that are sensitive to dust, moisture and pressure. As well as the projector of the LEDs to diffuse the lights appropriately while facing them in the general direction of the player, or anywhere the user would like the lights to face.

The general shape of the housing will be that of a standard guitar pedal, wedge shaped with a squared off front and back. This is to offer the proper angle for comfort for a player to interact with it from a sitting position with their foot. The body will be made out of a light grade aluminum. The light grade aluminum was selected based on our understanding that this piece should not receive any significant force on it, therefore it need not be able to withstand any significant force greater than dropping a microphone onto it. Figure 29 shows an isometric view of the product.



**Figure 29: LED Matrix Feedback Display**

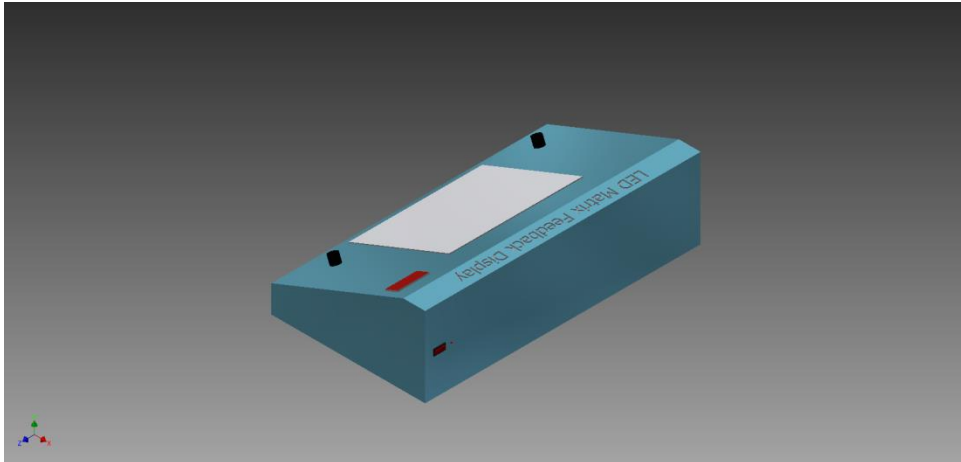
Centered on the body will be the LEDs. These LEDs are located on the printed circuit board that is below the surface of the housing. This is to allow for a diffusion panel to sit above the LEDs. This diffusion panel is used to blend the colors much like a 1970's dance floor did. It also keeps eyes off of unsightly LEDs that are not turned on, and lastly it will help keep the user from being temporarily blinded if they are to look directly into the LEDs. Below in figure 30 is the front view of the LED Matrix Display.



**Figure 30: LED Matrix Feedback Display Front**

Offset on both the left side and the right side of the LEDs will be two heavy-duty momentary switches. These will be standard switches used on guitar pedals. The switch on the left will be one that sends the LED MATRIX display back to tuning mode automatically. The right switch will toggle the display through its different display modes, but this toggling will never toggle it back to tuning mode.

In the top right of the unit will be two seven segment displays that will tell the user which tuning mode they are in. This will help the user cycle through the displays quickly ensuring that they reach the particular output they would like quickly and efficiently.



**Figure 31: LED Display Matrix Feedback Rear**

On the left hand side off of the front of the display will be an input jack that is 3.5 millimeters shown in figure 29. This is where the user will run the input cable to the LED Matrix Feedback Display, from whatever source they would like. There will also be a power jack and power switch located on the back of the unit shown in figure 31. This is located away from the input signal as to not cause any disturbances between the two of them. The bottom of the housing will have a material with high grip on it, most likely this material will be rubber feet with a wide enough area to offer no movement when the buttons are pushed with the feet.

### **3.3.3.6. Miscellaneous Components**

Some other items that are going to be required to complete the LED Matrix Feedback Display, but are not significant enough to mention individually are as follows.

1-channel 3.5 millimeter jack. This will be used to plug in the input signal from the source. For the main purpose of this project, the source will come directly from the G12 pedal, but in essence this source could be anything, including music that you are playing from a CD. We chose the 3.5 millimeter size since it's a very common size for auxiliary cables that people already own. It will also only need to be a single channel since the system can only handle one channel anyways.

Two momentary heavy duty buttons. These are briefly referred to when we were talking about the housing, but these buttons will be of the same quality that can be found on most guitar pedals. These buttons need to be able to stand up to the wear and tear of a user pushing on them with their feet, sometimes fairly hard as they are trying to act quickly to get the desired effect. These momentary switches need only be single pole single throw switches as they will only act as an interrupt to the main system.

A red status LED. This is an LED that will be placed on the back of the unit and will serve as a visual tool to help the user identify that power is reaching his system properly. This is key in letting the user know initially if there is a potential problem with his equipment, or with the power source.



A power rocker switch. This switch will also need to only be a single pole single throw switch. This switch will need to be able to handle 25 Watts of power as that will be the absolute maximum that the system will be able to provide, therefore all of that power would be passing through this switch since when this switch is closed it allows all power to flow to the system.

Seven Segment LED display. The system will call for a total of two seven segment LEDs. We do not anticipate even in the far off future having the need for more than 99 individual display modes, as this would take away from the usability forcing the user to scroll through too many displays to reach the one they may desire.

Capacitors. Nearly all electrical designs of this magnitude will at some point call for a capacitor or two. I want to only mention a couple significant capacitors that will be required. Two in particular will be required on the LED power side, one will be a large capacitor to offer much needed power during large power spikes to help keep the voltage level from dropping too low and potentially damaging components. This large capacitor will be accompanied by a smaller capacitor to ground that will help better filter out higher frequency noise, due to its size and it being a ceramic capacitor compared to an electrolytic capacitor like the larger capacitor. Each of the power pins going into every chip that we are implementing will also call for a decoupling capacitor as close to the leg of the power pin as possible, this will help keep most high frequency noise from causing interruption within the chip.

### **3.3.4. Printed Circuit Board Design**

Research for the printed circuit board for the LED Matrix Feedback Display has brought up some particular items to keep in mind while creating the design. These items are as follows.

Trace width. The current requirements through a trace will determine the trace width and weight of the material. The larger the current requirements, the larger the trace width needs to be. There are plenty of look up table to help calculate what these widths need to be based on the inputs you provide, such as weight, current and placement of width (whether it's an internal trace or external).

Size requirements. The printed circuit board will only be so big without the price jumping up significantly, therefore we must use our space efficiently with placement of parts.

Heat dissipation. Having parts many parts that require fairly large amounts of current running through them we must worry about heat dissipation and follow strict manufacturer design calls for large grounds beneath the chip with either solder or thermal paste being used to create a good contact for that heat to transfer to the board in order to create a larger area to be cooled.

Cross talking or inductance. A particularly troublesome design issue that may be harder to avoid in some cases. Having large planes of potential voltage separated by an insulator (the printed circuit board) can create large capacitors that will cause effects in the system that we will not want to see. Along with this comes not using the proper decoupling discipline to keep un wanted induced noise out of lines where it shouldn't be. This also

calls for minimizing traces to their minimum required length since the longer the trace is the more noise it will pick up due to EMF waves.

Time. If one thing needs to be followed its time. Manufacturing of a printed circuit board takes time, and expedited production comes at a hefty cost, therefore get the first versions of the board printed early as to allow ample time for testing and redesigning.

Patience. The board will have problems sooner or later, try to find the problems soon through testing. Even the experts have trouble when designing a new system.

### **3.3.5. Potential Drawbacks**

When you take something such as music that some people hold so near to them and try to improve upon it, you are bound to have some push-back. Whether that push-back comes from the musicians themselves, the instruments, or outside sources is another question. As a group, we wanted to address some potential issue adding the LED Matrix Feedback Display might bring about. These range from resistance from the user to power consumption. These potential issues are laid out here.

Musicians may want to live by the old saying, “if it isn’t broke, don’t fix it”. This has been seen come to fruition time and time again throughout American history of innovative products such as the early car industry. Any time a newer system such as VHS, DVDs or Blue-rays have come out, or more recently with Apple removing the headphone jack from their newest phone. The answer that we have for anyone that thinks this way, is to be on the leading edge of this, someone always has to be the first to try something out and discover its potential for greatness. They can be the user that finds a unique way to use the system that even as the designer, we couldn’t have imagined it used for.

Some Musicians may be afraid that it may throw off their ability to concentrate while playing. To this we say that like anything it may take practice, but before you know it, this may even improve your ability to understand what you’re playing, and be better able to tell if the note you were trying to play was achieved visually.

Power consumption users may worry about the extra use of power to run a bunch of lights just for enjoyment. Our answer to this is that with the design of modern LEDs, the power consumption is only fractional to running your standard incandescent lightbulb.

### **3.3.6. Similar Products**

Competitors are always something to worry about when trying to get a product out there. Luckily for us, there is no known competitor that creates a product such as the LED Matrix Feedback Display. That being said, there are several products out there that potentially can be adapted with a few extra bits of hardware to be similar to the LED Matrix Feedback Display. One of these products is the Adafruit 32x32 LED Matrix. This product comes as an already built matrix of LEDs that are each individually addressable to hold a particular color. The disadvantages to these LEDs is that their brightness cannot be individually adjusted. Another product that is similar is also manufactured by Adafruit, it is their Neopixel LEDs. These are a strip of their individually addressable

LEDs that can store their settings for each LED. These can be added together in series to create a seemingly endless array of LEDs.

These products offer the ability to have multiple LEDs each be controlled individually, but neither of which offers the controllability that our product offers. They would also both take some more hardware and software to get to a point where we would consider them a similar product.

## **3.4. Power Supply**

What is Power? Thinking about power, one thinks about inserting a plug into an outlet on a wall. In many cases, such as that one, power is something that is taken for granted. While plugging the device into an outlet, what happens? What kind of power is being supplied and why? Is the same amount of power being supplied to each device that is plugged it? How is it regulated and what kind of power supply is being used? These are just a few of the questions that are often overlooked about how power is being used in everyday lives. In this section, different types of power supplies, along with their advantages and disadvantages, will be discussed.

### **3.4.1. DC Power Supply**

A DC power supply is responsible for supplying a constant voltage to any load that is added to a unit.

#### **3.4.1.1. AC-to-DC Supply**

While functioning, the DC power supply changes from alternating current (AC) to direct current (DC). When an AC signal is first applied, a rectifier changes it to a DC signal. Then a regulator is used to change the DC signal to a desired output. A DC power supply can be manipulated into many desired output responses with different voltage ratings. This type of power supply can be powered by a DC or an AC source. Figure 32 below shows the process of changing an AC signal to a DC voltage. [8]

To relate a DC power supply to everyday life, one can think of a wall outlet that supplies 120V or 60Hz to any connected load. For example, since many logic device use a low voltage, a phone, iPad, laptop etc., can be charged using the power supplied by the outlet. Then the signal being received is manipulated into the desired output for each device.

In most cases having a DC power supply is necessary and essential for supplying constant DC power. Even though DC power supplies may be necessary for everyday use, they can also waste a lot of energy during the conversion process. Loosing so much energy as heat makes the DC power supply very inefficient.

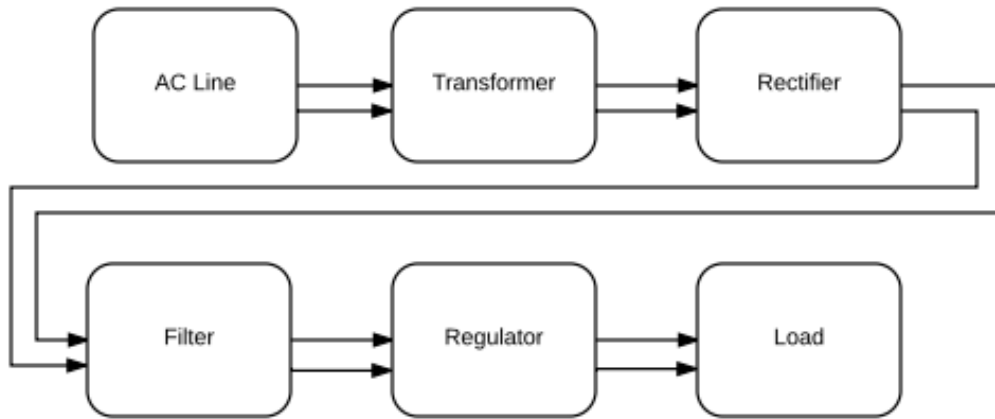


Figure 32: Block diagram of an AC signal

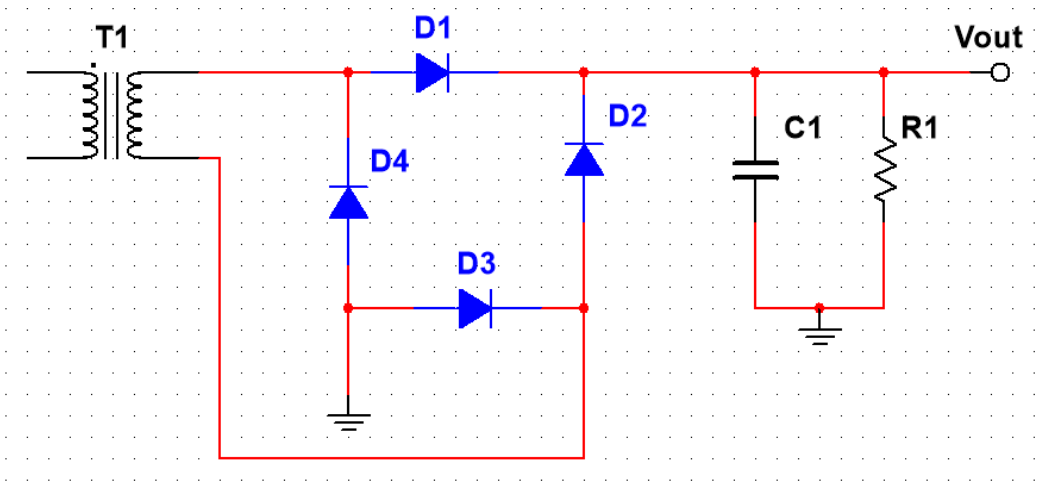


Figure 33: Full Bridge Rectifier

The circuit above in figure 33 shows a full bridge rectifier with a filter capacitor and a load resistor. 120 V/60 Hz is at the input and 120 Hz is at the output.

### 3.4.1.2. Linear Regulator

A linear regulator is used to help with the fluctuation that occurs when a DC power is supplied. It takes in an irregular DC voltage and converts it to a constant, lower DC voltage. As loads change for a circuit, the needed voltage also changes. It is important to have a linear regulator in the circuit to help account for the difference by varying its resistance. It will regulate the input to accommodate the output.

Linear regulators are great to use when it comes to limiting current because one of its functions is to limit current so that there will not be a huge current spike that will destroy a circuit. Another function that can be useful is, the reduction of noise present in the output. This can provide a more accurate reading. The disadvantage of using a linear

regulator is that it cannot supply and output voltage or a one that is opposite in polarity compared to the output voltage.

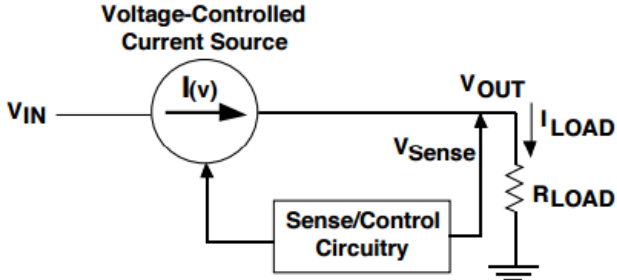


Figure 34: Basic Linear Regulator (Courtesy of Texas Instruments) [11]

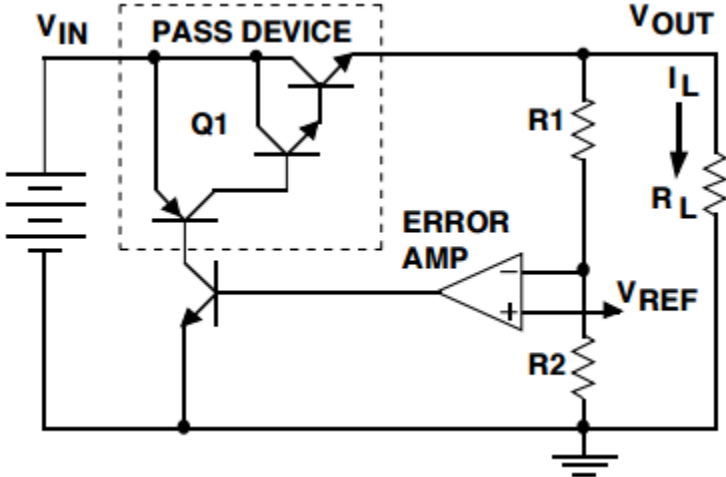


Figure 35: Linear regulator circuit diagram with a control loop (Courtesy of Texas Instruments) [11]

The two figures above were taken from Texas Instruments data sheet on Linear and Switching Voltage Regulator Fundamental Part 1. The basic linear regulator, as shown above in figure 34, has a sensor that detects any change in the output voltage. If any change is detected the current source is adjusted to gain the desired value. In the circuit, one can see that the output is being controlled by the feedback loop. The feedback loop has its own way of finding stability to compensate for any change that is occurring.

In the control loop circuit shown in figure 35, the voltage divider helps this circuit sense if there is any change to the output voltage. Since the two inputs of an op amp try to match each other and we want a positive voltage to cause the transistor connected to the output to turn “on” we tie the reference voltage to the positive terminal and the sensed output voltage to the negative terminal. Due to this difference in voltage, the output of the op amp will be the positive rail voltage until the two terminals have the same voltage. If, while using this circuit, there is a sudden change in load demand, the load resistance would adjust until the loop stabilized. [11]

Some advantages to using a linear regulator is low in complexity, cost and has low noise. If the G12 PedalVision requires step ups, along with step downs, the linear regulator may not be beneficial to use.

### 3.4.1.2.1. Adjustable Linear Regulator

One type of linear regulator is the adjustable regulator. An adjustable regulator allows the output voltage to be changed using various external resistor values.

#### 3.4.1.2.1.1. LM137

The LM317 is a three-terminal adjustable regulator. This linear regulator was considered for this design because of its wide range for the output voltage. Per its datasheet, it is current limiting and has thermal overload protection.

### 3.4.1.2.2. LDO Regulator

LDO, low-dropout, regulators are used to regulate DC when the input voltage is very close to the output voltage. Figure 36 shows the basic schematic of an LDO. If this regulator is used in conjunction with a switching regulator (as mentioned in section 3.4.1.3) it can greatly reduce the ripple noise of the system.

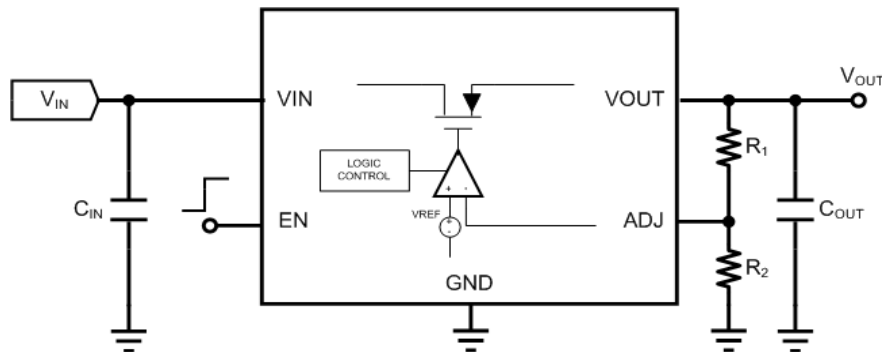


Figure 36: A basic schematic of a LDO regulator [5]

#### 3.4.1.2.2.1. MIC5265

The other linear regulator being considered was the MIC5265. It looked promising because it is used for portable devices. Not only are the applications appealing, it has a low drop out voltage and the supply voltage runs around 2.7 v – 5.5 V. Since the chip offers low output noise, it would make it easier to filter out all the unmercenary noise caused in the design.

After reading about both adjustable and low-dropout linear regulators, it appears the adjustable linear regulator would be more beneficial in this project. The reason behind decision is the adjustable linear regulator can be helped to drop down a wider range of voltage. This may be necessary if the switching regulator that is being considered doesn't from down the wall voltage to the desired about. Not only would the adjustable linear regulator be used to drop down the voltage, it would also eliminate most of noise

caused by the switching regulator. Some advantages to using a linear regulator is low in complexity, cost and has low noise. If the G12 Pedal Vision requires step ups, along with step downs, the linear regulator may not be beneficial to use.

Linear Regulator	Manufacturer	Voltage Range	Cost
LM317	Texas Instruments	1.25 V – 37 V	\$0.14520
MIC5265-1.8YD5-TR	Microchip Technology	2.7 V – 5.5 V	\$0.24

Table 16: Regulator Comparison (Courtesy of Texas Instruments)

### 3.4.1.3. Switching Regulator

Switching regulators can be used in place of linear regulators for more complex designs. When the regulator is switched off, it dissipates close to no power. Some advantages of using a switching regulator are that it is capable of being a boost, buck or an inverter. This allows the option for the output voltage to be greater than the input voltage. The switching regulator does not waste much heat, and is very efficient. This device would be beneficial to use in this project if a wider range of voltages needed to be accounted for.

There would be noise at the switching rate, but the pros may outweigh that one con. In a basic switching regulator circuit design, there will be an inductor, power switch and a diode. These three components can be moved around to gain the different functions of a switching regulator for the input and the output. Figure 38, shown below, from maxim integrated show the various topologies for a switching regulator.

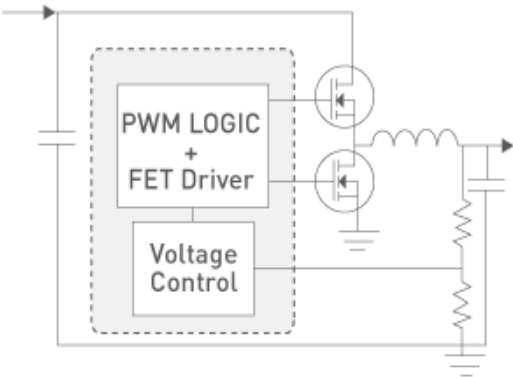


Figure 37: Basic Switching Regulator [9]

In a basic switching regulator circuit design, there will be an inductor, power switch and a diode. These three components can be moved around to gain the different functions of a switching regulator for the input and the output. Figure 39, figure 40, figure 41, and figure 42 shown below, from Maxim Integrated show the various topologies for a switching regulator.

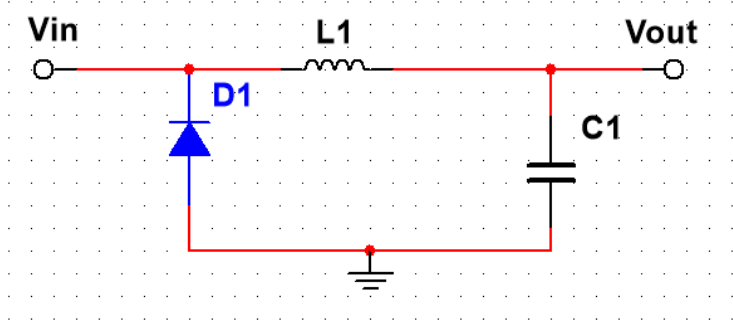


Figure 38: Buck Converter Topology

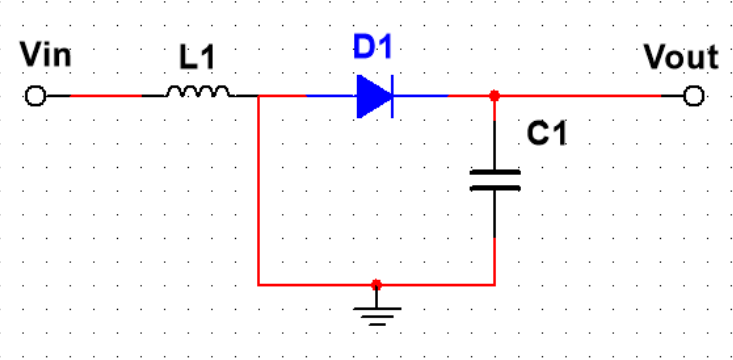


Figure 39: Boost Converter Topology

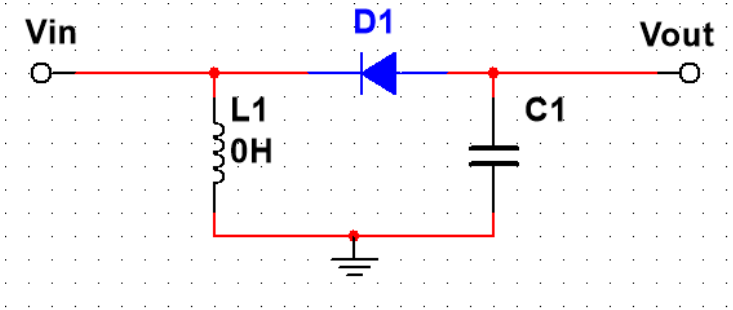


Figure 40: Inverting Topology

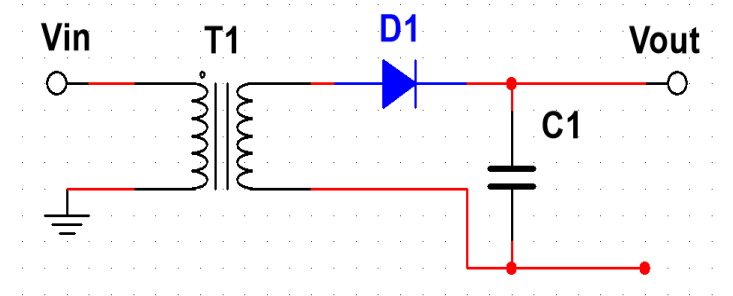


Figure 41: Transformer Fly-back Topology



### 3.4.1.3.1. MC34063 Switching Regulator

One of the switching regulators that is being considered for this project is the MC34063, A, Step-Up/Down/Inverting Switching Regulators. Per the regulator's datasheet, it operates from 3.0 V to 40V input, has a low standby current, is current limiting, has an output switch current as high as 1.5 A, an adjustable output voltage, and an operation frequency as high as 100 kHz. Figure 43 below, shows the bottom view of the MC34063.

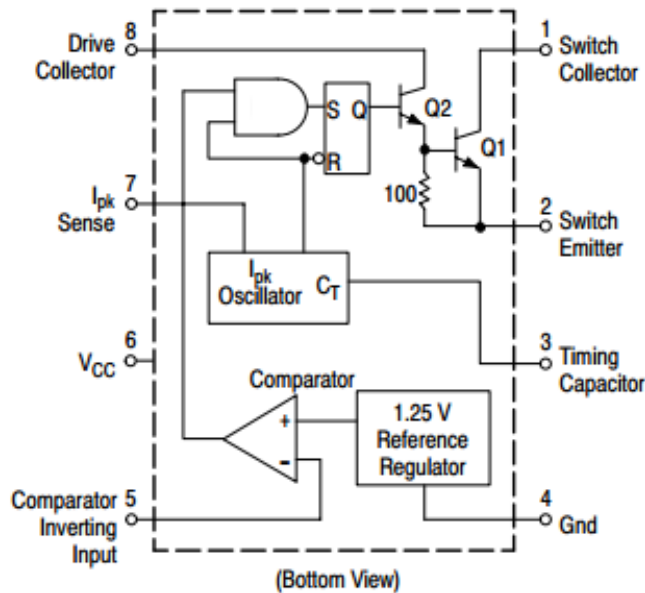


Figure 42: Basic Diagram for the MC34063 (Courtesy of Texas Instruments)

### 3.4.1.3.2. LM3578AMX Switching Regulator

Another option for the switching regulator that is being considered for this project is the LM3578MAX, A, Switching Regulators. Per its data sheet, this switching regulator has inverting and non-inverting inputs. It has a 1.0V reference at its inputs, along with operating voltages of 2.0V to 40.0V. It has an output current up to 750mA, with a saturation less than 0.9V. It can be used as a buck, boost, inverting and a single-ended transformer. In Figure 44, the pin diagram for the LM3578AMX is shown.

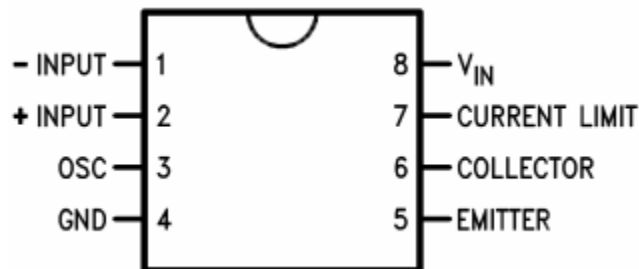


Figure 43: Pin Diagram for the LM3578AMX (Courtesy of Texas Instruments)

### 3.4.1.3.3. MIC3172YN Switching Regulator

Another one of the switching regulators that is being considered for this project is the MIC3172YN. Per its data sheet, the operating input voltage range for the MIC3172YN is 3.0V to 40.0v. It has a 1.25A internal switch rating, and it has internal cycle-by-cycle current limit. It is slightly limited because it only operates in most, not all, switching topologies. Table 17 below gives a comparison of the proposed switching regulators.

Switching Regulator	Manufacturer	Capabilities	Supply Voltage	Output Current
MC34063	Texas Instruments	Step-Up/Down/Inverting	3-40V	1.5A
LM3578AMX	Texas Instruments	Buck, Boost, Buck-Boost	2-40V	750mA
MIC3172YN	Microchip Technology/Micrel	Buck, Boost, Cuk, Flyback, Forward Converter	3-40V	1.25A

Table 17: Switching Regulator Comparison

After comparing all the basic specifications of the three switching regulators that were researched, the MC34063, is the one that will likely be chosen for this project. The wide range of input and output voltages of the MC34063 would be beneficial for this project.

All analog and digital effects in the G12 Pedal Vision Design should be running at around 9V. The LED Display that will be connected to the pedals will most likely run at either 12 V or higher. Since the LED Display will act as an accessory, load adjustment will also be something to consider.

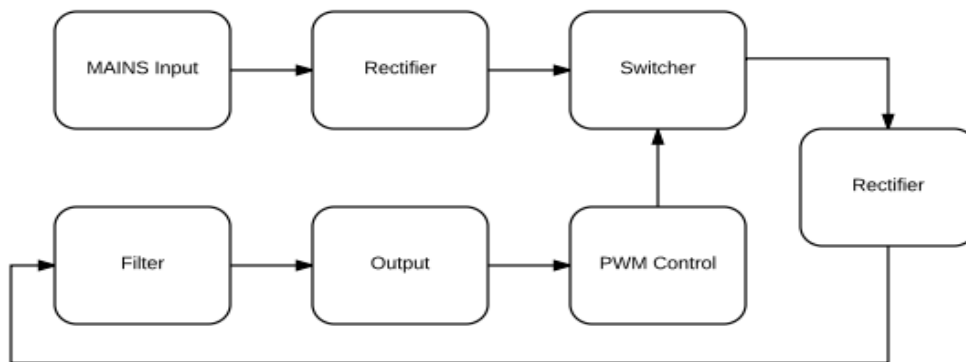


Figure 44: Switch Mode Power Supply Block Diagram

Figure 44 shows the block diagram of the general functionality of a switch-mode power supply. Switch mode power supplies are beneficial to use when a higher efficiency, and a smaller light weight size are needed. The down side to using a switch-mode power supply is that it can create a lot of electrical noise. Table 19 below compares the two types of regulators.

Regulator	Capabilities	Noise/Ripple/EMI	Input Range	Efficiency	Cost
Linear	Buck	Low	Small	Low to medium	Low
Switching	Buck, Boost, Buck-Boost	Low	Large	High	Medium to High

Table 18: Linear vs Switching Comparison

### 3.4.1.4. Voltage Divider

A voltage divider circuit is one of the fundamental ways to drop down voltage. Two resistors are placed in series and an input voltage is dropped down between the two resistors to create an output that is a fraction of the applied input voltage. Figure 45 shows a schematic for creating the voltage divider.

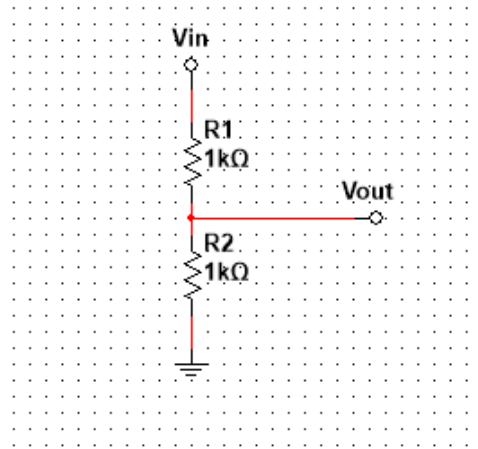


Figure 45: Voltage Divider

Voltage divider circuits come in handy with applications such as pots, and level shifts. They are easy to use in these applications because a small amount of voltage is being adjusted to accommodate any small change that may need to occur in a circuit.

For example, a pot consists of an adjustable voltage divider as seen in figure 46. The circuit below displays the wiper that is needed for the pot to work. If the wiper goes all the way to point 3, then the voltage is assumed to be 0 V and the device is off. If the wiper adjusted to the point 1 on the schematic, the voltage is assumed to be close to the input voltage, making the device be completely on. This application may be used to adjust the brightness of the LED's on the LED display. It would allow the user to adjust the intensity of the display. If a pot is used in the G12 Pedal Visions design, three pins will be dedicated to its functionality. The voltage, ground, and wiper functionality will all be supplied by the PCB.

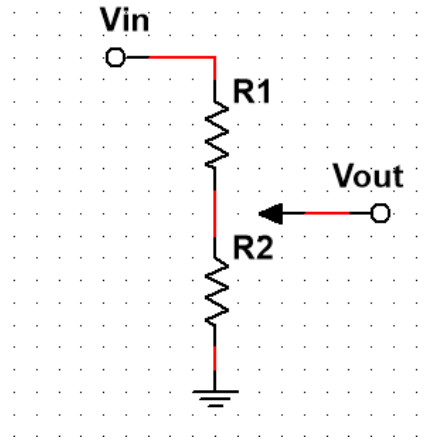


Figure 46: Potentiometer Example

Another beneficial application of a voltage divider is a Level shift. Level shifts come in handy after your voltage has been dropped down from the supplied voltage and then needs to be readjusted to a couple of different voltages. For example, if there are two chips on the board that need two different operating voltages, 3.3 V and 5 V, then a level shift would be used to supply the two different voltages at the same time.

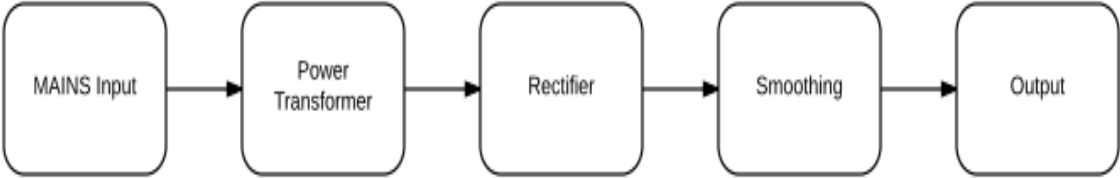
Voltage dividers shouldn't be used to supply power to loads. Dropping a higher voltage to one that is a lot lower is possible through voltage division, but not practical. If this technique is used, too much energy would be lost and it would burn out the resistors. If a large adjustment needs to be made to the voltages, it would be easier to use either the linear or switching regulator. Both types of regulators were discussed in 3.4.1.2 and 3.4.1.3.

### 3.4.2. AC Power Supply

Every outlet, MAINS input, that is supposed to power any device, supplies an AC supply. The voltage supplied through an outlet is dependent on its use and location.

When a switch is flipped to turn on lights or if your phone needs charging, 120 V is supplied from the outlet. Adjustments are made to the supply voltage for appliances. Appliances such as Dryers and ranges need 240 V supplied to them for functionality. For commercial locations 120-208 V is supplied. The 208 V is used for special equipment that may be used, such as equipment in an event hall.

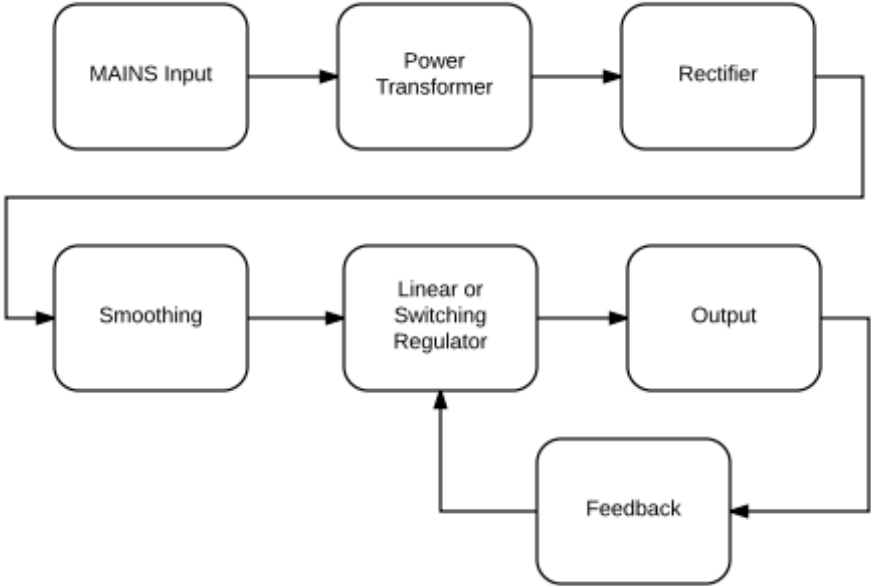
For residential areas, the AC power comes from a power pole. There are transformers on the poles that drop the voltage down to the desired range. For commercial use, the high voltage supplied from the power pole, is taken through a transformer to drop it down to 120-208 V. Instead of coming directly from a power pole, as it does for residential areas, the power for a commercial area is all tied into an electrical room where there is switch gear to the voltages 480V, 120V, and 208V.



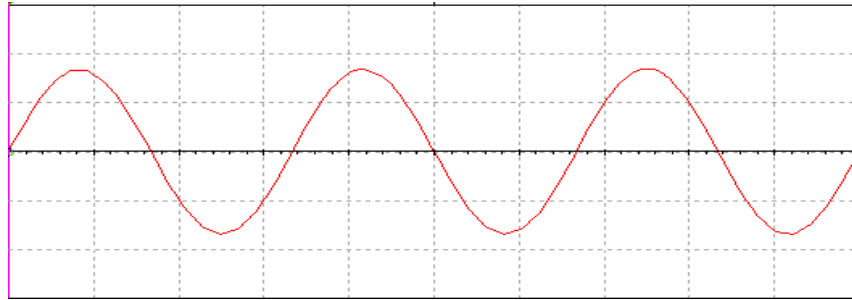
**Figure 47: AC Power Supply**

The block diagram above in figure 47 shows the general method of processing an AC Power Supply. First the signal needs to be dropped down, then rectified. After the rectification, the signal needs to be smoothed out, to reduce any discrepancies such as noise. Then the signal is ready to be used. The beauty in this design is that it can do the fundamental process of how to adjust and harness energy and use it for whatever is needed in everyday lives.

The block diagram below in figure 48 illustrates a regulated supply. The diagram is showing that the output voltage is continuously sampled to adjust the output voltage accordingly and to reduce any discrepancies or filtering.

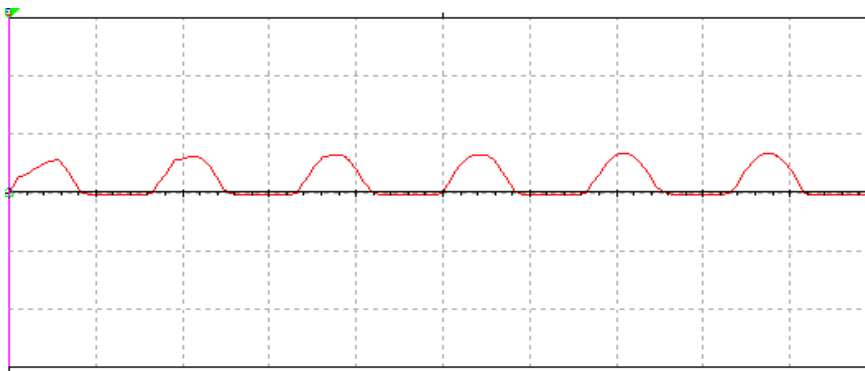


**Figure 48: Regulated Supply**



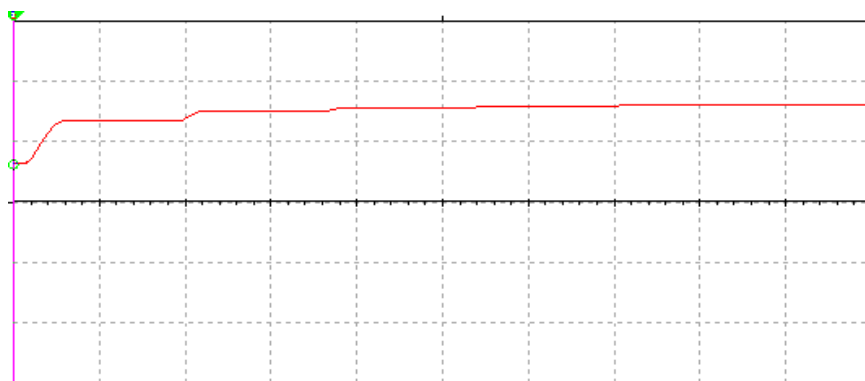
**Figure 49: Alternating Current Signal**

An alternating current signal that comes out of a Wall outlet is shown in figure 49 of the sine wave above.



**Figure 50: Rectified Sine Wave**

The sine wave is then rectified using diodes and the sine wave is adjusted and changed into a series of peaks so that only the positive signal goes through. This is shown in figure 50 above.



**Figure 51: Filtering after Rectification**

After the rectification, the wave still needs to be smoothed out. A capacitor is used to smooth out the signal, therefore creating a better quality of power supply. This is shown in figure 51 above.

### 3.4.3. Transformer

The job of the transformer is to step-down voltage. It takes changes a high voltage to a low voltage through a magnetic field. The higher voltage enters a coil that is around an iron core. The alternating current changes the strength of the magnetism. This process allows the voltage to go to an output wire, the secondary winding in figure 52, that has fewer coils. Since the secondary winding has a lesser number of windings than the primary winding, the voltage is dropped. Overall, alternating current power is transformed through the process so that it can be used to distribute power.

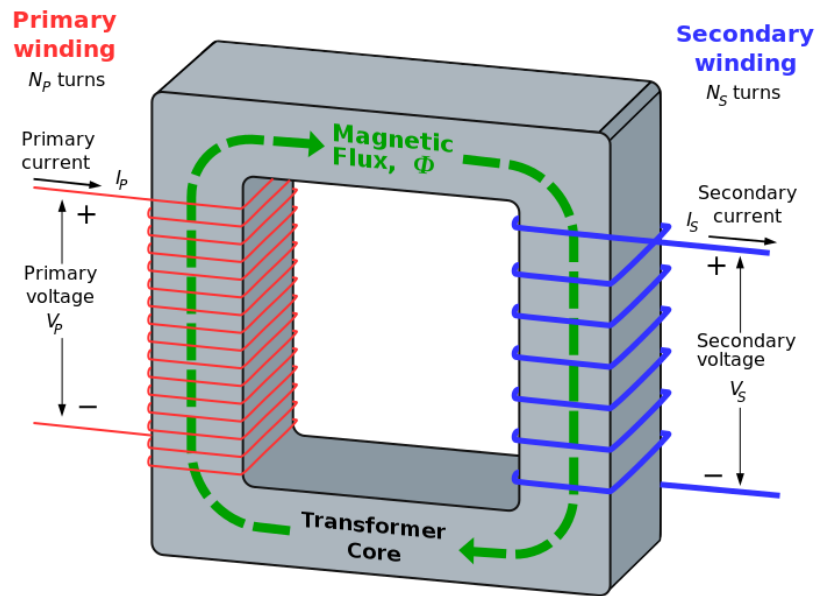


Figure 52: Basic Transformer Functionality

The transformer is a device that is commonly used in power lines. Higher power is cheaper to transport, therefore making the transformer cheap and easy to use when needed. Another benefit to using transformers is that they are reliable. A transformer will possibly be used in the G12 Pedal Vision design. It would be beneficial to use a transformer to drop down the wall voltage, which is 120V AC at 60 Hz, to a desired voltage range.

### 3.4.4. Variac

A Variac is simply a variable transformer. In a Variac the primary or secondary coils can be adjusted to vary the output voltage based on the input voltage. When thinking about how a variable transformer works, it helps to relate it to how a load is changed in a basic circuit. The change in load adjusts the functionality of the designed circuit accordingly. In figure 53, the number of primary coils can be adjusted, therefore adjusting the number of secondary coils. All of this is taking place while using one coil winding. An alternating current is still needed for it to function and just like a regular transformer, a time-varying magnetic field is used to transfer the power.

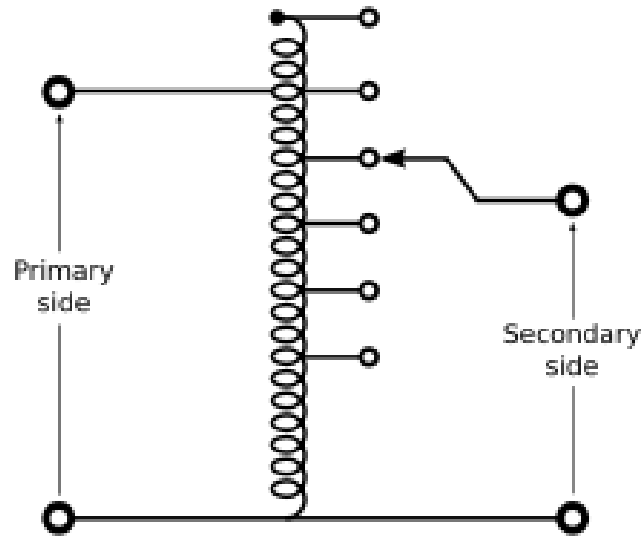


Figure 53: Variac

The Variac seems to be an easy way to convert a higher voltage to a lower one using time-varying magnetic fields. Even though it is easy to use, control and cost efficient, there are a few cons to using a variable transformer. It is less durable than a regular transformer, it cannot handle as much power as a regular transformer and it does not have any sort of isolation between its coil windings.

### 3.5. Power Distribution Design

While thinking of the power distribution design, some of the chips being used in G12 Pedal Vision were considered. Depending on each chip, operating voltage, the power may need to be distributed using different techniques to satisfy each condition.

#### 3.5.1. PCM3060

The PCM3060 ADC/DAC chip is a 24-bit asynchronous stereo audio codec with 96/192kHz sampling rate. Per Texas instruments, the chip has single-ended analog inputs and differential analog outputs. The operating voltage for this chip ranges from a minimum of 4.5V to a maximum of 5.5V for  $V_{cc}$ . For  $V_{DD}$  the voltage ranges from a minimum of 2.7V and a maximum of 3.6V. At full power down, it dissipates around 4.4mW of power.

#### 3.5.2. TMS320C6720

The TMS320C6720 Chip is a 32-/64-bit floating-point digital signal processor. From the core voltage to the voltage for the input and output, the voltage for this chip ranges from 1.2V to 3.3V.

#### 3.5.3. OPA164x

The OPA164x SoundPlus is a high performance, JFET-Input Audio Operational Amplifier that is made by Texas Instruments. This chip has a low noise and low



distortion. It has a wide voltage range of  $\pm 2.25\text{ V} - \pm 18\text{ V}$ . The chip is used for professional audio equipment and is also used for analog and digital consoles, making this chip applicable to the project.

### 3.5.4. LM13700

The LM13700 is a dual operational transconductance amplifier with linearizing diodes and buffers made by Texas Instruments. The chip needs around  $9.5\text{ v} - 32\text{ V}$  to operate. It is used for stereo audio amplifiers, current-controlled amplifiers and many more audio related applications.

### 3.5.5. TLC5955

TLC5955 is a 48ch, 16 bit LED Driver. Each channel can be adjusted and has a constant pulse width modulation, and color correction. The chip also offers constant brightness control for the LEDs. The chip needs around  $3.0\text{ V} - 5.5\text{ V}$  to run.

### 3.5.6. ATmega2561

The ATmega2561 is a chip made by Atmel is a microcontroller with either 16, 32, or 64 KB in-system programmable flash. The chip has advanced RISC architecture and it takes around  $4.5\text{ v} - 5.5\text{ V}$  to run.

### 3.5.7. Part Power and Cost comparison

Table 20 below shows the comparison of all of the aforementioned parts that will make up the G12 PedalVision.

Chips	Manufacture	Voltage Range	Cost
PCM3060	Texas Instruments	$4.5\text{ V} - 5.5\text{ V}$	\$6.51
TMS320C6720	Texas Instruments	$1.2\text{ V} - 3.3\text{ V}$	\$14.89
OPA164x	Texas Instruments	$\pm 2.25\text{ V} - \pm 18\text{ V}$	\$2.74
LM13700	Texas Instruments	$9.5\text{ V} - 32\text{ V}$	\$1.28
TLC5955	Texas Instruments	$3.0\text{ V} - 5.5\text{ V}$	\$.675
ATmega2561	Atmel	$4.5\text{ V} - 5.5\text{ V}$	\$16.38

Table 19: Part Power Comparison

## 3.6. General Harness Design

The wire harness for the power distribution is a crucial part of the design process. The wire harness for this project will most likely have a 64-pin connector. The connector will have pins dedicated to power, ground, each functionality of the analog and digital effects, and the LED display.

There are two types of connectors to consider for the design. The first being a connector that requires each wire to be soldered in for each pin. The second is when a pin is crimped onto the wire and placed into the connector.

Group 12  
G12 PedalVision

Soldering in each wire would be beneficial for testing purposes. If something was connected incorrectly or pin placement had changed, unsoldering the incorrect wire would be easy. On the other hands the crimped pins could also be viewed as an easy fix. A tool would be required to take out the incorrect pins.

Both types of connectors require specific tools and techniques. Based on experience, the connectors with the solder in wires would be easier to use and fix for G12 Pedal Vision. It would allow flexibility with different harness designs, in case anything else changed.

Smaller 9-pin connectors can be used when attaching/connecting a different power supply. For example, If the LED display needs a 12 V supply and the PCB needs around 30V or higher. The 30 V supply could be directly connected to the PCB, while the 12 V could come straight from the wall. The connector would allow a clean easy connection.

The PCB will be the main source of power for the potentiometers and possibly the LED display. Due to this, multiple grounds should be provided and connected accordingly. There will also need to be multiple power pins. Some of the power pins could be used for multiple potentiometers.

Based on experience, color coordinating the wires used and adding labels with the pin numbers will help with any troubleshooting that may be required. The cleaner the harness design, the easier it will be able to be fixed.

The harness will be one of the last things that is completely assembled for this project. It will need to be routed through the box to make a nice clean design. Zip ties can be used to old some of the bundles of wires together.

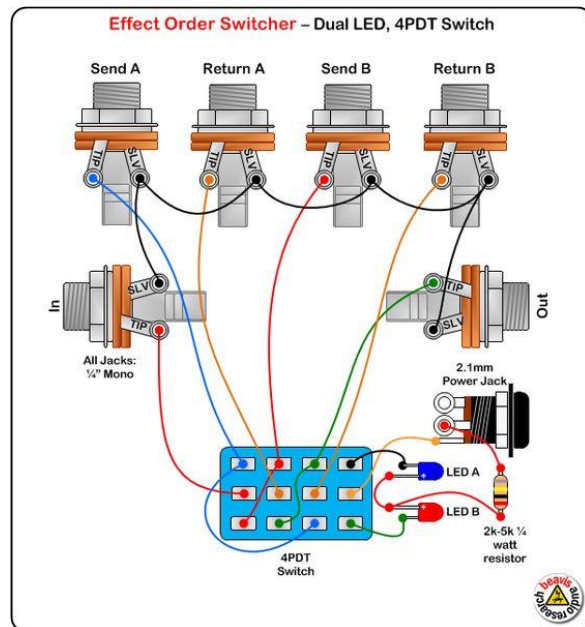


Figure 54: Example Wire Harness

The image above displays a very general overview of what a harness can look like. The harness isn't complete and shows basic connection. Instead of having a 4POT switch connect to the pedals, the G12 PedalVision will connect to the pedals.

## **3.7. Components**

### **3.7.1. Resistors**

Resistors will have various uses in this project. Resistors are known to limit current. If there is a constant voltage, the larger the resistance is, the lower the current will be. That relationship is known as ohm's law.

In this project resistors, will be used to make various voltage divider circuits to help make small adjustments to the voltage that are provided. The voltage divider could help drop down the voltage enough for the analog or digital effects.

Resistors will be used in accordance with the LEDs in the LED display. The current limiting aspect of the resistors will keep the LED from burning out.

Pull up resistors will ensure that all the circuitry will function correctly if the load is changed, power is disconnected or a high-impedance is introduced. There are also pull down resistors that can be used to pull to ground. For example, if a switch is open, the pull up will read high and the pull down will read low.

The values for each resistor will be determined based on what is required for each use. Values can be calculated, but may need to be adjusted during testing.

### **3.7.2. Capacitors**

Capacitors are components that are used to store electric energy in an electric field. A few different types of capacitors will be discussed below.

#### **3.7.2.1. Ceramic Capacitors**

One type of capacitor is the ceramic capacitor. In this case the ceramic material is being used as the dielectric. In resonant circuits, the ceramic capacitors, offer low losses and they also offer high stability. They are also more beneficial to use as coupling capacitors. Some other benefits are high efficiency for buffers and by-pass applications.

#### **3.7.2.2. Film Capacitors**

Film capacitors are constructed from two pieces of plastic that are then covered with electrodes. After they have been wound up into the desired shape, the terminals are attached. When the terminals are connected, the capacitors are complete and is cased. Unlike other capacitors, the terminals on the film capacitors are interchangeable.

The foil capacitors are made up of two plastic films as the dielectric, then each layer is covered with a metal like aluminum. The aluminum acts as the electrodes. The foil capacitor is use to handle high current surges.

The second type of film capacitor that can be used is a metalized film capacitors. Unlike the foil capacitor, two metalized films are used and the plastic acts as the dielectric. Some aluminum is then placed on both sides and acts as the electrodes. These capacitors are good to use in shorts circuits. The current surge rating for these capacitors is low, making the foil capacitors better to use.

### 3.7.2.3. Electrolytic Capacitors

Electrolytic capacitors are polarized and contain an anode. A specific type of metal is used an oxide layer is formed through the anodization process. The layer then acts as the dielectric. A cathode covers the oxide layer of the capacitor. Electrolytic capacitors have a higher capacitive voltage compared to the ceramic and film capacitors, making them useful for passing/bypassing low frequency signals and for decoupling and noise filtering. Electrolytic capacitors can come in metals such as aluminum, tantalum, and niobium. Table 22 below shows a comparison of different types of capacitors.

Type	Current Surge Rating	Used
Ceramic	Medium/High	X
Foil	High	X
Metalized	Low	
Electrolytic	Medium/High	X

Table 20: Capacitor Comparison

For this project, ceramic and electrolytic capacitors will most likely be used. After researching the various types of capacitors, these two seem to be the most practical for this project. The benefit to using the ceramic capacitor is the high stability. The electrolytic capacitors would be useful for their ability to decouple and filter noise.

### 3.7.3. Diodes

A diode is a semiconductor device that typically allows current to flow in one direction. Diodes have many applications, some of which are; logic gates, rectifiers, current steering, wave clipper, wave clamper, over voltage protection, and temperature measurements.

#### 3.7.3.1. Zener Diodes

Zener diodes are used because they can allow current to flow in the reverse direction. The current typically flow from the anode to the cathode. The reverse direction is permitted when the Zener voltage has been reached. Since Zener diodes can work in the break down region, it makes them useful to use for different applications.

Zener diodes are often used in many electrical devices. The advantages to using them in that they can stabilize power supplies and they can also provide a reference voltage for

many circuit designs. Another use for them is to protect the circuit from voltage spikes, such as electrostatic discharge.

### **3.7.3.2. Light Emitting Diodes**

The purpose of the light emitting diode is to emit light when it is activated. The applied voltage causes the electrons to combine with the holes with activates the energy, in this case photon. The energy from the photon determines the light that will be emitted. This is determined by the band gap of the semiconductor.

### **3.7.3.3. Rectifier Diode**

Rectifier diodes are used in rectifier circuits. Four diodes are used in a full bridge recruiter that is shown in the power supply section of this paper. The number of diodes used in a rectification design, determine the type of rectification the circuit will be performing. In applications rectification diodes are used to help adjust voltages according. Helping with these small adjustments allows the desired waveforms to be obtained.

### **3.7.3.4. Schottky Diode**

The Schottky diode is made out a semiconductor and metal junction. It has a very fast switching action due to its low voltage requirement. The disadvantage of using one of these diodes is the high leakage current. The high leakage current causes the temperature to rise, therefore leading to instability.

Schottky diodes are using for voltage clamping, they offer reverse current and discharge protection, and in switch mode power supplies. Out of the three, the switch mode power supply would be the most beneficial to this project. The low forward voltage of the diode, makes is beneficial to use in the power supply. Another benefit for this diode is that if it is put in series with a power source, it offers reverse polarity protection.

### **3.7.3.5. Tunnel Diode**

Tunnel diodes are known for their longevity and fast operations. They are used for frequency converters and detectors. Some other things they are used for are oscillators, amplifiers, and switching circuits.

### **3.7.3.6. Photodiode**

Photodiodes work when light is converted into current. The photodiode using the current gained form the movement of the photon. It can work in reverse biased and is useful in many electronic devices. The amount of measured light, allows them to figure out when they should and shouldn't work. Table 23 below compares the diodes that are likely to be used.

Type of Diode	Likely to be used
Zener	x
Light Emitting	x
Rectifier	x
Schottky	
Tunnel	
Photodiode	x

Table 21: Diode Comparison

After going through many of the diodes that are offered in today's world, it appears only three would make the cut for this project. The Zener diode is promising when it comes to its applications and functionality. The rectifier diodes can be used for the conversion of AC to DC. The circuit diagram in the power sections is a perfect example of the use of rectifier diodes. The photodiode may or may not be able to be beneficial to this project. Since it is used in many electronic devices, perhaps it can be used a sensor for part of the project. General and Zener diodes are probably going to be the main types of diodes that will be used.

### 3.8. Power Summary

Power is going to be a crucial part of this design, just like it would be for any other electronic design. Without the power, all functionality would be lost. To make sure every piece of this project receives the correct amount of power, the power design will be looked at carefully after all the pieces of the project are put together. As soon as everything is chosen after being tested, the power design will come in to play. As of now, it is possible to make a basic design, to turn make the project work. Calculating the exact power need will allow the project to work at maximum efficiency. The power overview provided insight in what could be done for the power for this project.

## 4. Standards and Design Constraints

The following standards apply to the audio circuits, data and communication systems and devices used for in the G12 PedalVision. The applicable standards for this product are as described in Table 24.

<b>Standard</b>	<b>Application</b>
<b>IEC 60958-1 Ed. 3.0 b:2008</b>	Digital audio interface- Audio, video and multimedia systems and equipment.
<b>BSR E1.50-201x</b>	Entertainment Technology - Safety Requirements for LED, Video & Display Systems
<b>IEC 60747-5-6 Ed. 1.0 b:2016</b>	Semiconductor devices - Part 5-6: Optoelectronic devices - Light emitting diodes.
<b>IEC 60268-9 Ed. 1.0 b:1977</b>	Sound system equipment. Part 9: Artificial reverberation, time delay and frequency shift equipment.
<b>IEC 60268-8 Ed. 1.0 b:1973</b>	Sound system equipment. Part 8: Automatic gain control devices - Audio, video and multimedia systems and equipment.
<b>IEEE 1149.1</b>	JTAG Standards. Circuitry built into an integrated circuit to assist testing and maintenance of PCB.

Table 22: Standards of G12 PedalVision

## 5. Project Hardware and Software Design

The following sections will discuss the specific design of each module of the G12 PedalVision.

### 5.1. Analog Design

The analog portions of the G12 PedalVision will be discussed in the sections to come. The design of the Input Buffer, Overdrive, Distortion, Equalization, and Compression units will be discussed. Although the overdrive and distortion units are similar in how they create their effect the design of each will be discussed separately, as they will have some variations from each other.

#### 5.1.1. Input Buffer Design

As discussed in previously the purpose of the input buffer is to ensure that the signal coming from the instrument is unaltered and passed onto the other effects as cleanly as possible. This circuit will also help to prevent pick up loading. For this input buffer we will be using the OPA1641 op amp as the heart of the input buffer. The input buffer will first have filter out any DC that may come from the instruments output. We need to make sure that we remove the potential DC as this shift in signal, whether that be positive or negative, will affect how the later circuits will manipulate the signal and this DC can damage the speaker that this product will interface with. This DC signal can come from active pickup instruments or other non-passive sources. In order to remove the potential DC we will place a capacitor, often called a coupling capacitor, before the input to the buffer. Since the primary function of the capacitor is to block DC its value does not matter too much. But if we place a resistor from the capacitor to ground we can filter out some of the very low frequencies we do not want in our tone by adjusting the resistor's value. The high pass input is shown in Figure 55 below. In order to ensure that the lower frequencies are not all removed we need to set our cut off frequency to be low. The cutoff frequency can be found by using the equation:

$$f = \frac{1}{2\pi * C1 * R1}$$

With this equation, knowing that DC is at 0 Hz we can set our cutoff frequency to be very low. This minimal filtering also helps us allow as many frequencies to pass above this cutoff frequency as possible. The reason this is desired is that since you do not want this circuit to color your tone, you want to ensure that it does not remove or amplify as many different frequencies as possible.

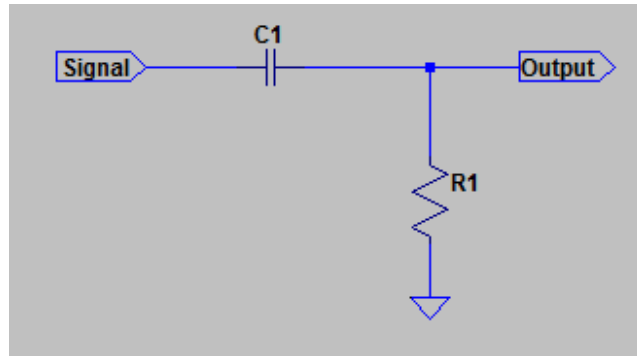


Figure 55: Coupling Capacitor and High Pass Filter

For this input buffer we connected this high pass filter to the positive terminal of our desired op amp. Since our op amp will be biased with 9v on the positive rail and grounded on the negative rail we need to bias the input to 4.5v. Although we initially did not want DC coming into the circuit, we want this 4.5v bias in order to allow our signal to swing without running out of head room. Since we need to bias the input of the op amp, the value of R1 needs to be a relatively high value in order to keep the current passing into the op amp as low as possible. Since the input to the op amp has a very high impedance the input impedance of our input buffer is just the value of R1 in parallel with the impedance of the op amp. But since the op amp's impedance is many orders of magnitude larger than the resistor, the equivalent input impedance is the value of R1.

Since this is an input buffer we do not want to add any gain, so we will connect the output of the op amp to the negative input of the op amp. This will create a unity gain buffer. By doing so, whatever appears at the input of the op amp will appear at the output. Finally, as seen in Figure 56, with the high pass filter connected to the unity gain buffer we will be able to pass the incoming signal cleanly to the output of the G12 PedalVision or to the input of the other effects. The capacitor at the output serves to remove the DC bias voltage that is added to the signal before passing through the unity gain buffer.



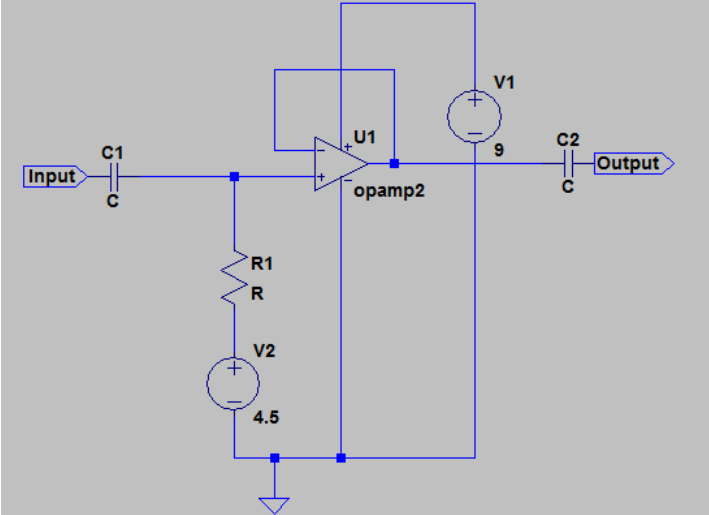


Figure 56: Complete Input Buffer Circuit

### 5.1.2. Overdrive Design

For the overdrive circuit we start with a noninverting amplifier as the base. For this overdrive circuit we will be using the OPA1641 op amp as the heart of it. For a non-inverting op amp the gain is equal to:  $G = 1 + \frac{R4}{R3}$ , but since this amplifier will be the base of the soft clipping circuit. In order to get the traditional “smooth” sound this type of effect has, and to prevent the aural perception of over amplification of the low notes played we will use a high pass shelving amplifier as shown in Figure 57. The gain of this amplifier is:

$$G = 1 + \frac{R4}{R3 + \frac{1}{C1 * S}}$$

As you can see the gain of this amplifier is now dependent on frequency, which is what we want in order to remove some of the “boom” that the lower frequency notes will have.

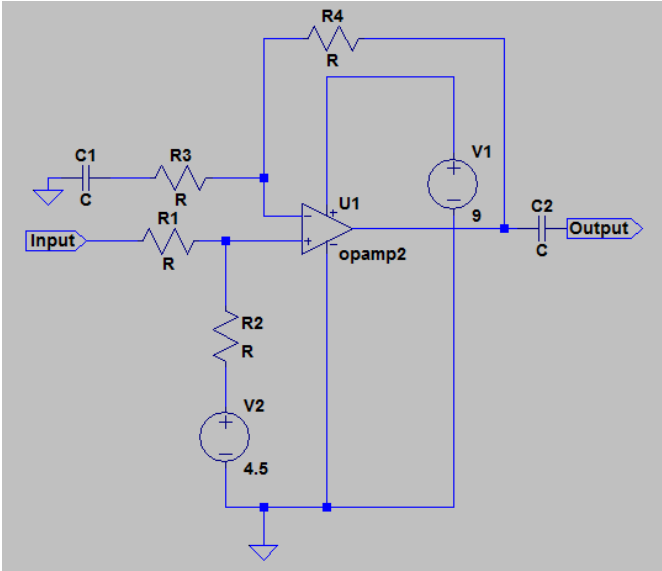


Figure 57: Basic High Pass Shelving Amplifier

In order to create the soft clipping which will create the overdriven sound we will place back to back diodes in the feedback loop of the amplifier. Another thing to think about is whether we want symmetric or asymmetric clipping, since we want versatility with this pedal we will place two similar diodes back to back in the feedback loop and a switch to toggle between the two configurations. Then we will place either two different back to back diodes in the feedback loop, or an uneven number of diodes in the feedback loop such as two diodes from the negative terminal to the output and one diode from the output to the negative terminal. Figure 58 below shows the configuration of the diodes in the high pass shelving amplifier. [17]

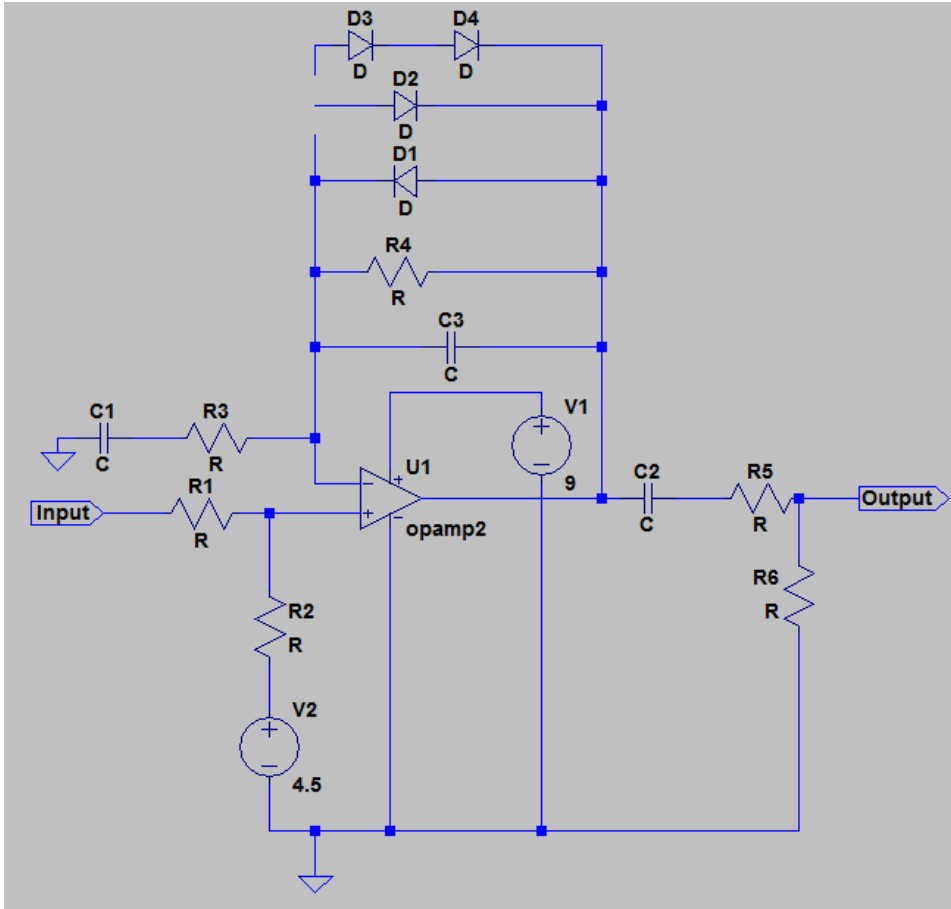


Figure 58: Overdrive Circuit with Switchable Diode Configuration

In order to remove the harsh higher frequencies we can add a capacitor from the negative terminal of the op amp to the output of the op amp. This allows high frequencies to pass through the amplifier without any amplification added to those high frequencies. By adding this capacitor we create a band pass type of filter due to the two capacitors in the circuit. Finally we add a capacitor followed by a voltage divider created by two resistors at the output. The capacitor C2 is there to filter out the DC voltage added to the signal due to the biasing before amplification. Then the two resistors, R5 and R6 are there to decrease the amplitude of the signal at the output. This helps with adjusting the volume of this specific effect and to help prevent clipping of the signal in later circuits if further

amplification is desired but the sound quality of this effect is desirable as well. The voltage of this voltage divider is proportional to the value of the two resistor values.

### 5.1.3. Distortion Design

Like the overdrive circuit the base of the distortion circuit will be a high pass shelving amplifier. For this distortion circuit we will be using the OPA1642 op amp as the base for its amplification stages. For the distortion circuit we start with a noninverting amplifier as the base. For a non-inverting op amp the gain is equal to:  $G = 1 + \frac{R4}{R3}$ , but since this amplifier will be the base of the soft clipping circuit. In order to prevent the aural perception of over amplification of the low notes played we will use a high pass shelving amplifier as shown in Figure 57. The gain of this amplifier is:

$$G = 1 + \frac{R4}{R3 + \frac{1}{C1 * S}}$$

The resistor numbers and capacitor numbers follow the reference designator of Figure 57 above. As you can see the gain of this amplifier is now dependent on frequency, which is what we want in order to remove some of the bottom end “muddiness” that the lower frequency notes will have. We will also place a capacitor in the feedback loop of the high pass shelving amplifier in order to remove the harsher higher frequencies from being amplified. We may need to further filter the higher frequencies later on but we will begin by only filtering the high frequencies during amplification.

In order to create the hard clipping which will create the distorted sound we will place back to back diodes from the output of the amplifier to the ground. But since we have biased the signal we need to remove the bias DC voltage before we try to clip the signal with the diodes, since without the removal of the bias the wave form will clip on one side but not on the other side of the wave form. We can remove the DC by placing a capacitor between the output of the amplifier and the diodes. This removal of the DC bias voltage will allow the distortion circuit to pass on a pure AC signal without any DC that could harm the other components of the following circuits. Another thing to think about is whether we want symmetric or asymmetric clipping, since we want versatility with this pedal we will place two diode configurations back to back to ground and a switch to toggle between the two configurations. First we will place two of the same diodes back to back to ground to create the symmetrical clipping, and then we will place either two different back to back diodes to ground, or an uneven number of diodes to ground to create the asymmetrical clipping. Figure 59 below shows the configuration of the diodes in the high pass shelving amplifier with the coupling capacitor as well. After the diode configurations there is a voltage divider created by a potentiometer at the output of the circuit in order to allow for volume adjustment of the distortion effect and to help prevent clipping of the signal in later circuits if further amplification is desired but the sound quality of this effect is desirable as well.

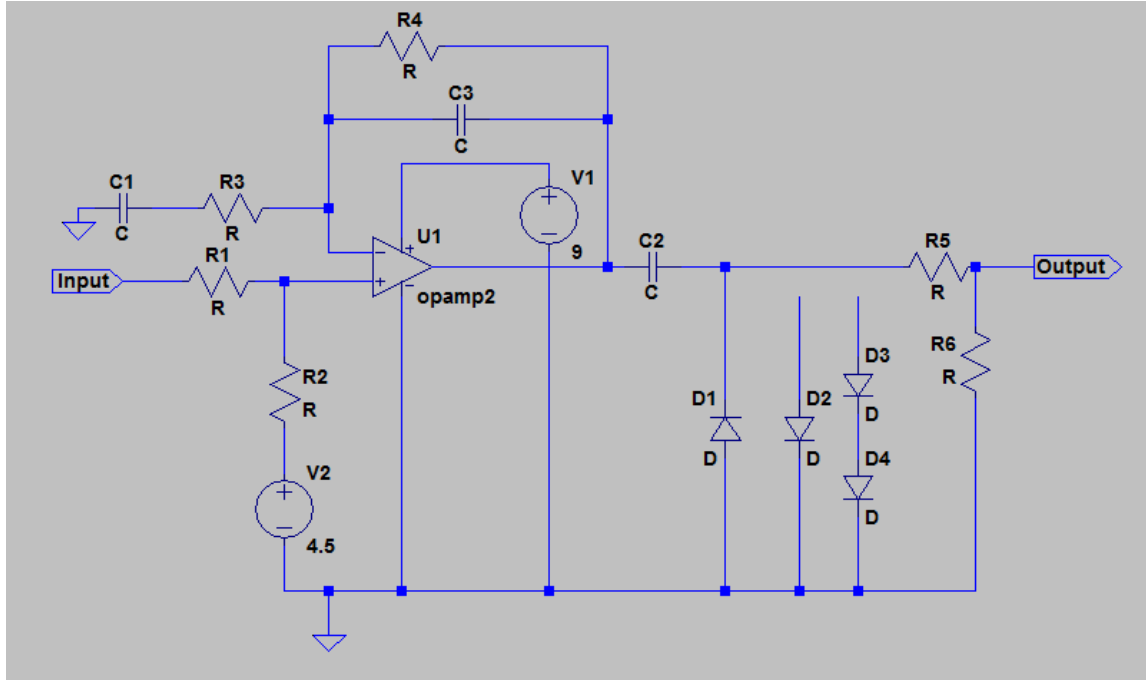


Figure 59: Main Distortion Circuit with Switchable Diode Configuration

Although this circuit will clip the signal well, if we add an additional amplification stage before the circuit shown above we can further clip the signal and get a more square like output signal. To accomplish this pre amplification we will add another high pass shelving amplifier to the input of the main distortion clipping circuit. The pre-amplification circuit will also have a capacitor in the op amp's feedback loop in order to remove the higher frequencies from getting amplified as well. The reason for removing a lot of the very high frequency components is that once these high frequencies are distorted they have a very harsh characteristic to them and are very unpleasant to listen to. Figure 60 below shows the complete circuit we will be using to implement the distortion effect.

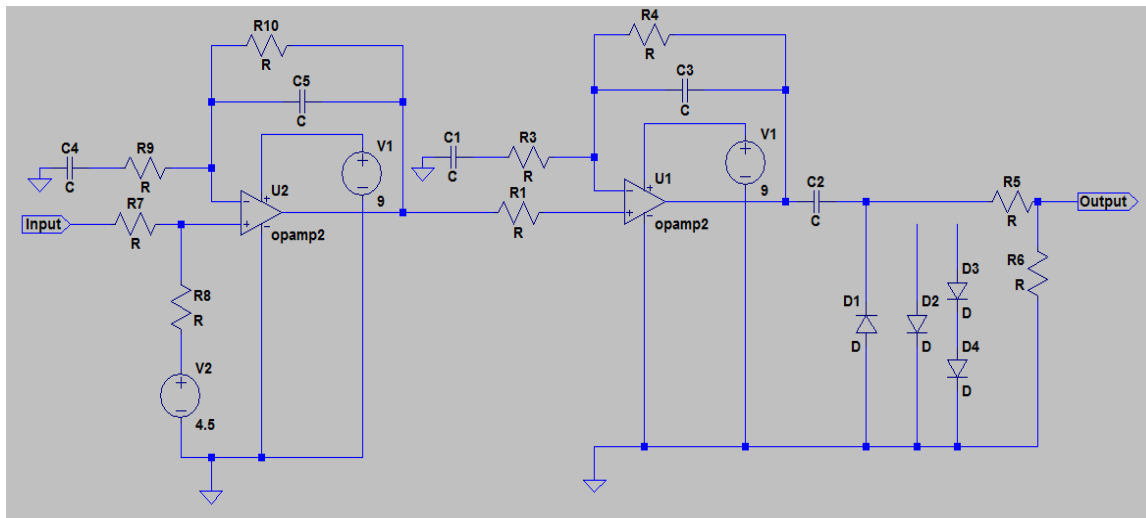


Figure 60: Complete Distortion Circuit

### 5.1.4. Compression Design

The design of the compressor utilizes a transconductance amplifier in order to fully amplify signals with small amplitudes and to attenuate or pass signals with large amplitudes. As more current flows into the input bias pin of the amplifier the more amplification occurs, while the less current flowing into the input bias pin of the amplifier the less amplification occurs. The first thing this circuit will do it to take the input signal and apply the original signal to the one of the two input terminals of the amplifier. Then the same input voltage will pass through a voltage divider to drop the same signal to the other terminal. This input configuration can be seen in Figure 61 below. This will cause the transconductance amplifier to always amplify the input signal.

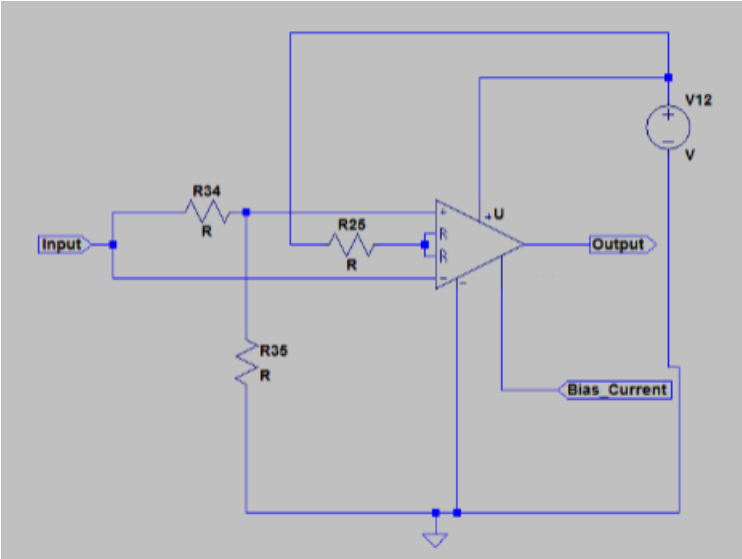


Figure 61: Compression Input

The way the amplifiers gain will be controlled is through the bias current being fed back into the transconductance amplifier. This can be accomplished through the implementation of a voltage controlled current source. This voltage controlled current source will also have a resistor that will allow for adjusting the ratio of the current being output with respect to the voltage being fed into the current source. The voltage controlled current source can be seen in Figure 62 below.

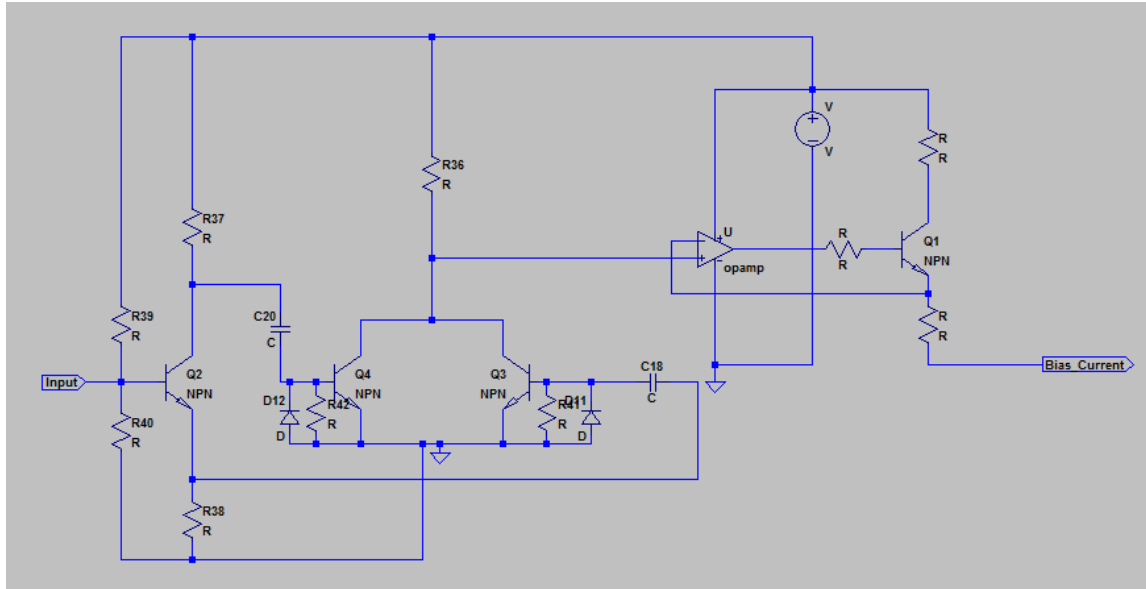


Figure 62: Voltage Controlled Current Source

In order to compress the signal we need to create a way to control the voltage of the voltage controlled current source. To accomplish this we need to sample the output of the amplifier and manipulate the output signal in order to turn it into a usable signal to control the input bias voltage controlled current source. In order to ensure that both the upper and lower halves of the input wave form are compressed we need to devise a way to allow for both cycles to affect the bias current produced by the voltage controlled current source. The best way to do this would be to pass the signal to an inverter so that the negative cycle of the input signal will now become the positive cycle at the inverter output. [23]

In order to create a smooth sound between the compressed output and the original signal we need to include a wet to dry mix. This can be done with a resistor between the output of the transconductance amplifier and the input signal. By placing this resistor between the input and the output we can determine how much of the original input versus how much of the amplifier output will appear at the output of the circuit. This mix will then be sent to an output buffer to then interface with the next effect in the signal chain. This can be seen in the full compression circuit in Figure 63 below.

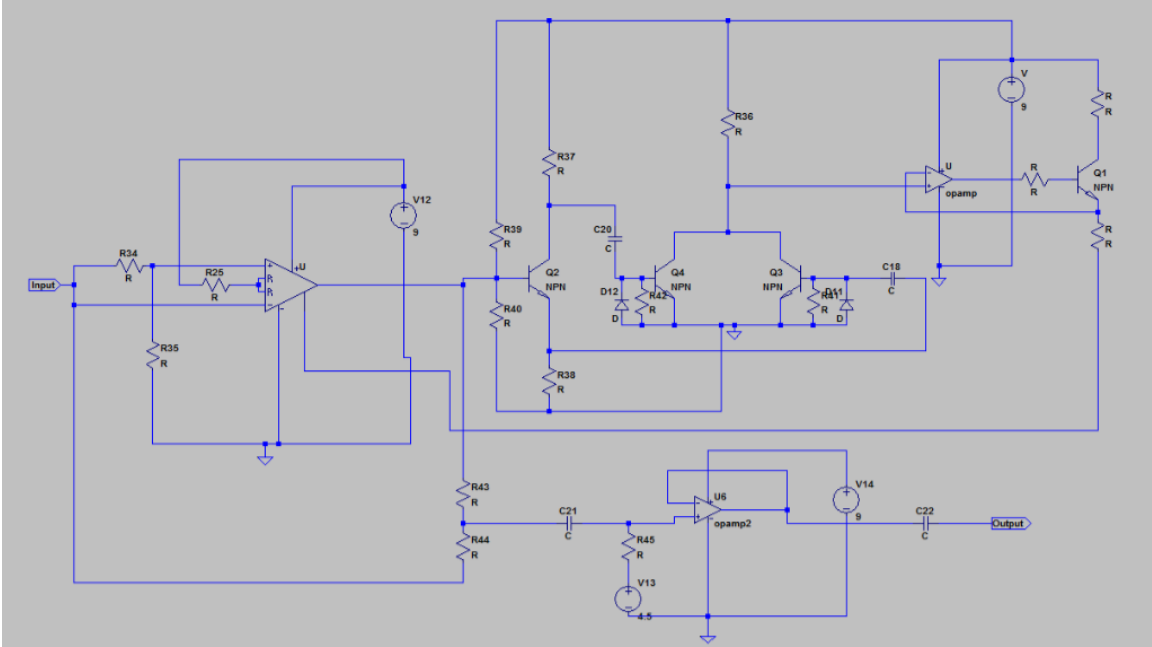


Figure 63: Complete Compression Circuit

### 5.1.5. Equalization Design

The design of the equalization will consist of a network of different types of filters. The filters will be arranged so that the user can toggle between the different filters and adjust the cutoff frequencies of each filter. The user will also have the option to bypass the filters and send the input signal directly to the output if they do not want any filtering done to their signal.

The first two filters we will have are the high pass filter and the low pass filter. The high pass filter is exactly same as the filter we use to couple our circuits together, it consists of a capacitors and a resistor. This filter works by allowing high frequencies to pass through the capacitor and lower frequencies are blocked. The low pass filter on the other hand works in the exact opposite way by swapping the position of the resistor and the capacitor in the circuit. The capacitor blocks low frequencies from reaching ground while the higher frequencies are allowed to pass to ground. The cutoff frequency is determined by the equation:  $f = \frac{1}{2\pi * C * R}$ . For a high pass filter all frequencies below this cutoff point are attenuated. While for a low pass filter, all frequencies above this cutoff point are attenuated. Figure 64 below shows the design of the low pass and high pass filter for this project.

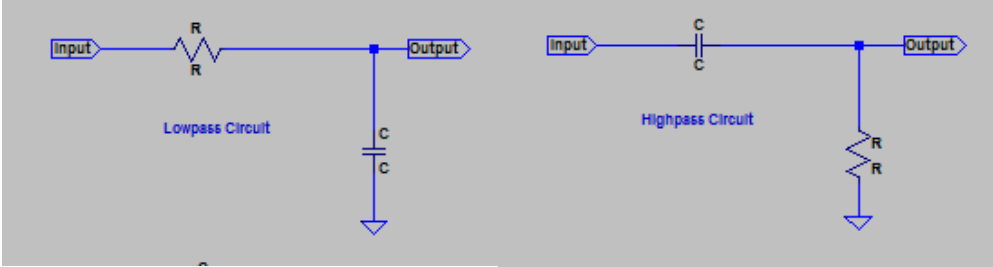


Figure 64: High pass and Low pass filters

The third option the user will have for their equalization will be a “mid-scoop” or band reject filter. This filter works by allowing the two ends of the frequency spectrum to pass but a certain range of frequencies in between the ends will be attenuated.

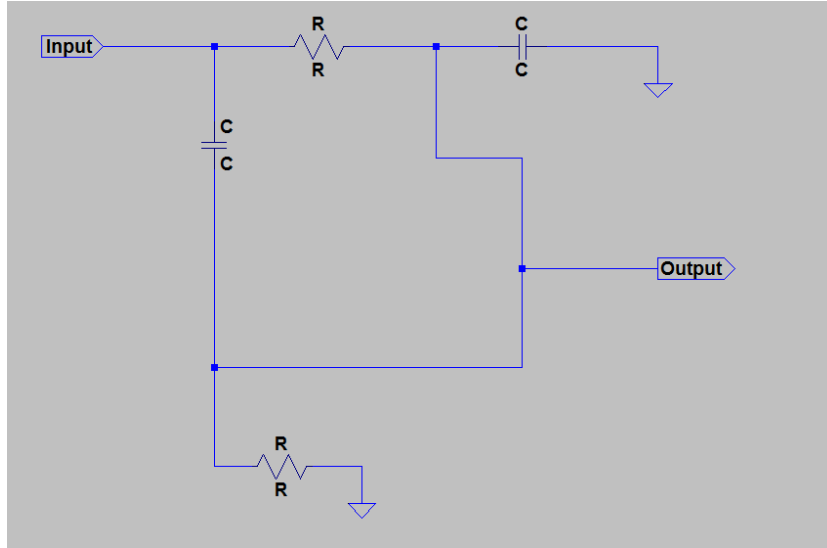


Figure 65: Mid-Scoop Circuit

The mid-scoop filter circuit in Figure 65 above works by taking the output of the passive low pass and passive high pass filter and combining the output of the two filters at one output. The filter will be adjusted by changing the resistance value of the high pass filter and the low pass filter. The depth, you could say, of this rejection is determined by how close or how far away the cut off frequencies of the high pass and low pass filter are from each other. The further away the cutoff frequencies are from each other, the deeper or more attenuated the middle frequencies will be. While the closer the cutoff frequencies, the less attenuation will occur at the middle frequencies. Although this could have been implemented by having a fixed cutoff frequency for either filter and placing a potentiometer between the two outputs, this reduces the flexibility of the circuit since you cannot shift the frequency range of attenuation or determine the depth of attenuation or lack thereof.

The final filter that the user will be able to use is a “mid-boost” or a band pass filter. The bass filter works by passing the frequencies in a specified range and attenuating the frequencies outside of that predetermined range. The design of this “mid-boost” circuit will consist of a low pass filter at the very beginning. The cut off frequency will be determined by the values of R and C as explained before. Since this low pass filter will then be connected to a unity gain buffer op amp circuit, we will need to bias the input signal after the low pass filter in order to allow the signal more headroom in the unity gain buffer circuit. Once the signal passes the unity gain buffer it is fed to a second op amp circuit that will amplify the filtered signal. Once the signal has been amplified by the second op amp circuit we can start to filter the higher frequencies in order to create a “hump” in the middle frequencies. The value of the cutoff frequency for the high pass filter is set by the values of R and C much like the low pass filter at the beginning of the



circuit. The purpose of placing the high pass filter at the end of the circuit is that this will also remove the DC bias applied to the signal at the beginning of the filter circuit. The bandwidth of the filter is determined by the cutoff frequencies of the two filters that make up this circuit. So if the two cutoff frequencies are far from each other, then the bandwidth will be pretty wide. If the cutoff frequencies are close to each other instead, then the bandwidth will be relatively small. This adjustment of the cut off frequency also allows the peak of the filter to be shifted to higher or lower frequencies depending on the values of the two cut off frequencies. Figure 66 below shows the complete band pass filter circuit.

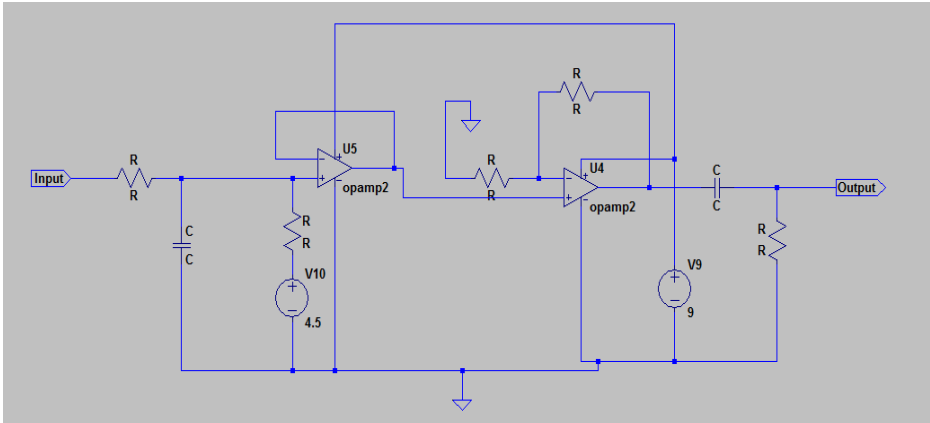


Figure 66: Mid-Boost Circuit

The final equalization circuit will allow for the user to decide whether they want to use a high pass, low pass, band reject, or band pass filter by flipping the switches available to them on the circuit. The user will also be able to bypass the equalization filter altogether if they wish to apply to filtration to their sound. Figure 67 shows the full equalization circuit and the switching options the user has at their disposal. [21]

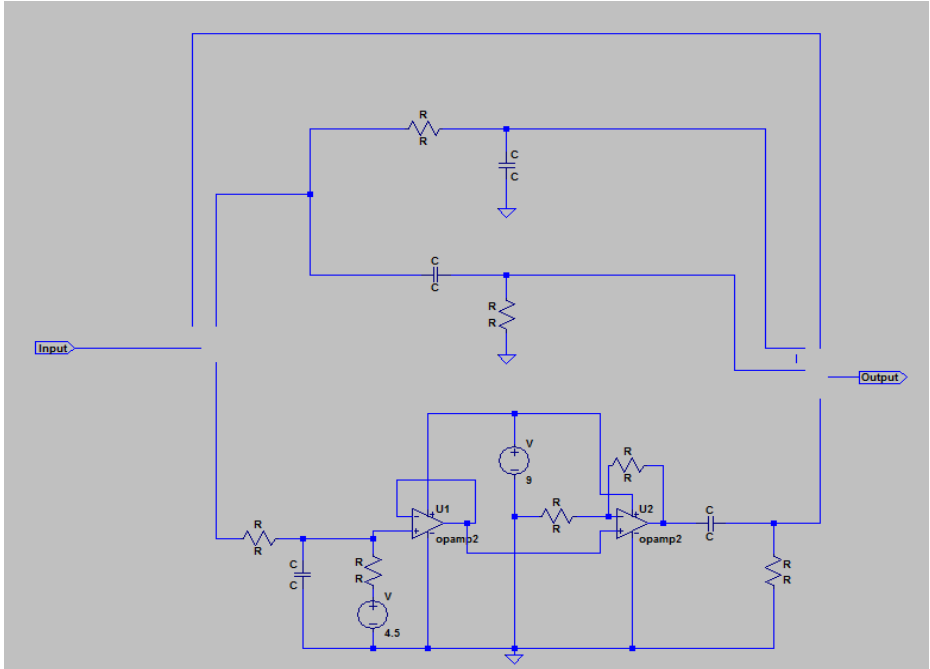


Figure 67: Full Equalization Circuit

### 5.1.6. External Effects Interface

Although the user will be able to accomplish many different effects with the G12 PedalVision’s analog and digital effects, they may want to add one of their own effects that they love or think would work well with this product. In order to for this ability to expand on this product there will be a capability to allow the user to decide if the output put of the analog effects will go directly to the next stages of the G12 PedalVision (such as the digital effects, LED module, or the output of the entire system) or to an external effects unit, then back into the G12 Pedal Vision for further processing or direct to the output. Since all of the DC biasing added to the signal within the G12 PedalVision will be removed before proceeding to the next stages, this added feature will not damage any external effects. In order to ensure a good quality sound a second input buffer will be used. The function of this second input buffer will work much like the input buffer at the very beginning of the G12 PedalVision’s signal chain. This will provide a high input impedance and a low output impedance. This will help with any potential “tone sucking” some other pedals on the markets may exhibit on the user’s tone quality. In order to remove any potential DC from the external effect unit, the input will have a coupling capacitor to remove any potential DC at the input. The output of the input buffer will have a capacitor as well in order to remove the DC bias that is added to the unity gain input buffer to allow for enough headroom on the upper and lower swing of the signal. Figure 68 illustrates the circuit for this feature’s implementation.

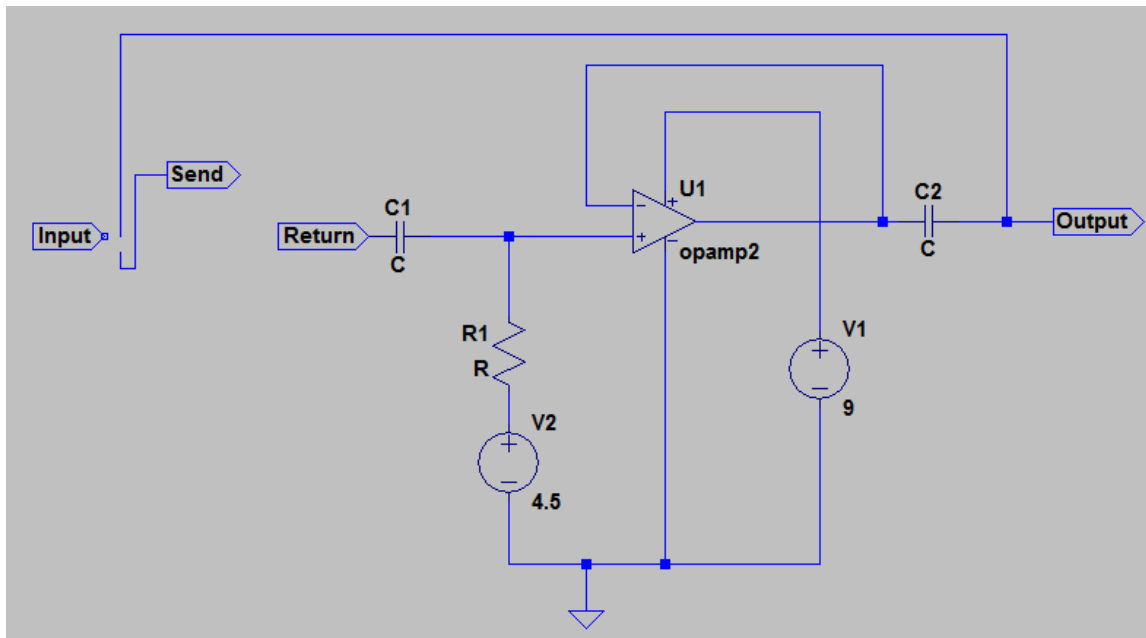


Figure 68: External Unit Interface

### 5.1.7. Toggling Mechanism

In order to activate each effect, we need to devise a way to connect the input signal to the circuit or bypass the circuit and move directly to the output. Since user may be playing his or her instrument while using the G12 PedalVision, they cannot toggle the desired effect on or off by hand. Due to this issue, all effects units have a foot switch to toggle

their effects on and off. Many large companies use switching circuits that take advantage of a momentary switch to toggle the desired effect as seen in Figure 69 from [electrosmash.com](http://electrosmash.com).

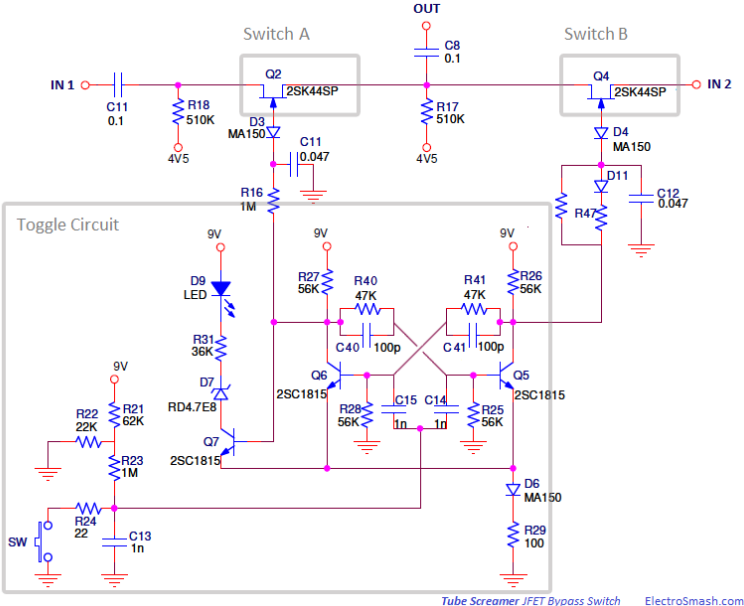


Figure 69: Switching Circuit (Courtesy of Electrosmash.com) [17]

The problem with these circuits though, is that since the input signal needs to run through the switching circuit regardless of if the effect is on or off. This would cause loss of tone depending on what the else was being interfaced with this unit. Due to this undesirable tone loss, some companies turned to 3PDT latching switched. The advantage of these foot switches is that they allow for something called “true bypass.” True bypass is the term many musicians use to describe the input to an effects unit being routed directly to the output without passing through any part of the effects unit’s circuitry. The disadvantage of the 3PDT switches is that they are very labor intensive since they require the leads to be soldered by hand as opposed to the switching circuits which can be done automatically during the manufacturing process. The other disadvantage of the 3PDT is the audible “click” heard when the effect is toggle “on.” While some may complain about the “click” it is not as audible when the user is playing through a song and distortion type effects are being used. Since the goal of this G12 PedalVision is to provide great tone, we will be using 3PDT footswitches for all analog effects. [17]

### 5.1.8. Analog Effects Order

One of the things that may seem unimportant is the order that the effects are placed. Although it seem like a superfluous task, it plays a major role in the tone of the musicians sound. The order we will implement the analog effects is as follows:

- Input Buffer
- Compression
- Distortion

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- Overdrive
- Equalization
- External Effects

The reasons for selecting this order are quite simple. The input buffer will be the very first thing in the signal chain of the analog effects due to the fact that it is supposed to prevent loading of the signal and pass the most accurate input to the rest of the system. The next effect in the signal chain would be the compression circuit. Compression would follow the input buffer since it will help to smooth out the input signal before more effects are applied to it. It is preferred to apply compression before any other effects because the larger the signal, or the harder the compressor is driven, the output signal will become smaller and smaller. This smaller signal will create the appearance of having a low volume, but once the compression effect is bypassed the apparent volume of the signal will be much higher. This can cause damage to the other pieces of equipment later down the signal chain. With this in mind that is why we are placing the compressor immediately after the input buffer. By doing this we do not get the apparent low output volume or the jump in apparent volume when the effect is bypassed.

Following the compression circuit we will have the distortion effect. By placing the distortion effect after the compression circuit we can feed the distortion unit a smooth signal that the circuit can easily manipulate without being driven too hard. This creates a smooth distorted sound, as opposed to a harsher tone due to the compressor evening out the amplitude before the distortion amplification stage.

Overdrive will follow distortion in the signal chain for the reason that distortion causes some compression. Since the distortion circuit's output is the forward voltage of the two diodes shunt to ground, the output will never really exceed that amplitude. By placing the overdrive after distortion, we are able to increase the apparent volume of the distorted signal by passing it through the gain stage of the overdrive circuit. On top of adding volume to the signal we will also be soft clipping the signal as well.

After the overdrive circuit we will have the equalization filter network. The equalization will be placed at this point in order to shape our signal's frequency components now that different harmonic components have been added to it through the distortion and overdrive circuits. This will allow for the user to shape their final tone before the signal is then fed to the rest of the G12 PedalVision.

The final circuit of the analog effects unit will be the external effects interface. This will be the last circuit on the analog signal chain because if the user wants to have the characteristic sound of the G12 PedalVision's analog units and then use a third party pedal before the Digital effects unit of this product, this "send and return" circuit will allow for easy interfacing with outside effects units.

The reason for the analog effects being placed before the digital effects is due to the fact that analog effects manipulate the signal by nonlinear means, such as integration and derivation. While the digital effects manipulate the signal through linear operations such

as addition, subtraction, multiplication, and division. Since the digital effects need to use ADC and DAC to manipulate and then reconstruct the signal, you do not get the exact same signal you input into the system. By doing the analog effects first you can get the exact signal at the output so you get the most true to the desired signal at the output. If the digital effects were done before the analog effects, the approximation that the digital system would apply to the output signal would create unwanted modulation artifacts once the signal passes through all of the analog effect units.

## 5.2. Digital Design

The design of the digital effects module is determined by the connections of each of the chips and switches, as well as the design of the software. The hardware design will deal with how the signal is connected into the chosen ADC/DAC pins, and get passed to the DSP chip. The software designs include pseudocode for creating the structure of the digital effects. The designs discussed in this section are not finalized by any means. The testing of the components will determine if the design is correct. The chosen components are currently being shipped and are not expected to get here before the draft deadline.

### 5.2.1. Hardware Design

The hardware design for the digital module is one of the most important parts of the entire system. Without the connections, the module will be unable to create the digital effects we are looking for. In figure 70 below are the pin connections for the DSP chip. When connecting to the DSP chip it must be made certain that all connections are correct, as well as no pins get soldered together. If a mistake like this occurs it may result in errors that may be hard to debug if the source is hard to find. To avoid this it is most probable that we will use a third party source to solder our small chips onto the PCB professionally. The pin descriptions and images can be found in Texas Instruments literature number SPRS370E.

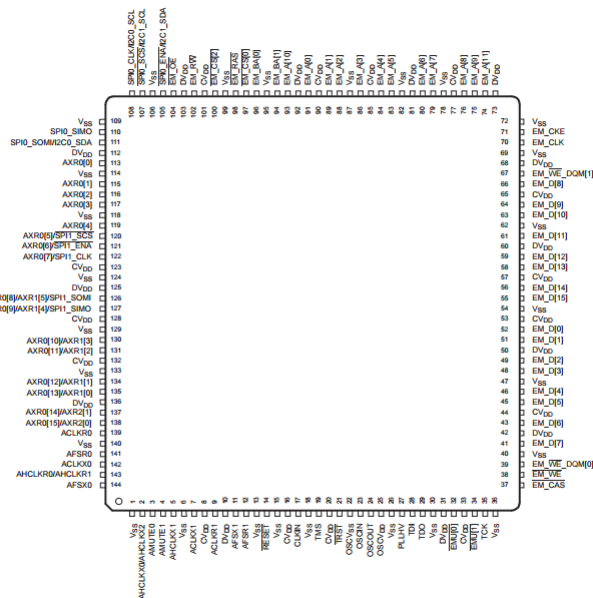


Figure 70: TMS320C6720 Pin Connections (Courtesy of Texas Instruments)

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G12 PedalVision

In order to flash the DSP chip to be programmed to the desired internal code, the JTAG pins will be used. These pin numbers are listed below in table 23. These pins are connected to the

Signal Name	Corresponding Pin Connections	Description
<b>TCK</b>	35	Test Clock
<b>TMS</b>	19	Test Mode Select
<b>TDI</b>	28	Test Data In
<b>TDO</b>	29	Test Data Out
<b>TRST</b>	21	Test Reset
<b>EMU[0]</b>	32	Emulation Pin 0
<b>EMU[1]</b>	34	Emulation Pin 1

Table 23: JTAG Pins (Courtesy of Texas Instruments)

If external memory is needed for the program that is running on the DSP chip, the External memory interface (EMIF) is needed. A memory device can be connected to the pins listed below in table 24. When these connections are made, the memory peripheral is able to be used as an extension to the internal memory. It creates a single memory space to be used.

Signal Name	Description
<b>EM_A[0-11]</b>	Address Bus
<b>EM_BA[0-1]</b>	SDRAM Bank address
<b>EM_CS[0-2]</b>	SDRAM Chip Select
<b>EM_CAS</b>	SDRAM Column Address Strobe
<b>EM_RAS</b>	SDRAM Row Address Strobe
<b>EM_WE</b>	SDRAM/Asynchronous Write Enable
<b>EM_CKE</b>	SDRAM Clock enable
<b>EM_CLK</b>	EMIF output clock
<b>EM_WE_DQM[0-3]</b>	Write Enable or Byte Enable
<b>EM_OE</b>	Output Enable
<b>EM_RW</b>	Mem Read/not Write
<b>EM_WAIT</b>	Wait Input
<b>EM_D[0-15]</b>	EMIF Data Bus

Table 24: EMIF Pins (Courtesy of Texas Instruments)

The pins listed in table 25 are used for the Inter-Integrated Circuit (I2C). These pins will be connected to the I2C bus which will allow for easy communication between the chip and a microcontroller. Since there are two serial busses that can be used, we will choose the one we will use when the building of the product starts. The data that is sent between master and slave is sent over the Serial Data pins. This data that is being sent will control the values for the different settings of each digital effect.

Signal Name	Corresponding Pin Connections	Description
<b>SPI0_SOMI/I2C0_SDA</b>	111	I2C0 Serial Data
<b>SPI0_SOMI/I2C0_SCL</b>	108	I2C0 Serial Clock
<b>SPI0_SCS/I2C1_SCL</b>	107	I2C1 Serial Data
<b>SPI0_SCS/I2C1_SDA</b>	105	I2C1 Serial Clock

Table 25: I2C Pins (Courtesy of Texas Instruments)

The next hardware component that must be designed is the ADC/DAC chip. Since the PCM3060 is currently chosen for the chip that we will be using, figure 71 shows the pin assignments for it. The pin diagrams and descriptions can be found in Texas Instruments Literature number SLAS533B.

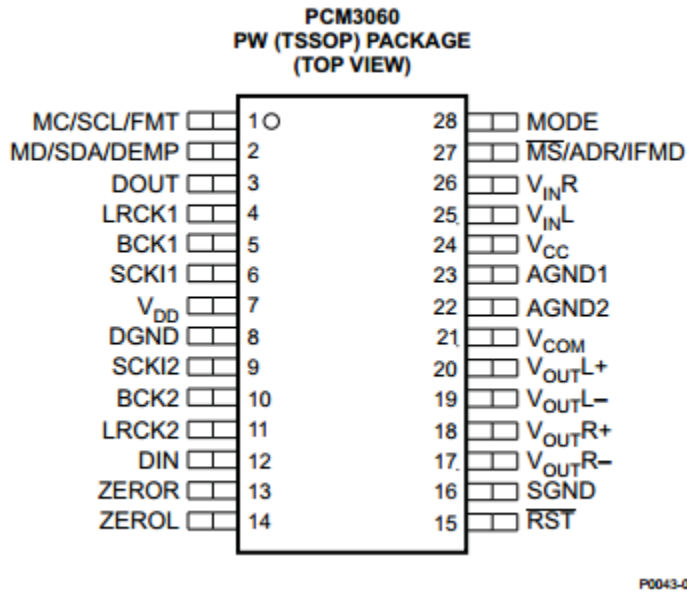


Figure 71: PCM3060 Pin Assignment (Courtesy of Texas Instruments)

The main pins that need to be discussed for the hardware design of the system are listed below in table 26. These include the input and output pins for both the analog and digital signals. Since this is a two in one chip, it will be simple to run the connections between this and the DSP inputs. An analog signal will come in through the Vin pins, and get converted to a digital signal. The digital signal will come out of DOUT, and go into the DSP chip. Once the modification is completed, the signal will come back to DIN, where it is then converted to an analog output.

Signal Name	Corresponding Pin Connections	Description
<b>AGND1</b>	23	Analog Ground 1
<b>AGND2</b>	22	Analog Ground 2
<b>DGND</b>	8	Digital Ground
<b>DIN</b>	12	Digital input
<b>DOUT</b>	3	Digital Output
<b>VinL</b>	25	Left Channel Analog input
<b>VinR</b>	26	Right Channel Analog Input
<b>VoutL-,+</b>	19,20	Left Channel Analog Outputs
<b>VoutR-,+</b>	17,18	Right Channel Analog Outputs

Table 26: PCM3060 Pins (Courtesy of Texas Instruments)

A block diagram of the digital signal modification module can be found in figure 7. The pins that were just described are the detailed design of the input and outputs of the system.

## 5.2.2. Software Design

The software design is the heart of the digital effects module. Without the software, there would not be any signal processing taking place. In order to create the effects the main process that needs to be designed is the delay line. All of the digital effects that are currently being created with the G12 PedalVision are created using the delay line. There are multiple ways of designing a delay line, and different languages can be used. For our purposes it is most likely that the C programming language will be used. In figure 72 below is an example of how the delay line code is designed. This code will be used as a starting point when designing our own code for the effects.

```
static double D[M]; /* initialized to zero */
static long ptr=0; /* read-write offset */

double delayline(double x)
{
    double y = D[ptr]; /* read operation */
    D[ptr++] = x;      /* write operation */
    if (ptr >= M) { ptr -= M; } /* wrap ptr */
    return y;
}
```

Figure 72: Delay Line Code [24]

When creating the delay/echo effect, this delay line is the only function that is really needed. To get the echo effect, the same value is passed in to the delay line multiple times with varying delay times and amplitude. By using this code, the effect will be able to simply call the delay function multiple times. The output of the function will be mixed with the original to create a realistic effect.



For the reverb effect, this delay line will be used to create a comb filter. The comb filter is shown above in Figure 14 as a part of the reverb block diagram. It consists of using the delay line, followed by multiplying by a gain value, and then mixed with the input signal and sent back through the delay line, as well as mixed with the output and the original input. Once again, the delay line is the meat of the comb filter. So the code above is the bulk of the design for the reverb effect as well. Also to create a more advanced reverb, a tapped delay line can be used. To code a tapped delay line, the code above can be modified to allow an output to be sent at random spots in during the delay.

The flanger effect is the different effect. The way this effect is created is by using a feedforward comb filter. This takes the original signal, and mixes it with a delayed signal that is multiplied by the gain. The delay is where the modifications need to be made. Since the flanger effect uses the delay line with a modulated delay time, it is necessary to make sure that the code can create the function of the delay. The modulation is created using the formula stated in Section 3.2.4.

The code for creating the effects is important to the actual output sound of the device, however, in order to allow users to change settings the communication software needs to be designed. To design this code, a software library for I2C communication must be used to communicate over the bus. A command will be sent to the slave from the master and write data into the specified address. When reading a message sent, the slave will read the message until the message is completed. The design of our I2C communication network must also allow communication between the microcontroller and the LCD display and each separate DSP chip. This design is show in the previous section for the hardware design. The communication to the DSP chip will allow the microcontroller to send the values that are set in the interface by the users to a specific register in the chip. This register will be read by the DSP chip to be used for setting specific values. This method of communication is simple, since the main commands are well documented, and easy to use. In order to make any i2c commands, the function call will begin with the start command. This will start the communication. The next is setting up the address for the slave device that is getting written to. After the slave device is set up to be sent to, the register address that is being written to needs to be set. After the data knows where to go, the command can be sent. Once data has finished sending, the stop command is sent. To read from the register, the address to read from must be set. Once this is set, read commands can be run until it gets passed in a 0. If multiple bytes need to be read in, then all except the last byte to be read should pass in a 1 on the read command. Once reading is complete, the stop command is run. [7]

## **5.3. LED Design**

### **5.3.1. Potential Programming Issues**

Before even writing a line of code towards achieving this goal there are some foreseeable issues or barriers that we will have to overcome to complete the task at hand. These range from a compatibility issue between hardware, to wants versus needs from the

programmer, or even differing of opinion on how to achieve a goal. We will confront and address these issues below with how we believe we will overcome them.

Whether loading the Arduino platform onto the ATmega2561 is best. The Arduino environment is one that we are very familiar with since we have used it many times to do independent projects, but loading it onto the ATmega2561 will call for a boot loader that we normally would not have to use. It may also cause the system to run at a slightly less efficient rate since with ease of use comes the costs of execution per instruction. Using the Arduino environment may also speed up programming time since we will not have to create our own libraries for bit heavy operations such as serial communication.

The registers for the TLC 5955 are each over 700-bits wide, we are concerned on how we will create a seamless, over 3000-bit wide, value to transfer into the four TLC 5955 connected in series. While it concerns us, we understand this is not an impossible feat as entire gigabyte transfers happen with 32-bit processors. It is unfamiliar territory but we are sure we can accomplish it.

We are unsure how we will represent the array of LEDs in the program. Some ideas on how to represent them is to create a two dimensional array in which each value works as each physical LED, and when it is time to update them you can easily update that array, and when it is time to update the TLC 5955 the software will iterate over the array combining them into one long strand to transfer out to the TLC 5955s in series.

### 5.3.1.1. Code Control Flow

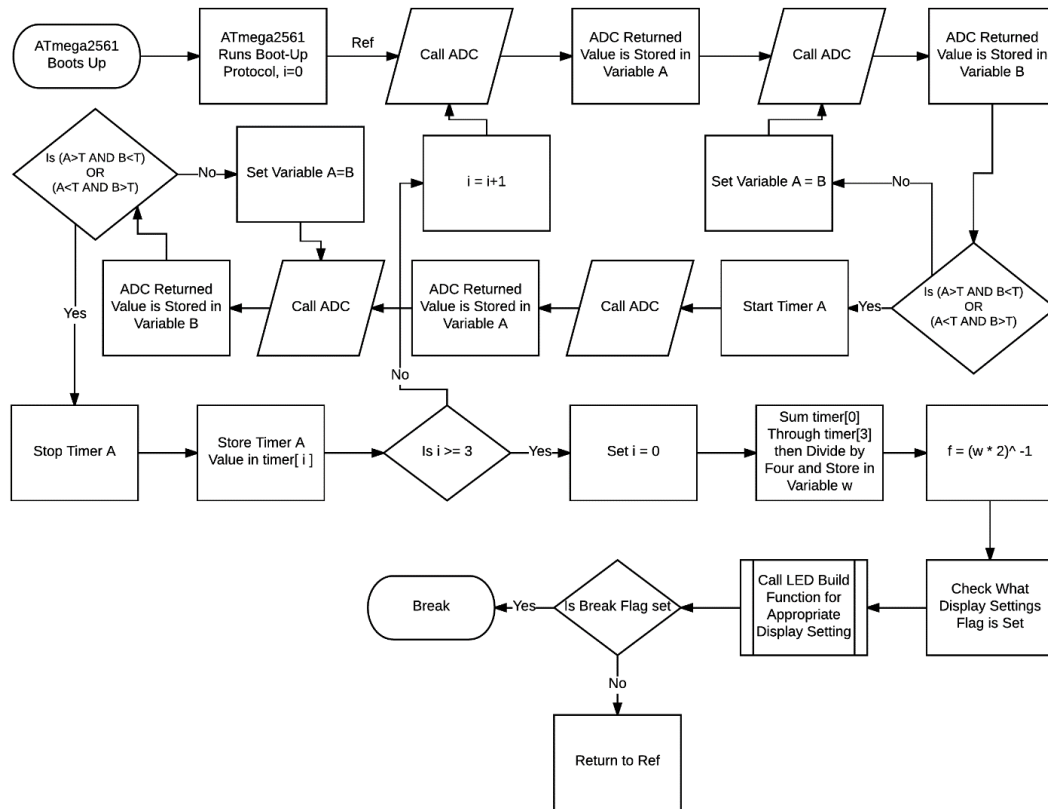


Table 27: Main Program Loop Running

### LED Matrix Display Selection Flow Chart

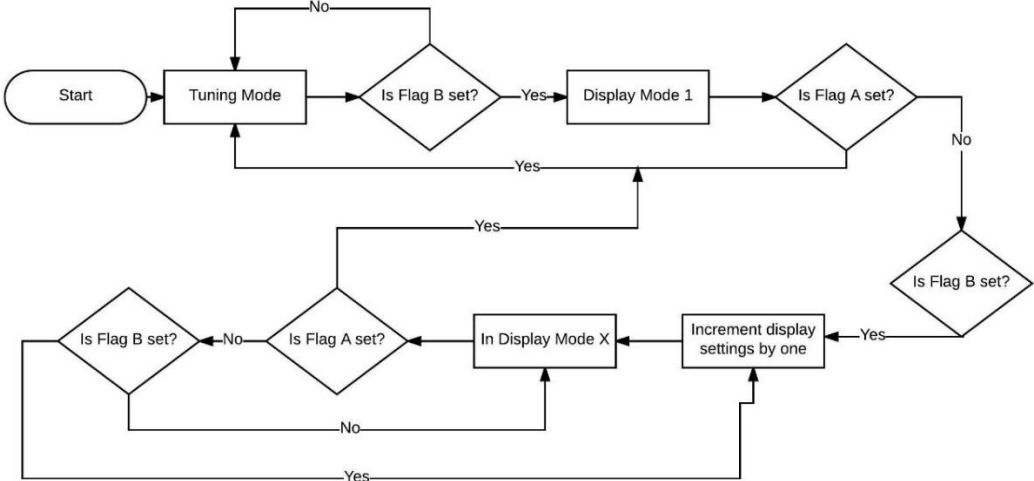


Table 28: Selection Flow Chart

### 5.3.1.2. Hardware

ATmega2561 Inputs:

- 1. Tuning mode button
- 2. Cycle button
- 3. Power

ATmega2561 Outputs:

- 1. 2 x seven segment display
- 2. Serial communication with the analog to digital converter
  - a. clock line
  - b. data line
- 3. Serial communication with the TLC 5955
  - a. Clock line
  - b. Data line
  - c. Enable line
- 4. Power status LED

PCM3060 Inputs:

- 1. Analog to digital converter clock (40 KHz)
- 2. Input signal from G12 pedal
- 3. Power

PCM3060 Outputs:

- 1. Serial communication with ATmega2561
  - a. Clock line
  - b. Data line

TLC 5955 Inputs:

1. Enable line
2. Serial communication with ATmega2561
  - a. Clock line
  - b. Data line
3. Power

TLC 5955 Outputs:

1. Serial data out (to be run in series with three other TLC 5955 laid out in series)
  - a. Clock line
  - b. Data line
2. 48 red, green or blue outputs per chip

The ATmega2561 will be the center of the LED Matrix Feedback Display. Doing most of the processing and timing. The ATmega2561 will regularly make requests of the analog to digital converter. The analog to digital converter will then pass back its information at which time the ATmega2561 will use its algorithm set forth in diagram x to determine the frequency of the incoming signal.

The algorithm used to calculate frequency is one that constantly calculates what the value of the incoming signal is, stores that value, and compares it to the next value that it stores. It compares them to see if the signal has passed the zero threshold. The zero threshold isn't necessarily zero volts, it is however the level at which the AC signal is centered around. This comparison is computed by checking to see if the first signal was above or below the threshold, and then seeing if the second signal is below or above the threshold respectively. If both are the case, then the signal would have had to cross through the threshold level since the signal is continuous and shouldn't not jump over this value at any time.

The moment the ATmega2561 has detected a crossing over the threshold level a timer is started. The ATmega2561 then begins the process over again, polling values, storing them, and doing this again to see if they cross the threshold. If the values do not cross the threshold, only the first value is removed and the second value is stored in the first values spot so that the next incoming value may be stored there. All the while the system tracks the average of the amplitude of the incoming values. This system will continue to repeat until the threshold is crossed again. It is at this time that the timer value is stored.

Once the timer value has been stored the process repeats itself another three times, each time storing the timer value in a unique variable. Once the four timer values have been obtained, the values are then averaged together to calculate half of the average period time. This average is very important to try and help prevent erratic outlier values from showing themselves in the output. The amount of iterations may also be adjusted during testing to obtain a truer output.

We can accurately say that the period would be obtained because as we said Nyquist sampling theorem tells us that at a sampling frequency of 40 thousand samples per second, we can accurately represent a sample of 20 KHz which is the maximum range

that a normal human can hear up to, and since we can accurately represent this sample, calculating when this sample crosses the threshold and timing how long it takes to cross that threshold again will give us how long it takes half of that wave to pass. Averaging multiple calculations together will only help reduce the amount of false readings, or insignificant readings we may get.

Once this averaged time is obtained, the average is then multiplied by two to obtain the average period time. The inverse of this average period time is then calculated to now obtain the average frequency of the samples. This along with the average amplitude of the wave is then stored. The ATmega2561 then calls the appropriate function, determined by which mode the ATmega2561 is set to, and passes in the average amplitude along with the average frequency.

Once the appropriate program is called upon, each program will take the values that is has received, run it into a look up table for values to set for its color out program, the program that sets the colors up for each LED, then call the serial out program which serially transmits the data to the TLC 5955. The difference between each of these called programs will simply be which lights get what color and intensity. All of which is created by joining reiterating parts of code to build large binary numbers to be shifted into the LED drivers.

As soon as the large data is transferred to the TLC 5955, the ATmega2561 enables the settings, this will activate the colors to actually show up for the users viewing pleasure. The program within the ATmega2561 then returns to calculate the frequency all over again.

The longest part of the entire frequency program will be actually obtaining the frequency, the reasoning is that sound waves travel much slower than the ATmega2561 can calculate, therefore majority of the time that it is polling the data, the data will not be crossing the threshold. But the moment that the data does cross the threshold, the ATmega2561 stores the data and sets the timer in a just a few microseconds. Then it begins waiting all over again.

The fastest that the ATmega2561 can compute the time between the crossing of a wave would be around the 20KHz range, and the slowest would be at the 20 Hz range. This will call for a code that adjusts its iteration time based on the speed of the first half period calculated. This is due to the fact that iterating the calculation of a 20 Hz frequency 4 times will take at the very least one-tenth of a second, which that kind of delay starts to becomes noticeable to the human eye around that range.

The communication between the analog to digital converter and the ATmega2561 will be the same as the communication speed between the ATmega2561 and the TLC 5955 which will all be at a 16 MHz clock rate. This will create a quick transfer of information from the analog to digital converter, to the ATmega2561 and then finally after some computation to the TLC 5955, where it will be translated to a visual representation to the user.

The buttons that are located on the housing of the unit will be tied to interrupt pins on the ATmega2561. These will run an interrupt service routine that will change the store settings to either tuning mode, if the left button is depressed, or it will cycle the settings through to their next setting if the right button is depressed. These were chosen to run as interrupt service routines due to their infrequent use. Also their interrupt service routines are able to be very short due to the necessary instruction only be a couple each. This becomes very useful since the timer would be running during any time that this routine is called, therefore minimizing the total time away from the frequency calculating program is imperative to maintain accuracy.

The input signal will be entering in through a 3.5 millimeter jack located on the left side of the unit. The signal will enter this jack and be met up by the analog to digital converter. Before it reaches the analog and digital converter the signal will be biased with the three volts positive. This will center the alternating signal about 4 volts for the analog to digital converter. There is no need to run the signal through an amplifier first since the analog to digital converter has a resolution of 24-bits over eight volts. This would leave its sensitivity down to the hundreds of nano volts, which is significantly more sensitive than we will need for this part of the project.

Having the input signal biased and wired up to the analog and digital converter allows the analog to digital converter to obtain a digital value correlating to the signal and pass it on to the ATmega2561 quickly. The analog to digital converter will use a clock supplied by the ATmega2561. This clock will be generated off of the ATmega2561 pulse width modulation pin at 48 KHz. It will then use the ATmega2561's master clock of 16 MHz to transfer the 24-bit value of the signal out to the ATmega2561. At this point the analog to digital converter is waiting for the next request.

Meanwhile on the other side of the ATmega2561 is the TLC 5955. The TLC 5955 can and will store its most recent display settings in its own internal registers. These registers can be updated all at once with the enable bit once all data has been shifted into the temporary registers. This is to allow all data to take effect immediately, saving the user from a moment of nonsensical madness. The data for the furthest TLC5955 will technically be the first data to enter into the first in line TLC 5955, this is due to the design of daisy chaining the chips in order to create 64 uniquely addressable LEDs. All four of the TLC 5955's enable pins will be tied together, as there will never be a foreseeable need to only update one chip at a time.

Since the TLC 5955 chips sink constant current across their outputs, there will be no need for current limiting resistors to be in line with each one of the LED legs. This current draw will create heat through the TLC 5955 chip that will be synced out through a large ground connected to the bottom side of the chip. This is a design necessity of the chip. We should not reach any limit of heat passing through the chip since this chip is designed to be able to handle all 48 outputs on at 30 milliamps a piece.

As for the power unit for the LED Matrix Feedback Display, there will be a status LED on the back of the unit where the power line comes in, this light will indicate whether or not the unit is receiving power. This is always a great tool for trouble shooting for the end user. The actual power supply will be a five-volt system capable of producing up to five amps for a total of 25 Watts of power. The average use of the system shouldn't pull anything near the 25 Watt line due to a few reasons.

All of the outputs will most likely never all be on simultaneously. The output colors have varying demands for power, where white light demands the most amount of power, and white light will not be a color commonly used in the programming. The global brightness will automatically be set to a lower setting, which will off hand turn the units off for so many cycles out of the total amount available.

We want to design the system to be capable of providing 25 Watts worth of power in case we decide to add a flashlight mode to the system, which would require the max amount of power to have all lights set to white light at full brightness, thus drawing its max power value of 19.2 Watts. This leaves us some room to grow in the case that we would like to add any further devices onto the system as is discussed in the expandability section.

## **5.3.2. Expandability**

### **5.3.2.1. Bluetooth Connectivity**

This connection would offer the user the ability to connect to the LED Matrix Display wirelessly through their own phone, tablet, or computer. They would need a companion application that would then work as a wireless remote for the LED Matrix Display, changing the display settings, and granting the user finer control of their own patterns that they may want displayed on the LED display. This Bluetooth system would be integrated directly within the housing of the LED unit connected to the ATmega2561 directly.

### **5.3.2.2. Wi-Fi**

Including a Wi-Fi module that works with the ATmega2561 would allow the chip to connect to an open network, or a private network that the settings have been saved to the unit, and obtain updates to the data wirelessly and without hassle to the end-user. This would add relatively low costs to the final product, but add years to the projected used life of the model, allowing developers to keep pushing out fun seasonal updates that keep the user engaging in the product time and time again at the very least to see what new has come out.

### **5.3.2.3. LCD Panel**

An LCD panel would offer a low cost alternative to Bluetooth connectivity along with the companion application. One would take the place of the other, although both can exist simultaneously, offering the user the option to connect with their phone or use a set of selection buttons and the low character count display to select their choice of settings.

#### **5.3.2.4. Touch Screen Display**

This again would work as a substitute for both the Bluetooth and the LCD panel, allowing the user to select differing display setting with ease. They wouldn't have to open up their phone and connect. All they would have to do is turn on the LED Matrix Display and the touch screen display will be on and ready to accept commands. This would eliminate the need for physical buttons as well that would accompany the LCD Panel.

#### **5.3.2.5. Humidity Sensor**

An integrated humidity sensor would not serve the display in creating a better display for the user, but instead would help increase the longevity of the display by shutting the system down and tripping a switch if it were to come into contact with water in particularly vulnerable areas, thus possibly preserving the internal components that are sensitive to humidity or liquid water.

#### **5.3.2.6. Temperature Sensor**

The Temperature Sensor would also serve as a tool to help preserve the longevity of the device by offering it a feedback system that can monitor the internal temperature of the units housing where heat will build up the most since all of the components are located within close proximity to one another and all of the LEDs produce a considerable amount of heat when added up. The sensor would keep track of the temperature potentially limiting the global brightness of the entire display if the temperature starts to reach a threshold, this will continue to happen until the temperature stabilizes. If this stability cannot be met by the time the temperature reaches a max threshold, the system will be shutdown.

### **5.3.3. Implementation**

The circuit is comprised of four inputs from the MCU. These connect to the serial in (SIN), serial clock (SCLK), latch (LAT) and gray scale clock (GSCLK). These are the communication lines that shift in data to the TLC 5955. Every time the serial clock is brought high, the data on the serial in is shifted into the storage register. The grayscale clock is an input clock that controls the speed at which the grayscale increments. The VCC pin is connected to 5 volts and the ground is connected to the ground connection of power source. Each LED is connected on their anode side to VCC in parallel, and their cathode side is connected to each respective output pin of the TLC 5955. No Resistor is needed in-line with the LED due to the fact that the TLC 5955 outputs constant current that is controlled by the settings sent to the TLC 5955 through the serial connection. Each of these LED outputs is labeled as such, OUTB0-OUTB15, OUTR0-OUTR15 and OUTG0-OUTG15. B correlates to the blue leg of the LED, R correlates to the red leg of the LED, and G correlates to the green leg of the LED. Each of the traces between VCC-LED and the LED should be as short as possible to reduce inductance of the circuit.

The TLC 5955 has a serial output that will output the 769th bit of the storage register when the serial clock is set to HIGH. This feature is what will allow us to connect 4 of



the TLC 5955's in series, enabling us to have 64 individually addressable LEDs with a total shift in bit count of 3076-bits.

A very important part of the circuit design is the thermal pad. The thermal pad is important because it allows the TLC 5955 to dissipate heat that it is producing due to a significant amount of current flowing through the device due to the LEDs that sink current through the device. The thermal pad on the device itself needs to be grounded through the PCB design in order to effectively work. Thermal vias must also be placed in order to transfer heat efficiently away from the device.

A well-regulated electrolytic capacitor must be used between the LED voltage source and its ground to reduce the voltage ripple to less than 5%. This is very important because if all of the LEDs are turned on simultaneously to a high current draw setting, then the voltage source might not be able to facilitate this kind of power surge.

A decoupling capacitor must also be placed between the VCC source of the TLC 5955 and its ground in order to filter out any high frequency noise that may disturb the chip while in use. This capacitor does not need to be electrolytic in nature.

The ground of the circuit needs to be as wide as possible as the total current through the chip can be as large as 1.53 amps.

#### Timing

When refereeing to a system that is a real time system, timing of devices used in that system is highly important. If it takes 3 seconds to render what frequency I have and then another second to output that on the LED display, then the user is going to have a poor experience with the product. This is why the TLC 5955 was considered for the job. With its ability to have a 25 MHz data transfer rate, and a 33 MHz clock for the PWM. The 33 MHz clock for PWM allows for the flashing of the LEDs on and off to not be noticeable to the human eye, while the 25 MHz data rate clock allows for the 3076-bits needed to set the grayscale settings to be transferred in at a total time of 123 micro seconds. Every update to the lighting scheme should take a total of 6152 clocks, 3076 clocks for the grayscale latch, and 3076 clocks for the control latch, this takes into account shifting into four TLC 5955s that are connected in series.

## 6. PCB

A PCB is printable circuit board. For this project, G12 PedalVision, will have an all-inclusive PCB. The analog, digital and LED display effects will all be routed from the PCB. The chips on the PCB will be programmed to gain the desired functionality from the project. The board will be responsible for regulating the power, taking care on any discrepancies that may or may not occur. Ideally, everything on the board will be prepared to withstand the worst possible scenario. To get to the point of full functionality, many steps need to be taken.

## **6.1. Design Process**

Before the PCB process is even considered, many other aspects need to be considered and completed. First and foremost, the project, along with all its functionality needs to be determined. After the components needs to be researched, tested and chosen. Then the PCB layout needs to be designed. After the layout has been designed and finalized, it needs to be sent out to a chosen manufacturer. The general overview of these steps is laid out in the sections below.

### **6.1.1. Component Selection**

After the project, has been determined, the next step is to choose the components. The components are found based on the desired functionality of the project. In this project the analog effects, the digital effects, and the LED display need to be considered. The component selection helps in determining the size of the PCB that will be needed. It also helps in determining the constraints that will need to be taken into consideration.

### **6.1.2. Eagle Layout**

Eagle PCB Design can be accessed by going to <https://cadsoft.io>. This will be the software used to design the PCB. It allows the user to choose how many layers the PCB will consist of. It will also allow the user to place the components and route the traces in the most efficient way possible. This is a crucial part of the design process. If the PCB layout is not looked at carefully, the whole project could fail.

In Eagle, every dimension can and will be taken into account for the design. Spacing is something that will need to be paid close attention to. Capacitors, for example, will need to be placed close to each other to help regulated the voltage going into the chip.

Level shifts may need to be used in case small voltage differences need to be adjusted. For example, one chip may need 5.5 v, while the other may need 9.0 v. The level shift will allow both voltages to be supplied simultaneously.

The schematic editor in Eagle will be extremely useful when it comes to determining the layout. Each pin will be named and traced to the exact location it will be placed. The PCB will be a combination of all the schematics.

While designing a PCB on Eagle, one can define whatever constraints are required for the design. Multiple modes can be used to help with the editing process. Any function used in the design process can be easily changed and manipulated.

Overall, this software will allow a fast and easy design process. The PCB for the G12 PedalVision will be responsible for making the project work. Eagles layout editor supports rounding edges, trackwidth and many other aspects of the design that need to be considered.

### **6.1.2.1. Trace Widths**

Trace widths, the minimum width of the copper, on the circuits board need to be calculated as part of the Eagle Layout process. It seems that .007 inches is usually the minimum amount of trace width that is required. Of course, this will be changed based on the design that is determined.

### **6.1.3. Finalization Process**

While finalizing the design, one must go through and make sure all the connections are made correctly. Another thing to go through all the components and make sure all the correct ones were chosen. The trace widths would need to be verified. Capacitors, pull up/pull down resistors are all placed and calculated correctly. After double and triple checking all the little things, the layout will be ready to send off to a manufacturer.

## **6.2. Manufacturers**

Choosing a manufacturer is the next step of the design process. For each manufacturer, must consider, cost, customer service, shipping time, quality, availability and reputation.

### **6.2.1. Seed Studio**

If Seed Studio is chosen as the PCB manufacturer then it will cost around \$9.90 for a basic PCB. This company gives the option for the user to order custom designs. One advantage of using Seed would be the half shipping cost and the two day delivery, if the customer was also an Amazon customer. The disadvantage to using this company is the online reviews. Many customers have not enjoyed the services provided by Seed Studio.

### **6.2.2. Osh Park**

Osh Park offers many deals, upfront. Per the company site, a two-layer board is \$5 per square inch and a 4 layer board is \$10 per inch. In both cases, only 3 copies could be sent and it would take around 12 business days. The time it would take to receive a revised board from this company is a huge disadvantage. If anything goes wrong with the board, waiting 12 days, just wouldn't be ideal.

### **6.2.3. PCBWay**

A two-layer board (100x100mm) would cost about \$10 from PCBWay. After researching this company, it seems to have the best user reviews and the fastest turnaround time for the price. This will be the company that group G12 will go with.

## **6.3. Surface Mount**

For a surface mount PCB the components are mounted onto the surface of the board. Pads on the surface are tied to the ground plane and the components are soldered down onto the pads. This type of design makes the PCB easy to troubleshoot. A surface mount PCB should be able to handle what this project requires, including all the constraints.

## **6.4. Thru-Hole**

For this type of PCB layout, holes are drilled through the PCB and the components are placed using solder. The disadvantage of using the using thru-hole is having to hand place the components. From user experience, the PCB have come without any of the components placed. This may leave room for more error, especially if a person isn't used to working with thru-hole components.

## **6.5. Testing Techniques**

During the design process 0 ohm resistors should have been added to each voltage rail. Adding these resistors would help trouble shoot each rail in case of troubleshooting. To test, one would remove one of the 0 ohm resistors. This would help with testing the connected rail. If the expected current is flowing, this rail would be not be the problem. To find out if it was the other voltage rail, the 0 ohm resistor would be reconnected and the 0 ohm resistor for the other rail would be disconnected. The same test could be done on this rail. By doing this test, one would be able to tell if the problem was with the board or it was with the components at the input.

Another way to test any problems that may occur is by using an oscilloscope. For example, a probe could be placed on the input and output of a component. The waveform that appears would be able to show if the problems lied within that region, witting those components.

Some general problems that can occur with the PCB are solder blobs, shorts, and improperly placed components. Most of these problems can be fixed using solder and a soldering iron. For surface mount components, one might have to look through a microscope. Choosing ball grind components may cause problems with trouble shooting. It would difficult to test the pins in the very middle of the chip.

For this project, a surface mount PCB will be used. This will help with troubleshooting the PCB if any problems occur. It will also be easy to be able to change components, if needed.

# **7. Project Testing Plan and Test Results**

## **7.1. Hardware Testing**

The G12 PedalVision uses various different components but the main components that will be used to accomplish the different tasks are shown in figure 73 below. Starting from the left side of the picture we have the DSP chips that will be used for the digital effects units. Moving to the right we have multiple ADC chips in order to convert our analog signal into a digital signal for the DSP chips to use. If we move over again we have single and double channel op amps for the analog effects. On the bottom right corner we have a dual channel transconductance amplifier for use in the compression circuit. Then finally in the upper right hand corner, we have the smaller LEDs for the LED matrix and large LEDs for the effect toggle indicators.



Figure 73: G12 PedalVision Main Components

### 7.1.1. Analog Testing

In order to ensure that the larger sub systems are completely functional we need to test the main components of each of the analog effects. To ensure that the operational amplifiers are working correctly we will test for the differential mode gain. Next we will create a simple non inverting amplifier to check for accurate amplification. We will also test the amplifiers while in a unity gain configuration in order to ensure they have the full audio frequency spectrum bandwidth. Once these main blocks are verified to be functional we will implement them into the larger sub systems to test the specific function of each analog effect. Figure 74 below shows the components used to build the analog effects.

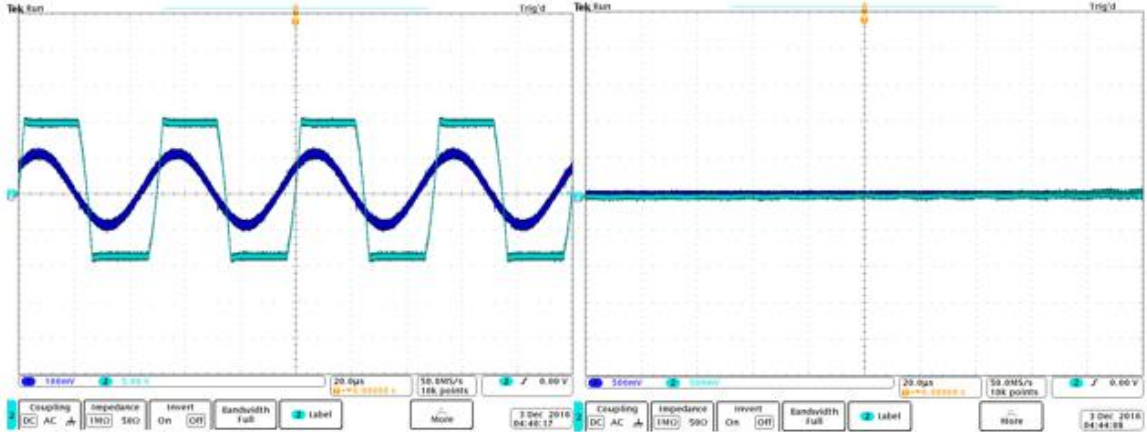


Figure 74: Differential Mode Test (left), Common Mode Test (right)

As you can see the chosen op amp is able to get very close to the rail voltage of 9V in the differential mode and has a small noise when tested in the common mode configuration. The amplifier is also able to provide a large gain without added clipping or distortion.

We also tested the inverting and noninverting amplifiers to see if we had accurate gain and if we had any distortion. For this test we used a 1k and 2k ohm resistor to set the gain. From figure 75 we can see that the inverting amplifier has a gain of approximately -2 while the noninverting amplifier has a gain of approximately 3. These results match what we expect and show that the op amp is functioning correctly.

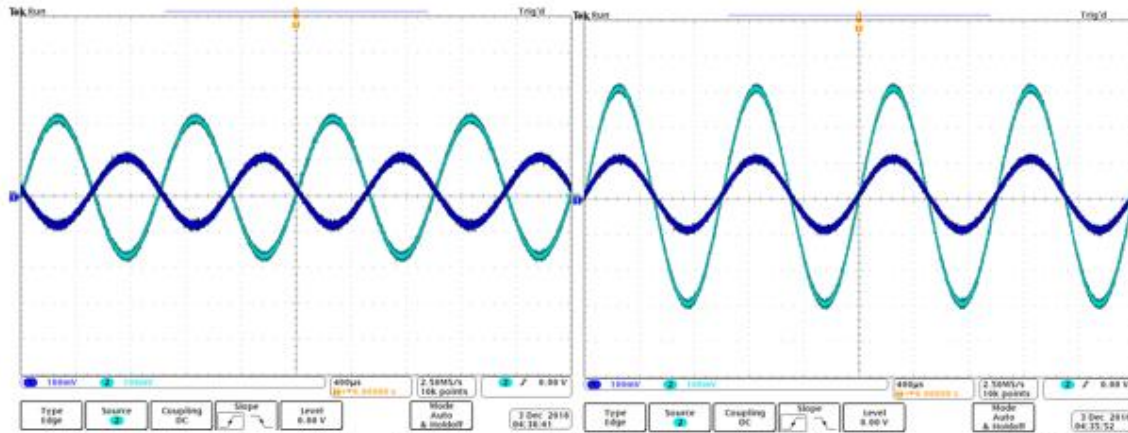


Figure 75: Inverting amplifier using OPA164x (left), Noninverting amplifier using OPA164x (right)

Next we tested to ensure that the op amp was able to match the input signal with the output signal at 1kHz and at the edge of the audio frequency, 20kHz.

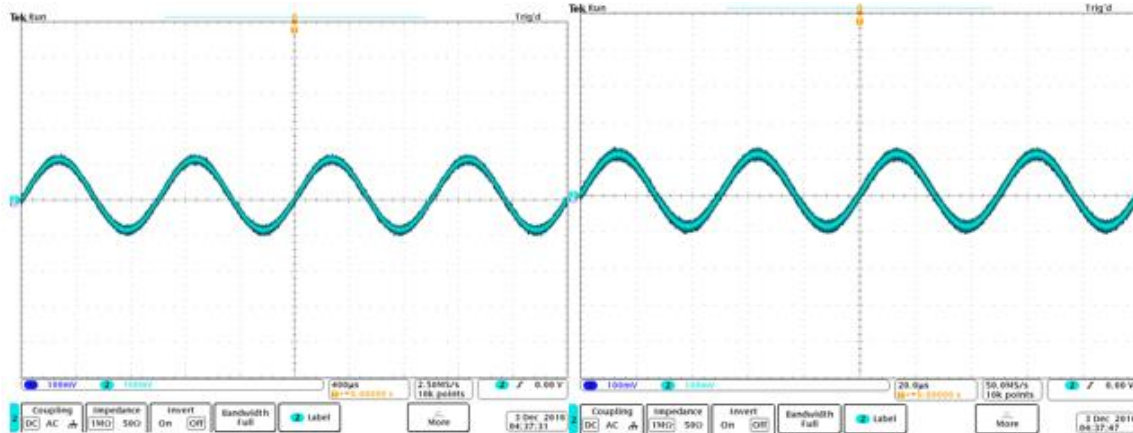


Figure 76: Unity Gain at 1kHz (Left), Unity Gain at 20kHz

From figure 76 the chosen op amp is able to accurately replicate the input signal to the output signal at 1kHz and 20kHz.

With the main components of the system tested we can begin to test the sub systems of the analog effects unit.

### 7.1.1.1. Input Buffer

In order to test the input buffer of the analog system we will do the following:

First we will supply circuit with the DC power it needs, 9V for the positive rail of the amplifier and 4.5V to bias the input. Next we will read what the output voltage is. Next we will input sine waves of varying amplitudes at 1kHz into the amplifier and read the output vs input on the oscilloscope. We will check to see how close the output signal matches the input signal. Ideally the input signal and the output signal should be identical since this input buffer should not change anything about the signal passing through it.

Our test results are as follows:

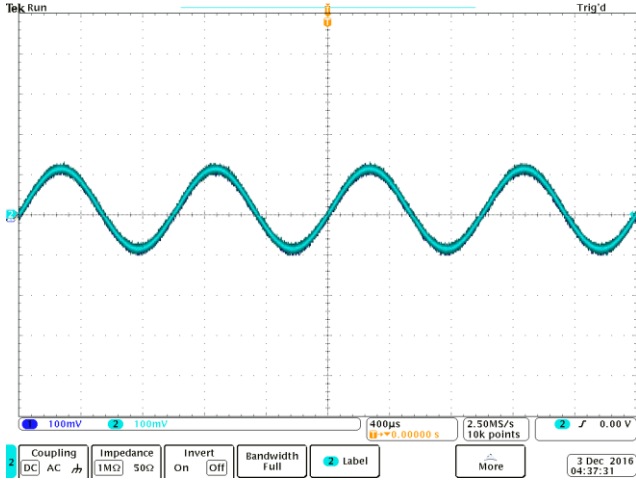


Figure 77: Input Buffer at 1kHz

From figure 77 the input buffer does a good job of passing the input signal exactly to the output at the standard frequency of 1kHz.

### 7.1.1.2. Overdrive

In order to test the overdrive circuit of the analog system we will do the following:  
First we will supply circuit with the DC power it needs, 9V for the positive rail of the amplifier and 4.5V to bias the input. Next we will read what the output voltage is. Next we will input sine waves of similar amplitudes at varying frequencies and read the output vs input on the oscilloscope. We will repeat this procedure at various gain amounts. At the very least we will test the circuit at minimum gain, moderate gain, and maximum gain. We will read the output vs input for each gain setting. We will also view the frequency characteristics by using the spectral analyzer function of the oscilloscope.

Using a simplified version of the overdrive circuit we were able to record the frequency characteristics of an overdriven ideal sine wave.

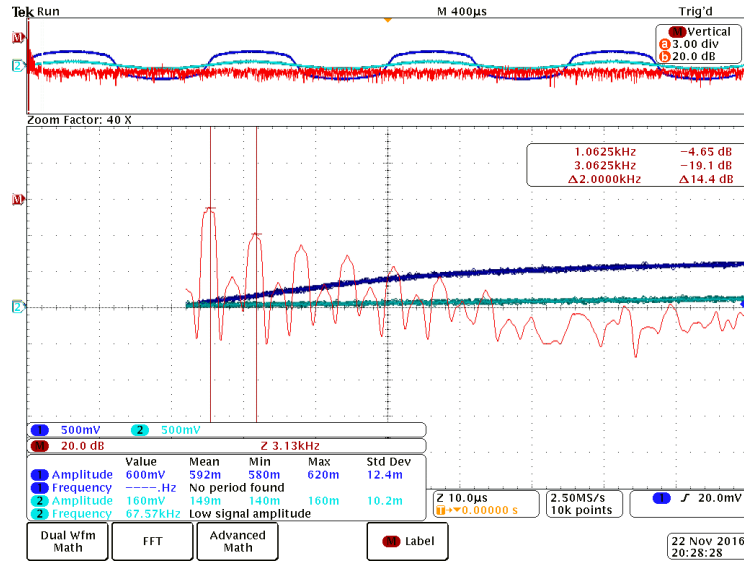


Figure 78: Ideal Overdrive Frequency

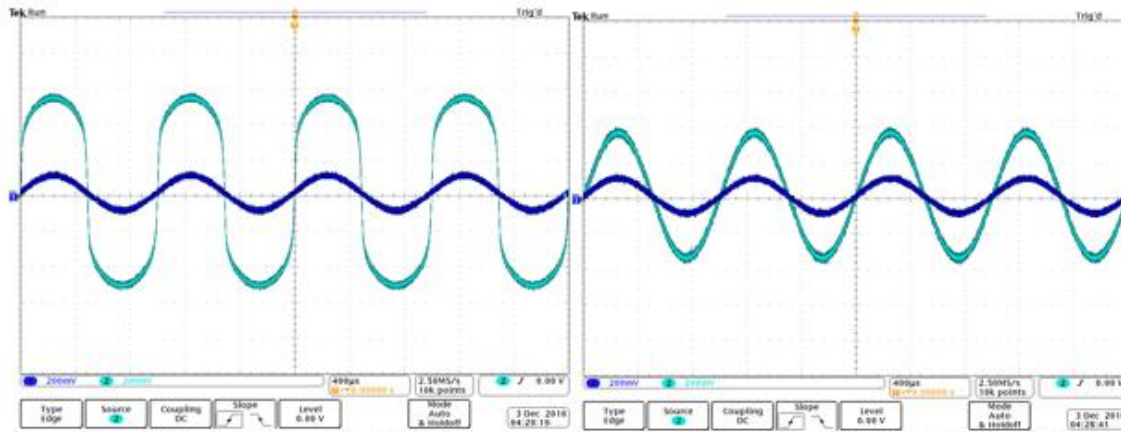


Figure 79: Overdrive High Gain (Left), Overdrive Low Gain (right)

From figure 78 you can see that the frequency spectrum looks similar to as if a square wave was being analyzed. These odd numbered peaks are what create the harmonic distortion. Figure 79 shows the input versus output waveform of the overdrive circuit with high gain on the left, and shows the same circuit with lower gain on the right. This shows that the gain setting of the circuit will determine how much clipping will occur with this circuit. As gain increases, more clipping will occur. The signal will eventually flatten out at about the turn on voltage of the diodes.

### 7.1.1.3. Distortion

In order to test the distortion circuit of the analog system we will do the following:  
First we will supply circuit with the DC power it needs, 9V for the positive rail of the amplifier and 4.5V to bias the input. Next we will read what the output voltage is. Next we will input sine waves of similar amplitudes at varying frequencies and read the output vs input on the oscilloscope. We will repeat this procedure at various gain amounts. At the very least we will test the circuit at minimum gain, moderate gain, and maximum



gain. We will read the output vs input for each gain setting. We will also view the frequency characteristics by using the spectral analyzer function of the oscilloscope.

Using a simplified version of the distortion circuit we were able to record the frequency characteristics of a distorted ideal sine wave.

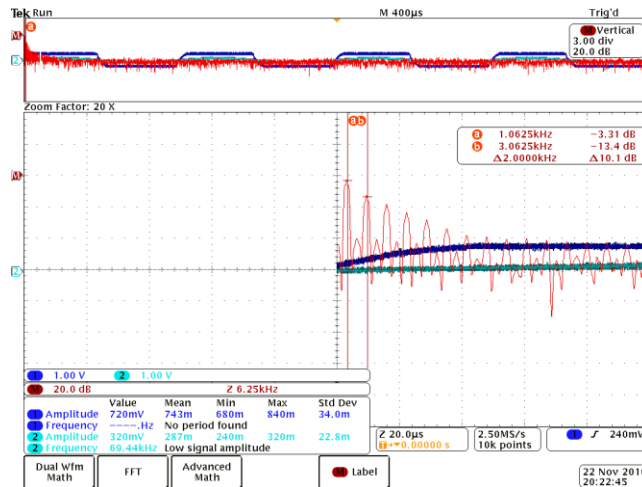


Figure 80: Ideal Distortion Frequency

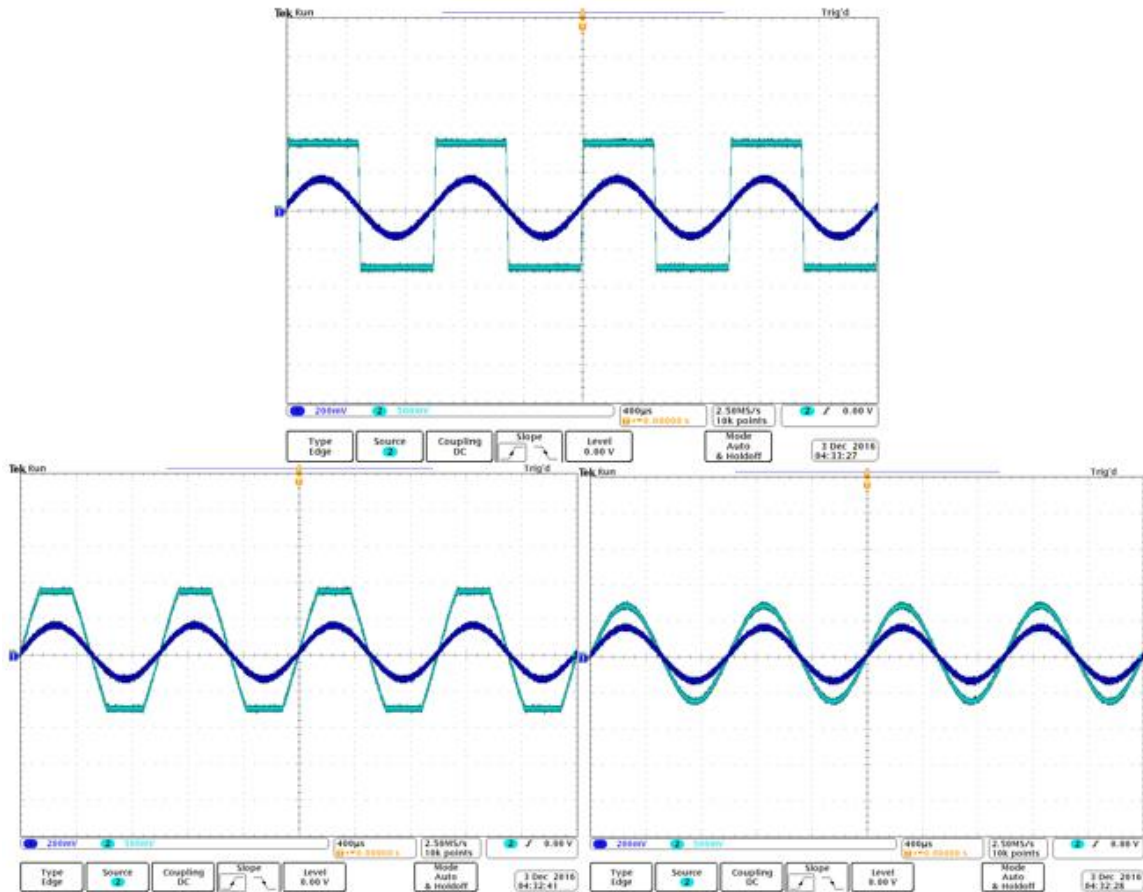


Figure 81: Distortion at Various Gains

From figure 80 you can see that the frequency spectrum of the distortion circuit. Like overdrive, the frequency components of distortion occur at odd integer values of the input signals fundamental frequency. These odd numbered peaks are what create the harmonic distortion. Figure 81 shows the input versus output waveform of the distortion circuit. The top waveform shows high gain, the bottom left shows medium gain, and the bottom right shows low gain. You can also see from the difference between the multiple waveforms in figure 81, that distortion creates some compression because the signal, regardless of the amplitude will clip and have a maximum output amplitude of about the forward voltage of the diodes. In this case the forward voltage and output amplitude is about 0.7V. You can also tell that the amount of gain is directly related to the amount of clipping seen at the output. The higher the gain, the more square the output wave form will be.

#### **7.1.1.4. Compression**

In order to test the compression circuit of the analog system we will do the following: First we will supply circuit with the DC power it needs, 9V for the positive rail of the amplifier and 4.5V to bias the input. Next we will read what the output voltage is. Next we will input sine waves of various amplitudes at a frequency of 1kHz. We will then read the output vs input signal on the oscilloscope. We will adjust the amount of current feeding back into the amplifier to see how the level of compression is affected by this change in current. Next we will adjust the ratio level of the output vs the input to see how the compression and mixing of the signals affects the total output wave form as well.

#### **7.1.1.5. Equalization**

In order to test the input buffer of the analog system we will do the following: First we will supply circuit with the DC power it needs, 9V for the positive rail of the amplifier and 4.5V to bias the input of the band pass filter. Next we will read what the output voltage is of the band pass filter. We will not need to read the output voltage of the low pass, high pass, or band reject filter since they are not active filters and do not have a bias voltage applied. Next we will apply various sine wave of similar amplitudes and varying frequencies to see how the different filters affect the output signal. We will apply the input signal and change which filter is selected for the same sine wave in order to view the output for filter at various frequencies. Finally, we will apply a sine wave of a predetermined frequency, then we will adjust the resistance of each filter to see how the change in cut-off frequency affects the output signal.

#### **7.1.1.6. External Effects Interface**

In order to test the input buffer of the analog system we will do the following: First we will supply circuit with the DC power it needs, 9V for the positive rail of the amplifier and 4.5V to bias the input. Next we will input sine waves of varying amplitudes at 1kHz into the amplifier and read the output vs input on the oscilloscope. We will check to see how close the output signal matches the input signal. Ideally the input signal and the output signal should be identical since this input buffer should not change anything about the signal passing through it.

Our test results are as follows:

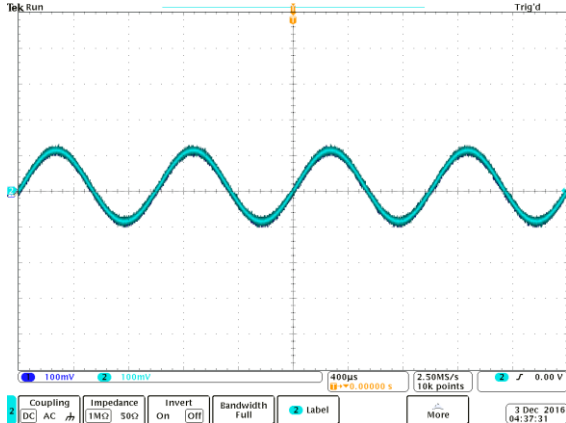


Figure 82: External Effects Interface at 1kHz

From our testing shown in figure 82 we saw that the amplifier we selected was able to accurately match in input signal at a frequency of 1kHz

### 7.1.2. Digital Testing

In order to test the digital effects we will have to have a working product. Because of this we will not be able to have test results in the current document. In order to show a test of the digital effects, we have probed a guitar in a lab using an oscilloscope. This oscilloscope was able to show the output that occurs without any effects applied, as well as show the modulated output waveform. Below in Figure 83 is the basic output waveform of a guitar signal.

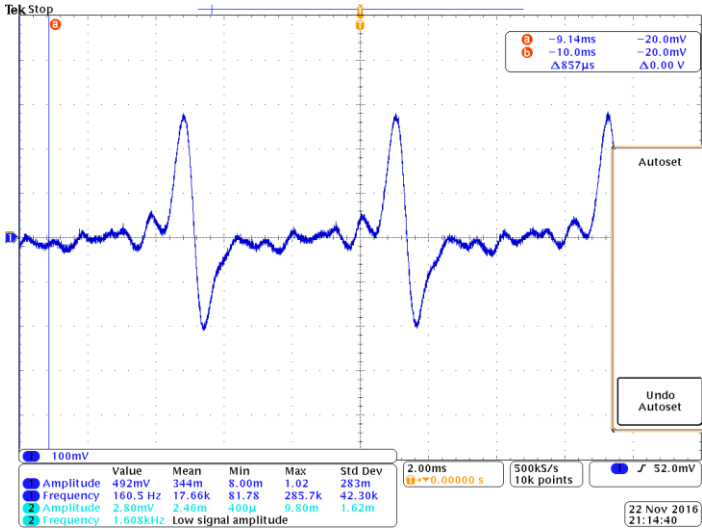


Figure 83: Dry Signal Output

#### 7.1.2.1. Delay/Echo Testing

In order to test the delay/echo effect, we used the TC Electronic Flashback Delay and Looper pedal. This is a commercial pedal that is available currently, and is a common

pedal used to achieve the delay effect. For testing purposes the output that was observed from this pedal will be a comparison when testing the digital effects of G12 PedalVision. If the output wave is able to match the lab testing with a commercial pedal, then the digital effects can be considered accurate. The output waveform for the delay pedal is simply an exact copy of the original output signal, except with a smaller amplitude. For this reason, the waveform is not pictured.

Along with testing the waveform of the output signal from the pedal, the actual sound of the pedal needs to be measured. In order for the sound of the effect to be tested the product needs to be mostly completed. At this stage we do not have the ability to test the sound of the output, so this testing must be done at a later scheduled date. When we are able to test this, we will simply need to have an experienced musician who can pick out differences in sound easily listen to both the output from the commercial pedal and the G12 PedalVision. The results will be updated when this testing is able to be completed.

It is also necessary to test that the values for the effect are being loaded into the DSP chip correctly. To test this the effect interface will be used to load different values onto the board. The procedure for testing will be to play the same note, while changing different values. If each value modifies the sound correctly then it can be said that the effect is working properly.

### 7.1.2.2. Reverb Testing

In order to test the reverb effect, we used the TC Electronic Hall Of Fame Mini Reverb with TonePrint commercial pedal. Using this pedal allowed for us to get an output waveform that we will use as a base when testing the G12 PedalVision. The output waveform of the commercial pedal is shown below in Figure 84. It can be observed that compared to the dry output signal in figure 83, this waveform has larger amplitudes at different parts of the signal. This occurs because the signal is sampled as it comes in and is mixed together initially with multiple echoes tapped at different lengths. See section 3.2.2 for more details about the reverb effect. To test the digital reverb effect that we will be implementing, this base waveform can be compared to the output from given when testing the G12 PedalVision.

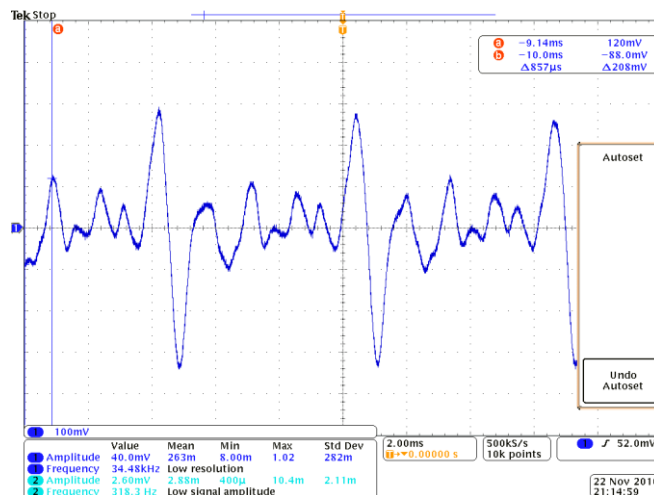


Figure 84: Output of Reverb Pedal

Along with trying to match the output waveform, the pedal must be tested based on sound. Just as we plan to test the delay effect, we will have an experienced musician listen to both the output of a commercial pedal and the output of our pedal. If the output sound is the same, then it will be considered to have passed the test. It is also necessary to test that the values from the effect interface are being loaded into the DSP chip. To test this, the process is the same as for the delay testing. We will simply change each effect and make sure that it modifies the sound correctly.

### 7.1.2.3. Flanger Testing

In order to test the flanger effect, we used TC Electronic Vortex Flanger Stereo Flanger Pedal. When probing this pedal in the lab we were able to test the output of the signal, as well as the frequency response of the output. In Figure 85 below the output waveform is shown. Section 3.2.4.1, Figure 21, shows and explains the frequency response graph. As one can observe in the figure below, the original signal is phase shifted by an amount that changes based on a few parameters. When testing the digital effect that will be used in the G12 PedalVision, the output waveform and frequency response will be compared to the results we received from probing the commercial pedal.

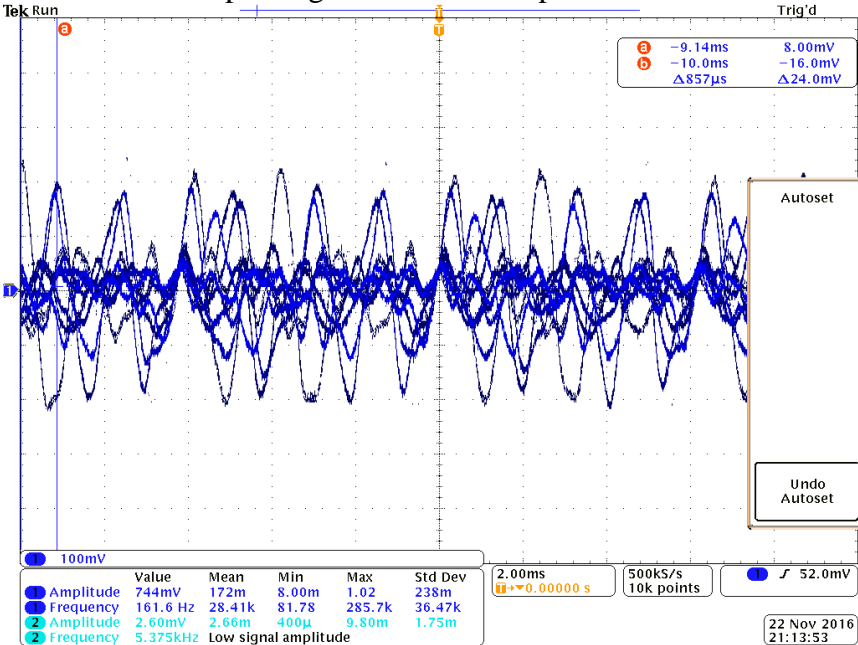


Figure 85: Output of Flanger Pedal

After making sure the waveform is correct, the next test that needs to be completed is to make sure that it sounds correct. The method for testing this will be to have an experienced musician compare the output sound of the G12 PedalVision to the output sound of the commercial pedal used in testing. The settings for the effect should also be tested by simply making sure the effect interface loads the correct values for each modification of the effect. If the modifications work as expected then the test will be considered passed.

## 7.1.3. LED Matrix Feedback Display Testing

### 7.1.3.1. LEDs

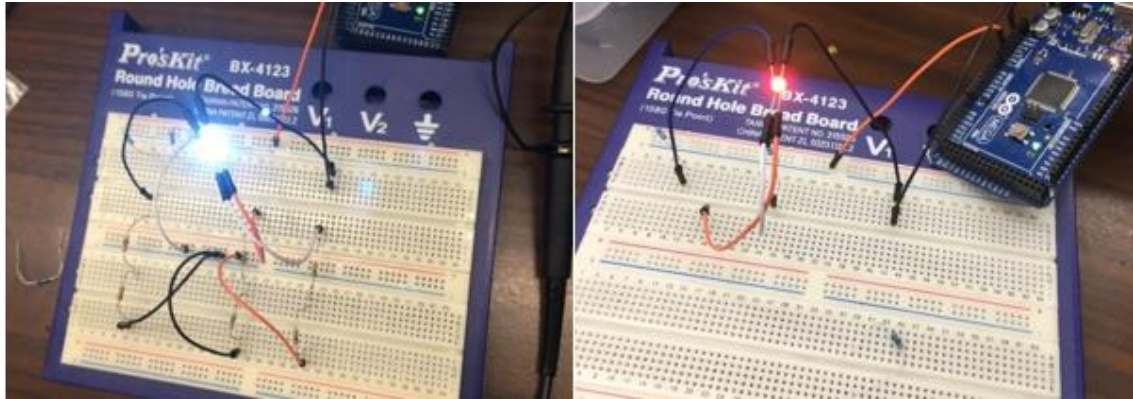


Figure 86: Left, All LED outputs on at 20 mA. Right, only red LED output on at 20 mA.

After the LED units arrived from the manufacturer we wanted to test them to verify that they functioned as advertised. We also wanted to verify the intensity of the light produced by the unit. Figure 86 shows the LED lighting up at different current values. These images were taken during testing.

To initiate our testing, we first soldered testing leads onto the contacts of the LED. This proved to be quiet difficult due to the fact that the contacts were extremely small and designed to be soldered using a solder paste and a heat gun. After this task had been completed with no bridging of contacts, a very important detail to ensure stability of the system and protection of all components, we hooked up the leads to the breadboard for them to be tested.

To each lead node we connected varying resistors to ground. These resistors are crucial in protecting the circuit as well as the LED from having too much current pass through it. In the final product these resistors will not be necessary due to the design of the TLC 5955 which has built in constant current capabilities.

To calculate the required resistance for each leg of the LED we first had to reference its data sheet for the voltage drop across each of the color LEDs. This information said that the red LED has a 2 volt voltage drop across it. The blue LED has a 3.3 volt voltage drop across it, and the green LED has a 3.2 volt voltage drop across it. With this information we used Ohm's Law to calculate the required resistance to keep the current in each leg at 20 milliamps with a 5 volt source provided from the Arduino Mega.

These values were 150 ohms, 85 ohms and 90 ohms respectively. After connecting all of the correct nodes, we hooked up the power leads and what we were given can be seen in figure x in the left image. The LED produced a white light with all of the leads maintaining the same current at the same frequency. The image on the right was

produced when the blue node and the green node were left as opens, only supplying power to the red LED. We did this isolated test for each color of the multi-color LED.

From our testing we were able to conclude that the device did in fact work as advertised. And that we are able to interface with it and control the outputs. The next steps in testing will be to attach each leg of the LED to a pulse width modulated signal as to verify all of the achievable colors that we are able to produce.

### 7.1.3.2. Frequency Code Testing

```

#define DEBUG false
#define on true
#define iterate 50
#define limit 50

int a = 0;
int b = 0;
int count = 0;
int temp = 0;
int increment = 0;
int val = 0;

unsigned int timer[2] = {0,0};
//unsigned int halfperiod = 0;
unsigned int timediff[iterate];

float halfperiod = 0.0;
float freq = 0.0;

void setup() {

  Serial.begin(9600);
  Serial.println("Begin communication");
  for(count = 0; count< iterate; count++)
    timediff[count] = 0;
}

void loop() {

  while((increment<limit) or (on))
  {
    a = analogRead(0);
    b = analogRead(0);

    if((a<=0 && b>0) || (b<=0 && a>0))
    {
      if(DEBUG)
        Serial.println("Crossed zero");
      timer[count] = micros();
      count++;

      if(count > 1)
      {
        if(DEBUG)
          Serial.println("Found a pair of crossing");
        timediff[temp] = timer[1] - timer[0];
        count = 0;
        temp++;
        if(temp >= iterate)
        {
          if(DEBUG)
            Serial.println("Found a frequency");
          for(val = 0; val<iterate; val++)
          {
            halfperiod = halfperiod + timediff[val];
          }
          halfperiod /= iterate;
          if(DEBUG)
            Serial.println(halfperiod);
          halfperiod = (double)halfperiod / (1000);
          freq = 1/halfperiod;
          freq *= 1000;
          Serial.println(freq);
          if(DEBUG)
            Serial.println();
          increment++;
          halfperiod = 0;
          temp = 0;
        }
      }
    }
  }
}

```

Figure 87: Frequency Monitoring Code. (continuation of code in right image)

Figure 87 above is the code created to roughly test out the theory of operation for the frequency monitoring system of the LED Matrix Feedback Display. The code is broken in half in the two images overlapping slightly for the viewer's aide.

We wanted to verify that the theory laid out in the code control flow chart would be able to calculate the frequency of an input signal accurately. These tests were completed using

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an Arduino Mega which uses a similar chip to the ATmega2561 which will be used on the final product.

Some limitation to our testing conditions were that the system we used did not use an off board analog to digital converter, but instead we used the integrated one on the Arduino Mega. The limitations this brought on were, low resolution of the polled input signal, along with slow reading time of the signal. This time was approximately 100 microseconds. These limits forced us to only be able to read in signals accurately up to a frequency of just under 1000 Hz.

After hooking the system up to a function generator we began testing the accuracy. We started ran the tests with input signals of 100 Hz, 500Hz and 900Hz to test the accuracy and the limits associated with them. We also tested these three frequency using varying amounts of iterations programmed to help with accuracy. The more iterations the more values are used to average in, therefore in theory, the more accurate the results. The data for these tests can be seen in table .

	20 Iterations			50 Iterations			100 Iterations		
	100 Hz	500 Hz	900 Hz	100 Hz	500 Hz	900 Hz	500 Hz	900 Hz	
	84.96	584.66	952.74	77.36	499.67	926.54	512.17	875.81	
	90.93	453.8	973.14	75.66	492.13	839.72	489.12	881.09	
	81.47	453.72	1006.04	73.98	509.03	966.82	486.98	872.54	
	71.32	499.6	973.14	75.1	499.47	872.54	497.28	962.65	
	97.6	453.72	1006.04	73.98	483.25	967.04	492.34	881.15	
	79.8	453.89	922.85	75.66	509.1	889.93	496.75	864.1	
	78.25	555.31	941.97	75.66	535.56	954.56	507.43	877.65	
	71.32	499.6	973.14	73.98	499.47	872.3	482.23	869.05	
	100.11	453.89	973.14	74.52	516.81	966.82	486.46	863.95	
	72.65	499.7	973.14	75.1	499.47	872.42	491.82	864.1	
Standard dev	10.47	47.20	25.72	1.06	14.38	48.84	9.45	29.40	
Mean	82.26	488.85	969.23	75.09	504.21	911.69	494.18	880.79	

Table 29: Data table for frequency tests

As you can see we did not complete a test for the 100 Hz frequency at 100 iterations, as this test would have taken a very long time due to the amount of data the program must test on a very slow oscillating wave. Another thing to note about the data is that while at 100 Hz and 50 iterations, the standard deviation is the lowest, the accuracy is off significantly for the mean value.

The most precise and consistent data that we obtained seemed to be along the 500 Hz frequency. This followed the expected output that the more iterations we have, the more accurate the reading should be. We can also observe that the mean value obtained for 500 Hz at both 50 iterations and at 100 iterations are both within ten hertz of the actual input



frequency. This is an impressively close result for the device used to obtain the frequency.

Further tests can be done with the actual analog to digital converter to obtain more accurate and faster values. A DC bias would also serve as a good tool for the next tests as this test did not include a DC bias and only took into account the fact that this chip cannot perceive any values below ground, therefore half of the wave would be perceived as zero volts, while the other half would grow from zero, and then tend towards zero volts. This may have caused some issues in the analog to digital converter that we may want to avoid next time.

The information that we obtained from this test that we were looking for is that the algorithm can work, it may need to be adjusted for slower input frequencies, this is because the number of iterations becomes significantly longer in time when the input signal is slower, but the system does show promise and success.

## 8. Packaging

Since functionality of not the only aspect a consumer will judge this product by, we will create a custom casing to allow for protection of the internal electronics and create a professional looking case. In order to create this case, we need to consider what we want to fabricate the enclosure out of in order to ensure that the product stays lightweight, sturdy and affordable. For this we will be looking at the two most common metals used for this type of manufacturing: Aluminum and steel.

The main things we want to compare between these two metals is weight, and price. We will also look into some other aspects of each metal that will aid in the selection.

To start, you can see in table 30 that aluminum is much lighter than steel which will allow the product to stay as lightweight as possible but at an increased price point.

Metal	Weight (lbs/in <sup>3</sup> )	Cost (USD/lbs)
Aluminum	0.09798	0.8865
Steel	0.28356	0.15

Table 30: Metal Cost vs Weight (metalprice.com, coyotesteel.com)

Since this purpose of this project is create a lightweight and inexpensive product, we will need to consider the final cost of the G12 PedalVision in order to make the best decision with regards to which metal to pick for the case of the project. Since weight should not be an issue since most of the weight will be from the electronics so far the biggest deciding factor will be the cost. The trade off will be price versus weight for this choice. Based on our values in table 30 we can see that a change of three times in weight means a difference of almost 6 times with regards to price.

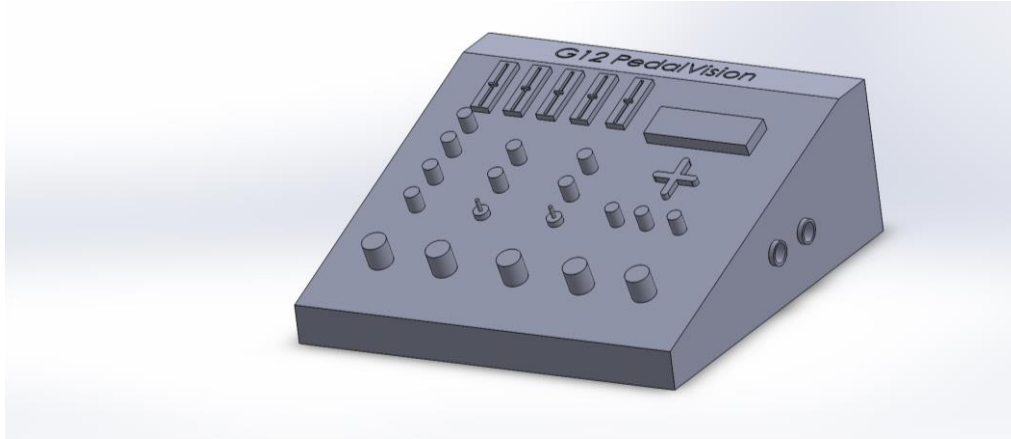


Figure 88: G12 PedalVision Isometric

## 8.1. Front Panel

The front panel of the G12 PedalVision will be where the user has access to all of the controls for each effect. Since this surface will be stepped on in order to toggle effects on and off, we need to consider the spacing between the control knobs and the footswitches. We need enough room between the footswitches and the controls to prevent the user from accidentally turning a control while attempting to toggle an effect, but not so spread out that it increases the size of the G12 PedalVision past reasonable constraints. Also the toggle footswitches will need to be far enough apart to prevent accidental toggling of multiple effects at the same time. These footswitches will be located towards the bottom of the G12 PedalVision to allow for easy unobstructed access. The front panel will have an LED bezel close to the footswitches to allow for a visual indication to the user if an effect is active.

Towards the top right corner of the front panel a display will be placed in order to allow the user to manipulate the digital effects and the controls for the corresponding effect selected. Below this display, three knobs will allow the user to adjust the digital effects settings.

On the left side of the G12 PedalVision are the various controls for the analog effects. Each set of analog effect controls will be laid out vertically from the footswitch to allow for easy association between the effect and their designated controls.

All switches and controls will be labeled using a contrasting color for easy readability. Each control knob will be labeled with their functionality for the analog effect, while the digital effects will be labeled with numbers in order to allow for easy associativity when the user is changing the setting using the display. Each switch will be labeled with the name of the effect to allow for fast recognition at a glance. LEDs will not be labeled as the player will see them light up once the corresponding switch is toggled on.



Figure 89: G12 PedalVision Front Panel

## 8.2. Side Panels

The side panels will give the user access to the inputs and outputs of the G12 PedalVision. On the right side panel of the G12 PedalVision, the user will have access to the input to the G12 PedalVision as well as the “Send” of the external effects interface. On the left side panel of the G12 PedalVision the user will have access to the output of the entire system as well as the “Return” of the external effects interface.

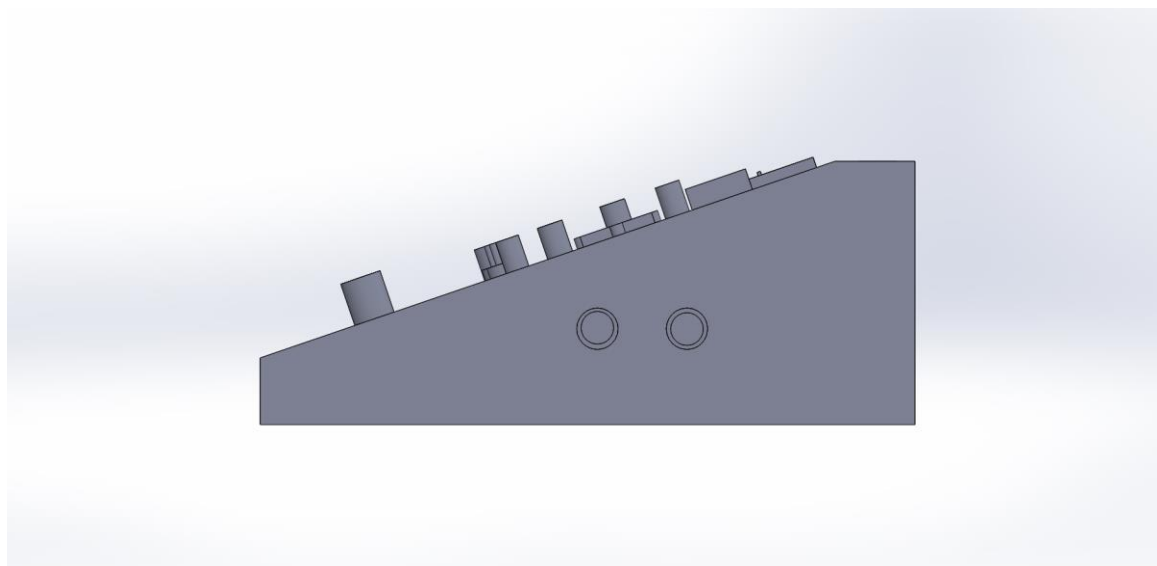


Figure 90: G12 PedalVision Side Panel

## 8.3. Bottom Panel

The bottom panel of the G12 PedalVision will serve as a form of isolation for the product. Rubber feet will be mounted to the bottom panel on the external side in order to prevent sliding when toggling the effects on and of via the footswitches. These rubber

feet will also provide some overall cushion when you “stomp” onto the box, this will in turn help increase the lifespan of the product by reducing the shock this effects unit will experience. The internal side of the bottom panel of the G12 PedalVision will have some risers in order to keep the circuit board from making contact with the bottom panel. We want to prevent contact with the bottom panel in order to prevent accidental shorts due to components or wires from our effects board contacting the metal of the bottom panel. We can also apply a thin insulation pad onto the internal side of the bottom panel in order to further isolate the G12 PedalVision’s electronics.

## **9. User Manual**

The following sections will provide detailed instruction on how to operate the different function of the G12 PedalVision.

### **9.1. Analog Compression**

The analog compression circuit allows for a wide range of different compression tones. The controls you have at your disposal for this compressor are:

- Ratio
- Sustain
- Attack
- Level

**On/Off toggle-** In order to activate the compression effect, depress the footswitch of the compression circuit one time. You will see the LED by the footswitch light up. This light will indicate that the effect is active. To turn the effect off, depress the footswitch once more. The LED should turn off, indicating that the effect is not active.

**Ratio-** This control allows the user to adjust the amount of compression heard at the output. This will work by mixing the input signal with the output signal depending on the set location of the control. With the control turned all the way counter clockwise no compression will be heard. As you turn the control clockwise, more compression will be heard.

**Sustain-** This control will allow the user to increase the amount of sustain heard. Sustain is the amount of amplification applied to the signal as it starts to die away. With the control turned completely counter clock wise, there will be minimal sustain. This minimal sustain will allow lower amplitude signals to fade away faster. As you turn the control clockwise, the more sustain will be added. This change will result in an increase in small signal amplification, thus creating a longer lasting pitch.

**Attack-** This controls how fast the circuit will react to the input signal. By having the control turned all the way counter clockwise, this will equate to a fast attack. As you turn the control clockwise, you will hear a slower attack. The faster the attack setting the more apparent the compression will be.

Level- This controls the total output volume of the effect. With the control turned all the way counter clockwise; no signal will be present at the output. As you turn the control clockwise, more of the total signal will be present at the output. This equates to a louder output.

## 9.2. Analog Distortion

The analog distortion circuit will allow you to get heavy tones and the added versatility of selecting symmetric or asymmetric clipping. The controls you have at your disposal for this compressor are:

- Distortion
- Clipping
- Level

Distortion- This control allows the user to adjust the amount of gain the amplifier applies to the input signal. With the control turned all the way counter clockwise, minimal distortion will be heard. As the control is turned clockwise, more distortion is heard at the output.

Clipping- This control will allow the user to select between symmetric or asymmetric clipping. With the switch in the left position symmetric clipping will be applied. Symmetric clipping will provide a smooth and mellow tone. With the switch in the right position, asymmetric clipping will be heard. Asymmetric clipping will provide a more aggressive and complex tone quality.

Level- This controls the total output volume of the effect. With the control turned all the way counter clockwise; no signal will be present at the output. As you turn the control clockwise, more of the total signal will be present at the output. This equates to a louder output.

On/Off toggle- In order to activate the compression effect, depress the footswitch of the compression circuit one time. You will see the LED by the footswitch light up. This light will indicate that the effect is active. To turn the effect off, depress the footswitch once more. The LED should turn off, indicating that the effect is not active.

## 9.3. Analog Overdrive

The analog overdrive circuit will allow you to get singing tones and the added versatility of selecting symmetric or asymmetric clipping. This is the perfect effect to add volume to any signal. The controls you have at your disposal for this compressor are:

- Gain
- Clipping
- Level

Gain- This control allows the user to adjust the amount of gain the amplifier applies to the input signal. With the control turned all the way counter clockwise, minimal gain will be heard. As the control is turned clockwise, more gain is heard at the output.

Clipping- This control will allow the user to select between symmetric or asymmetric clipping. With the switch in the down position symmetric clipping will be applied. Symmetric clipping will provide a smooth and mellow tone. With the switch in the up position, asymmetric clipping will be heard. Asymmetric clipping will provide a more aggressive and complex tone quality.

Level- This controls the total output volume of the effect. With the control turned all the way counter clockwise; no signal will be present at the output. As you turn the control clockwise, more of the total signal will be present at the output. This equates to a louder output.

On/Off toggle- In order to activate the compression effect, depress the footswitch of the compression circuit one time. You will see the LED by the footswitch light up. This light will indicate that the effect is active. To turn the effect off, depress the footswitch once more. The LED should turn off, indicating that the effect is not active.

## 9.4. Analog Equalization

The equalization circuit will allow the user to shape their tone many different ways. The controls available to them are:

- Filter Selection
- Low
- High
- Boost

Filter Selection- By adjusting the filter selection switch you can change which type of filter is active. With the switch in the first position, you will have the high pass filter selected. This filter will remove lower frequency components. With the switch in the second position you will have selected the low pass filter. This filter will remove high frequency components. With the switch in the third position you will be in bypass mode. This will apply no filtering. With the switch in fourth position, the band reject filter will be selected. This filter will remove middle frequencies creating a “mid-scoop” effect. With the switch in the fifth position you will have the band pass filter selected. This filter will filter the low and high frequencies of the signal creating a “mid-boost” effect.

Low- Low will adjust how much low frequency components will be in your output signal. If the control is completely counter clockwise, there will be less low frequency at the output. As the control is turned clockwise, more low frequencies are seen at the output.

High- High will adjust how much high frequency components will be in your output signal. If the control is completely counter clockwise, there will be less high frequency at the output. As the control is turned clockwise, more high frequencies are seen at the output.

Boost- Boost will allow the user to increase the gain provided by the “mid-boost” filter. When the control is fully counterclockwise, minimal gain will be applied to the “mid-boost” filter. As the control is turned more clockwise, more gain is added to the mid frequencies of your signal. This boost control will allow the user to be able to adjust how much their sound will cut through while playing with a band.

## 9.5. External Effects

The external effects interface will allow the user to connect an outside pedal to the G12 PedalVision. This interface will feed the signal from the analog effect to the outside unit, then back into the G12 PedalVision in order to be further processed. The control available to you is On/Off toggle.

On/Off toggle- In order to activate the external effect interface, place the switch in the upward position. The LED by the switch should light up. This light will indicate that the effect is in SEND mode. In SEND mode, the G12 PedalVision is outputting the analog signal to an external unit. To turn the SEND mode off, place the switch in the downward position. The LED should turn off, indicating that the effect is no longer in SEND mode. This means that the G12 PedalVision will send the analog signal to the digital effects toggle switch as opposed to an external unit.

## 9.6. Digital Effect Interface

On/Off toggle- In order to activate the compression effect, depress the footswitch of the compression circuit one time. You will see the LED by the footswitch light up. This light will indicate that the effect is active. To turn the effect off, depress the footswitch once more. The LED should turn off, indicating that the effect is not active.

Effect Cycle- In order to cycle through the effects that can be loaded into the chip, a button by the LCD display must be pressed. When pressed, the digital effect will cycle to the next available effect.

Settings Knobs- Each of the settings displayed on the LCD screen are labeled and correspond to the knobs. Turn the knobs clockwise to increase the value, and counterclockwise to decrease the value.

## 9.7. Digital Delay

The digital delay unit will allow you to get crisp echoing effects and fast slap back delays. This is the perfect effect to add that “wow” factor to any performance. The controls you have at your disposal for this compressor are:

- Delay
- Feedback
- Level

Delay- This control allows the user to adjust the amount of time between subsequent repetitions of the input signal. With the control turned counter clockwise, minimal delay

will be heard from when the pitch is played to when the pitch is heard again. As the control is turned clockwise, there will be a more noticeable delay time between the original pitch played and the repeated pitch at the output.

Feedback- This control will allow the user to select how fast the repeated signal will be attenuated. With the control turned fully counter clockwise, the volume of the repeated signal will be minimal. As the control is turned clockwise, the subsequent echos will have less attenuation applied to them. This will allow the signal to be repeated indefinitely if the control is fully clockwise.

Level- This controls the total output volume of the effect. With the control turned all the way counter clockwise; no signal will be present at the output. As you turn the control clockwise, more of the total signal will be present at the output. This equates to a louder output.

On/Off toggle- In order to activate the compression effect, depress the footswitch of the compression circuit one time. You will see the LED by the footswitch light up. This light will indicate that the effect is active. To turn the effect off, depress the footswitch once more. The LED should turn off, indicating that the effect is not active.

## 9.8. Digital Reverb

The digital reverb unit will allow you to get spacious and “full” sound without needing natural room acoustics to accomplish this. This will allow the user to sound as if they are in a large concert hall as opposed to a small dry room. The controls you have at your disposal for this compressor are:

- Decay
- Tone
- Level

Decay- This control allows the user to adjust the intensity of the reverb effect. With the control fully counter clockwise, the effect will sound very shallow, and will not have a lot of “dimension” to the overall sound. As the control is turned clockwise the intensity of the “dimension” of the overall sound will increase.

Tone- This control will allow the user to select tone of the effect. With the control fully counter clockwise the output signal will have a darker characteristic to it. This darker characteristic can be related to less higher frequency components in the output signal. As the control is turned more clockwise, more higher frequencies are heard at the output, making the effect sound “brighter.”

Level- This controls the total output volume of the effect. With the control turned all the way counter clockwise; no signal will be present at the output. As you turn the control clockwise, more of the total signal will be present at the output. This equates to a louder output.



On/Off toggle- In order to activate the compression effect, depress the footswitch of the compression circuit one time. You will see the LED by the footswitch light up. This light will indicate that the effect is active. To turn the effect off, depress the footswitch once more. The LED should turn off, indicating that the effect is not active.

## **9.9. Digital Flanger**

The digital flanger unit will allow you to get jet like flanges and outer space-like sound effects. This is the perfect effect to add dimension and uniqueness to any performance. The controls you have at your disposal for this compressor are:

- Delay
- Depth
- Level

Delay- This control allows the user to adjust the rate at which the comb filter will shift. This can be noted by the rate of oscillation of the characteristic “swoosh” sound. With the control full counter clockwise, the effect will be very slow. As the control is turned clockwise, the rate of oscillation will increase.

Depth- This control will allow the user to select how much “dimension” is heard in the output signal. You can think of this as how close and how far away the sound will appear. With the control fully counter clockwise, the effect will sound shallow and relatively “flat.” As the control is turned clockwise, more “depth” or apparent distance will be perceived.

Level- This controls the total output volume of the effect. With the control turned all the way counter clockwise; no signal will be present at the output. As you turn the control clockwise, more of the total signal will be present at the output. This equates to a louder output.

On/Off toggle- In order to activate the compression effect, depress the footswitch of the compression circuit one time. You will see the LED by the footswitch light up. This light will indicate that the effect is active. To turn the effect off, depress the footswitch once more. The LED should turn off, indicating that the effect is not active.

## **10. Administrative Content**

### **10.1. Conclusion**

The research that was completed for this paper was crucial for the authors before they are able to begin building the actual product. As an overview, the research workload was split into four parts, in order to give each group member the ability to focus research in a specific area. Although it was split up, conversation as well as collaboration was done in order to learn more about certain topics. The whole paper was a group effort, and each member was able to learn much from researching their topics.

The workload was split up into different modules. The first module was the analog effects design. This research was completed by Jose Ramirez. Kevin Leone primarily completed research for the digital effects module. Brian Boga completed the research for the LED Matrix, as well as the microcontroller that may be used for the digital effects as well. Ayesha Arif completed research for supplying power to each of the components of the design.

## **10.2. Milestone Discussion**

The initial project idea was selected very early on so role assignments were made relatively quickly based on assignment interest, prior knowledge or experience, an initiative. With the assignments initially assigned we began to research our assigned topics for about a week's worth of time. Since this project has three sub systems that can be run independently, we started to design and research the three sub systems as individual projects. The only exception to this is the input buffer of the analog portion of the project. The reason for this is that for amplification circuits a high input impedance and a low output impedance is required. The high input impedance is required in order to prevent drawing excess power from the source of the input. While the low output impedance is required to allow this project to be able to interface with any other system without causing any loading effects or damaging power draws. After the input signal passes through this buffer, then the three sub systems can be treated as independent from each other. Due to this independence the sub systems were designed in parallel.

With regards to the analog effects sub system, the first effect that was designed was the input buffer stage. The reason for the design of this stage first is to allow for the other effects units and third party products to interface with this device correctly. This design took very little time and was completed within a day. Although a time frame of one day may seem short, this was due to the simplicity of the input buffer design and simple unity gain function which is not responsible for coloring or altering the signal in any way. The next portion of the analog sub system that was researched was the equalization filter network. Research for this equalization filter network too about three days. This time frame was allotted in order to find previous designs and compare different techniques used in order to implement the same type of filters. There was a constant occurrence of tradeoffs between size and versatility being inversely related. In you wanted to reduce the save of the filter, then you needed to remove on of the potentiometers that would allow for you to change the desired attenuation frequencies. Due to the simple filters that comprise the equalization network the designing of the filters was relatively quick, what took some time was deciding between having larger filter area that allowed for more versatility, or smaller filter area and less versatility. Since the purpose of this project is to be versatility, we decided to have a slightly larger footprint but allow more control over the frequencies that the filter network will attenuate.

With the completion of the input buffer and the equalization filter network complete and simulations showing early signs of success, the over drive and distortion circuits were then researched. Due fact that these effect could be implemented in so many ways, a longer period of time was allotted for research. The amount of time spent researching and creating some designs for these two effects was a total of 16 days. The reason for this

amount of time was in order to allow time to research the different methods of implementation. Once research was complete in 10 days, we used the remaining 6 days to come up with the designs for the two effects. Since the overdrive and distortion effects are fairly similar the design process was relatively quick once one of the two effects designs were complete.

The final sub system of the analog system was the compression circuit. The time allotted for the compression circuit was approximately 24 days. Due to the level of complexity of this effect, more time was needed to research the circuit design. Upon research of this effect, not many different designs were found which made it difficult to create a unique circuit. A total of 16 days were used in order to attempt to research the effects functions, then the remaining 8 days were used to simulate various designs of the system. Due to difficulty in simulating the effects, the various parts of the total circuit were designed and simulated and the final circuit was physically implemented and probed on the oscilloscope using a breadboard test to verify complete functionality.

With the main components of the analog system complete, an additional sub system in the analog unit was added in order to allow for third party effects units to modify the output signal of the analog sub system of this project. This additional feature allows to external processing to be done before feeding the signal into the digital unit or the LED matrix later in signal chain of this project. Since all that was needed for this feature was a switch and a second input buffer, the research and design of this section was limited to 2 days. The system was then tested shortly afterwards.

The analog to digital conversion module was a very important part to the system. This module enables the input signal of the guitar to get converted to a signal that is understood by the DSP chips. If the conversion is incorrect or not precise, the signal that ends up in the output will become different from the desired sound, even if the algorithm for the digital signal processing is correct. This module was scheduled for a week of research on different options of how the ADC can be completed, two days for design of the circuit, and about a week for ordering and receiving parts. The testing was scheduled for four days since the process for testing is fairly simple. All that needed to be done was to see if the sound changes when a signal is passed through it, back to analog, then to the output. If the sound is the same, then it is working correctly.

The digital signal processing module is the key component to creating the digital effects. This module has multiple parts to be scheduled since each digital effect that is implemented is created differently. The process for modifying the sound is similar in a few ways, but different in its overall structure. For scheduling the module, the first thing needed to be completed is research, which is scheduled for two weeks. The first digital effect to be implemented is the delay/echo effect, which is scheduled for 5 days. Because the process for creating the other effects uses the basic ideas of the delay/echo effect, the implementation is scheduled for 3 days each. Testing of these effects cannot be completed until an interface is implemented. Once the interface is created the testing is scheduled for 4 days.

## Group 12 G12 PedalVision

Once all effects are implemented, there is still need of an interface to change the values of each effect in the DSP chip. Without the interface, loading the DSP chips with the correct algorithm to be run on the digital signal will be very difficult. Because of this research for this is done partially in parallel with research on the processing techniques, and is scheduled for about two weeks. This way once the DSP research is complete, the interface is scheduled to be completed around the same time. This allows testing to begin almost immediately. Implementation of the interface is scheduled to take a week, as well as another 5 days of testing and debugging the entire system.

The digital to analog conversion module was also a very important part to the system. This module enables the output signal of the DSP chip to get converted to a signal that is able to be passed to an amplifier and create the sound. If the conversion is incorrect or not precise, the signal that ends up in the output will become different from the desired sound. This module was scheduled for a week of research on different options of how the DAC can be completed, two days for design of the circuit, and about a week for ordering and receiving parts. The scheduled time is also done in parallel with time scheduled for the ADC module. The testing was scheduled for four days since the process for testing is fairly simple. All that needed to be done was to see if the sound changes when a signal is passed to analog, then to the output. If the sound is the same as the input, then it is working correctly.

The LED Matrix Feedback Display system was one that took many hours of research. Several weeks were spent sifting through countless parts on the Texas instruments website along with the Digikey website. Finding a chip to help us achieve our goal of fully controllable LED Matrix with the ability to read incoming signal and turn those readings into meaningful outputs was key. Many chips were viewed as able to get the job done, but none stood out as the chip we wanted to make this part of the project perfect. The proper chips needed to get the job done of course, but more than that, they needed to be able to get it done quickly, efficiently and with room to grow.

After finally finding the chips we wanted to use we shifted focus to the proper LEDs. This task was much easier and only took a solid weeks work to come to a decision. This decision was balancing on the intensity of the LEDs along with the power consumption. After battling for a short while on whether or not to go with a 2 milliamp LED or a 20 milliamp LED, our team decided that we would rather have a larger than life feel to the display, than have a product that is power friendly. With this in mind we selected the 20 milliamp LED knowing that it would provide us with a much better visual effect at a cost of power.

Now that all of the main chips had been selected, it was time to order all of the selected parts. This was probably the longest process of the entire project. Several of the chips selected were on back order, and therefore we had to wait several weeks for the chips to be shipped to us. On receipt of the ordered parts, an inspection was completed to ensure that all parts had arrived safely and were as expected. It was at this time that we could actually see how small some of these components were and that we would have some

issues testing some of the chips with contacts that were very small and close. These chips would require a breakout board to fully test.

With no time to design and produce a breakout board for some of the components before the production of this document, we began testing the LEDs and the frequency obtaining code for the ATmega2561. The LEDs worked flawlessly as we expected they would. This testing was able to be completed within one days' time.

The frequency code calculation showed some trouble originally, not wanting to output values that made sense to us, but after spending approximately 5 hours troubleshooting the code and rewriting the code, we were able to start obtaining outputs that were logical. These outputs were then logged for later interpretation. The data was found to support our theory of operation for finding the frequency of the input signal. While some test inputs proved more precise than others, all still showed a high level of accuracy.

Currently a printed circuit board design is being created for the breakout boards, along with the LED Matrix Feedback Display board.

The power supply had taken a few weeks to research. Most the hours spent focused on decided how to distribute the power. The different types of linear regulators were researched and a few were chosen from DigiKey and considered for the project. After speaking to peers and doing some more research, it was decided that a switching regulator would be used for this project. A few different chips, found on DigiKey, were researched and the one was chosen. The switching regulator may cause too much noise, how to fix that is yet to be determined.

Final power decisions are going to be made when exact components are chosen for the project. After the components are known, the power distribution can be decided and the total amount of power required by the PCB can be determined. After these specifications are able to be estimated, there aren't any problems that can be foreseen in the near future.

### **10.3. Budget**

The budget of this unit was more easily identified once the main sections of the project were designed. As each section was completed, a look up of potential parts could be done in order to compare the prices of the potential parts that could accomplish the desired task. In order to try to create a product that was at a competitive price point as compared to other products on the market we first estimated the cost of the parts we thought we would use. The as the main sub systems were designed we documented the actual price of the necessary parts. The initial cost estimate allowed for good initial budget projection but it was not until the parts where actually acquired that we had a clear vision of the end budget. Of course prototyping did increase the cost of developing this product but the total cost of the parts actually used for this product are showing that we should be on our way to staying under the \$500 price tag that comes with purchasing a fully digital product available on the market today. Table 32 below shows the cost breakdown of the G12 PedalVision.

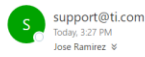
Group 12  
G12 PedalVision

<b>Part</b>	<b>Description</b>	<b>Vendor</b>	<b>Quantity</b>	<b>Price Per Unit</b>	<b>Price Total</b>
<b>OPA1641</b>	Single Channel Op amp	Texas Instruments	4	\$2.88	\$11.52
<b>OPA1642</b>	Dual Channel Op amp	Texas Instruments	1	\$4.20	\$4.20
<b>LM13700</b>	Dual Channel Transconductance Amplifier	Texas Instruments	1	\$1.36	\$1.36
<b>Diodes</b>		Digikey	8	\$0.10	\$0.80
<b>3PDT</b>	Footswitch	Reverb.com	5	\$3.59	\$17.95
<b>Misc</b>	Case		1	\$50	\$50
	Potentiometers	Digikey	12	\$5.34	\$64
<b>PCM3060</b>	ADC/DAC	Texas Instruments	3	\$6.15	\$18.45
<b>TMS320C6720</b>	DSP	Texas Instruments	2	\$15.95	\$31.90
<b>RRLCD204WB</b>	LCD Display	RioRand	1	\$10.99	\$10.99
<b>ATmega2561</b>	Microcontroller	Atmel	1	\$17.08	\$17.08
<b>TLC 5955</b>	LED Driver	Texas Instruments	4	\$6.02	\$24.08
<b>3.5x2.8 LED</b>	<b>BGR</b> LED SMD	Kingsbright	64	\$0.38	\$24.32
<b>MC34063</b>	Switching Regulator	Texas Instruments	2	\$0.63	\$1.26
<b>PCB</b>	Printed Circuit Board	PCBWay	2	\$30	\$60
<b>Total</b>					\$337.91

Table 31: Budget

# Appendix A – Permissions

RE: GEN, Email Technical Support, www.ti.com, N/A 1-2454463518



support@ti.com  
Today, 3:27 PM  
Jose Ramirez

Reply all

Inbox

Thank you for contacting the Texas Instruments Semiconductor Customer Support Center. This is an automatic acknowledgement of the receipt of your email. Your inquiry has been assigned Service Request# 1-2454463518 and is currently being worked by our Technical Support Staff.

If additional information becomes available or you have file attachments that you wish to submit, please reply to this email with that information and/or file(s). Please ensure the thread ID at the end of this email is included so your reply is automatically linked to your original request.

**Please note:** If there is no Service Request # listed above, then we are manually processing your inquiry.

Please consider using the forums at the E2E community site for assistance from the entire community, both Tiers and users. <http://e2e.ti.com/>

For helpful articles on development with TI processor products, please visit the TI EP Wiki at [http://processors.wiki.ti.com/index.php/Main\\_Page](http://processors.wiki.ti.com/index.php/Main_Page)

Looking for more information on our DSP and Analog products? We are offering free DSP and Analog Seminars, which deliver more information than ever before to make your job easier. For more information, go to <http://focus.ti.com/docs/training/traininghomepage.html>

Regards,  
Texas Instruments  
Semiconductor Technical Support  
<http://support.ti.com/>

[THREAD ID: 1-14LB07]  
[SR THREAD ID: 1-14LB066]  
[SYS: Acknowledged]

-----Original Message-----

From: jramirez94@knights.ucf.edu  
Sent: 11/20/2016 02:25:30 PM  
To: support@ti.com  
Subject: GEN, Email Technical Support, www.ti.com, N/A

[This Email Sent From: Email Technical Support  
<http://www.ti.com/guidesupport/docs/contactus.jsp>]

[This Email Sent From: Email Technical Support  
<http://www.ti.com/guidesupport/docs/contactus.jsp>]

[wfsegen]

[DATE / TIME (UTC): Sun, 20 Nov 2016 20:26:44 GMT]  
[CUSTOMER'S REGIONAL LOCAL TIME: 11/20/2016, 3:26:44 PM]

[Name: Jose Ramirez]

[Prefix: Mr.]  
[First Name: Jose]  
[Last Name: Ramirez]  
[Job Title: ]  
[Company: University of Central Florida]  
[Email: jramirez94@knights.ucf.edu]  
[Phone: 4077211070]  
[FAX: ]  
[Country: USA]  
[Address1: 16038 Birchwood Way]  
[Address2: ]  
[City: Orlando]  
[State: FL]  
[Postal Code: 32828]  
[Part# or Description: N/A]  
[Category: General Information]  
[Application: Consumer Electronics]  
[Design Stage: Prototype]  
[Estimated Annual Production: 1 units]  
[Production Date: Spring 2017]

[Problem:

I am a student at the University of Central Florida currently working on a senior design project scheduled to be completed in the spring of 2017. I would like to request permission to use the basic schematic of the voltage controlled amplifier circuit in the datasheet of the LM13700 as a starting point for our design. I would also like to request permission to discuss the different parameters of the following amplifiers in our documentation: OPA164x, OPA827, TL07xx, and the LM13700. ]

[Steps Needed to Recreate Problem:

N/A

]

Re: Permission to use image



ElectroSmash Team <electrosmash.com@gmail.com>  
Today, 4:06 PM  
Jose Ramirez: info@electrosmash.com

Reply all

Inbox

Hi Jose,

Of course you have our permission to use any of our images in your project. Good luck with it and all the best.

Ray,  
ElectroSmash Team.

On Sun, Nov 20, 2016 at 8:37 PM, Jose Ramirez <jramirez94@knights.ucf.edu> wrote:  
Greetings,

I am a University of Central Florida student currently working on a senior design project. I would like to use your JFET Bypass Switching image in my documentation as a reference design. Would it be possible for me to display this image in the documentation? The image I would like to use is attached.

Sincerely,

Jose Ramirez

# Group 12 G12 PedalVision



Kevin Leone | 'copyright@counsel@list.ti.com'  
Request for Permission to use Information in Scholarly paper

I am Kevin Leone, a Computer Engineering student at the University of Central Florida. I am currently in the process of writing a Senior research paper with my group for Analog and Digital Signal processing for audio applications.

I was wondering if we could use the images listed below:

- Image "Figure 1: Analog-to-digital conversion" from Literature Number: SNA098.
- Graphs in "Figure 13. Interpolation Example" from Literature Number: SLA523A.
- Pin Diagram in "Figure 2-9. 144-Pin Low-Profile Quad Flatpack (RFP Suffix)—Top View" from Literature Number: SPR370E.
- Pin Diagram in "Figure 33 illustrates typical circuit connection" from Literature Number: SLA533B.
- "PCM3060 PW (TSSOP) PACKAGE (TOP VIEW)" from Literature Number: SLA533B.

I was also wondering if we could use information from the datasheets and literature numbers below:

- Literature Number: SNA098 (ADC12EU050 Continuous-Time Sigma-Delta ADCs)
- Literature Number: SLA523A (High Speed, Digital to Analog Converters Basics)
- Literature Number: SPR370E (TMS320C6720 Datasheet)
- Literature Number: SLA533B (PCM3060 Datasheet)
- Literature Number: SBOS484D (OPA164x Datasheet)
- Literature Number: SBOS376I (OPA827 Datasheet)
- Literature Number: SLOS080M (TLO7xx Datasheet)
- Literature Number: SNOSBW2F (LM13700 Datasheet)

We will be making one hard copy, and one digital copy of our document to be turned in to our professor.

The images and information will be cited properly, and intended to use to aid in implementation of the chips in our system.

Thank you,  
Kevin Leone



Reply Reply All Forward

Kevin Leone | 'douglas@music.columbia.edu'

1:29 PM

Request to use Figure for Scholarly Paper

Hello Douglas,

I am Kevin Leone, a Computer Engineering student at the University of Central Florida. I am currently in the process of writing part of a Senior research paper for Digital Signal processing for audio applications and I was wondering if I could use one of the graphs presented in the "Music and Computers, A Theoretical and Historical Approach." The graph I would like to ask permission to use is Chapter 5, Section 5.2, Figure 5.5.

Here is the link to the page for quick access. [http://cmc.music.columbia.edu/musicandcomputers/chapter5/05\\_02.php](http://cmc.music.columbia.edu/musicandcomputers/chapter5/05_02.php)

Thank you,  
Kevin Leone

## Request to use Figures for Scholarly Article



GM Arts <gmarts@chariot.net.au>

Today, 12:44 AM

Kevin Leone



Reply all

You're welcome to use those Kevin, and thanks for asking.

Best wishes with your research.

GM Ars

**From:** Kevin Leone [<mailto:kjleone19@Knights.ucf.edu>]

**Sent:** Saturday, 3 December 2016 6:26 AM

**To:** [info@gmarts.org](mailto:info@gmarts.org)

**Subject:** Request to use Figures for Scholarly Article

Hello,

I am Kevin Leone, a Computer Engineering student at the University of Central Florida. I am currently in the process of writing part of a Senior research paper for Digital Signal processing for audio applications and I was wondering if I could use one of the Frequency Response Diagrams for the Phasing and Flanger guitar effects presented in the article at URL <http://www.gmarts.org/index.php?go=221#Phs>.

The figures will be properly cited.

Thank you,  
Kevin Leone



 Reply  Reply All  Forward



Kevin Leone | 'enquiries@dkimages.com'

3:17 PM

Request to use Image in Scholarly Paper

Hello,

I am Kevin Leone, a Computer Engineering student at the University of Central Florida. I am currently in the process of writing part of a Senior research paper for Digital Signal processing for audio applications and I was wondering if I could use Echo visual representation presented in the article at URL <http://www.dkfindout.com/uk/science/sound/echoes/>.

The figure will be properly cited.

Thank you,  
Kevin Leone



Request for use of Example code in Scholarly paper



julius.smith@gmail.com on behalf of Julius Smith <jos@ccrma.stanford.edu>

Sat 12/3, 7:54 PM

Kevin Leone

  Reply all

Hi Kevin,

No problem - have fun with your project

Cheers,  
- Julius

On Fri, Dec 2, 2016 at 8:01 PM, <kjleone19@knights.ucf.edu> wrote:

> Hello,

>

> I am Kevin Leone, a Computer Engineering student at the University of Central Florida. I am currently in the process of writing part of a Senior research paper for Digital Signal processing for audio applications and I was wondering if I could use the example C code from [https://ccrma.stanford.edu/~jos/Delay/Delay\\_Line\\_C.html](https://ccrma.stanford.edu/~jos/Delay/Delay_Line_C.html) as a starting point for my code. It will be used as an explanation for the design of a delay line which will be used for different digital effects.

>

> The code will be properly cited.

>

> Thank you,

> Kevin Leone

--

Julius O. Smith III <jos@ccrma.stanford.edu>  
Professor of Music and, by courtesy, Electrical Engineering  
CCRMA, Stanford University  
<http://ccrma.stanford.edu/~jos/>

Group 12  
G12 PedalVision

To  info-en@wikimedia.org x B

Cc

Image Permission Request

To whom this may concern,

I am a student at the University of Central Florida. I am requesting permission to use the image of the basic schematic of a LDO Regulator.

<https://commons.wikimedia.org/w/index.php?curid=22345641>

Thank you,  
Ayesha Arif

Cc

Image Permission Request

To whom this may concern,

I am a student at the University of Central Florida. I am requesting to use a few of the images found in datasheets in my senior design paper. The images are:

- 1) Literature Number: SNVA558; Figures 1 and 2
- 2) Literature Number: SNVS767E; Figure 1

Thank you,  
Ayesha Arif

To  sales-us@maximintegrated.com Bcc

Cc

Image Permission Request

To whom this may concern,

I am a student at the University of Central Florida. I am seeking permission to use an image from your website for my Senior Design Project.

The image I am requesting permission for is:

1.) basic switching regulator

<https://www.maximintegrated.com/en/app-notes/index.mvp/id/2031>

Thank you,

Ayesha Arif



Ayesha Arif

Today, 3:49 PM

info-en@wikimedia.org

  Reply all | 

To whom this may concern,

I am a student at the University of Central Florida. I wanted to ask permission for using a few of the images found on your site.

1.) The ideal transformer and induction law image.

<https://en.wikipedia.org/wiki/Transformer>

2.) The single-phase tapped autotransformer with output voltage range of 40%-115%.

<https://en.wikipedia.org/wiki/Autotransformer>

Thank you,

Ayesha Arif

## Appendix B – Datasheets

Device	Manufacturer	Literature Number
OPA164x	Texas Instruments	SBOS484D
OPA827	Texas Instruments	SBOS376I
TL072	Texas Instruments	SLOS080M
LM13700	Texas Instruments	SNOSBW2F
TMS320C6720	Texas Instruments	SNAA098
PCM3060	Texas Instruments	SLAS533B
TLC3955	Texas Instruments	SBVS237

Table 32: Datasheets

## Appendix C – References

- [1] "Chapter 5: The Transformation of Sound by Computer." Music and Computers. Music and Computers, n.d. Web. 05 Dec. 2016.
- [2] "Continuous-Time Sigma-Delta A/D Conversion." Springer Series in Advanced Microelectronics (2006): n. pag. Texas Instruments, 2011. Web. 05 Dec. 2016.
- [3] Marshall, Dave. "DAFX: Digital Audio Effects." (2011): n. pag. 10\_CM0268\_Audio\_FX. Cardiff University. Web. 5 Dec. 2016.
- [4] "Embedded DSP Processor Design." (2008): n. pag. Berkeley Design Technology, 2000. Web. 05 Dec. 2016.
- [5] "File:Low Drop Voltage Regulator.png." File:Low Drop Voltage Regulator.png - Wikimedia Commons. N.p., 2007. Web. 05 Dec. 2016.
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- [7] "Hi-Tech C I2C Master Example Code." Hi-Tech C I2C Master Example Code. HobbyTronics, 2011. Web. 05 Dec. 2016.
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