Pocket Amp

Group #4

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1. Executive Summary

This project developed from a need to meet the specialties of our various group members. Most of the electrical engineering members specialize primarily in the analog designs which a guitar amplifier would utilize. This project is heavy on circuit design with little to no moving parts, which lends to the abilities of all group members. However, other components of this project’s design require the expertise of a computer engineer such as control via a smartphone application over Bluetooth.

Our initial goal for this project is to pass Senior Design I with a grade of an A. Throughout the course of the project we want to gain insight in the design process. This project allows us to branch out from our classes and gain a different type of hands-on experience that we do not get to practice in labs. Our specific goal for the project is to design and build a lightweight, portable guitar amplifier which uses external speakers or headphones. We want to implement an easy-to-use user interface system for adjusting the amplifier’s controls. We would like to enable Bluetooth communications for interfacing with external devices.

In this day and age electronic devices are inseparable to Americans everywhere. The amount of electronics that are becoming day-to-day necessities in the lives of Americans is growing. Many of these devices started as compact versions of larger counterparts. Computers placed much of their functionality into smart phones, CD players were compressed into mp3 players, the trend continues with many other electronics. This project looks to continue this tendency by making a guitar amplifier pocket-sized.

Music is an ever-present part of many people’s lives. Many of the greatest inventions of the past century were quickly used for the storage and greater portability of music. Storable music has been compressed exponentially from Vinyl records to cassette tapes to CD and finally digital files. However, the portable nature of live, playable music has been slower to follow this trend. Even for electric guitars, while the guitar itself is largely portable, it's necessary additions, like amplifiers can be quite bulky. This bulkiness can be rectified with the use of the Pocket Amplifier. Musicians will be able to use the Pocket Amplifier to keep their music on-the-go with their busy lifestyles.

Traditional amplifiers rely on their own in-built speakers in order for the music produced by the guitar to be heard. While these speakers are useful when performing or for testing the acoustics of a room they add significant bulk to the amplifier. True musicians spend far more time practicing than they do performing, which the Pocket Amp is perfect for. By simply connecting their favorite pair of headphones with the Pocket Amp, musicians can practice their guitar skills and different amplifier effects with a portable setup. The use of headphones can allow the musician to practice in locations where the volume of their music would be prohibited. While headphones allow the musician to play privately, by simply connecting an external speaker, the musician can quickly be ready for public performance.

The Pocket Amp will also help musicians in its cost. Traditional amplifiers use vacuum tubes, large speakers, requiring large power sources, with fewer digital components. These impressive components can combine to form a very expensive product. The Pocket Amp is made primarily of transistor components, allowing a portable and cheaper product.

The Pocket Amplifier also brings amplifiers into the 2010's with the use of its companion app. The Pocket Amplifier needs far fewer external controls because its features and audio options can be altered via Bluetooth. Apps are becoming such a frequent occurrence these days that newer musicians can more easily familiarize themselves with the abilities of guitar amps.

The traditional guitar amp will always have a place for musicians seeking to play in public for large audiences. However, the societal and technological trend of compressing technology and making it more portable is not being met by traditional amplifiers. The Pocket Amp can improve the day-to-day function of musicians by being portable, private, comparatively low-cost, and easy-to-use.

Goals

- To earn a grade of A in Senior Design I and II
- To gain practical knowledge of real circuit design
- To gain valuable insight into the research process of product design
- To learn insights on troubleshooting our circuits.
- To learn how to implement our breadboard circuits onto a PCB circuit board.
- To design and build a portable and user friendly product
- To be able to fit the Pocket Amp within an actual pocket
- To produce a variety of guitar effects utilizing the Pocket Amp, including Overdrive, Tone Control, Reverb and other effects.
- To produce the various guitar effects with both digital and analog circuits.
- To provide an easy and user-friendly phone app, which controls the various options for the Pocket Amp
- To communicate with the phone app through Bluetooth
- To utilize a rechargeable battery system to power the Pocket Amp's various components

2. Specifications and Requirements

This section discusses the imposed requirements placed upon the project by its design, the constraints placed upon the project by industry and market standards and the specifications of this project that these requirements and constraints result in.

2.1 Requirements

The requirements for the Pocket Amp were derived from the research done in this document. Table 1 below outlines these requirements.

ID	Type	Requirement
E.1	External	1-hour battery life
E.2	External	Fit in a 5x6x3 inch pocket
E.3	External	Weigh less than 5lb
E.4	External	Have a 6.35mm “phone jack” audio input jack
E.5	External	Have a 3.5mm “mini phone jack” audio output jack
E.6	External	Will NOT have an onboard speaker
E.7	External	Will have an external charging port to charge the battery
I.1	Internal	Power a pair of headphones
I.2	Internal	Use an onboard MSP432 MCU to handle digital signal processing
I.3	Internal	Use MSP432 built-in ADC
I.4	Internal	Amplify the standard range of guitar and bass guitar sounds
I.5	Internal	Sample at least 12-bits with a frequency of at least 30kHz
I.6	Internal	Introduce less than 1% total harmonic distortion
F.1	Effects	Implement digital delay effects
F.2	Effects	Implement digital reverb effects
F.3	Effects	Implement digital echo effects
F.4	Effects	Implement digital flanger effects
F.5	Effects	Implement digital chorus effects
F.6	Effects	Implement analog distortion effects
F.7	Effects	Implement analog overdrive effects
F.8	Effects	Implement analog tone control
F.9	Effects	Implement analog volume control
U.1	UI	Communicate through Bluetooth with smartphone application
U.2	UI	App will be developed for Android devices
U.3	UI	App will control effects
U.4	UI	App will control volume
A.1	Admin	Unit production hardware cost will be less than $100
A.2	Admin	Unit prototype will be finished by the end of Spring 2017

Table 1. Requirements of Pocket Amp and App

The requirements cover physical external requirements, electrical internal requirements, guitar effect requirements, requirements pertaining to how the design interfaces with its user and other devices, and requirements for administrative design. The first category of requirement in Table 1 is external requirements. E.1 gives us a requirement of at least a 1-hour battery life of usage after being fully charged which is enough for the Pocket Amp to be useful as a portable amplifier but doesn't require expensive, large or heavy battery banks. E.2 and E.3 define the size and weight of the Pocket Amp, which will make it able to fit within a large-sized pocket (such as cargo pants or a small pocket in a backpack or purse). E.4 and E.5 define the physical input and output requirements of the Pocket Amp, namely having a 6.35mm "phone jack" type audio input from the guitar and outputting to a 3.5mm "mini phone jack" type audio output. These plug types were chosen since they are the de facto standard for their respective uses. Phone jacks are by far the most common guitar connector type and mini phone jacks are the standard for headphone output, to the point that there is generally no need to specify what kind of connector your guitar or headphones use. The next external requirement, E.6, specifies that there will not be a speaker on the Pocket Amp since the output will be headphone output. The final external requirement is that the Pocket Amp will have an external charging port to charge the battery with.

The next category of requirements is internal requirements for the Pocket Amp. I.1 through I.6 are requirements for the internals of the system. I.1 requires the Pocket Amp be able to power a pair of headphones. I.2 specifies that an MSP432 MCU will be used for all digital signal processing and I.3 specifies that the MSP432's onboard ADC will be used to sample the analog signal. I.4 specifies that the Pocket Amp will be able to handle the standard range of guitar and bass sounds, which range from 41Hz (Lowest E string common to bass guitars) up to approximately 5 kHz to 8 kHz harmonics. This is just the limit of the electrical response; I.5 specifies that the system must at least sample at 12-bits of audio depth and at least 30 kHz sampling frequency. The final internal requirement is that the Pocket Amp will introduce less than 1% total harmonic distortion upon the signal.

The next category is the guitar effects category, cataloging all the effects that the Pocket Amp will implement. F.1 through F.4 list digital effects to be implemented, while the remaining requirements list analog effects to be implemented. F.1 specifies the first digital effect, delay – just a time delay from input to output. The second effect, F.2, is reverb – the original sample mixed with very short delays to imitate the echo of a large room. F.3 is echo, which is the original signal mixed with a long delayed signal. F.4 is a digital flanger effect which is the signal with the same signal delayed a varying amount that's small enough that the signal's waves interfere. F.5 through F.8 are analog effects. F.5 and F.6 require that the Pocket Amp implement analog distortion and analog overdrive, respectively. F.7 requires the Pocket Amp to implement tone control to allow the user to adjust volume levels to equalize the sound across the spectrum. The final effect, F.8, is that the Pocket Amp will have analog volume control.

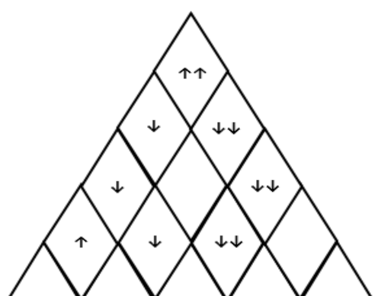
The next category is user interface requirements. This specifies how the users will be able to interact with the Pocket Amp. U.1 requires the App and Pocket Amp communicate through Bluetooth. The next requirement is that the app will be developed for the Android OS, largely due to the ease of developing applications for Android and the computer engineer's experience with Android. U.3 and U.4 require that the app will control effects and volume respectively. The final category is administrative, with simply specifying the unit cost of less than $100 and the final prototype deadline of the end of the Spring 2017 semester.

2.2 Specifications

The following specifications are selected based upon engineering and market requirements.

2.2.1 House of Quality

Table 2 below, represents the tradeoffs both between marketing requirements (sound quality, ease of use, portability and cost) and engineering requirements (efficiency, dimensions, cost, total harmonic distortion, and power output) and between individual engineering requirements.



		Efficiency	Dimensions	Cost	Total Harmonic Distortion	Power Output
		+	-	-	-	+
		>75%		<$100	<1%	<1W
Sound Quality	+	↑		↓↓	↑↑	↑
Ease of Use	+			↓		↑
Portability	+		↑↑		↓	↓↓
Cost	-	↓	↓	↑↑	↓	↓↓

Table 2. House of Quality

↑ represents a moderate positive correspondence
↑↑ represents a strong positive correspondence
↓ represents a moderate negative correspondence
↓↓ represents a strong negative correspondence

2.2.2 Block Diagrams

The high-level block diagram in Figure 1 shows the entire Pocket Amp system and how the major components will interact, along with how it will interact with the external components. This initial fundamental block diagram performs three main functions: integration, assignment and reduction. The block diagram integrates the different parts of the project into an understandable format. By outlining the different components, the diagram allowed the group members to understand how and where the components interact with one another. Next, the group utilizes the diagram to separate those different components and assign each to a different member. This leads to the third function of reduction. By assigning a different function to each member, the huge task of the project can be reduced to a manageable task that each person can accomplish and then integrate together for the final project.

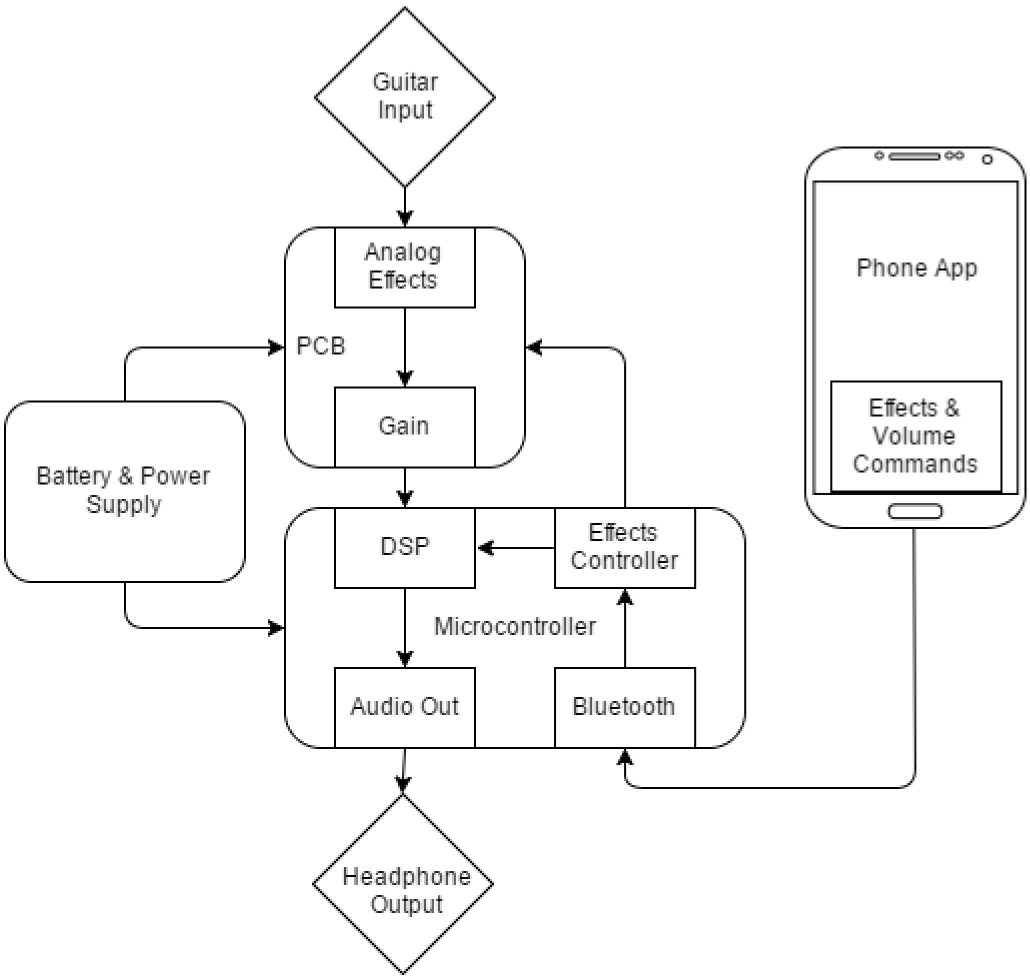


Figure 1. High Level Block Diagram

3. Research

In researching technologies for the Pocket Amp, we researched everything from guitar amplifier products, batteries, regulators, analog amplifiers, digital signal processing, etc. Everything that will be required for the construction of a Pocket Amp was researched to varying depth and written about. This section is a compendium of the knowledge we have attained through our research.

3.1 Similar Products

The first step to building a product is to identify a need, which requires a survey of what products are already on the market. This section identifies five product types which are relevant to the Pocket Amp's market. Modelling Amplifiers due to their DSP-heavy effects, portable amplifiers due to their size and battery power, phone app amplifiers due to our product being reliant on an app, guitars with built-in amplifiers due to the amplifier size, and effects pedals due to both size and form factor.

3.1.1 Modelling Amplifiers

Modelling amplifiers are modern digital amplifiers, pedals, sound boards, or software that is designed to model previous, generally vacuum tube driven, amplifiers or setups. Every amplifier setup has certain built-in tonal and harmonic characteristics in addition to the standard equalization and gain options that are usually connected to amplifiers. This is especially true with analog amplifiers due to their reliance on vacuum tubes to imperfectly amplify input signals. The tonal differences are also created by any systems upstream or downstream of the amplifiers itself, including pedal and effect boxes or the speakers themselves. Modelling amplifiers attempt to digitally recreate all the tonal features for many different setups for just a few hundred dollars – even when the original setups cost several thousand dollars to conceivably over a hundred thousand dollars. Strictly speaking, a modelling amplifier just requires basic DSP with presets that model a different setup. We will look at and compare multiple a low-end and high-end amplifier here to see what features are common.

The Fender Mustang III V.2 is a modelling amplifier built for more entry-level applications than most, coming in with a price tag of only $330. The main selling feature for this amp, like most amps, is the built-in speaker. The Mustang III features 17 amplifier models and 100 presets of effect and models. It also allows users to change effects on top of each amplifier model, allowing thousands of possible setups. The Mustang III measures 18x21x11 inches, though a large portion of this is the 12-inch speaker and the associated empty space required to house a speaker cone of that size and having a number of external controls greatly increases the amount of space required by the electronics.

The Kemper Profiler is a high-end profiling amplifier without a built-in speaker. This reduces the added-in tonal qualities brought in by playing through speakers. The Profiler is generally hooked up to a custom sound system or is used in studios to directly record off of. The Kemper Profiler achieves the modelling with high quality ADC's followed by powerful microcontrollers able to perform a wide range of time, frequency and tonal modifications on a live signal. The Profiler comes with hundreds of factory-set "rigs" which model everything from effect pedals, preamp, amplifiers, to the output speakers. In addition, the Profiler can be updated and more rigs can be added or current rigs modified by the users. The Kemper Profiler costs approximately $1800 to $2000 factory new, or $1600 to $1800 for a used profiler. While the Profiler has some similarity to our product since it is fairly small for band equipment, not having a speaker, and being almost entirely computer controlled, the Profiler is geared towards studios and studio-quality gigs and as such comes with a studio-quality price tag.

While modelling amplifiers are similar to our product, our main focus will not be in capturing audio setups from real world applications. We will allow the user create setups with what options are available to them. Since our output device will vary greatly it's also conceivable that users will have to tailor setups to match their setups. For example, studio-quality headphones, cheap headphones, and small speakers may need completely different tonal settings in order to create the sound the user wants. The main similarity between modelling amplifiers and the Pocket Amp is DSP-heavy applications handling effects and the size of some modelling amplifiers due to the reliance on software and MCUs.

3.1.2 Portable Amplifiers

Portable amplifiers are generally divided into two types, those with batteries and those that require an outlet. Outlet-powered portable amplifiers are generally just smaller amplifiers – usually approximately two cubic feet large – that generally are lighter. In addition, portable "tethered" amplifiers can sometimes have internal storage for cords and a more robust construction due to the need to move them constantly. Battery powered portable amplifiers provide a lot more mobility and are more closely related to our Pocket Amp, so research was focused on what these amplifiers provide.

A classic battery powered portable amplifier is the Pignose 7-100. The Pignose amplifier was invented in 1969 and is still popular in some music circles. The amp features a port for 6 AA batteries, a 5-inch speaker, a plug for AC current, an on-off switch, a volume control, and nothing else. While the Pignose can be run off AC or battery power, it is not rechargeable since rechargeable batteries back when it was invented were not nearly the quality we have access to now. Many Pignoses also feature two knobs to fit a guitar strap to – some musicians would carry it around with them while they were playing. Pignoses technically do not have any effects, but a distortion or overdrive effect is possible by maximizing the volume from the guitar and turning down the volume on the Pignose. This drives the internal amplifier to its rail, producing the overdrive effect. Many Pignoses also have a phone jack in the back allowing them to be used as an effect box or have audio recorded directly from them, as many musicians came to like the sound these tiny amplifiers produce.

A more modern portable amplifier is the Roland Cube series, available in a wide range of sizes and configurations. The Cube series was design for travel, all of the amplifiers in this series have a rugged and sturdy construction and are small for their power output. The largest of the series is the Cube-80, with an 80-watt driver on a 12-inch speaker cone, though this and many of the other larger Cubes require AC input. The Cube line has two sets of products which do boast battery power, which are the Micro-Cube and the Cube Street. Micro Cubes are small, cubic amplifiers with either a single 3-watt, 5-inch speaker or 4 4-inch speakers driven by 5 watts. The Micro Cubes feature can model 8 different amplifiers and contains 8 DSP-based effects in addition to equalizer controls and a tuner. The Street Cube is not a cube, but is a trapezoidal prism with speaker cones on the largest side. The Street Cube features 2 2.5-watt drivers, with one on each of the 6.5-inch speakers. The Street Cube is basically a reshaped Micro Cube, and has 8 different guitar amplifier models, 6 digital effects, and a built-in guitar tuner like the Micro Cube does. The battery powered line-up of Cubes all require AA's, though they are built to be powered off of rechargeable Ni-MH AA batteries (though they have to be recharges outside of the amplifier). The estimated battery life of these amplifiers is between 15 and 25 hours.

3.1.3 Phone App Amplifiers

Another popular trend for guitar amplifier substitutes is the use of guitar amp simulations on a smart phone. These simulations take place within a downloadable phone application. These simulations recreate the various effects that would normally be produced with analog circuits in a traditional amplifier. The guitar is connected to the smart phone through the use of a converter which takes an input from the guitar and then outputs it to the 3.5mm or USB input on the phone. For some of these products this converter also possesses audio outputs for either a 3.5mm headphone jack or a 6.35mm phone jack output for an external amplifier. Thus, these products are a combination of software and hardware, the software app, made functional from the hardware of the converter. Additionally, the converter and app are often separate products, which allows a degree of flexibility for the end user. These products produce a very compact product, which is able to reasonably simulate the output of a full guitar amplifier.

In function, these app-converter packages seek to produce many of the same benefits as the Pocket Amp. Portability, affordability and the user-friendly interface from a well-designed app are all goals that the Pocket amp and a guitar amp app seek to produce. The price of these phone apps varies and are usually an escalating price scheme after the base cost of the converter has been purchased. For example, after the initial purchase of the converter a free app can be used to simulate a specific amplifier. Then to purchase additional simulated amps, there are additional charges from the amp.

For converters, there are many options ranging in price. On the cheapest end of the spectrum is a simple converter called the JamUp Plug. This converter connects a guitar up to the audio input of a smart phone and can output audio to a traditional 3.5m audio jack. Reviews state that the audio quality is not high but for this simplicity the JampUp Plug retails for $20. One popular converter is the iRigHD2. This converter is more expensive at $99 but advertises as professional quality. The converter for the iRig2 is very flexible as it allows the user to not only output using both a headphone jack but also send a signal to a traditional amplifier. This allows the iRigHD2 to not only operate as its own amplifier but also as an effects box. The iRigHD2 connects to the phone through USB allowing for a faster digital connection.

One drawback of these converter-app combinations is the latency of the processing. When the signal enters the phone or tablet to be processed, modified and then sent back out, it might be delayed upon exiting. This delay can cause some discontinuity between the playing a chord and hearing it through the audio output.

3.1.4 Guitars with Built-In Amplifier

Guitars with amplifiers already built into the guitar are designed for a similar purpose as the pocket amplifier we are building. Guitars with built-in amplifiers were developed with ease of use and portability in mind. They are great for situations in which large equipment setups are not conducive. The built-in amplifier also allows a guitar to be played when there is not an electrical outlet available for a standard guitar amplifier to be plugged into. Most electric guitars have too soft of a sound without amplification for them to be heard in a crowded area or if they are being played in a band with other instruments, the electric guitar would be easily overpowered. Amplifiers allow the guitar's signal to be increased to the desired volume and let the guitar be heard when it would not be able to on its own.

One downfall of using an electric guitar with a built-in amplifier includes the number of effects that is capable of being implemented on the guitar is quite limited. However, it is possible to include some onboard effects such as delay, modulation and gain control. In most cases, the guitar is either not capable of having any more effects than that or it is simply not practical to add any more.

If a guitar has a decent tone and multiple effects with a built-in amplifier then it will easily cost hundreds of dollars. One goal of our pocket amp is to create an amplifier that is not only portable, but also economical.

3.1.5 Effect Pedals

Effect pedals lie between the guitar and the amplifier and provide some form of distortion or modification the signal. They are useful because they allow the artist to add an effect at certain times but not at others simply by pushing a button on the pedal with their foot, much like the pedals on a piano allow the artist to change the piano's output. However, because of the nature of the device having been designed it has been concluded that it would be more useful to the artist to integrate these effects into the amplifier, which can then be controlled through the smartphone application in order to keep overall size and weight down. There is a wide range of effects that can be implemented using pedals, ranging from subtle nuances that lightly color a sound to dramatic distortions that barely resemble the original signal. There is no set standard for classifying effects, however the common categories that they are often divided into are discussed below.

3.1.5.1 Distortion/Overdrive

Distortion and overdrive are often implemented by increasing the gain. The voltage of the signal is driven to the rails to produce a signal that clips at its peaks. Distortion is an extreme variation of overdrive, in this case distortion is a specific musical term and not any type of variation of a signal. Distortion aggressively flattens the peaks of a sinusoidal signal for a hard clipping effect while overdrive has a soft clipping effect that compresses the signal.

The overdrive effect was made popular originally by blues artists who sought after a “warmer” sound originally created by either soft or hard clipping that could be caused by slightly overdriving a vacuum tube amplifier or using an effects pedal to cause the amplifier it fed into to be slightly overdriven thereby causing the desired distortion. Although both soft clipping and hard clipping are considered to be types of overdrive the softly clipped variety is usually preferred as it maintains the integrity of the original signal and hard clipping is often noted to sound “cold” and “unnatural.” Because of this it is not uncommon to see soft clipping referred to as the overdrive effect and hard clipping referred to as the distortion effect.

3.1.5.2 Dynamics

Dynamics modify the volume and amplitude of the audio signal. A boost dynamic effect is often used during solos to increase volume. A compressor dynamic effect compresses the range of the amplitude of an audio signal, which effectively makes the louder volumes quieter and softer volumes louder. Since compressors increase the lower amplitudes of a signal, this often does add noise to the sound. A noise gate dynamic effect attenuates noise once the volume of a signal falls below a certain threshold by lowering the volume of the signal. Noise gate effects are the opposite effects of compressors; they are expanders. Noise gates soften the quieter sounds of a signal and increase the loud sounds. Most of the noise is on already on the soft side so this helps to eliminate noise.

3.1.5.3 Filter

Filters alter an audio signal by either strengthening or weakening certain frequencies. An equalizer is a set of filters such as the two bands that come with any basic home stereo equipment that can boost or cut bass and treble audio. A wah pedal is an effect that essentially alters the frequency spectrum of a signal; this allows manipulation of the tone and pitch of the audio signal.

3.1.5.4 Modulation

Modulation as an audio effect alters the tone of a signal by splitting the signal multiple times, varying the strength of some aspect of the signal, and then recombining the original signal and the modulated signal. A chorus modulation effect mimics having multiple signals at the same time by adding in to the original signal with a slight delay. A flanger modulation effect adds a variably delayed signal back to the original audio. A phaser modulation effect shifts the phase of a portion of the audio signal that creates a ripple effect in the sound. A ring modulator effect combines the original audio signal with a sinusoidal carrier signal that resonates. A tremolo modulation effect implements a rapid variation in the volume of the signal; however, the amplitude of the variation is small. A vibrato modulation is similar to the tremolo effect, except in vibrato the rapid variations are in the pitch rather than volume. This is produced by adding a carrier signal that generates frequency variations in the signal.

3.1.5.5 Pitch/Frequency

Pitch and frequency as an effect alter the frequency of a signal to change the pitch. A pitch shifter effect can raise or lower a signal by a set interval such as an octave. A harmonizer effect combines the original signal with a pitch shifter to create a harmony of multiple pitches. The harmonizer effect, especially when the pitch shift is not an octave, can be difficult to implement well if the musician is not familiar with music theory.

3.1.5.6 Time-based

Time-based effects produce delays in the audio signal. A delay effect can produce a single echo or multiple echoes. The two most common adjustments for a delay effect are the amount of time between repeats and the number of times that the echoes are repeated. A reverb time-based effect is when multiple echoes are added into the signal that gradually decay. A looper pedal is an effect that can record a signal and replay it later on.

3.1.5.7 Feedback/Sustain

Feedback and sustain are effects that are produced when an amplified signal is picked up by a microphone and played back through an amplifier, thus creating a feedback loop. Sustainer pedals boost a signal to prolong its length.

3.1.5.8 Tone Control

Controlling the tone or general sound of an artist’s guitar can change what comes out of the speakers to a great degree. Tone control generally refers to changing the magnitude response of an amplifier depending upon what range of frequencies the artist wishes to be most prevalent. For example, many bass guitarists want low frequencies to be most prevalent while attenuating higher frequency response to limit unwanted noise. Simply by changing the tone of an amplifier the overall effect of what the artist plays can be drastically changed.

3.1.5.9 Reverb

The reverb effect is meant to emulate the way sound waves naturally bounce off of hard surfaces before being heard. Because of this the listener may hear the original tone followed by several delayed versions of that tone as they bounce off of multiple surfaces, each adding a different delay depending upon the distance between the object and the listener. As time passes the reverberations should decrease in strength until they are essentially nonexistent.

3.2 Electrical

Discussed in this section are technological research topics relevant to the electronic design for this project including the power system and analog audio system.

3.2.1 Power Systems

The Power supply of a device is a difficult and essential step in the design of a product. Power systems often have the largest components and are interconnected with nearly every other component in the project’s design. Thus as the Pocket Amp is portable in nature and aims to be as user-friendly as possible proper power system design is essential.

3.2.1.1 Batteries

As the Pocket Amp is designed to be portable in design it was decided to use a battery to power the device. To increase the user-friendly aspect of the device, only secondary (rechargeable) batteries were given consideration.

3.2.1.1.1 Nickel Metal Hydride (NiMH) Batteries

Nickel metal hydride batteries began development in the 1960s but only became commercially available in the 1990s. Since then they have become one of the leading replacements for older nickel-cadmium batteries in rechargeable applications. Like NiCd, nickel metal hydride batteries use nickel as the positive cathode for the battery.

Nickel metal hydride batteries utilize nickel-oxide hydroxide to form the cathode in the battery. The anode is comprised of a metal hydride. This metal hydride works effectively as an anode because certain intermetallic compounds have the ability to capture and release hydrogen atoms in volumes much greater than the intermetallic compound’s own volume. Thus, the active component of the anode is not the metal-hydride but rather the hydrogen ions that is stored in the metal-hydride. The metal-hydride usually composed of two metals, the first being one atom of rare earth metals, to five atoms of a more common metal such as aluminum, cobalt, or nickel. The nickel-oxide hydroxide and the metal-hydride are separated by an insulating component and immersed in an electrolyte, usually potassium hydroxide (KOH). The electrolyte, and other components are placed in a steel container.

During the discharge cycle, the chemical reaction at the two terminals of the battery produce a voltage potential. At the cathode, the nickel-oxide hydroxide combines with water and a free electron to form nickel hydroxide and hydroxide ions. At the anode, the metal-hydride combines with hydroxide ions to produce free electrons, released hydrogen from the metal-hydride and water. The common cell voltage is 1.2V but decreases as the battery discharges down to 1 to 1.1 V. Under periods of over discharge, the NiMH battery undergoes a hydrogen cycle where water and free electrons are converted to hydrogen and hydroxide ions at the cathode and are converted back to water and free electrons at the anode. This means that the actual chemistry of the battery is not damaged by over discharging. Heat and pressure are generated by over discharging however, which can lead to damage of the battery if left for an extended time. Nickel metal hydride also possess a self-discharging characteristic like all batteries. However, the self-discharge on the NiMH battery can be significant and thus must be factored into its selection. Generally, this self-discharge characteristic can vary greatly with temperature, this can be from 0.5-4% per day at room temperature but can triple at higher temperatures. Certain more recent NiMH batteries are being marketed with lower self-discharge rates, though this can come at a trade-off with capacity.

Like nickel-cadmium batteries the nickel metal hydride has several options for charging the battery back to full-capacity. First the battery can be charged over slow charging. This can be done by applying a trickle current much smaller than the hourly capacity of the battery, which is usually about 10% of the hourly capacity. The battery must then be charged longer than its own capacity. For example, a 120mAH battery might need to be charged at 12mA for up to 15 hours. Also, nickel metal hydride batteries can still suffer from overcharging effects when with a very low trickle charge, especially if the battery is not in a room temperature environment. Thus, a timer is usually placed for slow charging a NiMH so that no overcharging characteristics occur. A recycling catalyst is sometimes placed as well in order to reduce and effects of overcharging.

Fast charging is similar to nickel cadmium, except it is more complicated as the chemical process of charging is exothermic. Fast charging is usually composed at around 120% of the batteries hourly capacity and charged for an hour. The risk of overcharging is much greater for fast charging and thus end-of-charge detection must be applied. The first method is to monitor how the voltage of the battery changes over the charging time. The terminal voltage of the battery drops slightly when the battery reaches maximum charge. This can be difficult to detect so another method is also used. The derivative of the voltage with respect to time is measured constantly and when the rate of voltage is zero, ideally max charge has been met and the charger shuts off. This method can result in prematurely shutting off the charger before fully charging the battery. Another method of end-of-charge detection is to monitor the temperature change of the battery. When the battery is charging, the energy being delivered into the battery is converted into the charge of the battery. When the battery reaches its full capacity then most of the delivered power will be converted into heat. By measuring the temperature of the battery the end of charge of the battery can be detected.

3.2.1.1.1.1 Advantages of Nickel Metal Hydride

- High Energy Density - Nickel metal hydride batteries possess one of the highest energy densities of common batteries. This density can be up to 40% greater than NiCd batteries and often doubles the capacity of Lead-acid batteries. However, lithium based batteries still do have a higher density.

- Deep Cycling - NiMH batteries can be cycled down to between 80% and 100% of their total capacity, depending upon the battery, without incurring any lasting damage.
- High Cycle Lifespan - Most nickel metal hydride batteries have a cycle lifetime of about 3000 cycles. Compared with Lithium-ion batteries which often have a lifespan of 300 to 500 cycles. This makes NiMH batteries exceptional for applications where long lifetimes are required.
- Robust Charging conditions - Electrochemically, nickel metal hydride batteries are resistant to abuse from overcharging and over discharging. The chemical process from both overcharging and over discharging produces a net zero reaction. None of the chemicals are turned into crystal structures or other permanently imperfect conditions. However, heat and pressure can be produced so overcharging and over discharging should be avoided.
- High Operating Temperature Range - NiMH batteries are able to operate in a varied range of temperatures. Recent nickel metal hydride batteries can operate from -30 ^{o}C to 75 ^{o}C. This has allowed NiMH batteries to be used in various applications in the automotive industry where temperatures can vary.
- Environmentally Friendly - Unlike lead-acid and nickel cadmium batteries nickel metal hydride batteries do not utilize heavy metals (which are toxic) in their construction. The nickel used in the construction is also easily recycled when the battery reaches its end of life.
- Storage Options - Nickel metal hydride batteries have no restrictions as to how they can be stored. Unlike some batteries, which must be fully discharged or charged before storing, NiMH can be stored almost indefinitely in any state.
- Stable Discharge Voltage - Like nickel cadmium batteries NiMH batteries keep a very consistent voltage throughout the discharge cycle. Usually, the voltage of 1.2V stays consistent until the very end of the discharge cycle. This consistency makes them useful for linear regulator design.
- Cost - While NiMH batteries were expensive at their onset, they are growing more affordable. Most nickel metal hydride batteries can be equal in cost to the nickel cadmium batteries and are significantly cheaper than lithium-ion batteries.

3.2.1.1.1.2 Disadvantages of Nickel Metal Hydride

- Memory Effect - Like their early cousin, nickel cadmium, NiMH batteries can develop of a memory of their charging voltage and if recharged at that spot too repeatedly can suffer past that point in their capacity. However, the memory effect is very much reduced in NiMH when compared to NiCd.
- High Self-Discharge Rate - Nickel metal hydride batteries suffer from a particularly high discharge rate. Early NiMH batteries could discharge up

to 5-20% within the first 24 hours and goes to 0.5-4% daily from this point. This severely limited the usefulness of NiMH for products that are not used daily. However, advancements have been made since the introduction of the nickel metal hydride battery, and certain low discharge NiMH batteries can retain 70% of their capacity when stored for over a year.

- Overcharging Difficulties - While the chemical conditions of overcharging a NiMH battery are not damaging, the produced heat and pressure of overcharging can damage both the battery and its environment. Because the charging reaction is exothermic in nature it requires complicated charging schemes to ensure that overcharging does not occur. These charging schemes add complications to the batteries' use and often require removable battery slots.
- Coulombic Efficiency - Coulombic efficiency refers to the ability of the battery to convert its active materials into energy able to discharge. This is found by comparing how much energy it takes to completely charge a battery versus how much energy is produced from that complete charge. NiMH batteries usually have a coulombic efficiency of around 65% which pales in comparison to Lithium battery's 99%.
- Low Cell Voltage - Often the cost of a very consistent voltage output is a low voltage output. Cell voltage for the nickel metal hydride is usually only about 1.2 volts, much lower than many other batteries. Because of the low voltage output several cells must be placed in series which can add several complications.
- Rare Elements in Cell Production - While only limited amounts of rare earth elements are necessary for the design of the battery they are still very limited in quantity. Material like Lanthanum can be difficult to obtain and often increase the cost of the battery.

Reasoning Behind Nickel Metal Hydride's Rejection

Nickel metal hydride batteries are popular options for small electronic applications having a good capacity with a constant discharge rate. However, in spite of these advantages, nickel metal hydride batteries have not been selected for this project for several reasons. First the low voltage output is undesirable for this project. The required voltages for this project range from 3.3V to 6V and the 1.2V cells are not sufficient for powering the project. Adding extra cells in series increases the complexity of the charging circuit greatly. Charging the nickel metal hydride battery is a complicated process and would add many complications to the design of the power system. Most likely, the design would require the end use to remove the batteries from the Pocket Amp and place them in an external charger for charging, which does not reflect the user-friendly portion of this device. Finally, the high-self discharge rates limit the battery's usefulness drastically. As the Pocket Amp is not a daily needs device which would need charging every day, the end user may find themselves needing to charge the battery each time they desired to use the device.

3.2.1.1.2 Lithium-Ion (Li-ion) Batteries

Lithium-ion batteries are a relatively new development in battery technology. Extensive research into lithium batteries truly began in the 1970s, early developments were quite expensive but held promising results. The first commercially available lithium-ion battery became available in the early 1990s. Since the 1990s lithium-ion has become the most common type of battery for portable consumer electronics.

Lithium-ion batteries come in many variations for their electrode composition. The negative electrode (the anode during discharge) is generally composed of carbon, while the positive electrode (the cathode during discharge) is formed of a lithium-ion. Various combinations of lithium and other metals are used but the most common is a lithium-cobalt-oxide. A lithium-ion is used instead of pure lithium metals is due to lithium metal's inherent instability. For example, if pure lithium metal interacts with water it can explode violently. Thus, Li-ion was substituted for pure lithium despite its lower specific energy. The electrolyte in lithium-ion batteries is generally a lithium salt combined in an organic solvent. As in all batteries this electrolyte forms the conductive pathway between the positive and negative electrodes. Often different carbonates are used to form the organic solvent to give the electrolyte the ability to form an SEI. The solid electrolyte interphase (SEI) is formed around the negative electrode during the charging cycle by the combined organic solvent. This SEI interface allows ionic conductivity but is electrically insulating. The SEI protects the electrolyte from further decomposing around the negative electrode during later discharging cycles. Li-ion batteries can also utilize polyoxyethylene based electrolytes which can be either liquid or solid. Like NiCd batteries Li-ion batteries are often wrapped in cylindrical shape to create a larger surface area.

During the discharge cycle, on the cathode the lithium-ion is ionized forming Li^+ and a free electron. On the anode side Li^+ and a free electron are absorbed by the carbon in the anode. This constant oxidation and recombination forms a voltage potential between the cathode and the anode. The common cell voltage for a Li-ion battery is 3.6V, which is much higher than most cells of similar size. This incredible cell voltage is due to in part, the volume of the lithium atom. Lithium is the third smallest element which allows many more Li^+ ions all carrying positive charges in a smaller space. Over discharging a lithium-ion battery can be damaging to the battery, causing lithium-cobalt-oxide to supersaturate with Li^+, and free electrons. This saturation forms lithium-oxide and cobalt-oxide, often this reaction is irreversible. However, different protection circuits (often prepackaged with the battery cell) are designed to prevent over discharging the battery. During the discharge cycle most lithium ion batteries will stay at 3.6V for about 80% of the discharge, providing a very consistent voltage.

Charging a lithium-ion battery has two primary stages, a constant current and a constant voltage stage. At first a constant current is applied to the battery, this current is anywhere from half to equal to the battery's hourly discharge rate. This current continues until the voltage of the cell reaches its designed threshold.

After stage one the battery is approximately 70% charged and can be used directly from this point. After stage one, a constant voltage is applied. This constant voltage is equal to the maximum cell voltage, multiplied by the number of cells. This constant voltage is applied until the current drawn by the battery reduces down to about 3% of the constant current applied in stage one. Often a lower threshold voltage is chosen than the maximum cell voltage, so the battery is never 100% charged. Indeed not completely filling a Li-ion battery prolongs its overall lifespan. Overcharging a Li-ion battery can cause dangerous stresses to the battery. Even a constant trickle charge to a full lithium-ion battery can cause the positive electrode to form metallic plating from the lithium which can be very dangerous in a thermal runaway situation.

While lithium-cobalt-oxide is the most popular choice of lithium-ion, other types of lithium metal combinations do exist and each have their advantages and disadvantages. Lithium-cobalt-oxide is the most popular choice for small cell portable applications. This is due to lithium-cobalt-oxide's high specific energy making it ideal to power items such as cell phones. However, Li-cobalt cells due tend to have shorter life spans and a smaller temperature tolerance range. Another disadvantage of Li-cobalt is that it cannot be charged in excess of its hourly capacity, which prevents charging the battery in less than an hour. Another popular option is lithium manganese oxide ($LiMn_2O_4$). $LiMn_2O_4$ possesses a very low internal cell resistance, which allows the cell to be fast charged, and also allows it to provide larger discharge current. These aspects make the $LiMn_2O_4$ cell ideal for power tools and various electric vehicles. There are dozens more variants of the lithium-ion cell, from lithium iron phosphate to lithium titanate, each with different advantages and disadvantages.

An additional variant of lithium-ion batteries is the lithium polymer battery or lipo. Lithium polymer batteries differ from other lithium batteries primarily due to the electrolyte that is used. Lipo batteries use a gelled electrolyte which allows the casing of the lipo battery to be more flexible in how it is packaged. Thus, a lithium polymer battery generally comes in a flatter package than traditional lithium-ion and does away with the hard, circular packaging of a traditional cell. This packaging allows the lithium polymer battery to be lighter and smaller than a normal lithium-ion battery.

While lithium-ion batteries are widely used and thoroughly researched, the inherent instability of lithium metal can cause hazards in certain aspects. If the battery is overcharged or very another reason is heated beyond the battery's rated capacity, thermal runaway can occur. Thermal runaway is the process where the battery cell's temperature increases uncontrollably. This produces gas and thus a dramatic increase in cell pressure which must then be vented. If the gasses are unable to vent, then the pressure built up could result in an explosion of the battery. Puncturing the battery cell can also be dangerous as this generally creates a short circuit between the anode and cathode resulting in the formation of lithium metal plating which immediately reacts with any moisture in the area causing dangerous reactions. However, all of the dangerous reactions of a lithium-ion battery are under misusage and do not occur under proper operation.

3.2.1.1.2.1 Advantages of Lithium-ion Batteries

- High Cell Voltage - Lithium-ion batteries have a very high cell voltage of around 3.6V depending upon the ion composition. This cell voltage triples that of Nickel based batteries and allows only one cell to be used where NiMH might use three cells for an equivalent voltage.
- Light Construction - As stated earlier, lithium is the third lightest element in existence. The extensive use of lithium in lithium-ion batteries makes the construction very light, this allows the batteries to be used in very small scale portable applications.
- High Discharge Rates - For short periods, lithium-ion batteries can provide a very high discharge current, certain types can provide 40 times their hourly discharge rate, for extremely short periods. This allows Li-ion batteries to cope with extreme start-up currents common in many applications.
- Deep Cycling - While Li-ion batteries are not tolerant of 100% discharge most batteries can be discharged down to 80% of their capacity allowing their stored energy to be effectively usable and permitting more start-up currents.
- High Coulombic Efficiency - The coulombic efficiency rate of most Li-ion batteries is 95% or greater. This means the vast majority of the energy being stored into the battery will be used to supply energy later and energy is not wasted by simply charging the battery.
- Numerous Variants - The high number of lithium-ion variants allows for a customization that is not found in other battery types. By selecting the appropriate version, users can find a different lithium-ion battery suitable for their tasks.
- No Memory Effect - Unlike nickel-based batteries, lithium-ion batteries do not suffer from memory effect. Regardless of the battery's level upon starting the recharging cycle, Li-ion batteries will not develop the tendency to drop in output voltage once reaching a remembered voltage. As the battery develops no memory, Li-ion batteries do not require reconditioning and can tolerate microcycles, which is useful in portable applications which might be charged at any point in the discharging cycle.
- Long Cycle Lifetime - Lithium-ion batteries support a long cycle life, depending upon the variant of lithium-ion a cycle life of 3000 cycles can be supported
- Self-discharge rate - Unlike nickel metal hydride batteries, Li-ion batteries have a low self-discharge rate, to the point where many lithium-ion batteries can still possess a charge after a decade of storage. This self-

discharge rate allows the Li-ion battery to be flexible in its storage solutions.

- High Energy Density - The high energy-density of lithium-ion batteries is integral to many of it important applications. This density allows individual cells to possess an hourly capacity greater than 1000Ah. The high energy-density of Li-ion batteries also allows them to provide large power currents. For example, 4 times the amount of lead-acid batteries must be used to equal the same amount of lithium-ion batteries.

3.2.1.1.2.2 Disadvantages of Lithium-ion batteries

- Cost - As a newer technology lithium batteries are generally more expensive than many counterparts. The use of cobalt in many variants of lithium-ion batteries also raises the cost significantly. This factors most greatly in high-power applications where many cells must be used.
- Shipping Restrictions - Due to explosive nature of lithium-ion battery's failure modes, shipping lithium batteries has several restrictions. For small orders, usually another item must be ordered with the battery, and in large scale orders the shipping methods can be labeled as a hazardous method. This raises the overall cost of the battery and can cause the shipping time to be extended.
- Hazardous Failure Modes - Due to the flammable nature of the electrolytes used in Li-ion batteries its failure modes can be hazardous. If the circuit is overcharged, or the cell is punctured or crushed, explosive venting can occur. Thankfully, these failure modes tend to be rare, especially if the battery is handled properly. Additionally, almost all Li-ion batteries are prepackaged with protection circuits to prevent dangerous overcharging.

Reasoning Behind Lithium-Ion's Acceptance

Lithium-ion various advantages make the ideal choice for the Pocket Amp. Most of the voltage requirements for the amp are at 3.3V, which is very close to the lithium-ion's 3.6V output. The light lithium construction lends itself to a relatively light battery, which is helpful for portability. The high energy-density of lithium-ion allows for large capacity single cell batteries, which will give the Pocket Amp a long battery life, especially considering the low current draw of most of the loads. Protection circuits are built-in to most battery cells and allow an easier design for fast charging. Lithium polymer variants can be used to create a lighter and smaller package. Finally, the vast majority of the failure modes for the battery are regulated to user error, and the Pocket Amp will isolate the battery from the user, preventing any user tampering from occurring. These advantages fulfill the desire for a user-friendly, portable experience that is desired from the Pocket Amp. For these advantages a lithium-ion polymer battery was selected to power the Pocket Amp.

3.2.1.2 DC/DC Converters

DC/DC converters are used to step up or down from DC voltages. This is required due to the various voltage requirements of the different parts, leading us to need several rails at multiple voltage levels. In addition to providing changes in voltage, the regulators will do what they were named for – they will regulate the voltage of our rails. The two main types of DC/DC regulators are linear regulators and switching regulators, both with pros and cons and both potentially having uses in the Pocket Amp.

3.2.1.2.1 Linear Regulators

Linear regulators are simple devices designed to output a constant voltage given an occasionally varying voltage supply. Linear regulators operate in a very similar fashion to a traditional voltage dividing circuit. Linear regulators use a simple voltage feedback system in order to dynamically adjust its resistive components to ensure a constant voltage output. This feedback regulates a variable resistor to dissipate voltage from the input. Linear regulators generally create this variable resistor with three components, a BJT Power transistor, an error amplifier, and a current amplifier. The circuit on Figure 2 shows how these components are interconnected.

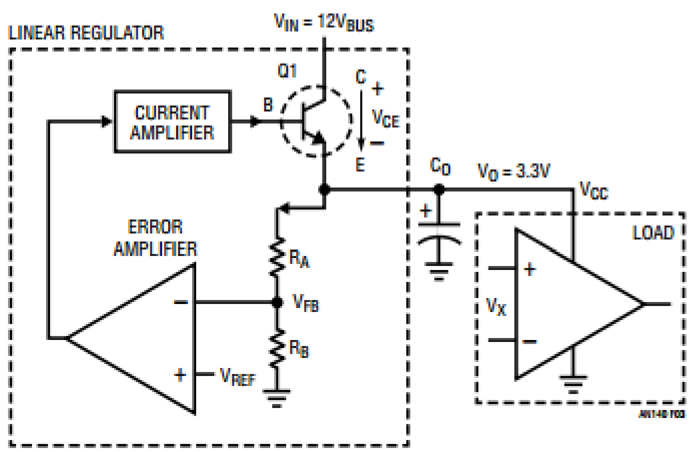


Figure 2. A Linear Regulator Implements a Variable Resistor to Regulate Output Voltage
Permission Pending from Linear Technologies

The error amplifier samples the input voltage through the resistor network and compares the input with a reference voltage. The output of the current amplifier is amplified and used as the base current for the power transistor. If V_{cc} lowers, then the error amplifier will automatically push more current into the base of the transistor causing the overall V_{ce} to drop and the V_{cc} voltage to rise. The opposite process occurs when V_{cc} begins to rise to high, the error amplifier provides less current causing V_{ce} to rise and V_{cc} to lower again. This constant feedback provides a stable output voltage to the load.

The linear regulator is a very time-tested device, since the original inception of semiconductor devices to the onset of switching mode power supplies, the linear regulator was the primary device for power supply circuits. The simplicity of the linear regulator allows for a very simple design process, requiring relatively few passive components and no need for any inductors.

Unfortunately, the linear regulator can consume a fair amount of power. The power transistor fundamentally acts as a resistive component. Because of this the voltage dissipated can be quite large if the voltage shift is large or the current draw is significant. The entirety of the dissipated power is released as heat like a resistor, so linear regulators often need compensation for creating so much heat. Additionally, the power dissipated due to thermal power loss of this design means that the linear regulator is a less efficient device and that linear regulators can only operate as a step-down DC/DC converter.

Linear regulators have an additional drawback which is dropout voltage. In order for the feedback circuit to continue to operate, the power transistor must be in the forward active mode. Forward active mode ensures that current continues to flow from the input to the load. If the input voltage is too close to the output than the power transistor will drop into cut-off mode preventing any form of voltage to pass through.

While these drawbacks can prevent linear regulators from many high efficiency applications, linear regulators are still an excellent option for the Pocket Amp. The digital components of the project only require a 3.3V input which is only a small step-down from the nominal 3.7V produced by the battery. This results in a nominal efficiency of 90%. With such a relatively high efficiency, the benefits of the linear regulator outshine its downsides. Unlike most switching regulators, no inductors will be necessary with linear regulators, resulting in a cheaper and smaller design. Linear regulators are also easier to design requiring less external components. Finally, linear regulators produce less overall noise as there are no transient currents except for the initial start-up.

3.2.1.2.2 Switching Regulators

Switching regulators are similar to linear regulators in that they utilize a power transistor to form a voltage drop between the input voltage and load thus regulating what voltage level the load receives. However, this transistor is not operated linearly but rather in a discrete switching mode. When the switch is on, the voltage across the transistor is quite small and current is being passed through. When the switch is off minimal current is passed through and voltage is blocked. These discrete switching allows very little power to be lost during the regulation process, allowing for very high efficiency.

The first type of switching regulator considered for the design of the Pocket Amp is the buck converter. Buck converters possess similar components to a linear regulator, but utilize a MOSFET transistor instead of a Bi-polar junction transistor, include an inductor for voltage storage and several diodes for transistor protection. A buck converter schematic is shown in Figure 3.

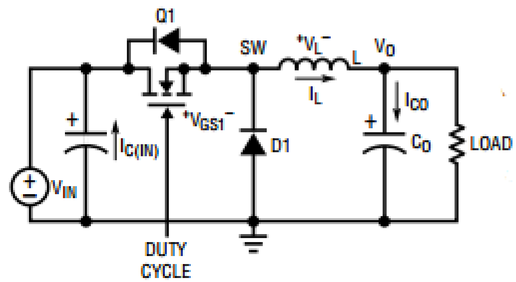


Figure 3. Buck Switching Converter Schematic
Permission Pending from Linear Technologies

The buck converter operates in two modes, each corresponding to the on or off position of the transistor. When the transistor is on, the regulator is in inductor charging mode. The switching voltage at V_{SW} becomes equal to the input voltage and the inductor begins to charge up. When the transistor is off, the regulator is in inductor discharging mode. No input voltage reaches the inductor and the inductor begins discharging current into the load, with the diode forming the return path back to the inductor. The inductor is responsible for limiting the current coming from the power switch protecting the output from an instantaneous and possibly damaging current change. Output voltage is formed by the following equation.

$$V_{o(DC)} = avg(V_{SW}) = V_{in} \frac{Time_{on}}{Switching\ Period}.$$

The ratio between the between the time on and the switching period is the duty cycle of the regulator. With a relatively fast switching frequency the output voltage is generally a DC output with mV rippling.

While it would be preferable to have a device that dissipates no loss in power when converting voltages that is impossible. Buck regulators do have several methods of power loss. The first losses come from DC conduction. These losses of are voltage drops across the power transistor, the inductor and the diode. Thus, the power losses through DC conduction are formed by the equation

$$P_{DC\_losses} = P_{loss\_Q1} + P_{loss\_L} + P_{loss\_D1} = I_o^2 * R_{DS(on)} + I_o^2 * R_L + I_o * V_D * (1 - D).$$

In many cases the diode consumes the most amount of current so it is often replaced with another MOSFET. This conversion forms a synchronous buck converter. The other losses formed in a buck converter are the AC switching losses. While an ideal MOSFET would instantaneously switch from saturation mode to cut-off this is not realizable. During this time current and voltage is still being passed through as a short AC transient response. This AC transient response causes losses in the magnetic windings of the inductor across the body diode. Calculating these losses is very difficult due to its transient nature, but in general they are quite small.

The buck regulator is a very efficient method of lower DC voltage and regulating it against shifts in the voltage level. Many buck regulators can provide an efficiency of 90% or greater even while supplying low load currents, and are more suited to the greater drop in voltage that the lithium-ion battery undergoes. Buck regulators have a very long history of use since they became popular devices in the 1960s. All of these considerations make the buck regulator an excellent choice for the digital loads of the pock-amp.

Another form of switching regulator worth considering is the boost regulator or boost converter more appropriately. Unlike like linear regulators, switching regulators possess inductors which give the ability to store energy in the regulator itself through the inductor. This energy storage allows the switching regulator to not only cause a voltage drop by a voltage boost. A boost converter is created by taking the same components which form a buck converter and altering their topology. Where a buck regulator has the inductor and MOSFET in series with a diode in parallel, the boost regulator has the inductor and diode in series with the MOSFET in parallel. Figure 4 shows the basic configuration of the boost converter, with the switch being a power MOSFET in general.

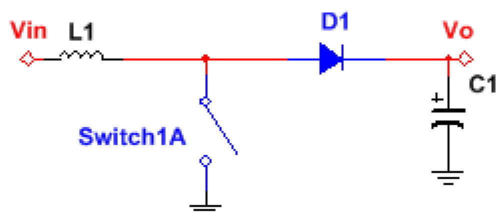


Figure 4. Boost Regulator Topology
Created by Trent Coleman using Multisim.

Boost converters undergo two phases like buck converters. The first is the charging phase, which occurs when the MOSFET is on, like a closed switch. During this phase the input voltage causes current through the inductor to rise at a linear rate. The diode prevents the capacitor from discharging this growing current so it is effectively stored in the inductor. Next is the discharge phase when the MOSFET turns off. The energy stored in the inductor is released as current proportional to the voltage of the inductor. The steady state voltage of the inductor equals to zero between charging and discharging, which means that the average current is also in a steady-state behavior. From this knowledge, we can form a steady state equation to calculate output voltage. $V_o = \frac{V_{in}}{1-D}$. Where D is the duty cycle which is equal to the time the MOSFET is on divided by the switching period. Given that D is always less than or equal to 1 the output voltage is greater than the input voltage.

The efficiency of the boost converter is largely decided the same way as a buck converter. Power is lost across the inductor, the MOSFET and the rectifying diode with the rectifying diode being the primary culprit for the losses. These losses can easily be reduced by swapping out the diode with Schottky diode or another MOSFET for even greater efficiency.

Another type of switching converter worth noting for this project is the inverting buck-boost converter. Once again by swapping the topology of the inductor, MOSFET and the rectifying diode, a switching converter can produce a bucked or boosted inverted voltage. Often the topology of the inverting converters is both buck and boost depending upon the values of the surrounding passive components. The inverting topology is created by swapping the terminals of the rectifying diode and capacitor. Figure 5 displays the basic layout of the inverting topology.

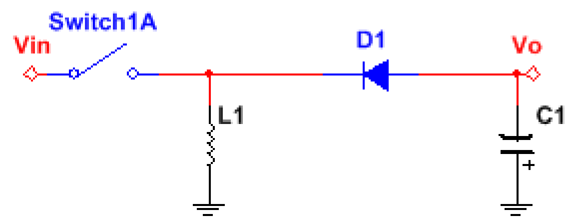


Figure 5. Inverting Converter Topology
Created by Trent Coleman using Multisim.

While in the on-state, input voltage is provided to the inductor, which begin the charging process of the inductor. During the onstage the output voltage is supplied entirely from the charged rectifying capacity, which also has had its voltage terminals reversed, providing a negative voltage. The off-state occurs after the on-state, the MOSFET is off and the inductor begins to discharge its stored energy into the load through the rectifying circuit. Because the rectifying diode is reversed, positive current is directed towards the ground, this entails that the voltage across the rectifying capacitor will be negative, forcing a negative voltage upon the output.

There is another switching converter which is able to raise the voltage level. This converter is known as a charge pump. While both buck and boost converters utilized inductors to store and transfer energy in order to lower or raise the voltage level, charge pumps utilize capacitors for energy storage. While most buck/boost converters utilize only a single switch, charge pumps utilize multiple switches for their design, with a set of switches for each stage. Unlike an inductor converter which must utilize a MOSFET for switching, charge pumps may utilize passive diodes or MOSFET for its switching. An oscillator works for each stage to charge the different capacitor for that stage. Figure 6 represents the process of an inverting two stage charge pump.

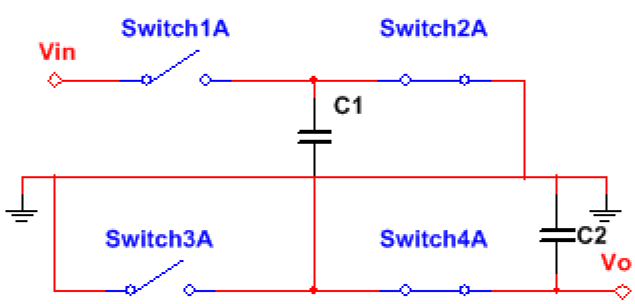


Figure 6. Inverting Two Stage Charge Pump
Created by Trent Coleman using Multisim.

The first two switches Switch1A and Switch3A are closed charging the first capacitor until its voltage is equal with the output. At the same time Switch2A and Switch4A are open during this stage enabling all of the input to be across the first capacitor. When that first half of the cycle ends the switches swap states, Switch1A and Switch3A close and Switch2A and Switch4A open. As this occurs the first capacitor discharges into the second capacitor, the reservoir capacitor. Now the reservoir capacitor is wired inversely to the initial capacitor causing a negative output voltage. This allows this charge pump to act as an inverter. Charge pumps are able to produce regulated voltages by the inclusion of an internal comparator which again compares the input to a certain reference voltage. The comparator is then used to drive the oscillator and switch the voltage as needed to regulate the output. Charge pumps can also produce buck/boost configurations which operate in a similar manner to inductors switching regulators. Charge pumps are sometimes preferred to inductor DC/DC converters as the lack of the inductor and external diode reduces the overall cost of the regulator by also decreases the size of the regulator. These benefits are extremely useful when space and cost is a consideration but comes at a cost of output current. While charge pumps are able to raise voltage levels, they usually can only sustain an output current measured in mA, usually less than 250.

Boosting the power supply is an essential component for the Pocket Amp’s power supply design. The battery selection of a lithium-ion only provides 3.6-3.7V which is insufficient to supply the op-amps of the analog section. While both charge pumps and inductor converters can function for step-up applications the inductor boost converter is the optimum choice for this project. Charge pumps are more specialized in a boosted by unregulated output, where almost all inductor boost converters are inherently regulating. Inductive boost converters also have a longer period of use making tutorials and resources on their application easier to find and generally more credible. Thus a boost converter is necessary to supply the rail voltages of the op-amps.

The input of the attached guitar for the pocket amp inherently produces both negative and positive voltages. Thus in order to properly amplify the signal in its purist form, dual supply op-amps were chosen in order not to clip any part of the input signal. In order to supply the negative rails some form of inverted output must be created. While there are other methods of inversion, through charge pumps or inverting flyback transformers, the inductive switching boost inverters are superior for this project. Having a dedicated inverting regulator allows both the positive and negative rails to have isolated regulation, and also allows each of the rails to be separately managed if some form of rail clipping is desired for analog effects.

3.2.1.3 Methods of Isolation

Electrical isolation is an important phase of power supply design. In order to prevent noise and other interference from the power affecting the end load, electrically separating the load from the source is sometimes necessary. Two methods of isolation were considered for this project. Opto-isolators and transformer flyback switches.

Transformer flyback switches or flyback converters provide isolation by combining a transformer with a MOSFET switching device. A rectifying network is then placed immediately after the second side of the transformer. Figure 7 outlines the basic topology of most transformer flyback networks.

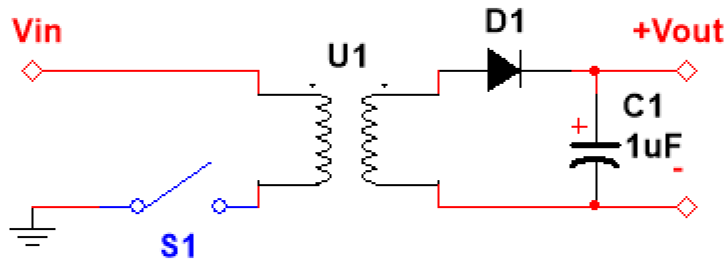


Figure 7. Flyback Converter Topology
Created by Joshua DeBaere using Multisim

Like other switching converters, flyback converters operate in multiple stages. The first stage occurs when the MOSFET is on, forming a continuous flow from the input across the transformer coils to the ground. During this stage, energy is stored into the transformer coils and in the magnetic flux like inductors. The secondary coil voltage becomes negative which is prevented from flowing due to the regulator. During this phase the capacitor discharges to provide the load with voltage. The second stage begins by turning the MOSFET off and preventing a continuous path to ground on the primary side. When the MOSFET turns off the stored energy of the transformer begins to discharge into the rectifying network. The secondary side of the transformer releases positive voltage which the rectifying network passes to the load and utilizes to charge the capacitor.

The flyback converter can be utilized to buck and boost voltages like other switching converters, but can also create electrical isolation between the source and the load. This isolation is created by converting the electrical energy of the source into magnetic energy in the transformer coils and then back to electrical energy for the load. Unfortunately, flyback converters can be complicated devices to design, as they utilize a small-scale transformer. If a standard transformer is not available for the desire design then one will have to be built, a difficult process in its own. Flyback converters also have issues with size and cost, as transformers are substantial devices in terms of component cost and overall size.

The other considered method of isolation is the opto-isolator. Opto-isolators are comprised of a light emitting diode in series with the source, paired next to a phototransistor in series with the load. The diode produces light which is then immediately converted back to electricity. Opto-isolators are devices specializing in preventing high surge voltages as the LED can only produce so much light and the photodiode can only produce so much electricity from that light. Additionally, the opto-isolator is one-directional by nature as the LED and photodiode can only perform their respective functions. This combined with its ability to pass DC voltages makes it a very efficient form of isolation.

Ultimately, the design of the power supply was deemed to not require electrical isolation from the source. This is because the source is a lithium-ion battery, which outputs a practically consistent voltage without large amounts of noise or other fluctuation. Neither is there a tremendous amount of noise put into the power supply through any other source in the design. Electrical isolation is required for different sources of power. For a motor power source or an AC power source electrical isolation is almost mandatory. For these types of sources both opto-isolators and flyback converters form efficient methods of isolating the noise of these sources from reaching the load.

3.2.2 Input Filtering

Many sources of noise exist between the output of the guitar and the input of the amplifier including the most notable mains or 60 Hz hum. 60 Hz hum is caused by the ever present electromagnetic oscillations at 60 Hz from the outlets, lights, and wires running through the walls of almost every building in modern society. Because guitar output usually stays above 80 Hz, this noise can easily be filtered out however it is important to understand that there are many resonant harmonics that occur at higher frequencies and as such these high frequencies should not be attenuated like a band pass filter would. Due to these two constraints, the input filter should be of a high pass form.

Several methods of input filtering exist that will satisfy these requirements. The most elementary of these methods is a passive high pass filter. However, this filter would have to be of a very high order to have a steep enough roll off to pass 80 Hz and attenuate significantly at 60 Hz which will decrease the stability of the pass band and are more complex, possibly requiring more components than an active filter. The second method would be a notch filter using the Sallen-Key topology. This method would impose a notch at a specified frequency, in this case 60 Hz and, with a high enough quality factor, pass 80 Hz well. However, a well-known disadvantage of the Sallen-Key family of filters is that as quality factor increases sensitivity in gain also increases. Finally, the Fliege topology notch filter exists, as explained by “More Filter Design on a Budget” by Bruce Carter at Texas Instruments. This filter is limited to unity gain but does not have a sensitivity at higher quality factors and will therefore allow a higher quality factor to be used to realize the stringent amplitude response requirements for the Pocket Amp.

3.2.3 Effects

This section discusses the relevant technology and methods of creating the effects, both analog and digital, that will be used in the design of the Pocket Amp.

3.2.3.1 Tone Control

Tone Control or equalization helps the artist realize the tone or general sound characteristics that he wishes to produce and helps the listener tune the music they hear to their particular tastes. For example, an artist or listener who prefers a heavy bass response will either boost the lower frequencies or cut the higher and mid frequencies. By doing this the general sound profile of the music being produced or listened to tends towards the lower frequencies as, in either case, the amount of energy produced by the lower frequencies will be higher relative to the energy of higher frequencies.

The tone control circuit does this by either “boosting” the amplitude of the response to certain frequencies or by “cutting” or attenuating the response at a frequency. Tone control networks for guitars often split the frequency spectrum of a standard tuned guitar, 80 Hz to 20 kHz, into four sections as can be seen in Table 3 below. The first section which is often referred to as the “Bass” section ranges from 80 Hz to 250 Hz. Increasing the energy in these frequencies rewards the listener with a heavier, often referred to as richer, sound however too much energy can make the music sound muddy or overpowering. The second section is called the “Midrange” section and contains frequencies between 250 Hz and 2 kHz. By amplifying these frequencies there is often no discernable positive effect, however amplifying them too much results in a tinny or sharp sound. For this reason, some tone control circuits leave out controls for the midrange. The third common frequency section is called the “High Mids” and ranges from 2 kHz to 6 kHz. Boosting this section produces what is often called a clearer sound however it can often lead to fatigue in the listener. Finally, the “High Frequencies” are composed of those frequencies of between 6 kHz and 20 kHz. It should be noted that there are no fundamental harmonics of a guitar in these frequencies, only higher order harmonics. By boosting these frequencies, the music takes on what is referred to as brighter sound that keeps the music from sounding dull. However, too much amplification in the high frequencies is blamed for causing unwanted fatigue and a sharp or brittle sound. It should also be noted that many frequencies in this range are outside what the average listener can physically hear and are often times more felt than heard.

Tone Section	Frequency Range
Bass	80 Hz – 250 Hz
Midrange	250 Hz – 2 kHz
High Mids	2 kHz – 6 kHz
High	6 kHz – 20 kHz

Table 3. Correlation of Tone and Frequency Ranges

Some of these frequency bands are able to be combined to simplify the design of a tone control circuit for a given amplifier. Some circuits use only two bands Bass and Treble that can be either boosted, cut or both and leave the mid bands as a reference, since amplifying them typically doesn’t produce positive effects. Other designs use three bands, Bass, Mid and Treble, so that the mid band response can be adjusted as well. However, these designs often only cut or attenuate frequency bands and the inclusion of the mid band is out of necessity for if the listener wishes to produce a sound heavy in either bass or treble and therefore must lower the mid band and the other, unwanted, band. Finally, some use all of the bands or even split the previously defined bands further so as to give the listener or artist more control over the output. These final types of tone control circuits are often only found in professional or studio grade equipment however due to their larger size and higher cost.

3.2.3.2 Overdrive

There are several existing ways of creating an overdriven effect including overdriving a tube amplifier, driving an Operational Amplifier to its rails or using a clipping network consisting of a pair of diodes. In each case the ability to vary the level of overdrive needs to be included because some artists may wish for more, less or no overdrive.

Undoubtedly the simplest way to create the distortion associated with the overdrive effect is to use a tube amplifier and drive it until its performance starts to break down. However, this limits the choice of amplifier to a vacuum tube amplifier and limits the ability for the artist to control volume unless multiple amplifiers are used in conjunction with one another which, in the case of tube amplifiers, would quickly become too costly and too large for this project.

Another way to produce an overdriven effect is to use a simple clipping network after the signal has been amplified. By using two opposing diodes and two variable DC voltage sources as shown in Figure 8 a reliable and variable clipping distortion can be produced.

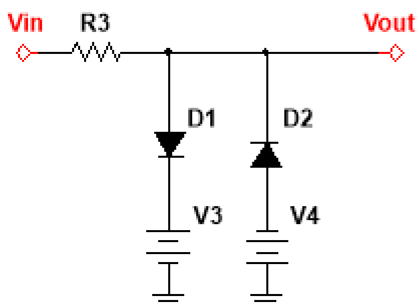


Figure 8. A simple clipping circuit
Created by Joshua DeBaere using Multisim Student Edition.

The network in Figure 8 is governed by the fact that D1 will only pass current when the input voltage passes a threshold.

$$V_{in} \geq V_{T,1} + R3 + V3$$

Likewise, D2 will only pass current when the input voltage falls below a threshold

$$V_{in} \leq V_{T,2} - R3 - V4$$

As such, assuming the current flowing into Vout is very small because it is the input to an Operational Amplifier, the signal Vin will be preserved between these two ranges and clipped off by the dissipation of voltage through the resistor when current is allowed to flow through the diodes once the threshold for turning one of them on is met.

While this method is certainly more practical for small form factor and low cost applications due to its simple, compact nature it does require two variable voltage regulators which are often fairly large due to the need for a heatsink and not very efficient.

3.2.3.3 Reverb

Reverberation, or more commonly known as reverb, is one of the most often used audio effects used when playing a guitar. Reverb is when an echo is created from a sound or signal by reflecting the sound repeatedly after the original sound is produced. The reflections of the sound quickly condense after the impulse and then the pulses of sound decay until they fade out completely. It is easiest to hear the echo when the signal is an impulse or short burst of sound that dies off quickly enough that the diminishing repetitions of the signal can be more clearly heard.

The Sabine and Norris-Eyring equations are two very popular equations for approximating the amount of time that an audio signal takes for its reverberation to decrease by 60 dB. Using the average distance that a sound wave will travel before it is reflected off of a surface in a rectangular room where *V* stands for the Volume of the room, *S* stands for the surface area of the room, *c* is the speed of sound and τ is the mean time until the sound is reflected which yields

$$n(t) = \frac{t}{\tau} = t\frac{cS}{4V}$$

Using RT_{60} as the reverberation time, which is the amount of time that the audio takes to decrease by 60 decibels. The Norris-Eyring formula approximates that

$$RT_{60} \cong -\frac{0.161V}{S\ln(1-\alpha)}$$

where α is the proportion of energy that is absorbed at each reflection. The Norris-Eyring equation works under the assumption that the absorption constant is the same throughout the room. A more accurate formula that doesn't use the assumption regarding the absorption coefficient is

$$RT_{60} \cong -\frac{0.161V}{S\ln\left(1-\frac{1}{S}\sum_i S_i\,\alpha_i\right)}$$

If the most accurate reverberation time approximation is not necessary, then the Norris-Eyring formula can be further simplified into the Sabine equation. The Sabine equation assumes the α is small and that $\ln(1-\alpha) \cong -\alpha$ which yields

$$RT_{60} \cong \frac{0.161V}{S\alpha}$$

The absorbtion coefficient of any material must be between 0 and 1. For a room desired to produce strong reverberations the material of the walls would be designed so that the absorption coefficient would be very close to zero. This would allow the Sabine equation to give a viable approximation of the actual reverberation time. If the Sabine equation was used to approximate the reverberation time of a large open space or a padded room the assumptions that were used to develop the equation would be invalid which would yield an incorrect time for the audio to decay by 60 decibels. These equations will come in handy when designing the reverb effects that we want to provide.

There are several different avenues that could be undertaken to create a reverb effect. Of the different varieties of reverb, they can mostly be divided into two categories: analog reverb and digital reverb. The three main analog implementations of a reverb effect include chamber reverberators, plate reverberators, and spring reverberators.

Chamber reverberators are just what it sounds like, they use a room such as a chamber to create an echo. Usually a speaker is used to produce a sound that is then reflected off of the walls and picked up by a microphone that is aimed at the opposite wall. This type of reverb is clearly impossible for a portable guitar amp because a small size is what is desired.

Plate reverberators are composed of a metal plate that vibrates when a signal is produced by a transducer onto it. A contact microphone picks up the motion of the vibrations on the plate. The contact microphone outputs a signal that can be combined with the original signal to create the reverberation effect. The most obvious reason that a plate reverberator would be impractical for our guitar amp is the same as the chamber reverberator in that the size is not feasible.

Spring reverberators are the third main type of analog reverberators. Spring reverberators are similar to plate reverberators in that they also utilize transducers to create and receive the vibrations used to produce the reverberation. A signal vibrates the spring and then is picked back up at the other end of the spring, this new signal is then added back into the original sound to produce the reverberation. Using a spring reverberator would be the only feasible analog implementation of reverb in our portable guitar amplifier, however we decided against using this because a digital implementation is simpler.

A reverb effect is also possible to create in the analog domain using operational amplifiers. At the output, a summing amplifier is used to add multiple delay amplifiers into the original signal to create one output signal. The operational amplifier on the left side of the circuit in Figure 9 below takes the input signal and delays it. After the signal, has been delayed it is added back into the original signal through the summing amplifier on the right side of circuit to produce a reverb signal. The reverb signal that is produced from the circuit in Figure 9 below however only has one echo. Only one echo in the signal can barely even be considered a reverberation since the basic definition of reverb is multiple echoes after an input signal. To implement the type of reverb effect that is commonly used by musicians several delays would need to be used. For every delay that is desired to be added to the signal, an additional operational amplifier must also be added to the circuit. This would yield a very large and unrealistic circuit for a relatively simple concept.

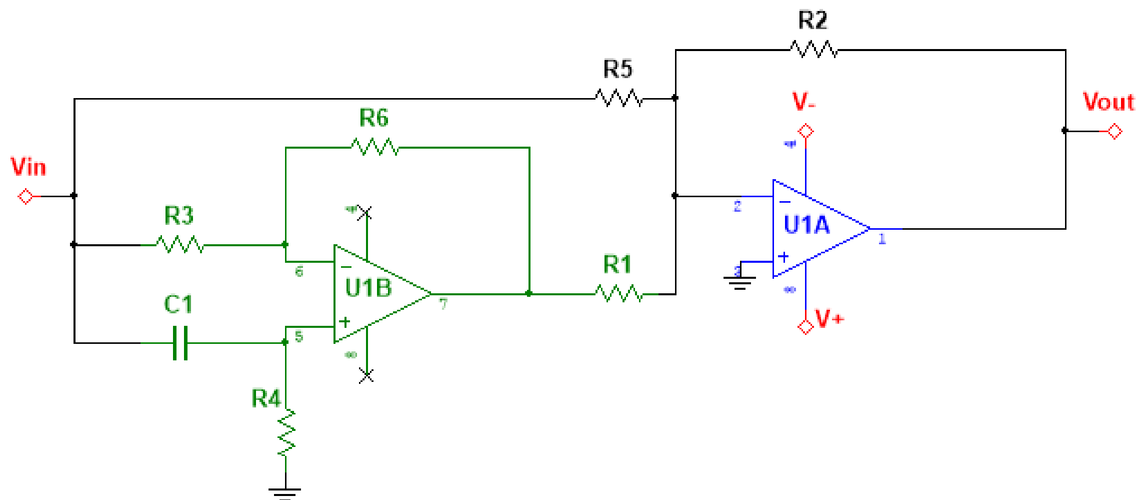


Figure 9. Reverb Effect Using Op-Amp Delays
Created by Joshua DeBaere using Multisim Student Edition

With all of the above implementations of a reverb effect in mind, creating a reverb effect using analog technology would be impractical for our purposes. These executions all require a large physical implementation of materials that are not suitable for a small-scale product. A digital solution to creating a reverb effect in our portable guitar amp is essential. The two algorithmic representations of reverberations that were considered are Schroeder's Reverberator and Moorer's Reverberator. Schroeder's Reverberator, which was developed by Manfred Schroeder in 1961, is a reverberator that is implemented in the digital domain. This reverberator, which is shown in Figure 10 below using a block diagram of filters and in Figure 11 below using a block diagram of the transfer function.

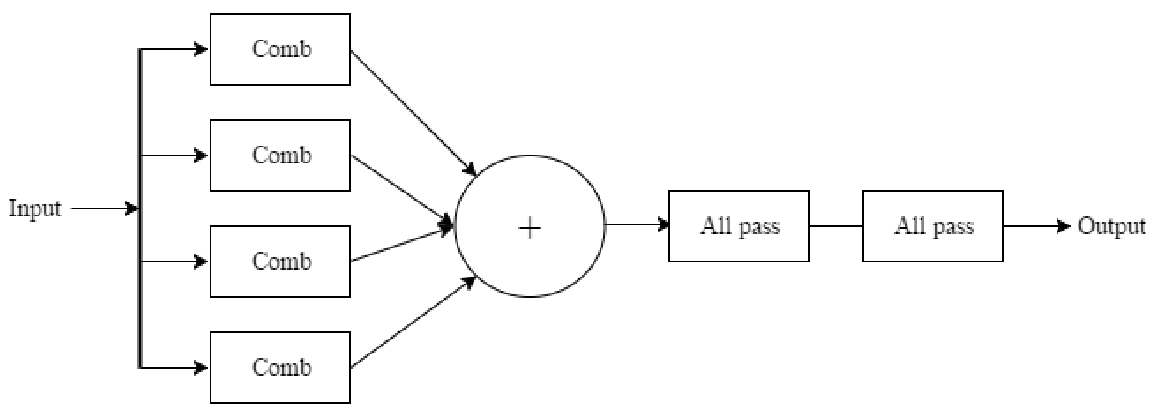


Figure 10. Schroeder's Reverb Filter Block Diagram
Courtesy of Manfred Schroeder

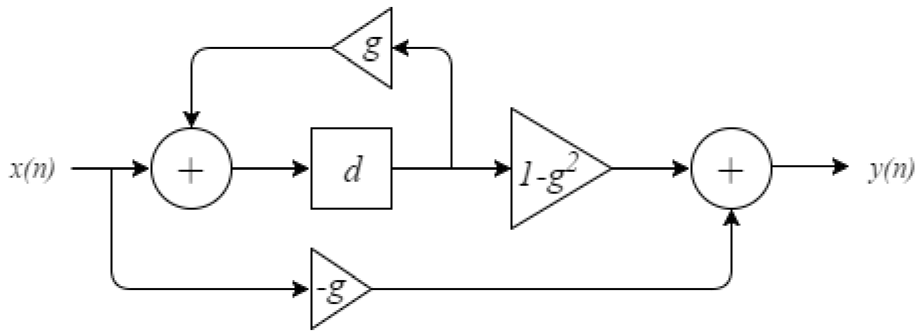


Figure 11. Schroeder's Reverb Transfer Function
Courtesy of Manfred Schroeder

Moorer's Reverberator, which was developed by James Moorer in 1979, is another reverberator that is implemented in the digital domain. This reverberator, which is shown in block diagram format in Figure 12 below, breaks reverb into two parts consisting of the early reflections and the late reflections. The early reflections are the echoes that are heard just after the initial impulse; these reflections are related to the size and shape of the room as well as the material that makes up the room. In a smaller room the reflections of the sound will return sooner than reflections of sound in a larger room so the frequency of oscillation would be higher in a smaller room. The material of the room affects the reverberation of a signal. This is easily shown in a room with metal walls having the sound reflect many times before dying out when compared to a room with fabric-covered walls having the sound absorbed rather quickly allowing little time for the sound to reverberate.

The later reflections have less of a correlation with the room that they are created in; these reflections oscillate more frequently and sporadically since they are out of phase with each other. The later reflections decay exponentially.

This algorithm combines a finite impulse response (FIR) filter to simulate the early waves of reverberations. The initial FIR filter is cascaded with low-pass parallel comb filters to simulate the later reflections. The parallel comb filters are utilized because they are useful in simulating an exponentially decaying impulse response.

In general, reverberation has a tendency to attenuate at high frequencies. This attenuation is reflected in the algorithm by using low-pass filters with the comb filters to achieve the desired effect. The final Z^{-d} delay that is implemented in the system just before the output which allows the late reflections to begin just after the early reflections. This yields a more natural sounding output signal because the later reflections are not meant to be initiated at the same time as the early reflections.

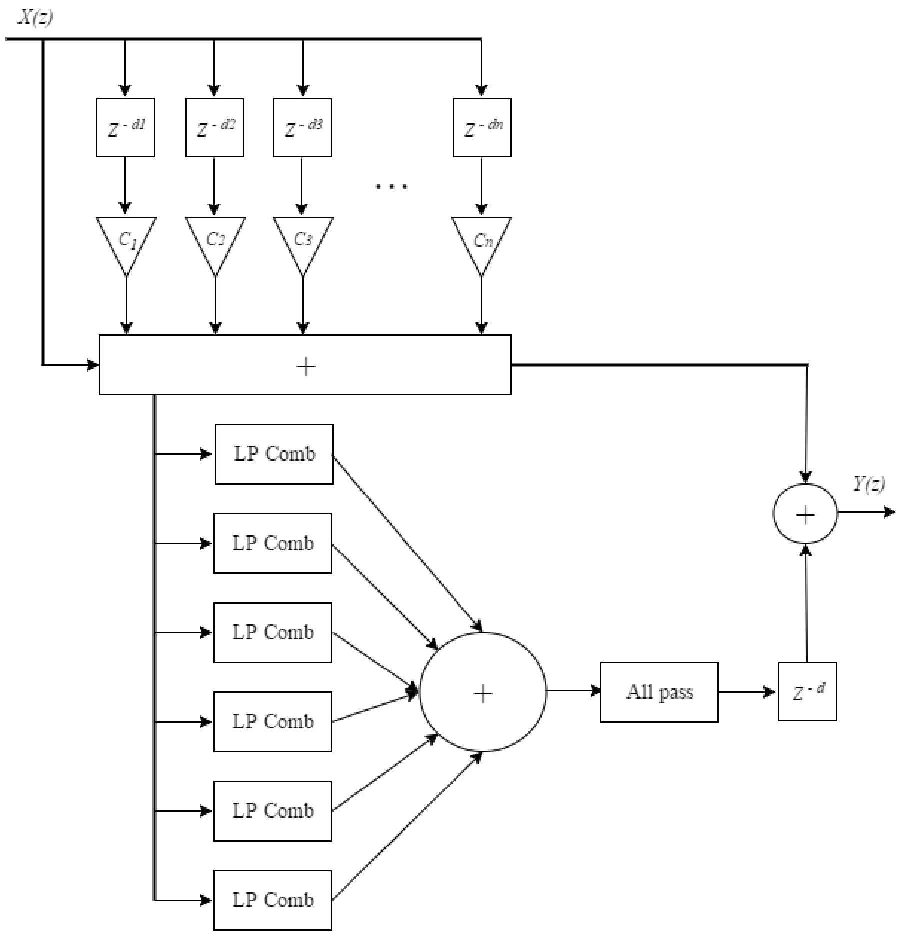


Figure 12. Moorer's Reverberator Block Diagram
Courtesy of James Moorer

3.2.3.4 Flanging

The flanging effect is a very common effect used in music, especially in the rock genre, that is used to create a “whooshing” sound. To implement this sound digitally, an audio signal is split into two pieces and the copied signal then has a variable delay added to it. The delay added to the audio signal to create a flanging effect is in the range of zero to twenty milliseconds; however, the delay is most often between five and ten milliseconds. These delays are very short compared to the standard time that is delayed before an echo begins. A standard delay effect is usually greater than fifty milliseconds after the initial impulse. The delay implemented during flanging is indiscernible as a distinct echo to the human ear because of slight time between sounds.

The delayed signal that is added back into the original input has a variable period which can be created using a low frequency oscillator. Since there is only a small delay between the two signals, at certain frequencies the signals are 180 degrees out of phase from one another and the signals nearly cancel each other out. The resulting signal dips at periodic frequencies which lends to the name “comb” filter since the frequent dramatic dips in the signal resemble the notches in a comb. As the frequency is raised and lowered the signal the peaks sweep up and down through the frequency spectrum.

A digital implementation of flanging using the time domain instead of the frequency domain can build off of the digital implementation of an echo, except using a variable delay instead of a fixed delay. A simulation of a flanging effect is shown below in Figure 13. The simulation shows the divided signal with the variable delay implemented and then added back into the original signal.

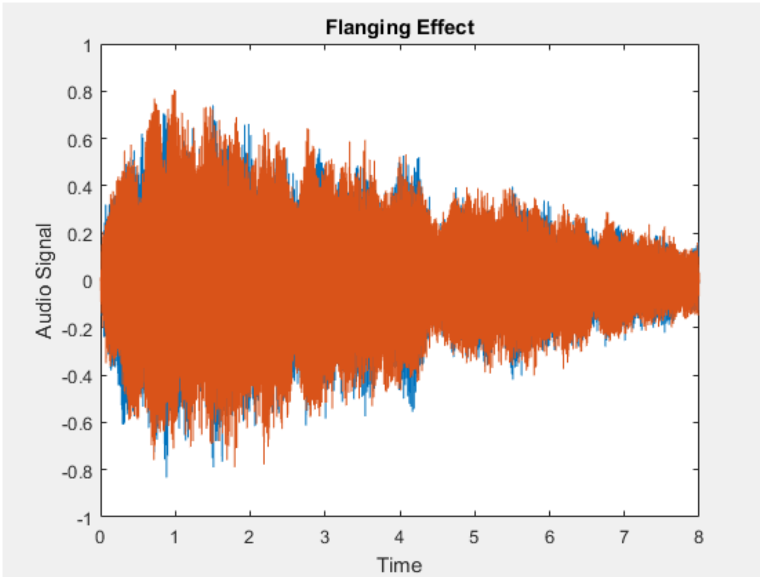


Figure 13. Simulation of Flanging Effect
Created by Marissa Kane using MatLab

3.2.3.5 Delay

A delay effect is one of the simplest audio effects that can be implemented in the digital domain. This effect is created by first delaying an input signal by a specific amount of time. Next, the delayed signal multiplied by some gain is combined back in with the original input signal. This will yield a difference equation shown below that is also represented in Figure 14.

$$y(n) = x(n) + ax(n-d)$$

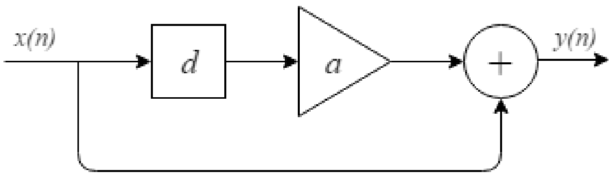


Figure 14. Block Diagram of Delay Effect
Created by Marissa Kane

3.2.3.6 Echo

An echo effect is implemented in a very similar manner as a delay effect; in fact, it is a form of a fixed delay. The distinguishing factor between a standard delay and an echo is that an echo generally repeats multiple times with the volume of the audio decreasing exponentially with each echo. To get more than just a single delayed echo, a feedback loop must be incorporated which can be demonstrated with a difference equation of

$$y(n) = ay(n-d) + x(n)$$

A sustaining echo is actually very similar to the early reflections in a reverberation effect. After an initial impulse signal periodic delayed repetitions of the original signal, produced at a decaying volume are very similar to the same form of oscillations that are heard at the beginning of a reverberation. An echo effect is at a significantly slower rate than that of a reverberation effect which is what produced the main difference in their sound. The echo effect is drawn out in a block diagram below in Figure 15.

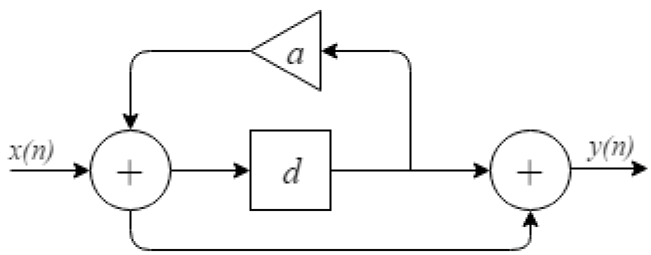


Figure 15. Block Diagram of Echo Effect
Created by Marissa Kane

3.2.3.7 Chorus

A chorus effect is when one instrument is made to sound like there are multiple instruments playing at the same time. When more than one person plays the same type of instrument and the same music at the same time, there are always slight fluctuations in the timing, tone, pitch, and other qualities of the sound, because two people cannot realistically be perfectly synchronized with each other. The sound of a chorus playing a piece gives a much fuller tone to the music and the slight variation creates a harmonic sound. The chorus effect is actually a variation of the flanging effect. One implementation of a chorus effect is simply with the same digital signal processing as a flanging effect, except the delay value is increased to about 20 to 40 milliseconds instead of a flanging effect where the delay value is approximately between 5 and 10 milliseconds. This small change in the implementation of the effect allows for an entirely different sound. The longer delay in a chorus effect than in the flanging effect produces the imitation of two or more instruments playing at the same time.

In Figure 16 below, the block diagram for a chorusing effect is shown. The input is split and the signal is then combined with a variable delay implemented by a low frequency oscillator. The delayed signal is added back into the original to create some feedback. The delayed signal is finally combined back into the original signal to produce the outputted chorus effect.

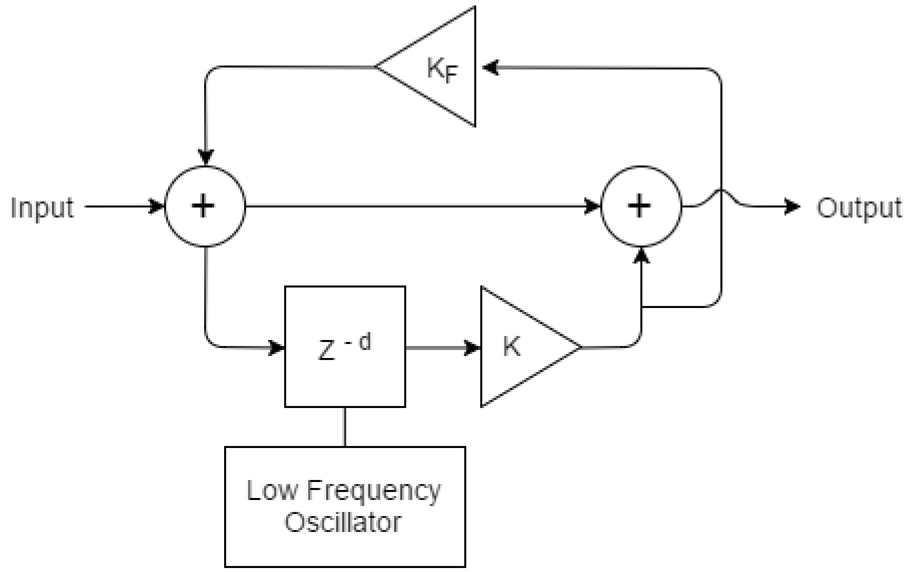


Figure 16. Chorus Effect Block Diagram
Created by Marissa Kane

3.2.4 Volume

The ability to control the volume of the sound is inherent in any audio amplifier. A system to control the gain of the audio must therefore be implemented in our portable amplifier. Once again, the decision to be made is whether to control volume in the analog domain or the digital domain.

Analog implementations of volume control also have their drawbacks. The signal must be passed through a potentiometer that allows variable attenuation to control the volume. Traditional mechanical potentiometers have a logarithmic response allowing more control of louder volumes. Common “logarithmic” potentiometers do not actually have a logarithmic response. It is considerably cheaper to have a pot consisting of two regions of different resistances that divide the voltage to create a stepwise response as a rough approximation rather than to have a true logarithmic potentiometer. This is not a very accurate approximation, but in many cases a perfectly logarithmic potentiometer is not necessary.

To implement the audio, a connector is required to transmit the sound from the guitar to the amplifier. There are a variety of connectors of all different sizes with various capabilities. The most common analog audio connectors come in three sizes: 2.5mm, ⅛” (3.5mm), and ¼” (6.35mm). These connectors, often called audio or phone jacks, typically have between two and five contacts ranging from TS connectors to TRRRS connectors. The ‘T’ stands for tip, the ‘R’ stands for ring, and the ‘S’ stands for sleeve. The two-conductor TS connector is available in all three sizes and is used for unbalanced mono audio. The three-conductor TRS connector is also available in all three sizes and is used in balanced mono or unbalanced stereo audio. The four-conductor TRRS connector and the five-conductor TRRRS conductor are both less widely used so a variety of sizes of the connectors are less readily available with the 3.5mm being most common. TRRS and TRRRS are generally used with microphone input or have analog video on one of the connections but are less standardized. The mono tip-sleeve (TS) ¼” connector is widely used for audio amplifiers because more than two conductors are not necessary for this purpose. The general convention of connectors is that the male plug is on the cable and the female socket is mounted on the equipment.

3.2.5 Amplifiers

There are two main types of guitar amplifiers: vacuum tube amplifiers and solid state amplifiers. Each type of amplifier has its own advantages and disadvantages for various purposes.

3.2.5.1 Vacuum Tube Amplifiers

Vacuum tube amplifiers are highly prized for their “warm”, authentic sound that, while it is not a perfect reproduction of the input signal, is favored by many contemporary and classical artists and such amplifiers still feature prominently in professional grade audio devices used by top tier musicians. However vacuum tube amplifiers are fairly large, commonly ranging from 1” to 6” tall, and require a high voltage to heat up the cathode enough to excite a release of electrons as needed for amplification.

3.2.5.2 Solid State Amplifiers

Solid state amplifiers are those that use transistors or Operational Amplifiers to achieve the desired amplification. These amplifiers are very small, often less than 0.2” square, and inexpensive both in terms of cost and power consumption. However solid state amplifiers are often cited as being “too perfect” and lacking the “warm” slightly distorted sound of their tube counterparts that many artists feel makes them sound less natural.

Because of the size requirements of this project vacuum tube amplifiers must be eliminated as they are simply too large and require too much power to operate. Due to this decision, this section will focus on solid state Operational Amplifiers only.

Many specifications factor into the performance of an Op Amp in an audio network such as the Gain Bandwidth Product, Slew Rate, Offset Voltage, Nose Voltage, Total Harmonic Distortion, Output Current and Price.

3.2.5.3 Gain Bandwidth Product

The Gain Bandwidth Product commonly abbreviated ‘B’ is a metric that details the maximum gain multiplied by the bandwidth of the amplifier. The larger the Gain Bandwidth Product is the better that amplifier will perform for most audio applications as it suggests that the amplifier will be able to operate over a wider range of frequencies. When considering the number of harmonic frequencies generated by guitars this becomes particularly important.

3.2.5.4 Slew Rate

The Slew Rate describes the ability of the amplifier to match the slope of a signal. If the slope of the input signal is steeper than the maximum slew rate of amplifier, then the output signal will be unintentionally distorted creating audio artifacts not intended by the musician. Because there are so many frequencies present in the output of a guitar or other musical instrument the slew rate for audio amplifiers is recommended to be quite high. Naturally this is also dependent upon the output voltage as a signal may be amplified to exceed the slew rate of almost any amplifier. Most audio amplifiers have a Slew Rate above 5 $\frac{V}{\mu S}$.

3.2.5.5 Offset Voltage

Offset voltage describes the output voltage of an Operational Amplifier due to only the internal imperfections and mismatches of that Op-Amp, that is to say the output voltage when the differential inputs share a common voltage. It can also be used to describe the corresponding voltage that, when applied to one of the input terminals of the Operational Amplifier, will result in an output voltage of zero Volts. Ideally the output voltage would be zero in this scenario so as not to introduce a DC offset to the signal being amplified however most audio amplifiers have an offset voltage in the range of tens of microvolts to a few millivolts.

There are several ways to correct for offset voltage imperfections in an Op-Amp, the first of which is to apply a corrective DC voltage to the input of the Op-Amp. This method is straightforward however it is generally difficult to control that voltage source without introducing sufficiently more imperfections. This renders this solution impractical outside a controlled system such as a laboratory with dedicated power supplies. The second solution is to use a potentiometer and voltage divider network in place of the DC voltage source and then tune the potentiometer until the output is zero when the inverting and non-inverting inputs to the Op-Amp are common. While this is certainly simpler than the first and much more practical it is still susceptible to variances of the resistors and potentiometer due to temperature, use and other outside effects. The third way is to capacitively couple sensitive stages of the amplifier in order to filter out all DC components of the signal. Because we do not expect any DC offset in audio signals this is a viable option that is simple and effective.

3.2.5.6 Noise Voltage

The noise metric is a measurement of the noise that the Operational Amplifier itself will introduce into the signal that it is amplifying. The sources of this noise may be, for the most part, thermal noise of each discrete component that makes up the Op-Amp, electromagnetic interference from outside sources, or fluctuations in supplied power. Naturally in an application where most signals will be fairly low power, such as those coming from a guitar output, and highly sensitive to any noise distortion a low noise voltage is preferred, especially when considering how many Op-Amps and their corresponding additional noise voltages that signal must traverse. Because of this most audio amplifiers will have a measured noise voltage of less than 10 $\frac{nV}{\sqrt{Hz}}$ in order to keep the signal to noise ratio of the finished product low.

3.2.5.7 Features

There are several features that manufacturers advertise with their Operational Amplifiers. Only those features that are found in the amplifiers being considered for this project will be discussed. These features include Burr-Brown Audio design, High CMRR, and Shutdown. Many Op-Amps from Texas Instruments are advertised with Burr-Brown Audio since the company Burr-Brown was acquired by TI. Burr-Brown is a well-known and highly regarded manufacturer in the audiophile community and as such the use of their designs in an amplifier gives it credibility and, generally, good performance. A High Common Mode Rejection Ratio means that the ratio of differential voltage gain to common mode voltage gain is high which suggests that the common mode gain is very low, ideally it will be zero therefore making the CMRR infinite as seen in the following equations.

$$CMRR = \left|\frac{A_d}{A_{cm}}\right|$$

$$CMRR_{ideal} = \left|\frac{A_d}{A_{cm,ideal}}\right| = \left|\frac{A_d}{0}\right| = \infty$$

Because the common mode voltage gain is so low in an Op-Amp with a high Common Mode Rejection Ratio that amplifier is the better that amplifier will filter out signals common to both inverting and non-inverting input. These signals are usually the product of noise therefore a high CMRR can reduce unwanted noise in the amplifier output. The final feature highlighted by any of the selected Op-Amps is the Shutdown feature. This feature enables the Op-Amp to enter a power saving mode when not in use. However, since all of the Operational Amplifiers used in this design will be in almost constant use, save when the artist pauses momentarily, it is not predicted that this feature will be of substantial benefit and will therefore be not impact the decision of Operational Amplifier selection as highly as other features will.

3.2.5.8 Total Harmonic Distortion + Noise (THD+N)

Harmonic Distortion is caused when an amplifier introduces harmonic frequencies into a signal that did not exist in the original signal. When the amplifier does this it may cause distortion and inaccurate reproduction of the signal at the output. This distortion is due to the non-linear behavior of active components and as such the amount of harmonic distortion depends upon which of these devices are being used, that is to say there will be a tangible difference between the harmonic distortion caused by a BJT and that caused by a MOS device. Because harmonic distortion affects the additional higher order harmonics added by an amplifier the term Total Harmonic Distortion can be used to describe the overall effect of the distortion due to each harmonic frequency not originally in the signal. THD+N is an even more useful metric as it describes almost all of the sources of distortion as it encompasses the sum of not only the harmonic distortions but also those due to noise. As such this is a very important parameter to consider when designing for a project that requires low levels of distortion of the original signal when required.

3.2.5.9 Output Current and Voltage

Because the output voltage of this amplifier will rely on the voltage supplied by the rails, not by the amplifier in and of itself, the maximum output voltage parameter becomes much less significant. For this reason, this section will focus on the maximum output current and power an amplifier can supply.

The output current metric of Operational Amplifiers is important to consider when selecting what kind of load the amplifier will drive. Obviously the higher the output current the larger possible load the amplifier is capable of supplying power to at a reasonable volume. Because this project is concentrating on driving headphones this section will assume that impedance ranges of the load will be those of headphones. Most generic, consumer grade headphones have an impedance of around 25 Ohms which would require only a low current to drive. However, there are studio and professional grade headphones that have impedances ranging from 50 to 600 Ohms that would require a much larger output current to produce a signal at a decent volume. This does depend upon the sensitivity of the headphones which is usually measured in decibels per milliwatt however since we are considering output voltage to be a non-factor as explained above we can directly relate the volume of the output and the headphone input current. Generally, the lower the impedance of the headphones the higher the value of sensitivity while the higher the impedance the less the amplitude of the output will change with respect to input current. This can be expected since the lower an impedance is the higher the current flowing through it will be for a given voltage.

3.2.5.10 Price

Naturally, price is a factor to consider when selecting an Operational Amplifier. In the case of this project the final product is meant to be an affordable amplifier and as such component costs should be kept to a minimum while keeping performance at reasonable levels.

3.2.6 Audio

Audio interfacing comes in two parts, analog-to-digital conversion and digital-to-analog conversion. These two interfaces are how audio is discretized from the analog domain into the digital domain and back again. While it’s not entirely necessary to convert to digital, amplifiers can perform many more effects if they make the change to digital.

3.2.6.1 ADC

An analog-to-digital converter (ADC) converts a continuous analog signal into a digital signal composed of bits. The most common digital output that an analog-to-digital converter outputs is a two’s complement binary number. ADCs are generally implemented as integrated circuits. To converter a signal from the analog domain to the digital domain, the signal has to go through the processes of sampling and quantization. Sampling converts the analog signal into a discrete-time signal. Quantization approximates each discrete value that is often represented as fixed-point words. Once a signal is quantized, this introduces quantization error that is produced from the necessary approximations in the process. The quantization error of a conversion can be lessened by increasing the number of levels that the signal is converted into. For every sample that is taken of a signal, those sampled values are quantized. Analog-to-digital converters are characterized by its bandwidth, which is defined by its sampling rate.

3.2.6.2 DAC

A digital-to-analog converter (DAC) converts a digital signal consisting of bits into a continuous analog signal. There are six main parameters that determine the suitability of a DAC for a specific application: physical size, power consumption, resolution, maximum sampling frequency, accuracy, and cost. Audio converters generally have a low speed and a high resolution. The resolution of a converter is the number of bits of depth that the DAC can reproduce. The audio bit depth is the number of bits in each sample. Audio signals are stored in digital data and with respect to a guitar amplifier; some of the effects that will be implemented on the signal will be executed in the digital domain. Total harmonic distortion (THD) is another measurement of accuracy of a DAC. This is the amount of noise and distortion that is introduced to a signal when it is converted. The total harmonic distortion and noise is expressed as a percentage of the total power. THD characterizes the linearity of audio systems and the accuracy of their reproduction. THD_F is commonly used as the notation for the percentage of THD in audio distortion specifications where V_n is the RMS voltage of the *n*th harmonic and n=1 is the fundamental frequency.

$$THD_F = \frac{\sqrt{V_2^2 + V_3^2 + V_4^2 + \cdots}}{V_1}$$

3.2.6.3 Audio Sampling and Bit Depth

The audio sampling rate and bit depth will directly affect our audio quality. Audio sampling rate is the frequency that the ADC will sample and hold the input voltage steady. Audio sampling is the frequency that the DAC will adjust its output voltage level. Bit depth is the number of bits per sample, which corresponds to the range of values that can be expressed between the maximum and minimum voltages and determines the dynamic range of the system.

Audio sampling follows the Shannon-Nyquist theorem where perfect reconstruction of a signal can be achieved by sampling at twice the rate of the signal. For audio, most frequencies are between 80Hz and 10kHz and the extent of human audio is approximately 40Hz to 20kHz. This means that we can perfectly reconstruct the entire range of human hearing with a sample rate of 40kHz. CD audio set the de facto standard for audio quality sampling by setting the sample rate to 44,100Hz, allowing some frequencies that most humans will be unable to hear in order to potentially capture more high-order harmonics.

Since the higher frequencies cannot be sampled properly, they show up as extremely low frequency noise which is audible to humans after sampling and converting back to analog. Higher sampling rates are sometimes used in professional audio equipment (namely 88,200Hz to allow for down-sampling to CD rates) which reduces the distortion caused by miss-sampled higher frequencies. In addition, low-pass filters are used to attenuate any higher frequencies before and after sampling.

Audio bit depth is not, as many people claim, simply the “resolution” or choppiness of the output wave. Any commercial DAC will be anti-aliased to some degree and the discrete nature of the output wave hidden. Instead, bit depth is directly related to the dynamic range of the output signal. The dynamic range is the decibel rating between the lowest volume signal and the highest volume signal, with every bit equally 6db of dynamic range. A properly set up sound system will have the minimum signal volume correspond with what’s called the “noise floor” of the room. The noise floor is just all of the background noises combined. Studios will have exceptionally low noise floors, homes will have moderate noise floors, and places like outdoor venues will have very high noise floors. The maximum volume of the signal is the noise floor plus the dynamic range, for example, the background noise of a quiet residential room is generally around 40db, so a bit depth of 8 would give a maximum volume of 88db, which is about the noise of a car idling heavy and is approaching the need for hearing protection for extended listening.

The standard bit depth for audio applications is the audio CD with 16-bits giving us 96db above the noise floor. This gives our quiet house a maximum db rating of 136db, which is well within hearing instant damage territory. Some DACs have much higher bit depth with many high-quality ones boasting 24-bits, resulting in 144db of dynamic range. This alone is as loud as a jet taking off next to you and will instantly cause pain and hearing loss. Shifting up the volume to account for the noise floor would result in death to any listeners within the formerly quiet residential house. 24-bit DACs don’t actually make use of the extra bit depth, and instead lower their minimum volume to allow the user to drop the volume below the noise floor of the environment and still hear the audio just fine. This reduces the amount of adjustment that is required by the end-user, and only marginally improves the quality of the audio over a DAC with slightly fewer bits.

3.3 Computer

The computer components researched consist of the components from phone to the interface to the chips. This includes the phone platforms, communication between the phone and Pocket Amp, processors for the Pocket Amp, and inter-integrated circuit communication protocols.

3.3.1 Processors

Due to the low-power nature of this system, the processor architecture must have been designed with energy efficiency in mind. The entire system will be run off of batteries so long-term power consumption must be limited if possible. Short-term power consumption will vary greatly as effects are turned on or off. This leads us to require many power modes and modularity in our chip. In some modes, the processor will essentially hand off the digital signal to the DAC and keep listening for Bluetooth instructions. In some modes, the processor will have to be real-time modifying a digital audio signal.

In addition to power requirements the real-time nature of audio requires us to have a fast processor with enough processing power to accomplish any effects while still controlling the hardware and the communication systems of the system. Any lag due to background tasks or due to not having enough processing power to accomplish effects would cause undesirable skipping, distortion, or pauses in audio output.

In addition to processing power constraints, architecture word size and clock rate is also of a concern for audio processing. ADCs generally output 10-24 bits per clock cycle, at a clock cycle rate of about 16-96kHz. A processor with small word sizes (mainly 8-bit processors) will not be able to store ADC output in a single word and would end up with exponentially more instructions to perform on a given audio sample. A processor with a clock rate low enough would make real-time audio processing impossible as the processor would either have to throw away samples or would be slowly falling farther and farther behind the audio.

Because of the electrical and processing power constraints, the main focus of our research was on ARM processors with a few non-ARM processors selected for comparison. In addition, we considered some field-programmable gate arrays due to the ability to create and modify effects through logic gates. The ARM processors we researched were picked due to their widespread use in the prototyping and IoT applications, giving us plenty of development tools. The non-ARM processors we researched were also ones with well-developed prototyping tools.

3.3.1.1 ARM

ARM processors are known for being low electrical power for the amount of computing power they deliver. Due to their widespread use in everything from Internet of Things to phones to laptops to server arrays, ARM processors are well-developed technology.

3.3.1.1.1 MSP432 (Cortex-M4F)

The TI MSP432P401R processor is a 32-bit 48MHz ARM Cortex-M4F processor. The MSP432 is a low-power ARM processor with built-in support for many basic analog and communication functions. The standard operating voltage range is low but very large, ranging from 1.62v to 3.7v. Being battery driven, this processor would be able to take a large range of voltages regulated from the battery and not worry about drops due to load. The processor has built-in low power modes with the lowest being 25nA.

The MSP432 has a built-in 24 channel, 14-bit differential ADC offering up to 1 million samples per second. As for communication, the MSP432 offers up to 4 I^2C

connections, 8 SPI connections, and 4 UART connections. The MSP432 has 256kB of Flash memory, which is approximately 64 thousand ARM assembly instructions - more than sufficient for our purposes. In addition, the MSP432 contains 64kB of RAM. The chip itself is only 3 to 4 dollars from TI.

The TI MSP432 Launchpad uses the same chip, allowing for rapid development without having to first develop a development board for the microcontroller. The MSP432 Launchpad allows us to program and debug the MSP432 while it's on the board, then disconnect all of the Launchpad features to treat it like a large breakout board to test the functionality. After debugging and testing, we have the choice of desoldering the programmed chip from the Launchpad and resoldering it on our board or just programming the chip on our board without the debug features of the Launchpad. The Launchpad kit is inexpensive, at only 13 dollars for the kit directly from TI.

The MSP432 seems a great candidate for our project, as it is very cheap, very efficient, more than fits our requirements, and has an onboard ADC. The addition of having Launchpad kits, TI's development tools including Code Composer Studio IDE, and the relative popularity of ARM and the MSP line makes this an easy chip to develop for.

3.3.1.1.2 Atmel SAM3X8E (Cortex-M3)

The Atmel SAM3X8E is the processor the popular Arduino Due board is based on. The SAM3X8E operates at the wide range of 1.62v to 3.6v. Operating with the standard ARM 32-bit instructions it has a max frequency of 84MHz. The SAM3X8E has an ADC/DAC module with 16 12-bit ADC channels and 2 12-bit DAC channels, solving our need for analog conversion. The SAM3X8E contains 4 SPI connections, 2 I^2C connections, and 5 UART connections for inter-chip communication. SAM3X8E has 96 kByte of RAM and a half megabyte of non-volatile memory. The smallest package offered for this chip is 9mm by 9mm, with the largest being 20mm by 20mm. As far as specifications are concerned the SAM3X8E far exceeds anything we would need.

The Arduino Due is the main consumer application of the SAM3X8E, which gives us a well-maintained development toolset through the Arduino platform. Though it is an Arduino, it is the only ARM-based Arduino product, and as such does not benefit fully from the ARM community and platform. The Arduino Due is on the more expensive side of development kits for low-power, at 50 dollars for the board. Besides basic programming and power regulation, the Arduino Due doesn't offer any extra features over a breakout board of the SAM3X8E chip.

The SAM3X8E chip overshoots our requirements for a microcontroller, and the 8 dollars per chip price reflects that. The integrated ADC/DAC gives makes this board very competitive for our project, but the 8-dollar price tag and large package size detracts from this choice.

3.3.1.1.3 TI Sitara AM335x ARM Cortex-A8

On the more powerful side of ARM microcontrollers is the TI Sitara AM335x, which is the chip the BeagleBone board uses. The BeagleBone is capable of using lightweight Linux distributions, making higher level tasks such as programming and communication extremely easy for us. Due to the overpowered nature of the Sitara line for this application, only the AM3351 was considered, being the lowest power of the line.

The AM3351 is a 1GHz processor with 288kB of cache memory split between L1 and L2 cahches, 64kB of on-chip RAM as a shared L3 cache, and 176kB of on-chip non-volatile memory. The onboard 12-bit DAC only provides 200k samples per second and is designed for resistive touchscreen applications, not audio applications. The processor allows for inter-process communication and is built for time-sharing operating systems, not real-time applications.

Due to the design change from MCU focus to processor and OS focus, the low-power Sitara was deemed an unfit processor for our application, and similar higher-powered ARM processors and MCUs were not explored further. The BeagleBone price tag of 80 dollars was also deemed too high, though the chip cost of 5 dollars is reasonable for our product.

3.3.1.2 Non-ARM

There's a vast number of architectures available for microcontrollers besides ARM. They come in a wide range of power levels, bit widths, frequencies and with varying levels of support for hardware and firmware.

3.3.1.2.1 MSP430F5x

The MSP430 line was heavily considered due to the team's familiarity with the system. The MSP430 is a non-ARM, 16-bit RISC architecture designed for low-power applications. The maximum clock frequency is a relatively low 25 mHz. The MSP430 does have a 12-bit ADC, though due to the small instruction size and low clock rate, computation power would be limited. The major specification draw to the MSP430 is the average power consumption of 5.8mA when running at full execution speed from non-volatile memory (worst case scenario) and low power modes all the way down to .1 μA at a shutdown mode.

Due to the low processing power the MSP430 would most likely not be able to perform sampling, effects, and output without getting dangerously close to being overloaded. Due to the possibility of not being able to implement CPU-heavy effects, we did not select the MSP430.

3.3.1.2.2 ATmega328P

The Atmel mega328P MCU is the basis for the Arduino Uno board, which is the standard Arduino board. The mega328P is only 8-bits, has just 2kB of RAM, and operates at 20MHz. The MCU does contain an ADC, but is only 10-bits and samples 15k times per second, requiring a discrete ADC for any actual audio application. The extremely low specs would be unable to handle even the most basic effects without discrete ADC and DAC chips, and would never be able to process data fast enough to handle most digital effects. The Atmel mega328P would not work with our application.

3.3.1.2.3 Intel Edison

The Intel Edison is Intel’s introduction to Internet-of-Things markets. The Edison is not a single MCU, but is actually a “computer on a module” where a very small CPU and a small MCU are integrated with RAM and Memory along with other components. The entire package uses a Hirose 70-pin connector to attach to the 35x25x3 mm board. The CPU is an Intel Atom System-on-a-Chip with two dual-threaded cores running at 500MHz. The MCU onboard is a 32-bit Intel Quark running at 100MHz. The Edison has built-in Bluetooth, WiFi, 1Gb of RAM, and 4Gb of memory in addition to having a lightweight Linux distribution already onboard the CPU and a real-time operating system built in the MCU.

The Intel Edison solves many of our design issues, with communication and low-level programming already done for us. Building effects onto a multi-core processor would be extremely easy and using the onboard MCU to control analog effects would also be easy. The drawbacks are that it is power hungry (3.3v - 4.5v and requiring 7mA at standby with Bluetooth) and costs 46 dollars for just the chip. In addition, while most MCUs have operating temperatures up to 60C or higher, the Edison caps out at 40C operating temperature — requiring us to consider large heat sinks or active cooling to keep such a power-hungry component cool in a case full of amplifiers and batteries.

3.3.1.3 FPGA

Field-Programmable Gate Arrays are by-far the most power digital signal processors in the microcontroller realm. Due to the vast number of digital and analog effects that can be implemented by FPGAs, many real-world high quality audio products use FPGAs, from sound mixers to modelling amplifiers, FPGAs are an extremely great choice for audio. The team initially considered FPGAs for this application but due to costs of high quality FPGAs and low quality FPGAs having lackluster development environments and questionable performance for audio, FPGAs were decided against for this project. In future iterations of small portable amplifiers such as the Pocket Amp, FPGAs would make an excellent choice for audio effects.

3.3.1.3 Overview

Table 4 below outlines the ARM and non-ARM processors we’ve reviewed. The final decision was made to select the MSP432 for our processor, since it appeared to be the most suitable for our application. The wide range of operating voltages, use of ARM 32-bit architecture, low-cost and power consumption, and excellent development environment lead to this being the best choice for the project.

Processor	V_{Low}	V_{High}	A_{Min}	A_{Avg}	MHz	Form Factor	Pins	Cost
MSP432	1.62v	3.7v	2.5nA	3.84mA	48	LQFP (PZ)	100	$3
ATSAM3X8E	1.62v	3.6v	2.5μA	???	84	LQFP/TFBGA	100/144	$8
Sitara AM3351	~1v *	~1.5v *	3mA	???	600	S-PBGA	268/324	$5
MSP430F5x	1.8v	3.6v	0.1μA	4.9mA	25	LQFP	80	$5
ATmega328P	1.8v	5.5v	15μA	4mA	20	PDIP/TQFP	28/32	$2
Edison	3.15v	4.5v	7mA	???	500	Hirose	70	$46

*Requires many voltage inputs of varying voltage ranges

Table 4. Processor Comparison
Created By William McKenna

3.3.2 Interfaces

The inter-chip communication system is extremely important to the design of our system, as it allows information to flow between integrated circuits. These interfaces are used in everything from digital potentiometers to microcontroller communication. While many types of inter-IC communication protocols exist, we will be researching the two that allow multiple devices per bus since having five to ten chips hooked up to serial port pins on our microcontroller would be difficult.

3.4.2.1 SPI

Serial Peripheral Interface (SPI) is a de facto standard developed by Motorola. It is similar to serial ports in the way data is transferred, but has an additional clock signal line and one selector line per “slave” device. SPI is a master-slave protocol; one device is selected as the master and every other device in the setup is a slave device. SPI is a full duplex communication protocol. A slave and master can both send and receive information at the same time - though in practice it is usually used as half duplex where one is sending and one is receiving. SPI is a synchronous protocol, where communication is always structured around a clock signal.

Physically, SPI is characterized by $(n - 1) + 3$ communication wires, where n is the number of devices, therefore (n - 1) is the number of slave devices in the network. The master connects to every wire in the SPI network, while slaves do not. Master wires are SCK, MOSI, MISO, and SS0 through SS(n - 1). For example: a network of 12 devices will have one master and 11 slaves. The master’s wires will be SCK, MOSI, MISO, SS0 through SS11. Each slave has four wire inputs, SCK, SDI, SDO, and CS.

The SCK wire is the serial clock signal. The serial clock signal is driven by the master device. All communication takes place on a clock signal edge, with four modes of operation. The four modes determine whether the clock is active-high or active-low and whether communication happens on the rising edge or falling edge of the clock. To maximize stability of signals, data is transmitted on one edge of the clock signal and is read on the other. For example, transmitting on the rising edge of a clock signal allows a half wavelength of the clock for the signal to stabilize before it is read. Because of this design slower clocks will lead to fewer errors in communication due to impedance or capacitance issues.

From the view of the master the two data lines are the MOSI (master out slave in) and the MISO (master in slave out) signals. From the view of each slave devices the MOSI is SDI (serial data input) and MISO is SDO (serial data output). These two half-duplex lines create the full-duplex serial data bus of the SPI interface.

The SS wires are slave select wires. Slave selection is usually active low connections, where the device the master is communicating with will have a low voltage on the respective SS line when that device is selected. Since only one slave can be selected at a time, SS wires are usually run from a demultiplexor, which reduces the number of pins the master has to use to control the slave devices.

The standard setup of SPI is that each slave and master maintain a shift register and optional buffer register of an agreed upon number of bits, usually 8 to represent a full byte. The input from the other device is shifted into the shift register (commonly called SSPSR for synchronous serial port shift register) as the output from the device is shifted out of SSPSR. After SSPSR has been fully shifted through (8 clock cycles for a byte) it is read to the buffer SSPBUF and what was in SSPBUF is written to the SSPSR. The two devices at this point have effectively communicated their respective byte to each other.

3.4.2.1.1 Pros of using SPI

- Few wires for few slaves
- Fast due to synchronicity
- Simple hardware and software implementation
- Master-Slave scheme matches our overall design
- Not very susceptible to line capacitance or inductance

3.4.2.1.1 Cons of using SPI

- Wires scale with number of slaves
- Requires well-formed data packets (can’t just send an arbitrary amount of data)
- Full-duplex is not needed in almost all cases and requires two wires

3.4.2.2 I2C

I^2C was originally developed for Philips ICs and eventually released as a public communication specification. I^2C consists of two wires that connect all devices. Though it does have a master-slave design, I^2C can have multiple masters (though masters cannot communicate to each other through the I^2C system). I^2C

has a higher-level protocol stack than most IC communication systems; I^2C requires every packet to contains a start condition followed by an address frame followed by multiple message frames and finally a stop condition.

I^2C is physically implemented with an “open drain” configuration. The bus is kept at a high voltage until someone uses it, at which point the user pulls the line to a lower voltage. This allows multiple masters to attempt to transmit at the same time; bus contention for listening or transmitting will not damage the hardware.

The open drain configuration also allows for communication between devices that use different but similar voltages. As long as both devices can detect the difference between the high voltage and low voltage it doesn’t have to have voltage signals shifted to each device’s voltage levels.

I^2C protocol overhead is generally low, but not non-existent. For every byte sent over the bus, there is an additional ninth bit used for acknowledgement of the data. In addition to the 11% overhead caused by the ACK bit, every packet must have an address frame of a 7-bit address and an additional read/write bit. Some implementations of I2C allow for a 10-bit address, which requires two full address frames. Worst case scenario for overhead is transmitting one byte of data with 10-bit addresses. Of the 27 bits being sent, only 30% is actual data being transmitted. Best case is one 7-bit address frame and an arbitrary, extremely long data stream being sent. This scenario provides approximately 89% data being transmitted, where the main source of overhead is the 9th bit of every frame being the ACK bit.

The two wires of I^2C are SCL and SDA. SCL is the clock signal, usually driven by the currently transmitting master though it can be forced low by a slave to signal to the master to slow down data transmission. The slave forcing the clock low causes the master and slave to stop sending and receiving data which allows the slave to catch up with processing data from the master. SDA is the data line which, as described above, is a half-duplex active-low line.

To demark a complete packet, I^2C requires a “start condition” and a “stop condition.” The start condition is nobody is currently using the SCL signal (nobody is transmitting so it is held high) and a master bringing the SDA signal low. This initiates the start of a packet; at this point the master that pulled the SDA signal low knows that he has control of the I^2C bus. After pulling SDA low the master begins the SCL signal, and the slaves start listening for the address. At the end of a packet the stop signal is SCL held high, and SDA transitioning from low to high. The choice of this as a stop signal is that no transitions should occur during the SCL high period, and SDA going from low to high can only be achieved by every device letting SDA go high. This prevents erroneous stop signals or for greedy masters to force a stop signal.

In addition to starting and stopping a packet, the I^2C specification deals with handling failures or edge cases such as repeated starts, invalid addresses, and arbitration of bus contention between masters. A master getting disconnected from the network while transmitting will unintentionally produce a stop signal, due to the nature of the stop signal. For bus arbitration, the master that is either faster (on SCL) or is transmitting more 0's on SDA gets priority. The master that does not have priority must stop transmitting and allow the other master to finish. Bus contention rarely occurs even with multiple masters since the masters must first sample the SCL to check for current transmissions. Repeated starts are handled by using the previous slave address as the current address and treating the new frame as a new data transmission to the slave.

3.4.2.2.1 Pros of using I^2C

- Only requires 2 wires
- Addressing allows a very large number of devices to be hooked up to
- Flexible bus voltage
- Multiple masters (though we are unlikely to use this)

3.4.2.2.2 Cons of using I^2C

- Complicated protocol stack leads to high software overhead
- Half-duplex, controlled by master requires us to poll any input sensors
- Active low bus is susceptible to line capacitance and line inductance
- Address collision occurs with similar chips, addresses are at least partially selected by manufacturer

3.3.3 Phone Platforms

There are three smartphone platforms with over 1% market share according to comScore's March 2016 report on smartphone market shares (www.comscore.com: "comScore Reports January 2016 U.S. Smartphone Subscriber Market Share"). Researching development for a phone platform with fewer than 1% of a market share would be illogical as creating an app that fewer than 1% of smartphone owners could use would severely limit our market impact. In addition to accessibility to customers, smartphones with smaller market shares will generally have less matured development tools due to smaller developer bases. The distribution of apps to customers is generally more difficult on smaller market share devices – the Apple App store, Google Play Store, and Windows Phone Store have been working with many thousands of developers to streamline the process of getting apps to customers. In addition to customer concerns, the engineering team currently owns only Android and Apple devices and as such, developing for other systems will increase the cost of the Pocket Amp development as the team would have to acquire devices to test and build on.

3.3.3.1 Android

Android is Google's Linux-based phone operating system currently deployed on approximately 53% of all US smartphones, along with many other devices ranging from tablets, televisions, desk phones, netbooks, laptops, and many other devices with a processor and user interface device. Development for Android is done through Google's Android Studio. Apps are distributed through the Google Play Store, or can be directly downloaded and installed on phones without root access being required for most. Due to the vast array of products that run and support Android, there is quite a wide array of hardware and software setups of Android systems.

Developing for Android is done mainly through Google's official Android Integrated Development Environment (IDE) called Android Studio. Android Studio is cross-platform compatible on Windows, OS X and many Linux OS's. Android Studio was designed to be an all-encompassing IDE; it contains a full build system including the ability to change built packages while running, an emulator for many different types of Android devices, testing and profiling tools, and support for many team development systems like GitHub and Google Cloud. Android studio is largely a coding IDE where the main developer interface is a code editing screen with intelligent code completion, error catching, and other tools expected out of a modern full-featured coding IDE.

Getting apps on the Google Play store is easy and cheap. To be able to publish to Google Play a developer must register for a Google Publisher Account. Registering requires giving Google basic information such as name, email address, physical location, then reading and agreeing to Google's Developer Distribution Agreement for the physical location of the publisher, then the publisher must pay a $25 registration fee. The registration fee is fairly low to not inhibit developers (almost any app will make at least $25) but is high enough that it deters people from making spam or junk accounts. Google also recommends setting up a payment account if the app is not free, if it has ads, if it has subscriptions, or if it has in-app purchases. After setting up these two accounts a developer can easily publish to Google Play, where there is a basic automated check for appropriate description of the app and check to ensure there is no copyrighted information in the description or logo or screenshots. After this, an app can be removed from Google Play for a variety of reasons, such as low ratings or inappropriate content.

Android program packages, called apks, can also be distributed directly to Android users through downloads to their phones. This is generally done for apps that do not conform to Google's terms of service for various reasons. This also removes Google's support for the app, namely the apps won't update automatically. The user of the phone must also allow external installs on their phone. This is an easy setting to change and the change is prompted by the phone when the user attempts to install the apk, but is another small hurdle to avoiding the Google Play store.

To combat the issues of having Android on such different hardware setups, Android provides tools to alter the app based on the hardware it is installed on or limit what hardware can install the app. This solves the issue of users attempting to install apps on refrigerators or other devices based on Android; the device can only install things that are compatible with its hardware. In addition to blocking hardware from installing, apps can resize or otherwise change themselves to better accommodate the hardware they are one. In addition to device compatibility, Android grants apps almost no access to the hardware they are on without requesting it, which usually results in either the OS granting the permission, the user being asked at install-time, or the user being prompted at run-time for whether they want to grant the app the requested permission.

3.3.3.2 iPhone

Apple iPhones make up 43% of the US smartphone market. The iPhone runs on Apple's iOS and delivers apps through the App Store. Development for iPhone apps is done through Apple's iOS SDK in either Objective-C or Swift, a proprietary programming language and proprietary scripting language respectively. The iOS SDK contains Apple's Xcode, full-featured IDE for OS X environments, which is used for the consumer-level programming for iOS. The iPhone SDK requires an OS X environment of some kind to be run. Options besides buying a Mac OS X laptop include running server versions of OS X in a virtual machine or using a cloud OS X service. All of these options are expensive for creating a developing environment.

To put an app in the Apple App Store requires a developer to pay a $99 yearly subscription and share 30% of the profits of the app with Apple. In addition, if a user requests a refund and Apple grants it to them then the developer must pay back 100% of the app's cost to the user – Apple keeps the 30% of the price. Apple also has the strictest removal policy and has occasionally exercised their right to remove apps that don't even conflict with their app store policy.

3.3.3.3 Windows Phone

The Windows Phone only boasts less than 3% of the market share of smartphones, so research into this platform was limited as the potential impact of this platform is limited. Windows Phone is Microsoft's smartphone operating system, which closely mirrors the Windows Operating System's design and release scheme (for example Windows 7 launched around the time of Windows Phone 7). Windows Phone apps can be developed for Windows OS simultaneously using what's called the Window's Runtime. Apps are developed using C#, Visual Basic.NET, C++/CX, or HTML5/Javascript and generally are tested and developed using Microsoft's Visual Studio IDE with Windows Phone Extension Tools.

Windows Phone Store acceptance guidelines are stricter than many. Before an app is even accepted it must be reviewed by Microsoft employees to ensure the quality of the app and verify it does what it says it will do. Microsoft will take 20% of all revenue generated by an app, and does charge a yearly fee of $99 dollars to developers to be able to submit any apps. If a developer submits more than 100 apps that do not generate revenue, then there will be a $20 fee per app submission. Windows also has the strictest content policy, banning any apps that include adult content including even alcohol or tobacco use.

3.3.4 Communication Platforms

There are many communications platforms that can bridge the gap between phone app and Pocket Amp. This section summarizes a few selected technologies relevant to communication, including Bluetooth, Infrared, general radio communication, and ZigBee.

3.3.4.1 Bluetooth

Bluetooth Classic and Bluetooth Low Energy are two low energy radio communication standards managed by the Bluetooth Special Interest Group. Bluetooth is used for piconets – personal area networks of two to eight Bluetooth devices – and generally for computer peripherals and audio applications. Bluetooth Classic is the main branch of the Bluetooth standard, used for connection oriented communication like audio streaming or most computer peripherals. Bluetooth Low Energy is an offshoot of Bluetooth classic that is more low-energy application-centric by focusing on client/server ephemeral connections instead of maintaining constant connections. Bluetooth is implemented in almost every Android phone, making it an excellent choice for our application since it will not require any peripherals to communicate with the Pocket Amp.

The Bluetooth standard is examined in-depth in the Standard's section of this document, but an overview is supplied here. Bluetooth is a radio communication within the 2.4gHz ISM band, with frequency hopping in Classic and a mixture of frequency and time domain modulation for Low Energy. Bluetooth Classic has a very basic connection oriented stack while Low Energy implements more layers allowing for higher levels of abstraction to overcome the issues of a connectionless client/server relationship. Classic and Low Energy both implement security to encrypt and authenticate connections, though for our uses this is not extremely important. Both Bluetooth and Low Energy would work for our application but due to the energy requirements and the type of data we will be sending Bluetooth Low Energy seems a very good candidate for the Pocket Amp.

3.3.4.2 Infrared Radiation

Infrared technology is a very useful communication and control system that is utilized mostly in short-range transmissions, but it can also be used for some medium-range systems. Infrared communication is often classified as optical communication, however this is actually a misnomer. Optical technology actually refers strictly to visible electromagnetic radiation, while infrared radiation is invisible. Infrared communications work by emitting a narrow beam of infrared radiation that is being rapidly switched on and off. The modulation of the signal is the factor that helps to filter out any unwanted infrared radiation such as ambient sunlight or non-natural lighting. The infrared radiation from lighting changes slowly so the receiver can easily differentiate that from the desired quickly oscillating signal. A photodiode is used at the receiving end of the signal to convert the infrared radiation back into usable electric current. Infrared radiation is also incapable of passing through walls. This could be a disadvantage for some systems, however that is not an issue for our purposes. The pocket amplifier is meant to be used in the same room as the user, if not within arm's reach, or even in the user's pocket. The incapability of infrared radiation to pass through walls can even be a benefit for our portable amplifier because any infrared radiation in surrounding rooms would not be able to interfere with our signal. This is especially useful if the pocket amp is being used in an area with a high population density where there could be many infrared radiation signals being communicated to various platforms. Using an infrared radiation communication system, the data exchanged could reach a maximum speed of about 100 Mbps. This data speed is more than sufficient for our purposes.

Infrared radiation communication systems generally operate in two different modes: line-of-sight mode and diffuse mode. In line-of-sight mode, the infrared beam must be pointed directly at the receiver and the path to the receiver must be completely unobstructed. This would be a very difficult mode to implement in our communication because the user would have to ensure that the amplifier is always in a direct line of sight of the control system. If the amplifier was moved even a little bit, this would inevitably obstruct the path of the beam of infrared radiation and the control system would not be able to communicate with the amplifier. The diffuse mode, which is also sometimes called scatter mode, for infrared radiation communication systems would be significantly more feasible. The diffuse mode allows the source and the receiver to not necessitate being in a perfectly direct line of sight of one another. Instead of a narrow beam of infrared radiation being sent as in line-of-sight mode, a broader wave of scattered radiation is sent out and is able to be reflected off of other surfaces before reaching the receiver and it will still be accepted. This is a considerable advantage in that the source and the receiver only need to be pointed in the same general direction of each other and not directly at each other. Even though the diffuse mode is much better suited for the needs of our pocket amp, the user would not be able to put the amplifier in his or her pocket and walk around with it. If the user put the amplifier in his or her pocket this would disrupt the signal and it would be unable to communicate with the control system. We chose not to use an infrared radiation communication platform, in large part because the system would be based on a directional component that is not suited for our pocket amp.

One of the other factors taken into consideration for not choosing infrared radiation as a communication platform is the fact that the effect of the infrared radiation on the human body are still not entirely known. Infrared radiation at elevated levels could cause flash burns on the eyes or infrared cataracts. While the pocket amplifier is in use there would be a constant infrared radiation stream towards the control system that the user would be in close proximity of. To ensure the safety of the user, an infrared radiation communication system will not be used.

3.3.4.3 Radio Frequency

A radio frequency communication system is another one of the control systems that we considered to use as the communication platform for our pocket amp. Radio frequency devise generally operate within a range of approximately 3 kHz all the way up to 300 GHz. Radio frequency transmission does have a great advantage in that it can transmit data at speeds near the speed of light. However, there are significantly more constraints that need to be considered and tailored to in the design of our pocket amp when a radio frequency communication system is used. Radio frequency signals can travel far greater distances and can be communicated through walls and other barriers that may come in between the source and the receiver. In general, this would be a benefit, however for the purpose of our pocket amp this is actually a disadvantage. The amplifier and control system will always be in close proximity of each other so a long-range communication system is unnecessary. The ability to transmit the signal through walls is considerably more of a disadvantage than an advantage for us because then the signal may encounter significantly more interference than if a communication system that cannot travel large distances and through barriers is used. This added capability does not even provide the pocket amplifier with any benefits so its addition is considered a hindrance.

If a radio frequency communication platform were to be used, steps would need to be taken to ensure that all regulations are met regarding emission levels and usage for our desired frequencies. The Federal Communications Commission has many standards in addition to any other regulations from other agencies would all have to be met. The FCC does allow the use of ISM bands (industrial, scientific, and medical radio bands) from 902 to 928 MHz, 2.4000 to 2.4835 GHz, and 5.725 to 5.875 GHz for unlicensed communication equipment.

3.3.4.4 ZigBee

ZigBee is a wireless mesh network that is another possibility for us to use as a platform for our pocket amp's communication system. The ZigBee Alliance, an organization comprised of hundreds of businesses and corporations, developed the ZigBee protocol. The protocol is based on a specification from the Institute of Electrical and Electronics Engineers (IEEE) standard 802.15 regarding the operation of and standards related to wireless personal area networks. IEEE 802.15.4 is the standard that defines the operation of low-data-rate wireless personal area networks. Standard 802.15.4 details the physical layer and media access control layer for low-data-rate wireless personal area networks. The physical layer describes the frequency, power, modulation, and other wireless conditions. The media access control layer describes the format in which the data is handled. This standard is the foundation for ZigBee's development. ZigBee operates in the unlicensed ISM bands and is generally less expensive than other platforms like Bluetooth or Wi-Fi. Of the ISM bands used for this platform, the 2.4 GHz is the most commonly used frequency band worldwide for this application, but it also uses the 915 MHz band in the United States. ZigBee is also a simpler system than Bluetooth and Wi-Fi.

ZigBee is ideal for implementations of systems with low data rates that require a long battery life. Communications systems implementing this platform use very small amounts of power so batteries can have life spans of several years. ZigBee has many pre-developed applications that include a wide variety of specialized operations. This would considerably ease the difficulty of designing and building a communication system since there are many automations, systems, and applications that we can model our pocket amp's communication system on. ZigBee has a standard data rate of 250 kilobits per second when it is using the higher frequency 2.4 GHz band. If the lower frequency 915 MHz band is used for operation then the date rate will be approximately 20 kilobits per second. The downside of ZigBee for our purposes is what makes this platform great for other systems. The low data rates are not compatible with our requirements and the wireless mesh network is not as suitable as other options.

4. Design Details

Designing the Pocket Amp consists of compiling all our known information from our research and creating plans for a product which fulfills all our requirements. Due to the skill division in our team between electrical engineers and a computer engineer, the design was also split into two sections. The software design section covers all the code from microcontroller to phone application. The hardware design covers all the electrical components including power systems, amplifier systems, and all the effects.

4.1 Software Design

The Pocket Amp App will take all the users input and settings for the Pocket Amp and translate them to Bluetooth packets which get sent to the MSP432, which in turn turns the packets into appropriate settings for the hardware components and applies any digital effects to the signal from the guitar. To ease conceptualization of the software, the entire project's software can be broken into three categories. The "app user interface" category is all the user interface design done in Android Studios which the user directly interacts with. The "app backend" is the behind-the-scenes code which takes the user's input and creates the Bluetooth packets that the Pocket Amp can understand. The final category is on the other end of the Bluetooth, where the MSP432 interprets the packets and sends out the appropriate signals.

4.1.1 App User Interface

The app user interface will be a simple GUI built in Android Studio for Android smartphones that allows users to select what effects will be applied and control any options for the effects. The GUI will be designed using components that restrict the user to only be able to input valid values and will be designed to be intuitive to guitar and bass players. These two goals will be mutually necessary – the two will help each other towards their respective goals. Using graphical components such as volume dials or light-up pushbuttons helps users that are already familiar with those types of controls from physical amplifiers use the Pocket Amp.

4.1.2 App Backend

Initially, for the transferal of settings from app to Pocket Amp a well-formed object transferred by a packet was decided upon. Due to the simplicity of JavaScript Object Notation (JSON), a modified version of JSON would be easiest to implement. At its simplest, JSON is a data interchange format where an object and its members, elements, arrays, values, etc. are parsed into text then at a later point parsed from text to object again. This object to text to object serves our purpose, but using an even more lightweight implementation constrained to just the syntax we need will make our Bluetooth communication even simpler, smaller, and therefore faster.

JSON implements two structure types: objects and arrays. Objects and arrays are fully polymorphic with all JSON data types, which include objects and arrays. For example, an array can contain an object with an object inside it. Arrays are simply lists of values separated by commas and enclosed in square brackets. While we could encode all of our data into an array it would be very sparse and contain mostly 0's or whichever null value we choose. JSON objects are enclosed in curly brackets and are string and value pairs separated by commas. Each value can be a string, number, an object, an array, a boolean, or a null value. The string before the value in a JSON object denotes the name of the value. For example, an object for a text style:

{"font":"arial", "size":12, "bold":true, "alignment":"center" }

For our application, JSON is more robust than it needs to be. JSON as it stands can represent polymorphic arrays and objects, but since we control the types of options being sent we can control the data types that are being sent. Arrays do not help us, so in our implementation they are not needed. Polymorphic objects do not need to exist in our implementation, since we don't need to send any objects to the Pocket Amp within another object. This leaves us with a standard object, with strings, numbers, boolean, and null values. For us, null values do not make sense; if we don't want an option on we will set it to false or leave it out of the object. Strings are unnecessary since we will not be displaying text and any options can be encoded into numbers.

Our final design for our modified JSON leaves us with curly bracket notation surrounding an arbitrary number of name-value pairs separated by commas. The possible values are arbitrary-length numbers and boolean values. The length of each object will vary depending on what options are selected, and will be generated by the app when it sends a packet. Spaces will be stripped out of the modified JSON string in order to save transmission space. An example object with volume, overdrive, and equalizer information follows:

{"volume":12,"overdrive":true,"bass":12,"midrange":45,"treble":0

The Android API contains an android.bluetooth package which contains classes that manage Bluetooth functionality in apps developed through Android Studio. The Bluetooth package requires Bluetooth permission called "BLUETOOTH" on the device being run, which is usually asked and granted during post-install setup. Some classes within the package require an administrative Bluetooth permission tagged "BLUETOOTH_ADMIN" to run. The Bluetooth package contains all the classes required for Bluetooth Low Energy to find devices, pair and find paired devices, connect to and manage sockets, transfer data, and act as generic attribute (GATT) servers or clients.

The JSON strings over packets would work well for a constantly connected implementation of Bluetooth or an implementation of Bluetooth that establishes a connection, sends a large amount of data, then breaks the connection. Android Studio has native application support for GATT client/server connections, which it implements through multiple profiles for BLE. With a GATT client/server connection, the Bluetooth Low Energy stack implements a series of attributes, which the client can read and write to. The lower layers of the stack take care of the exchange of attributes. To the app, the exchange is simply opening a connection and telling the Bluetooth stack what attributes are to be changed.

Both of these methods have pros and cons to their use in the Pocket Amp. Implementing modified JSON and forcing a connection and sending a packet has less reliance on the Bluetooth stack and is fairly simple to implement in the software of the app and the Pocket Amp. Using GATT client/server profiles would leverage any efficiency in the Bluetooth stack implementations and let the Bluetooth stack deal with malformed attribute requests, but would be entirely dependent on finding or writing firmware which will perform all the tasks required for the client/server connection for us. If we cannot find working firmware, implementing GATT transactions would quickly become time or cost prohibitive.

Due to the potential for either method working or not working, we will first attempt to use GATT profiles built in to Bluetooth Low Energy. If the support is not readily available we will switch to developing for packet sending, still targeting Bluetooth Low Energy but with a secondary backup plan of using Bluetooth Classic due to its widespread support.

4.1.3 MSP432 Code

The MSP432 code will be an interrupt-driven extremely basic operating system used to support a core of DSP functions. The DSP functions will take up a majority of the code size and clock cycles in runtime. The only functions besides DSP functions will be to manage communication, power management, and hardware management.

4.1.3.1 Main Loop

All of the non-DSP functions will be entirely interrupt driven in order to reduce the need for polling, spin-waits, or any other wasteful overhead that takes clock cycles away from DSP functions. Any incoming communication from the Bluetooth module will raise an interrupt and the Pocket Amp will cease playing while the interrupt is handled. This is to avoid creating potentially damaging or unpleasant output signals by having samples piling up or overwriting current samples. The MSP432 will receive the Bluetooth input, output the appropriate signals, change the appropriate internal states, clear the sample cache, then re-initialize the sampling and DSP functions.

The primary functionality of the MSP432 will be contained in one loop. This loop will wake when it is time to sample the ADC, send the output sample to the DAC, take in a new sample, apply any DSP functions, then go to Low Power Mode 0 (LPM0) until a new sample is taken. LPM0 reduces the power consumption by turning off the processing cores of the MSP432 but takes less than 4 clock cycles to return to active mode. Since having dropped samples is something to be avoided, we will attempt to drive the MSP432 at less than 90% maximum load, leaving at least 100 clock cycles on average to be in LPM0. Thus, if the load increases by up to 11% of nominal max load the MSP432 can continue to function and the switch to LPM0 will have at least some power savings and not interfere with the function of the Pocket Amp.

The samples from the MSP432's built-in ADC will be 14-bits wide, which allows the samples to fit into a queue of 16-bit halfwords. Since ARM processors can align data to byte, using halfwords will allow the MSP432 to fit two samples into every 32-bit word in order to save on memory. Most queues in computer science are implemented through linked lists, which have higher processor overhead than most other data structures but allow more abstract data types and lists of arbitrary and fluctuating length. This implementation would work for the Pocket Amp but has unnecessary features and thus overhead. Since we have a static number of samples to keep track of, our samples can be stored in an array. This array will be accessed by a pointer to the latest sample and when a new sample arrives the pointer will move ahead one space and overwrite the oldest sample. Accessing any sample will be done in constant time; just add the offset of the data to the pointer and wrap around if it goes beyond the end of the array. This is far faster than the linear time complexity of accessing a linked list element.

DAC output will be done through I^2C to a dedicated 12-bit DAC. Since the MCU must send the sample through I^2C this creates some overhead for every sample sent. For every byte of data sent (which will be 2 in our case) there is one overhead ACK bit and for every sample sent there will be a 7-bit address, R/W bit, and an ACK bit. This gives us an efficiency of 44%. For every 12-bit sample we send we need 27 bits sent over the bus. For this communication to occur regularly and avoid having fluctuating output sample rates, the DAC write will occur immediately after the wakeup of the MCU.

4.1.3.2 DSP Functions

The design of the DSP functions will take every possible precaution to reduce the clock cycles with minimal compromise to the audio or effect quality. This will be done by reducing the clock cycle cost of every DSP function and reducing overhead. To reduce overhead the DSP functions will each be discrete functions called from the main loop if and only if the DSP function is to be needed. Using a logical check before calling a DSP function will keep the MSP432 from caching or loading functions that ultimately don't affect the audio. Every DSP function will have access to its own static variables which are set by the Bluetooth interrupt function and have direct access to the array of samples. This reduces the number of parameters passed by the function thereby reducing the number of things pushed and popped from the stack and reducing the function call overhead.

4.1.3.2.1 Delay

The delay function is not actually going to be a function or subroutine. Instead, the delay effect will just be an offset variable telling the other functions an offset to perform. This will also adjust the depth the other functions can sample to. To give the other effects a reasonable sample size, the delay function will be limited to 250 milliseconds (11,025 samples). The amount of delay will be decided by the user and input through the app.

4.1.3.2.2 Reverb

The design that was chosen for the reverb effect is modelled after James Moorer's reverberator. This circuit which is shown in Figure 12 in Section 3.1.5.9 Reverb is by far the most realistic to implement in our amplifier, because any analog circuit that was considered would consume an outrageous amount of space. Moorer's design of reverberator was chosen over other digital designs because this type of reverb is also the most natural sounding. This reverb has the same quality of early and late reflections that is heard in popular tube amplifiers. The combination of FIR filters and comb filters are simple to implement in digital signal processing.

The reverb function will have a few presets for different room sizes, corresponding to different presets of delays and attenuations. Each of the presets will be stored in an array which represents the amplitude of the sample and the length of the next delay. The reverb function will always use 20 samples, but if a preset does not use all 20 samples then it will use an amplitude of 0 and a delay of 0 for the next sample. This allows the reverb function to always use the same number of clock cycles and have the same formula within the function and not require logical checks or long recursive calls. Figure 17 shows a pseudocode implementation of how this type of function will work.

```
Define ratio power of signal, amplitude = [1, 0.6, 0.2, 0.3...];
Define next delay in samples, nextDelay = [12, 42, 123, 1, ...];

start Reverb
    Set outputSample to sample[0];
    Add 60% of sample[12] to outputSample;
    Add 20% of sample[54] to outputSample;
    Add 30% of sample[177] to outputSample;
    ...
    return outputSample;
end Reverb

```

Figure 17. Pseudocode for Reverb
Created by William McKenna in Visual Studio Code

4.1.3.2.3 Flanging

The digital flanging function will have to take two samples, sum them together, and check for 14-bit overflow. The two samples will be taken a varying time delay apart, from a predetermined range of time delays that will be cycled through linearly. The slightly delayed sample will be attenuated by a small amount in order to not completely attenuate or double the original signal when the waves align. The amount of attenuation will be 5%, in order to keep it minor but still attenuate the secondary signal a large amount. The secondary sample will be delayed by between 2 milliseconds and 20 milliseconds, with the exact delay length adjustable by the user. A pseudocode example of the flanging effect sweeping from 2.5 milliseconds to 20 milliseconds in 2-sample increments is seen in Figure 18

The user will be able to change the shortest delay, the longest delay, and the time to sweep in between the two. The user will be able to select between 2 milliseconds and 9.5 milliseconds for the lower end of the delay and 10.5 milliseconds to 20 milliseconds for the higher end of the delay. This will limit users from conflicting the two time delays. The user will also be able to select the rate of sweep between the two times, which will be converted to a rate of samples per second. When the flanger is moving towards longer delays the rate will be added to a floating-point value holding the current delay and when it is sweeping downward the value will be subtracted. The float will be converted to an integer value to pick the delayed sample, which will ensure that the delay sweeping is linear and not choppy due to the integer truncation.

```
Define minimum delay in samples, delayMin = 110;
Define maximum delay in samples, delayMax = 882;
Define samples the flange moves per sample,
                              sampleSweep = 2;

start Flanging
    Set outputSample to sample[0];
    Add 95% of sample[currentDelay] to outputSample;

    if sweeping upwards
        Add sampleSweep to currentDelay;
    if sweeping downwards
        Subtract sampleSweep from currentDelay;
    Check for sweeping up or down next call;

    return outputSample;
end Flanging
```

Figure 18. Pseudocode for Flanging
Created by William McKenna in Visual Studio Code

4.1.3.2.4 Echo

The echo function is a sampling of the current input, added with a varying degree of samples from before the current sample. The previous samples decay off at an exponential rate at a set delay time. The power level and delay time will be set by the user and stored in a variable accessible by the echo function. The number of delays possible will be bounded by our limited number of samples remaining after the delay offset and a limit of 30 milliseconds to keep it from interfering with the waves and to limit the processing overhead. For example, a 10-millisecond delay with and echo of 100-millisecond with a power of .2 would have the 10-millisecond delay sample, plus a sample delayed by 110 milliseconds at power level .2, plus a sample delayed by 210 milliseconds at power level 0.04, until it passes 500-milliseconds. Figure 19 shows the pseudocode for and an echo effect with 40% volume for each echo and a delay of 50 samples.

```
Define volume of echo,        power = 0.4;
Define echo delay in samples, delay = 50;

start Echo
    Set outputSample to sample[0];
    Add 40% of sample[50] to outputSample;
    Add 16% of sample[100] to outputSample;
    Add 6.4% of sample[150] to outputSample;
    ...
    return outputSample;
end Echo
```

Figure 19. Pseudocode for Echo
Created by William McKenna in Visual Studio Code

4.1.3.2.5 Chorus

The chorus effect is almost identical to flanging in implementation but has a longer time delay between the original and the previous sample. In addition, the sweeping amount is generally smaller and the sweep rate is usually slower. The minimum time delay for chorus is 20 milliseconds (882 samples) and the maximum is 40 milliseconds (1764 samples). Figure 20 shows a pseudocode implementation of chorus with the maximum delay range.

```
Define minimum delay in samples, delayMin = 882;
Define maximum delay in samples, delayMax = 1764;
Define samples the flange moves per sample,
                              sampleSweep = 2;

start Chorus
    Set outputSample to sample[0];
    Add 95% of sample[currentDelay] to outputSample;

    if sweeping upwards
        Add sampleSweep to currentDelay;
    if sweeping downwards
        Subtract sampleSweep from currentDelay;
    Check for sweeping up or down next call;

    return outputSample;
end Chorus
```

Figure 20. Pseudocode for Chorus
Created by William McKenna in Visual Studio Code

4.2 Hardware Design Details

4.2.1 Amplifiers

In this section the nature of the amplifiers that will be selected for this project is discussed and chosen. It should be noted that the Vacuum Tube amplifiers have been eliminated from discussion due to their size and cost. Transistor amplifiers have also been eliminated as, while they are certainly less expensive than Operational Amplifiers, their non-ideal properties and more complicated support networks make them a less appealing option when the relatively low output current of an Operational Amplifier will suffice to drive the relatively high loads associated with headphones makes transistor amplifiers unnecessary.

4.2.1.1 LME49723

The LME49723 Operational Amplifier from Texas Instruments is the lowest cost Op-Amp under consideration at $0.50 for a dual package for $0.25 per amplifier. This amplifier is marketed as an Ultra-Low distortion, low noise and high slew rate amplifier with a high Common Mode Rejection Ratio (CMRR) of 100dB making it ideal for audio applications. The manufacturer also claims that its 25 mA maximum output current is enough to drive a 600Ω load. Because this amplifier has the smallest output current of any other amps considered, this suggests that this metric may not be as important as others and therefore will not be considered as highly as other significant measurements.

The price of this amplifier makes it attractive, except it does have the highest Total Harmonic Distortion (THD) and one of the largest noise voltages of any Op-Amp considered for this project. This project will be utilizing multiple cascaded amplifiers so this may become an issue. The slew rate of this amplifier is also quite small at 8 VµS compared to other amplifiers being considered, however since most voltages in this project will be under 1 Vp except in the Overdrive effect stage where voltage between the first and second amplifiers will be significantly higher. The slew rate is a useful measurement when designing a circuit, however once again we consider it to be less important than other measurements in audio applications.

As can be seen in Figure 21, the THD and noise voltage of the LME49723 is at its lowest approximately between 100mV and 1.5V at less than 0.001% THD+N combined. Every indication gleaned about the output of a guitar suggests that the output will mostly stay within this voltage range, however it may occasionally drop below 100 mV when it is being played softly. If the voltage drops below this level the THD+N will rise. However, even at 20 mV, which is the lowest measured output voltage of the guitar in this project, the combined distortion is only 0.005% for this Operational Amplifier.

4.2.1.2 OPA1662

The OPA1662 Bipolar Input Operational Amplifier from Texas Instruments is marketed as low power, low noise, low distortion, and able to operate over a wide supply voltage range which makes this dual channel amplifier appealing for audio applications, especially as it bears the Burr-Brown name that has become synonymous with audio excellence. The manufacturer boasts a very low 1.5 mA required supply current for each channel, or a total of 3 mA for the dual channel OPA1662 package which makes it very attractive for portable devices where it will make the most of the available power supplied from a limited source. It accomplishes this all while maintaining a cost of $0.75 suggested by TI for the dual channel package or $0.375 per amplifier making it seem to be a very cost effective option for high performance audio.

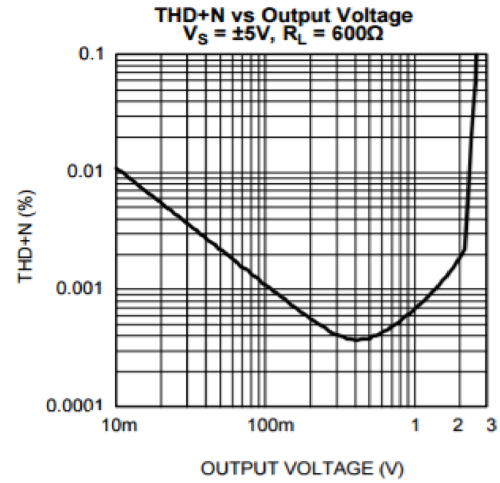


Figure 21. LME49723 Harmonic Distortion vs. Output Voltage
Courtesy of Texas Instruments

While the THD+N vs Output Amplitude curve of the OPA1662 is fairly similar to the lower priced LME49723 the OPA1662 however the 0.00006% THD+N advertised by TI was measured at 3Vrms output voltage and a frequency of 1 kHz. However, the more same metric may be gleaned from Figure 22 at a more applicable 1Vrms output and a load of 600 Ω to 2 kΩ to be approximately 0.0001% for frequencies under 500 Hz. It should be noted however that at 20 kHz, the maximum frequency this project is concerned with, the total distortion rises sharply to 0.002%. While it does rise significantly it is still comparably better than the LME49723 in this respect. This amplifier also enjoys excellent channel separation that the LME49723 does not in order to keep the two amplifiers in the package from interfering with each other and producing even more noise. This can be seen in Figure 23 below which shows that for the frequency range required by this project of under 20 kHz the crosstalk is at a gain of lower than -120 dB. While this figure assumes an output voltage of 3 Vrms and a closed loop gain of 1 it is still an indication of strong performance.

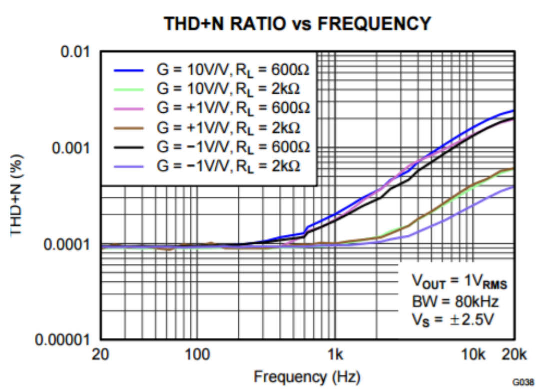


Figure 22. OPA1662 THD vs. Frequency
Permission Pending from Texas Instruments

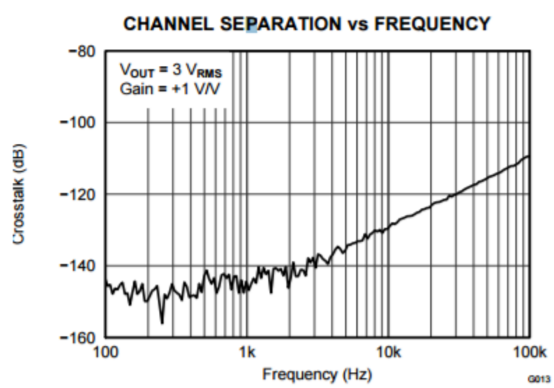


Figure 23. OPA1662 Channel Separation vs. Frequency
Permission Pending from Texas Instruments

4.2.1.3 LM4562

The LM4562 Dual Channel, Ultra-Low Distortion, Low Noise, high Slew-Rate amplifier from Texas instruments was chosen for consideration due to its high common mode rejection ratio of 120 dB and low, consistent total distortion characteristics that would make it suitable for a project where load impedances may vary such as being suitable for a wide range of headphones from consumer to professional grade. This amplifier is also claimed to be able to drive a 600 Ω load to accommodate some of the highest impedance headphones on the market. This Op-Amp also enjoys a very low claimed THD+N of 0.00003% at an output of 3 Vrms and 1 kHz frequency with a closed loop gain of 1.

While this amplifier seems exceptional on paper the total distortion at lower rail voltages deteriorates quickly as seen in Figure 24. Based upon this figure the THD+N of this Op-Amp will not be as low as previously considered options and, when coupled with a substantially higher price point of $1.16 for the dual channel package or $0.58 per amplifier, this effectively rules the LM4562 out of consideration for a project that will likely require the biasing of its Operational Amplifiers with less than the voltage that it will require for this amplifier to perform as advertised.

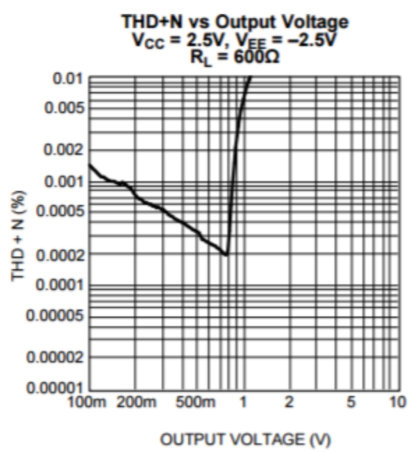


Figure 24. LM4562 Harmonic Distortion vs. Output Voltage
Permission Pending from Texas Instruments

4.2.1.4 LME49720

The LME49720 Dual Channel High Fidelity Operational Amplifier enjoys an advertised 0.0003% THD+N at a closed loop gain of 1, output voltage of 3 Vrms and frequency of 1 kHz as well as a low noise voltage of 2.7 nVHz and a very low offset voltage of 0.7 mV maximum.

All in all, it shares many of the features of the LM4562 including its high slew rate of 20 VµS. However, it does have a significantly lower output current of 26 mA although as previously discussed it is suspected that this will not have a significant effect on this project. Unfortunately, this Op-Amp does share the

LM4562's subpar performance at lower bias voltages. While the price is somewhat lower at $1.10 for the dual channel package or $0.60 per amplifier it is still double the price of some of the amplifiers that perform better than it as far as total distortion at low bias voltages.

4.2.1.5 LME49724

The LME49724 High Performance, High Fidelity, Fully-Differential Audio Operational Amplifier from Texas Instruments is marketed for ultra-high quality audio applications and boasts a Common Mode Rejection Ratio of in excess of 100 dB, a THD+N of 0.00003%, an exceptionally low noise voltage of 2.1nVHz, a very high slew rate of 18 VµS and a very high 80 mA maximum output current that can drive a 600 Ω to 52 Vpp with sufficient rails. It also enjoys the shutdown feature explained in the Research section of this paper. While its maximum offset voltage is slightly higher than some previously discussed amplifiers it is still at a respectably low 1 mV.

As shown in Figure 25, the LME49724 as a very flat THD+N vs Frequency response with the rails at ±2.5 V, the output voltage at 500 mVrms over a range of load impedances. For all frequencies relevant to this project the total distortion due to the amplifier will be approximately 0.001%. While this performance is not as good as if the rails were set higher, ±15 V or ±18 V for example, it is still very good. When comparing THD+N to output voltage the performance of the LME49724 does not deteriorate as much as some of the other considered Op-Amps as shown in Figure 26 however, it is only improved by 0.0001% which will likely be imperceptible to the average or even more experienced listener.

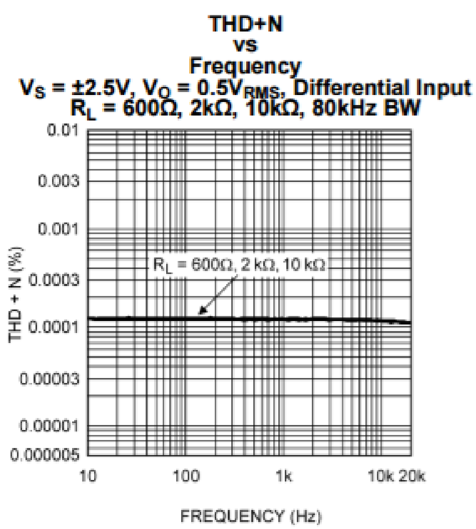


Figure 25. LME49724 THD vs. Frequency
Permission Pending from Texas Instruments

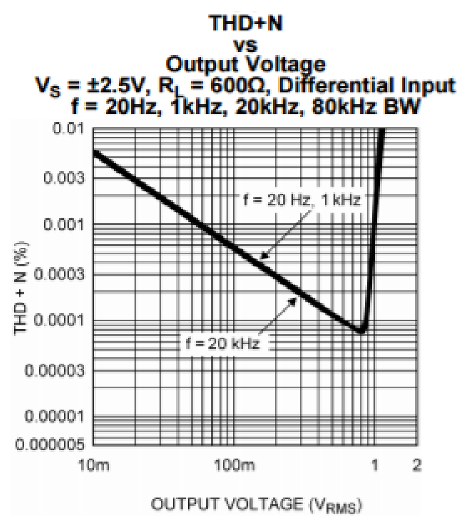


Figure 26. LME 49724 THD vs. Output Voltage
Permission Pending from Texas Instruments

The choice of the LME49724 will come down to price and whether it's marginally better performance at low bias voltages and frequencies will justify the cost at $1.42 for a single amplifier package, easily the highest cost per amplifier considered so far.

4.2.1.6 OPA1632

The OPA1632 High-Performance, Fully-Differential Audio Operational Amplifier from Texas Instruments is advertised as having among the lowest harmonic distortion with a THD+N of only 0.000022% and a noise voltage of 1.3 nVHz as well as an exceptionally high slew rate of 50 VµS and an exceptionally high Gain Bandwidth product of 180 MHz. This low distortion is partly due to the Fully-Differential nature of the OPA1632 which, according to the manufacturer, reduces even-order harmonics and increases the Common Mode Rejection Ratio.

However, the THD+N characteristic figures given in the datasheet for the OPA1632 assume rails of ±15 V and an output voltage of 3 Vrms which may indicate that at lower offset and output voltages this amplifier will not perform as well as indicated, an assumption supported by the findings regarding other Op-Amps explored earlier in this section whose performance at such required voltages was sufficiently less than advertised at higher voltages.

Once again, the price of the OPA1632 is quite high at $2.23 for a single channel package. Unless this project required the high gain bandwidth product of this amplifier it is unlikely to be chosen. This is a very unlikely event as there is a well-defined and relatively small bandwidth that the amplifier must reach its full performance within and, even then, with only a relatively low voltage.

4.2.1.7 OPA827

The OPA827 Low Noise, High Precision, JFET-Input Operational Amplifier from Texas Instruments is easily the highest cost Op-Amp in consideration at $2.23 for a single channel package. The manufacturer justifies the high price with the fact that this amplifier has an exceptionally low 0.015 mV maximum offset voltage and high resistance to changes in temperature as well as a very low biasing current of merely 3 pA which does make this amplifier an attractive choice for a portable application by reducing the power requirements of the system.

Where this Op-Amp falls short of previously considered options is in noise voltage, in this case 4 nVHz and, as previously mentioned, the high price. For these reasons, unless the system requires low offset voltage and low current draw, it is unlikely that this amplifier will be chosen. However, if it is chosen it still has a respectable gain bandwidth, slew rate, and THD+N as advertised, however like many of the previous Op-Amps considered these are not necessarily representative of the performance of this amplifier at lower bias and output voltages as no figures are provided to indicate such an assumption.

4.2.1.8 LME49726

The LME49726 is a high current, low distortion amplifier produced by Texas Instruments. This Op-Amp has an extremely high maximum output current of 230 mA while keeping respectably low distortion levels of 0.00008% THD+N. This amplifier also has a much narrower supply voltage range of only 2.5 V to 5.5 V which suggests that performance at the levels that this project requires may be superior to some of the options explored already. However, by examining Figure 27 it can be seen that regardless of offset voltage there is a large peak in THD+N when a load of 600 Ω is applied.

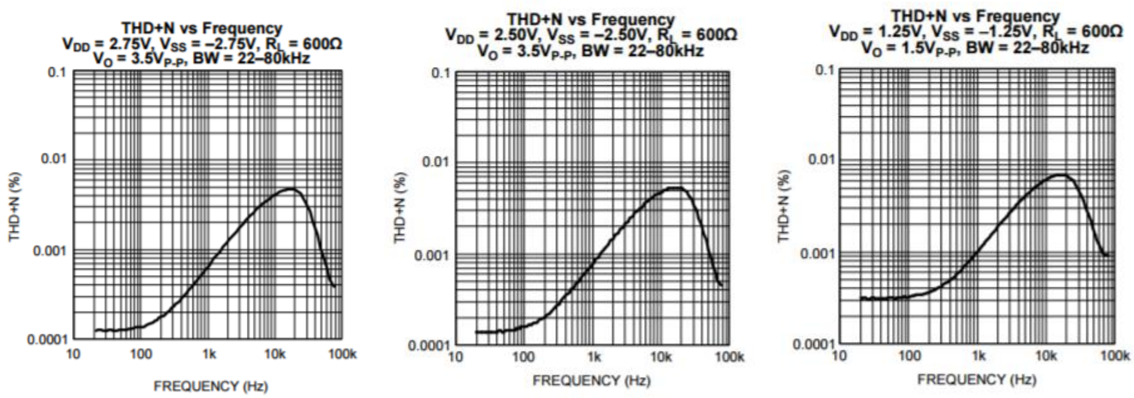


Figure 27. Effect of Rail Voltages upon THD vs. Frequency
Permission Pending from Texas Instruments

It should also be noted that this amplifier has the worst noise voltage, gain bandwidth, and slew rate of any in consideration at 6.9nVHz, 6.25 MHz, and 3.7 VµS respectively. Because of these this Op-Amp is unlikely to be used as anything other than perhaps the output stage, and only then if the output current required to drive headphones is sufficiently large which upon further examination of modern headphones it is unlikely to be. Price for this option would be $0.55 for a dual channel package for a per-amplifier cost of $0.275.

4.2.1.9 OPA1611 / 1612

The OPA1611 High-Performance, Bipolar-Input Audio Operational Amplifier from Texas Instruments has a very low noise voltage at 1.1 nV/(Hz)$^{.5}$, the lowest in consideration, and total harmonic distortion at 0.000015% as advertised, again the lowest out of all Op-Amps considered. This amplifier also draws merely 3.6 mA while outputting a maximum of 30 mA. The OPA1612 is merely a dual channel package version of the OPA1611 with low crosstalk.

This Op-Amp has a very attractive THD+N vs. Output voltage curve which, as shown in Figure 28 is a full order of magnitude lower than any of the amplifiers considered previously. The magnitude of the THD+N vs frequency curve in Figure 29 is also approximately an order of magnitude lower between 0 Hz and 2 kHz than that of other considered Op-Amps. While the cost is more than others at $1.75 for the single channel OPA1611 and $2.75 for the dual channel

OPA1612, for $1.375 per amplifier, if total distortion and noise is an important metric in this project that cost may be justified by the high performance of these amplifiers.

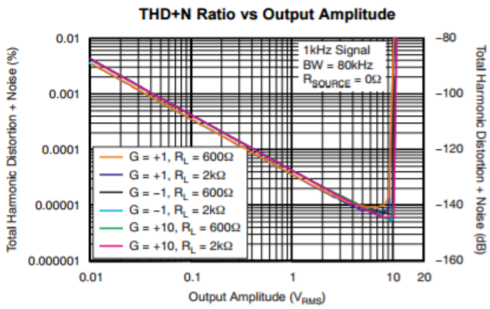


Figure 28. OPA 1611 THD vs. Output Voltage
Permission Pending from Texas Instruments

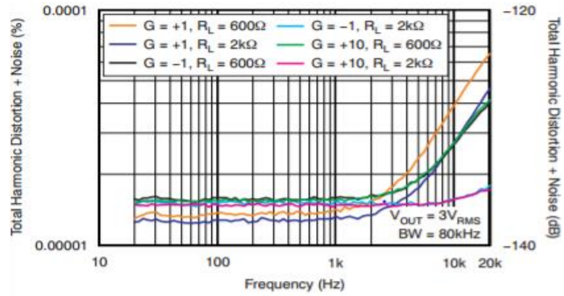


Figure 29. OPA 1611 THD vs. Frequency
Permission Pending from Texas Instruments

The OPA1611 and OPA1612 do not have the highest gain bandwidth product, slew rate or output current but all are respectable with corresponding values at 40 MHz, 27VµS and 55 mA respectively. While these amplifiers may not be the best at any of these three metrics they will almost certainly be sufficient for this project while keeping the total distortion and noise volume very low. As long as the price is justifiable these amplifiers are good all-around performers that are also have very good audio characteristics.

4.2.1.10 Amplifier Selection

Since it seems that many of the higher performing Operational Amplifiers may simply not be needed in this project due to its low output voltage and current requirements, portable nature, and low biasing voltage of somewhere around ±5 V the obvious considerations for the final selection would be those with attractive prices and audio performance. The Op-Amps fitting these criteria are the LME49723, OPA1662, LME49726 and OPA1611 / OPA1612. A comparison of these Op-Amps is shown in Table 1Table 5. Among these the OPA161X family definitely has the strongest audio performance with their exceptionally low distortion levels but are the most expensive. This may be justified by the fact that in this design many Op-Amps will be cascaded to produce the final amplifier. In such a design the distortion would only be compounded by each additional Op-Amp and as such an Operational Amplifier with low distortion should be used. The LME49723 should be eliminated with easily the highest THD+N value as should the LME49726 with a much higher noise voltage, both of which would only be amplified by each additional stage.

Between the final two Op-Amps in consideration, the OPA1662 and the OPA161X family, there are major differences. The OPA1611 has a better gain bandwidth, slew rate, offset voltage, THD, output current and noise voltage however these may not factor into the overall performance of the final amplifier as much as previously thought and these high values may not be required by this application, in which case the OPA1662 with a much lower cost would be the prudent choice.

Product Number	B (Gain Bandwidth Product)	Slew Rate	Max Offset Voltage @ 25C	Vn (Noise Voltage)	Features	THD+N	Output Current	Price
	A*ω	V/μS	mV	nV/√Hz		%	mA	USD
	+	+	-	-		-	+	-
LME49723	19	8	1	3.2	N/A	0.002	25	0.5
OPA1662	22	17	1.5	3.3	Burr-Brown Audio	0.00006	40	0.75
LM4562	55	20	0.7	2.7	N/A	0.00003	42	1.16
LME49720	55	20	0.7	2.5	N/A	0.00003	26	1.1
LME49724	50	18	1	2.1	Shutdown	0.00003	80	1.42
OPA1632	180	50	3	1.3	N/A	0.000022	85	2.23
OPA827	22	28	0.015	4	Burr-Brown Audio	0.00004	30	3.75
LME49726	6.25	3.7	2.5	6.9	N/A	0.00008	350	0.55
OPA1611	40	27	0.5	1.1	Burr-Brown Audio, High CMR	0.000015	55	1.75

Table 5. Comparison of Operational Amplifiers
Created by Joshua DeBaere

4.2.2 Input Filtering

This section considers the methods proposed in section 3.2.2 Input Filtering that may be used to filter the output of the guitar when it is received by the Pocket Amp. Doing so will produce a more accurate representation of notes that the artist plays. Discussed will be various filtering methods to clean up the signal supplied to the amplifier and which method will ultimately be selected.

4.2.2.1.1 Passive Filters

Undeniably the cheapest and simplest form of filtering is the passive filter. An uncomplicated and inexpensive high pass passive filter can be constructed as seen in the following Figure 30 and its small signal response can be seen in Figure 31 with the sensitivity of the resistance R considered.

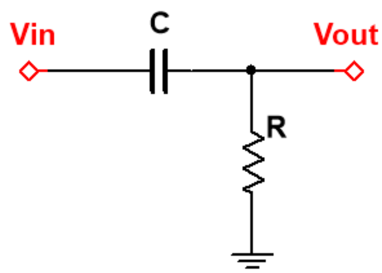


Figure 30. A purely passive High Pass Filter
Created by Joshua DeBaere using Multisim Student Edition

By choosing values of R and C to satisfy $f_c = \frac{1}{2\pi RC} = 60\ Hz$ the filter in Figure 30 will attenuate by 3 dB at 60 Hz and climb to a gain of essentially 0 dB by approximately 500 Hz although the higher the frequency, the closer to 0 dB it will be.

Advantages of this design include cost and simplicity as well as an acceptable sensitivity to changes in the values of R and C as can be seen in Figure 31

where a parameter sweep is run to compare the magnitude and phase response of the ideal and worst case scenario values for the value of R assuming a 5% resistor is used. However, it does have a relatively slow roll off and only attenuates by 3 dB at 60 Hz which may or may not be problematic depending upon how much energy is at that frequency in a particular location.

Naturally higher order passive filters can be made by cascading the network in Figure 31, an example of which can be seen in Figure 32. By using the same values for $C = C_1 = C_2$ and $R = R_1 = R_2$ the filter in Figure 32 will now ideally attenuate by 9 dB at 60 Hz and should display a quicker roll off. However, it does introduce more components which both increases size of the filter and increases the potential deviation caused by tolerances in manufactured components. These two reasons would imply that anything beyond a second order passive filter would be less practical and more sensitive than this project dictates.

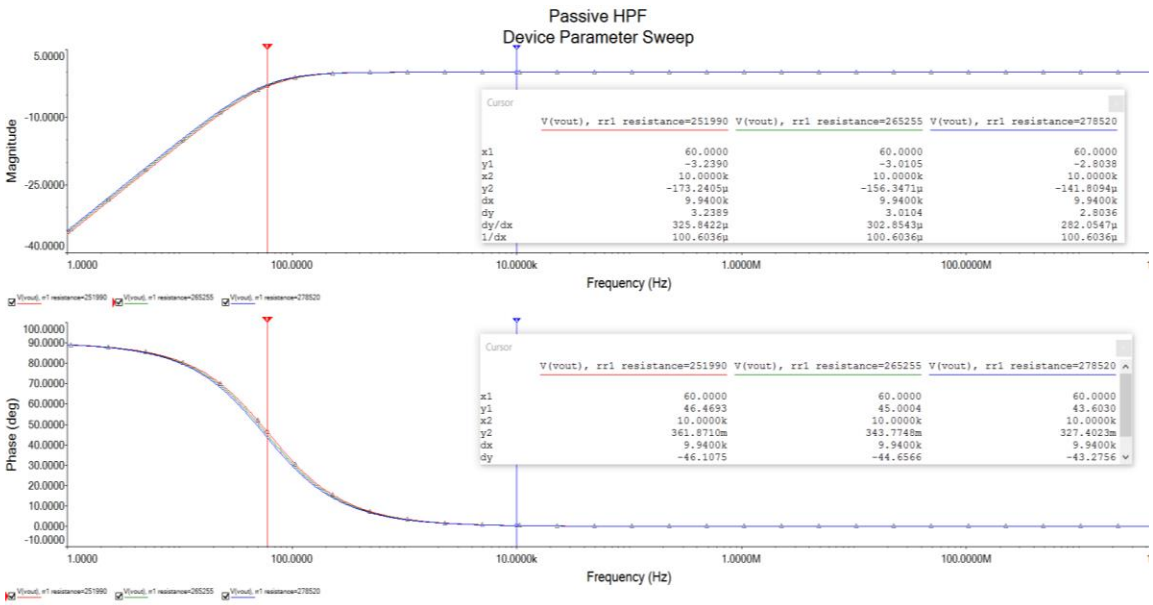


Figure 31. Sensitivity of Passive High Pass Filter.
Created by Joshua DeBaere using Multisim Student Edition

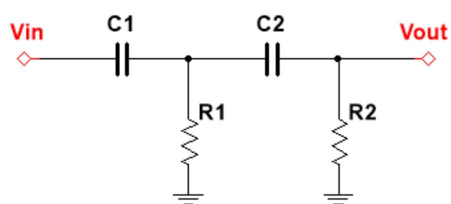


Figure 32. A second order passive High Pass Filter
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4.2.2.1.2 Active Low Pass Filters

Introducing active components into filter design gives the filter the ability to produce some gain. The addition of gain adds function to the filter, possibly allowing it to assume the role of a successive stage of the amplifier if possible to reduce size and cost. The downside to this is that it may cause the voltage level of the noise to increase and interfere more strongly than before.

4.2.2.1.3 Passive Notch Filter

Notch filters impose a drastic attenuation at a specific frequency that shall be referred to as ω_p that appears as a notch in the magnitude response of that filter. The filter in Figure 33 is a “Twin-T” notch filter cascaded after a simple first order passive high pass filter to generate a high pass notch filter. This solution is simple in design and inexpensive in construction being governed by the following equations:

$$f_c = \frac{1}{2\pi C_5 R_4} = 40\ Hz$$

$$f_{notch} = \frac{1}{2\pi RC} \text{ where } R = R_1 = R_2 \text{ and } C = C_2 = C_3$$

$$R_3 = \frac{R}{2} \text{ and } C_1 = \frac{C}{2}$$

While this filter attenuates greatly at 60 Hz, often close to 70 dB of attenuation, and attenuates below 60 Hz although to a lesser extent, it also attenuates a large amount at 80 Hz, a frequency which represents one of the fundamental harmonics of the standard tuned guitar.

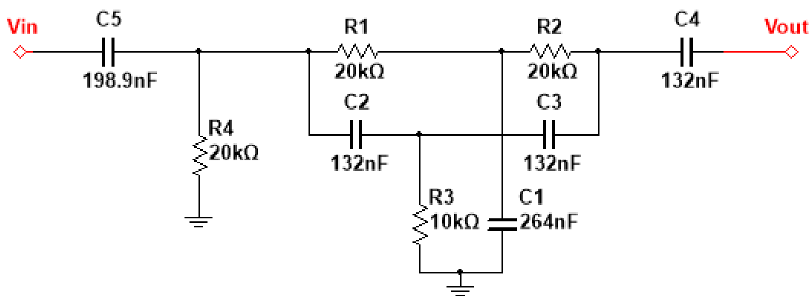


Figure 33. A High Pass 60 Hz Twin-T Notch Filter
Created by Joshua DeBaere using Multisim Student Edition

4.2.2.1.4 Active Notch Filter

Because of its sensitivity at higher quality factors that would be required to pass an 80 Hz wave and attenuate a 60 Hz wave the Sallen-Key topology will not be an effective input filter for this project. As such this section will focus on the Fliege topology of active notch filters. The filter in Figure 34 is a standard Fliege topology notch filter cascaded after a low pass filter with a cutoff frequency at 20 Hz to attenuate the low frequency input. This filter is governed by the equations below and results in the magnitude response in Figure 35. As can be seen in the response the attenuation at 60 Hz is by far enough to quiet any noise from the mains at that frequency and all frequencies below 60 Hz are also attenuated in order to eliminate any noise that is not produced by a standard tuned guitar. It is also planned to have a way to bypass the low pass filter if the artist or listener wishes the frequencies below 60 Hz to be present in the output. It should also be noted that because the Fliege filter is always a unity gain filter outside of its rejection band the pass band is very stable and without ripples that may become noticeable after amplification.

$$f_C = \frac{1}{2\pi R_7 C_3} = 20\ Hz$$

$$f_{notch} = \frac{1}{2\pi RC} = 60\ Hz \text{ where } R = R_3 = R_4 = 26.528\ k\Omega \text{ and } C = C_3 = C_2 = 100nF$$

$$Q = \frac{R_Q}{2R} = 5 \text{ where } R_Q = R_1 = R_2 = 265.52\ k\Omega$$

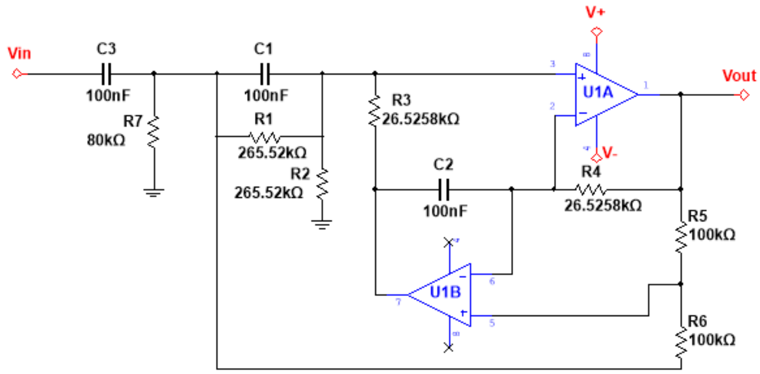
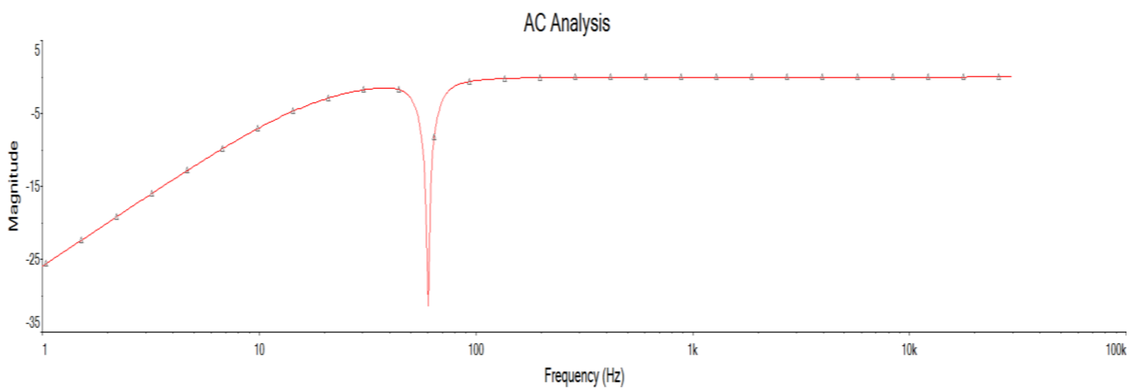


Figure 34. Fliege High Pass Notch Filter
Created by Joshua DeBaere using Multisim Student Edition



Created by Joshua DeBaere using Multisim Student Edition
Figure 35. High Pass Fliege Notch Filter Response

4.2.2.1.5 Input Filter Selection

Because of its response curve and ability to attenuate at 60 Hz while leaving 80 Hz relatively un-attenuated without producing a high sensitivity and variance in gain the Fliege notch filter cascaded with a first order high pass filter has been selected. While it is the costliest possible design, requiring two Operational Amplifiers instead of zero or one in the other possible designs, it is believed that this topology will be much better suited to the requirements of the Pocket Amp and will therefore result in the best experience for the artist or listener. It is true that a design using the Sallen-Key topology could have used digital potentiometers to adjust for component tolerances, resulting in an even larger attenuation at 60 Hz however it is unlikely that this dramatic of an attenuation will be needed since the majority of the noise at 60 Hz will be coming from electromagnetic interference produced by the mains wires running through the walls of every modern building in the United States of America and not by a direct connection to the mains network unless the device is being played while being charged.

4.2.3 Effects

This section explores the potential and selected networks to be used to create the various effects available in the Pocket Amp. Each section herein will explore several possible methods for the corresponding effect and select one of them.

4.2.3.1 Tone Control Method Selection

A commonly used tone control circuit called the Baxandall Tone Control Circuit is a compact and effective method of controlling the overall frequency response of the amplifier. This method allows for attenuation, in the case of either the passive or active variant, or amplification, in the case of only the active variant, of two frequency bands called "Bass" and "Treble." An example of the active version of the Baxandall Tone Control Circuit is shown in Figure 36.

The network as shown in Figure 36 can both attenuate and amplify high frequency signals, those above 7 kHz, and low frequency signals, those below 75 Hz. If this network is to be used, the corner frequency of the low pass filter formed by R1 and C2 when the wiper of VR1 is at its lower terminal, the one common with C2 and R2, must be changed as the common range of high frequencies begins at 2 kHz instead of 7 kHz.

The Baxandall tone Control Circuit attenuates to a great degree, a claimed 20 dB attenuation, at low and high frequencies and has the ability to amplify at these frequencies as well if the active form is used, however it only allows two channels of control. While the mid tones will rarely be amplified the artist looking for a very heavy bass or very heavy treble response will wish to attenuate the middle frequencies in the 250 Hz to 2 kHz range. From a price standpoint this design requires one digital potentiometer per range of frequencies that can be controlled for a total of two digital potentiometers and an Operational Amplifier as well as several passive components whose cost will be considered trivial in comparison.

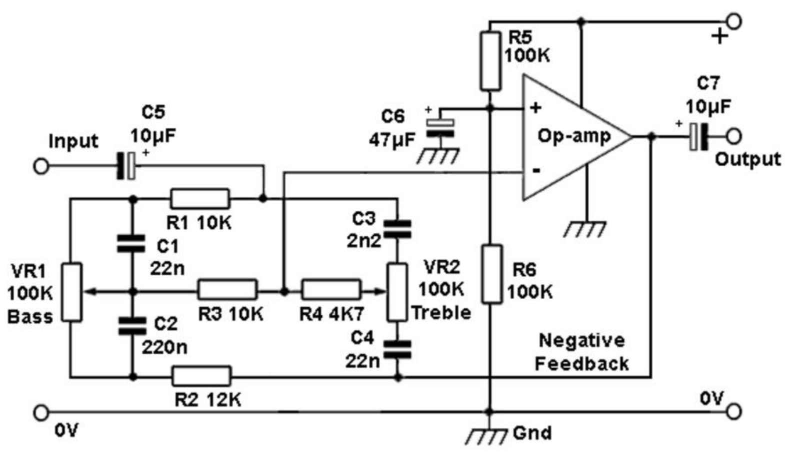


Figure 36. The Baxandall Tone Control Network
Courtesy of Eric Coates.

An alternative to the Baxandall Tone Control Circuit is a simple three channel filter with a summing amplifier as shown in Figure 37. This design does have a higher variance in terms of individual component values however it also allows the artist or listener control over a third channel, the middle frequency channel. This will allow the artist or listener to reduce the energy in the lower order harmonics at the output. Changing tone in this network is also simple as it relies on one digital potentiometer per channel just like the Baxandall Tone Control Circuit, however in this case it will require a total of three. This network still requires only one Operational Amplifier despite adding the third control channel. It does cost slightly more in terms of passive components and space on a pcb, however this is to be expected as it gives more control.

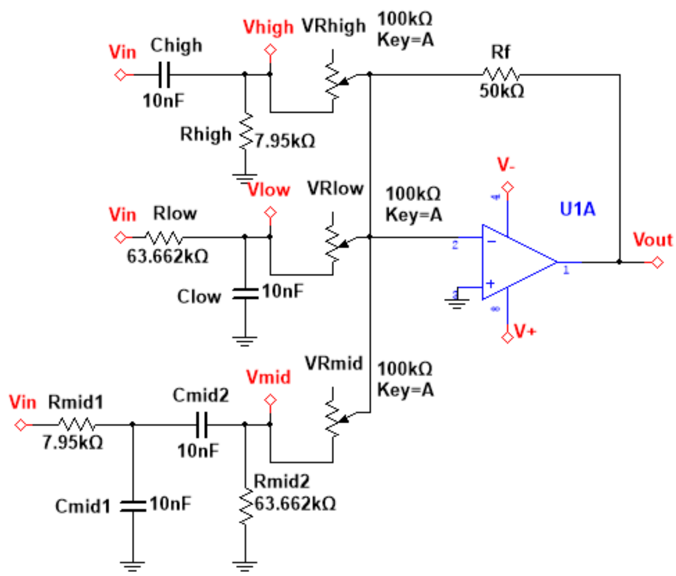


Figure 37. Three Channel Tone Control Network with Boost and Cut Capability
Created by Joshua DeBaere using Multisim Student Edition

The Three Channel Tone Control Network presented in Figure 37 has a magnitude response that can be seen in Figure 38 where section (a) shows the range of attenuation and amplification due to changing the potentiometer VRhigh in Figure 37, section (b) shows the range of attenuation and amplification due to changing the potentiometer VRmid in Figure 37, and section (c) shows the range of attenuation and amplification due to changing the potentiometer VRlow in Figure 37. It should be noted that these potentiometers will be allowed to range between 20 kOhms and 100 kOhms. Their lower bound is determined by the lowest resistance that will enable the three networks to function independently while the upper bound is set by the constraints of the selected potentiometers. It should also be noted that the resistor Rf in Figure 37 can be changed or even made a variable resistance. Doing this would change the maximum attenuation and maximum amplification of the tone control circuit if the network design calls for such a change.

While the Baxandall Tone Control Network is undoubtedly a more compact and streamlined solution the Three Channel Tone Control Network will ultimately allow both the designer and the artist or listener more control. In the case of the designer it will allow for changes to be made easier if the corner frequencies, maximum and minimum attenuation, and maximum and minimum amplification. In the case of the artist or listener it will allow them to control the mid band frequencies whereas they would not be able to with the Baxandall Tone Control Network. For these reasons, the Three Channel Tone Control Circuit has been selected for use in the Pocket Amp.

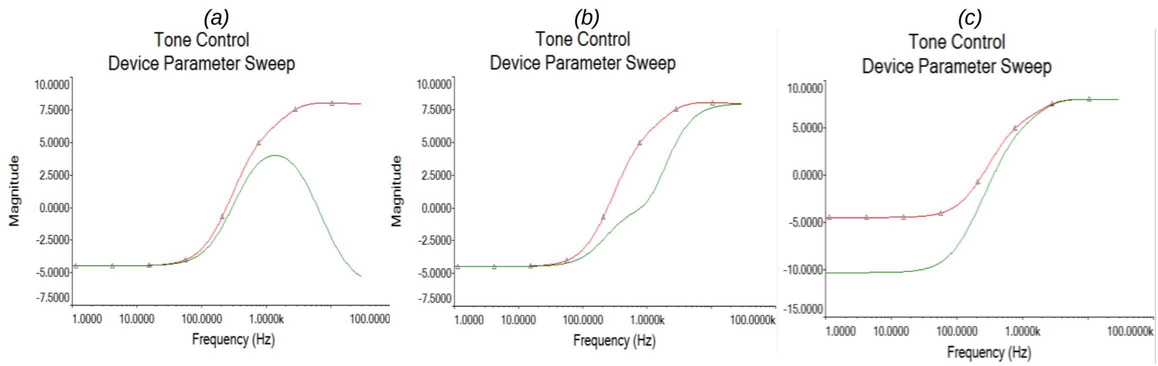


Figure 38. Magnitude Response of Three Channel Tone Control Circuit
Permission Pending from Texas Instruments

4.2.3.2 Selected Overdrive Method

While the diode clipping network explored in section 3.2.3.2 Overdrive of this report is simple it requires multiple variable DC voltage sources which would have complicated the power supply system for this project and would require variable voltage regulators which would drive the cost up significantly. For these reasons this design option was not selected.

If the single amplifier driven to its rails was used as proposed also in the section 3.2.3.2 Overdrive it would likely output a voltage at the rails at some point. This output would be at least ±5 Volts which would be well outside the maximum input voltage of the Analog to Digital Converter in the subsequent stage which is given by Vcc + 0.3 Volts or approximately 3.6 Volts. For this reason, this design was also rejected.

Because this project requires overdrive without significant gain in order to protect the Analog to Digital Converter from high voltages another option is required. This design makes use of two successive Operational Amplifiers with gain K and $\frac{1}{K}$.

The first amplifier will be driven to the rails using the gain to control the level of clipping and the second will be used to return the amplitude of the signal to one that will not damage subsequent components such as the microcontroller. This solution would be more expensive than the diode clipping circuit but would allow for much better control over the level of clipping simply by adjusting the gain instead of requiring variable voltage regulators. An example of this solution can be seen in Figure 39.

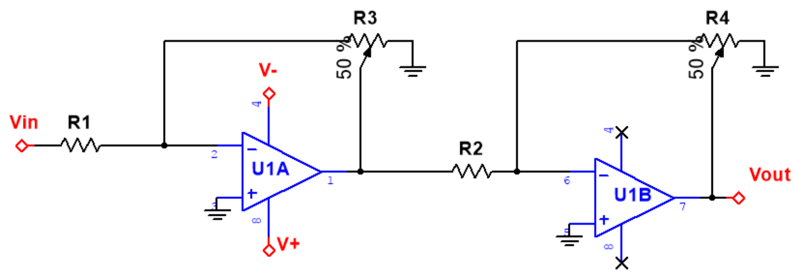


Figure 39. Overdrive with Cascaded Operational Amplifiers
Created by Joshua DeBaere using Multisim Student Edition

By changing the resistance of the variable resistors R3 and R4 the level of distortion caused by driving U1A to its rails can be changed while U1B attenuates to restore the same approximate voltage thereby providing the ADC a relatively stable, low voltage signal. To do this there must exist a relationship between the variable resistances R3 and R4 and the resistors R1 and R2 as shown in Equation 3. This equation will be implemented using the microcontroller to provide a signal to each variable resistance controlling the resistance seen by the feedback loop in the Op-Amp.

$$K = \frac{R_3}{R_1} = \frac{1}{\frac{R_4}{R_2}} = \frac{R_2}{R_4}$$

While this design is more expensive in terms of component cost and space on the PCB it satisfies the requirements of this project and therefore it has been selected for use in the final design.

4.2.4 Volume

Many guitar and general audio amplifiers make use of a transistor power amplifier or volume stage. However, the principal reason that this technology is used is its ability to produce a very high output current that is capable of driving fairly low impedance speakers, which are usually measured at less than 10 Ohms, to reasonably high voltages and therefore reasonable volumes. Technologies with a lower maximum output current simply cannot attain the volume needed for the signal to be heard and understood. However, in this case the impedance is much higher, ranging as high as 600 Ohms in some cases. This means that a much lower output current can be used to produce volumes of reasonable amplitude.

In this case, volume will be controlled by a simple inverting amplifier with variable gain. The gain will be controlled by using a variable resistor as the feedback resistor. Because the Pocket Amp is designed to be used with a pair of headphones instead of a speaker, the output current does not need to be as high as with a traditional guitar amplifier. Therefore, it is advantageous to use an Op-Amp due to its more ideal characteristics than a transistor amplifier. If necessary, a high output current operational amplifier can be used in this case however, as noted in section 4.2.1 Amplifiers, even the lower output current Op-Amps considered for this project are marketed as being able to drive a 600 Ohm load for audio applications.

4.2.5 Power Supply Design

The supply of power to the other fundamental portions of the Pocket Amp underwent several iterations before reaching the final design. The first step was to create a fundamental block diagram from which to begin creating the individual components of the design. The next step was to approach each component individually and select the necessary IC for that component’s function. Finally, with the ICs selected, the different passive components can be selected to form the necessary circuits.

4.2.5.1 Power Supply Fundamental Block Diagram

The decision to powering the device with a battery supply was easily reached as the device is portable. Next, a rechargeable battery was desired as it adds to the challenge of the design and aids in the functionality of the device. Other team members were questioned to gain a fundamental understanding of the different components that would require powering from the battery supply. Through these questions, the voltage and requirements of each component were known and the fundamental needs of the power system were established. The power supply would need to supply two sections, each composed of different devices and requiring different voltages. The first section is the analog components primarily consisting of operational amplifiers that provide filtering and analog effects for the Pocket Amp. These op-amps require both a negative and positive voltage to power the rails of the op-amps. The second section is primarily digital components: the MSP432 microprocessor, Bluetooth chip component, the Input/Output power for the MSP432. From these questions Table 6 was developed to roughly show the different power requirements.

Component	Voltage Requirement (V)	Current Requirement(mA)
Various Op-Amps	±2.25-18	±4.5 (per channel)
Bluetooth (nRF51822)	1.8-3.6	120 (maximum)
MSP432	1.6-3.7	4 (at 48MHz)
I/O Power	3.3	Negligible

Table 6. Initial Power Requirements

By utilizing this table, a fundamental block diagram (Figure 40) was constructed in order to make further refinements to the design.

With the fundamental block diagram completed, each of the components can be worked on separately, gaining greater detail as each component is finalized.

Working from largest component to smallest, the proper choice for the battery was established. Both nickel metal hydride and lithium-ion batteries were considered, each having different characteristics advantageous aspects for the Pocket Amp. However, after consideration the lithium-ion proved to be the superior choice primarily as its output voltage of 3.6V is much closer to the 3.3V required by the digital components. Additionally the analog component went through a design phase which required a clipping circuit and thus a voltage bias to alter the clipping level. Additionally, it was decided to add some form of buffer stage to each digital component. With these design choices in mind the fundamental block diagram was altered and given greater detail. Voltages of each components were added and the smaller necessary components. Figure 41 reflects many of the changes for the second fundamental block diagram.

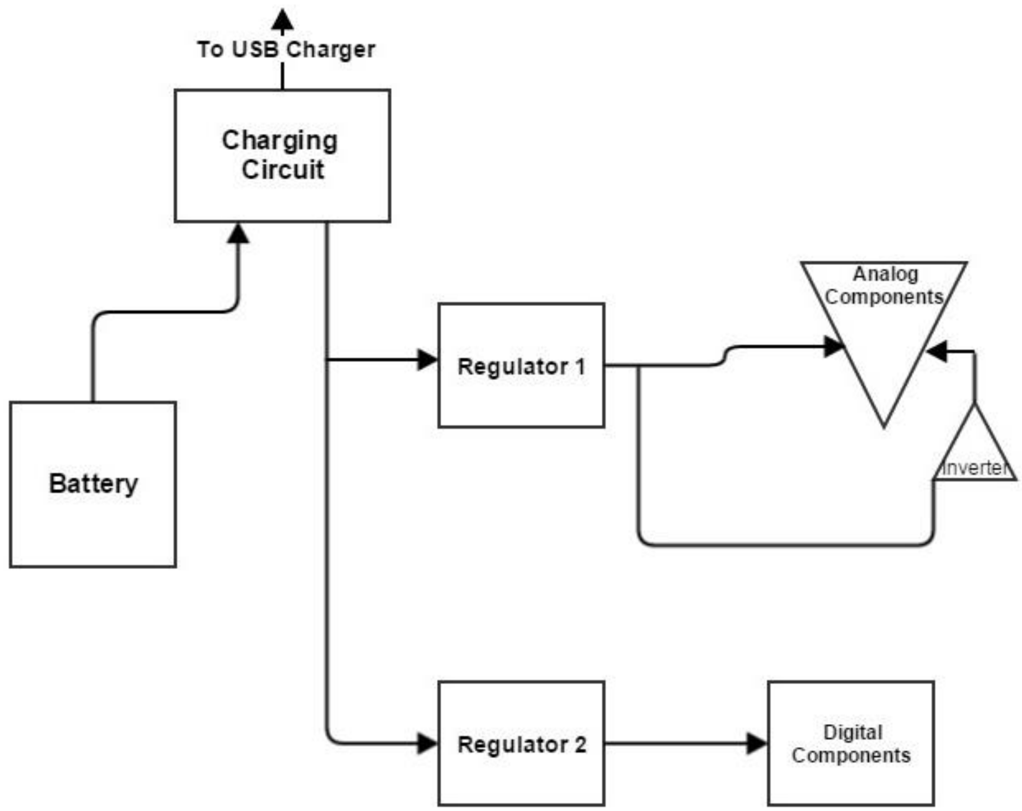


Figure 40. Fundamental Power Block Diagram Iteration #1
Created by Trent Coleman

With the creation of the details block diagram the different regulator components were selected to meet the different sections requiring different voltages. For the step-up regulator required for the op-amps a boost switching converter was selected. The inductive boost regulator was selected over other step-up DC/DC converters such as a charge pump for its higher efficiency, especially for low-current loads, and time tested use in power supply design. The options for the step-down regulator were more difficult as both LDO linear regulators and buck switching regulators were very suitable for the voltage reduction. LDO's generally provided a smaller package and do not require an inductor in their construction. Additionally, LDO's generally produce less noise as they are not constantly switching and producing noise. Buck switching regulators sport a higher efficiency and are more able to handle the larger drop as the lithium ion battery goes from 4.2V to 3.3V. However, in this application the buck regulator sports only a marginally higher efficiency. The nominal voltage of 3.7V for the battery will still produce an efficiency greater than 90% when utilizing an LDO regulator.

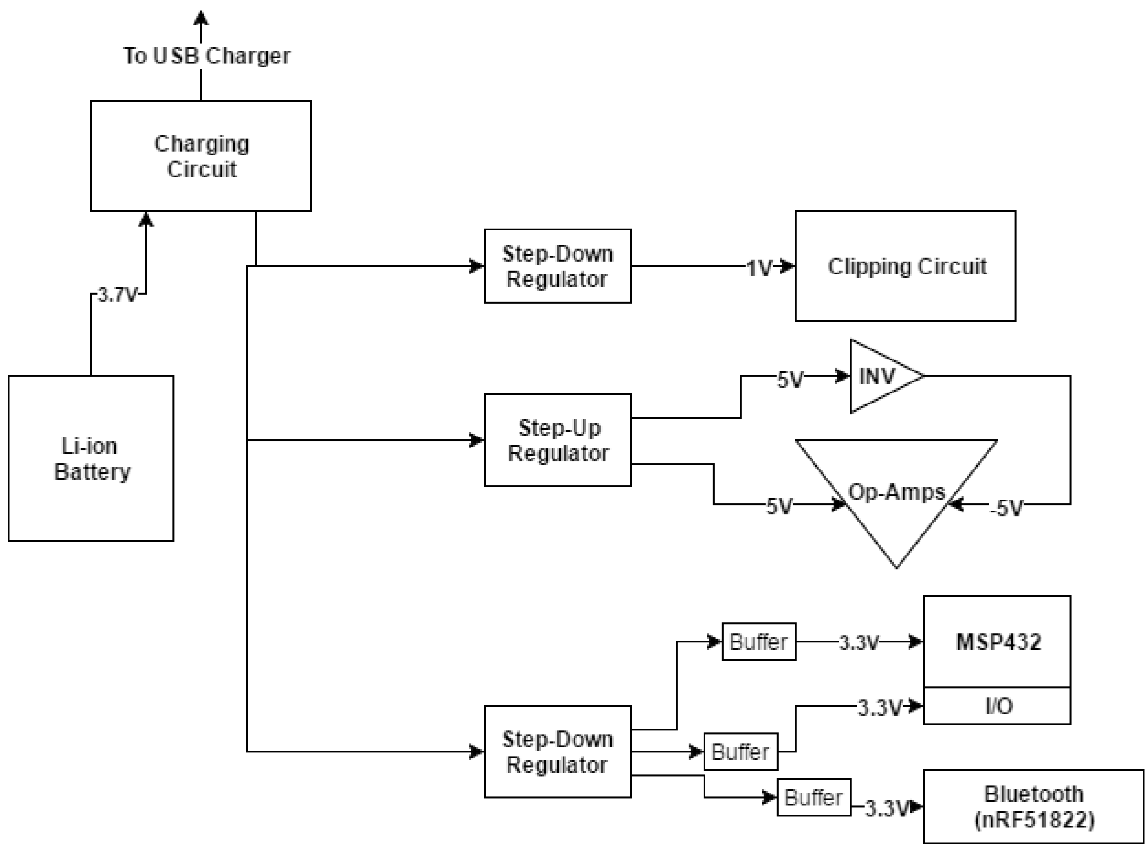


Figure 41. Fundamental Power Block Diagram Iteration #2
Created by Trent Coleman

Further research was also done for the isolation of the digital components. Unfortunately, a traditional op-amp buffer would have proved impractical in nature as then their own rail voltages would have to be provided for them. Optoisolators were considered for their ability to transfer electrical energy to light and then back to electrical energy, thus isolating one line from another. Another method of isolation was provided through transformer flyback devices, which transfer electrical energy to magnetic energy and then back to electrical to form electrical isolation. However, after consulting other electrical engineers and observing the isolating requirements of batteries, no isolation was deemed necessary for the digital components. Furthermore, the analog designer developed a different method of clipping without utilizing a 1V bias voltage, and found more ideal op-amps which required a higher rail voltage.

To complete the final design for the fundamental diagram the inverted rail voltage was considered. Unfortunately, traditional op-amp inverters would prove impossible to use, as an inverter does not operate if its rails and its input are provided by the same source. Again opto-isolators were considered as inverting buffers. However, the best design was decided to use an inverting buck-boost switching converter to drive the negative rails directly, giving them a more direct access to the battery power source. With all of these final design considerations established the final block diagram design (Figure 42) was formed.

The fundamental block diagrams provided a structural basis for the rest of the design. By creating the these blocks the complexity of the design can be greatly reduced. The fundamental blocks allow the design to be compartmentalized reducing the design greatly. This compartmentalization allows each component to be handled individually, without the rest of the larger design being affected. With the final block diagram created, attention to the individual design choices can be established and then after each active component is created, the individual passive components can be chosen.

4.2.5.2 Step-Down Regulator Selection

The first component to be chosen is the step-down regulator. The first decision for this regulator was to choose between linear regulation and a switching converter. Switching converters provided an option of higher efficiency, and no dropout voltages. Linear regulators provide an overall cheaper design while creating less noise.

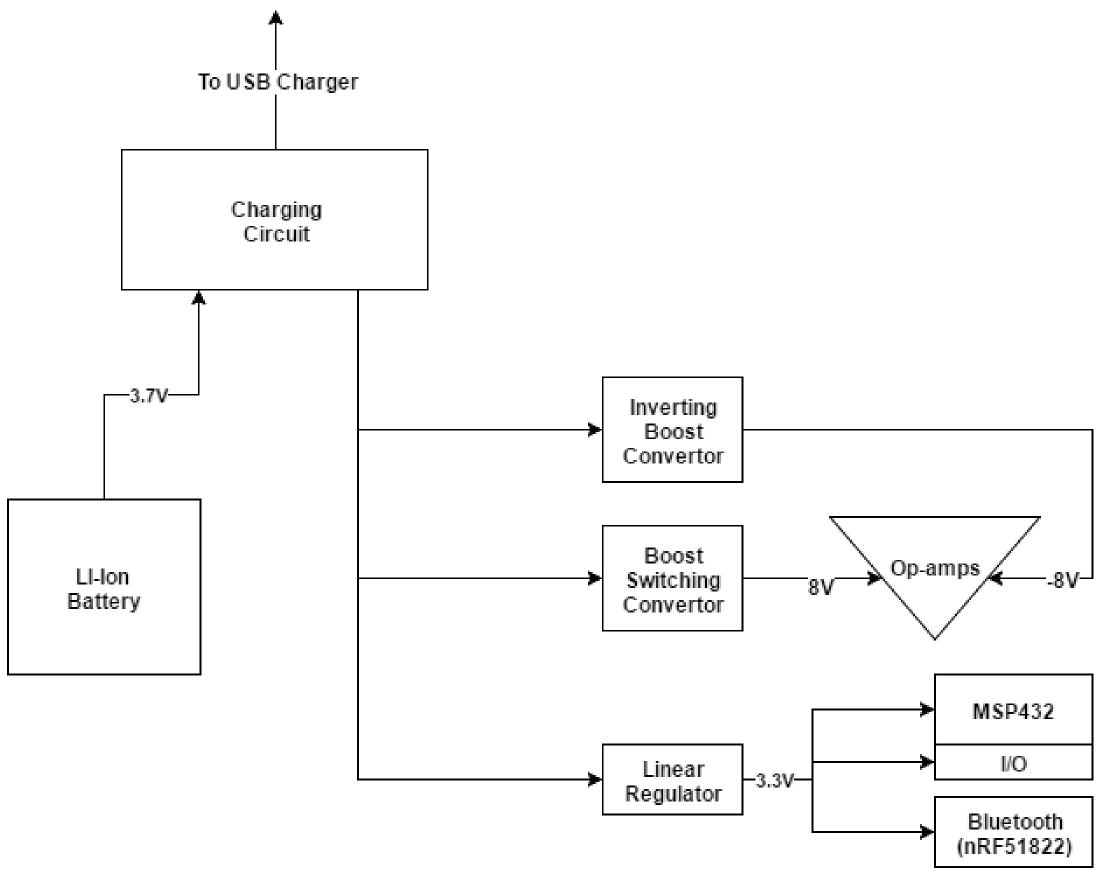


Figure 42. Fundamental Power Block Diagram Final Iteration
Created by Trent Coleman

4.2.5.2.1 TPS62160

One considered IC was the Texas Instruments TPS62160 Buck converter. The 62160 boasts an efficiency greater than 90% at an output current of 500mA, and easily be configured for a 3.3V output. This high efficiency at low load currents was essential for supplying the digital components, as even at maximum draw they draw less than half an amp. The buck converter would be an excellent choice for a large step-down in voltage with low output currents. The 62160 is able to receive a large input voltage range of 3 to 17 volts, which makes it an able converter for large step-down voltages. However, despite these admirable qualities the TPS62160 is not the best choice for this project. The inductor necessary for the switching converter greatly increases the device's footprint of the PCB board. The switching regulator will also create more noise when compared to a linear regulator. These detriments led the rest of the selection process to focus on LDO regulators.

4.2.5.2.2 TPS776

The next regulator under consideration was the Texas Instruments TPS776. The TPS776 is a LDO regulator which supports an input voltage range of 2.7 to 10V and can create a regulated output from 1.2 to 5V. Additionally, the 776 only supports a maximum output current of 500mA. These parameters ensure that the current draw for the digital components is easily met by the regulator, but ensures that the excessive currents will not be able to cause damage to those components. The 776 also supports fixed output voltage of 3.3V, enabling a more consistent output and requiring less external passive components. These features made the TPS776 a very suitable candidate for the step down regulator but was not chosen as the primary regulator. The TPS776 was selected as a back-up option in case the chosen regulator proved unsuitable for unforeseen reasons.

4.2.5.2.3 LP2989

The final and ultimately chosen regulator was the Texas Instruments LP2989. The LP2989 is a LDO regulator, which accepts input voltages from 2.1 to 16V and outputs voltage from 2.5 to 5V. The 2989 holds all the advantages of the TPS776 while gaining some essential features. In addition to low maximum output current, and fixed output voltages, the 2989 supports very low noise down to $18\mu V_{rms}$ and ceramic capacitors. The low noise is a very desirable feature as noise is very detrimental to the audio signals that are being transmitted throughout the rest of the Pocket Amp. Ceramic capacitors are also very useful as they are cheaper, safer and easier to use and design then non-ceramic capacitors. The LP2989 also comes in very useful pin configuration both 8 pin SOIC and 8 pin VSSOP. These packages are very common and easy to find breakout boards for, aiding in the breadboard testing and eventual troubleshooting. An additional useful feature the LP2989 supports is an error flag which triggers high when the output is 5% below the desired voltage. This flag can be fed to the MCU to form an easy battery depletion circuit.

As a TI product, the LP2989 is also easy to simulate utilizing the free TIWebench online tool. This tool greatly easies the design process as the input and output characteristics can be entered into TIWebench and the surrounding passive component values will all be generated for the proper output. In addition to full circuit schematics, the TIWebench tool allows users to simulate efficiencies, output waveforms, heat mapping and other vital simulations for creating the physical design.

The following circuit design was created utilizing the TIWebench tool. Assuming an output current of 0.2A a consistent output of 3.2915V at steady-state conditions. Input and output capacitors of 4.7µF are implemented to protect both the regulator and the following circuit. A 10nF capacitor is placed as a bypass, this capacitor is instrumental in reducing the noise generated from the regulator. A 330 kΩ resistor is used for the error signal. The output voltage of 3.3V is automatically decided by purchasing the 33 model of the LP2989 which is preset to output 3.3V. The next figure (Figure 43) shows this designed schematic with a test load 16.5 Ω.

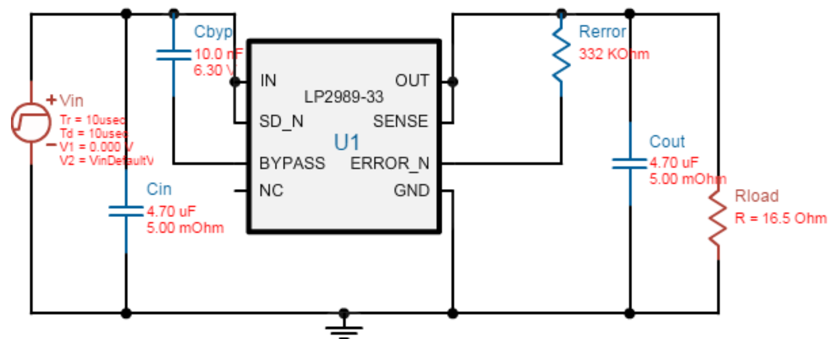


Figure 43. LP2989 Schematic
Created by Trent Coleman Using TIWebench Power Designer

Additionally, various output and input waveforms can be generated to simulate the proper output of this circuit. Start-up and steady state responses are some of the most important simulations as they show the initial and normal operation of the circuit. TIWebench allows its users to easily create such output graphs and use them for analyzing the responses of their circuit. The two waveforms below were simulated using the schematic above and show that both the startup and steady state responses will easily suffice for this circuit. These output waveforms are displayed in Figure 44.

In addition to the output waveforms important graphs comparing the various parts of the circuit can be formed in TIWebench. Most of these graphs are useful for comparing the various power and current outputs but most effectively utilized for measuring the efficiency of the circuit. The efficiency graph for the LP2989 circuit in Figure 45, clearly shows an efficiency of roughly 80% when the battery is started at 4.2V. However, when the battery reaches its nominal voltage of 3.6V an efficiency of 90% or greater appears.

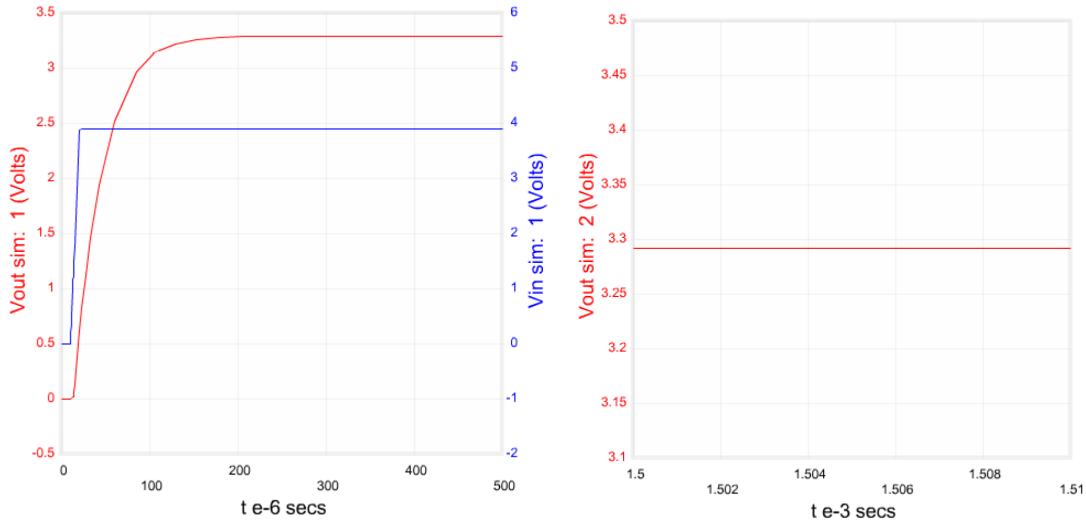


Figure 44. LP2989 Startup and Steady State Outputs
Created by Trent Coleman Using TIWebench Power Designer

4.2.5.3 Boost Converter

The next IC component to select was the boost converter. This boost converter would be required to supply the positive rails of the Op-amps of the Pocket Amp and thus must be able to boost the battery output to at least 5V. Three different boost converters were considered for providing this output voltage.

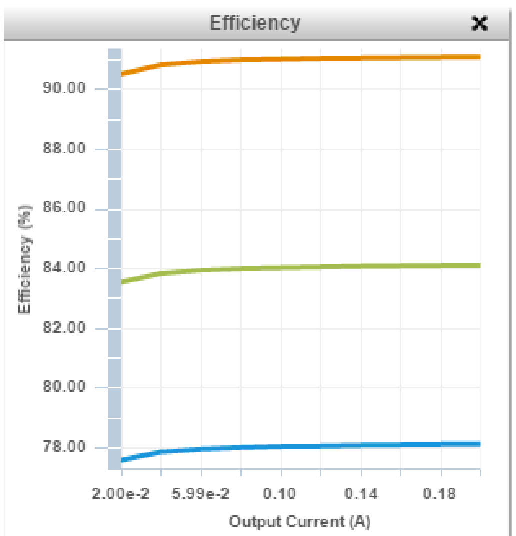


Figure 45. Efficiency of the LP2989 Circuit
Created by Trent Coleman Using TIWebench Power Designer

4.2.5.3.1 TPS61256

The Texas Instruments TPS61256 is a high efficiency step-up converter capable of outputting a well-regulated 5V. The 61256 can produce its output from an input voltage range of 2.5 to 4.85V which easily with the battery's ability to produce. Additionally, the 61256 is able to keep a high efficiency even while providing a low output current, which is very important, as the op-amps do not draw a large amount of current. While the TPS61256 would have been sufficient for an excellent 5V output two major factors disqualified it from the design of the power supply. The first disqualifying factor is that the TPS61256 is locked-in at outputting only 5V. Because 5V is the absolute minimum voltage it was decided that a higher voltage should be provided, so that if an issue arises and the voltage drops the op-amps are not immediately turned off. The second disqualifying factor is that the TPS61256 is only available in a packaging known as DSBGA. DSBGA is a surface mounted wafer level CSP that boasts the smallest footprint for I/O count. While this package is very small, it is difficult to find breakout boards in, making prototyping and testing very difficult.

4.2.5.3.2 TPS60150

The TPS60150 is a 140 mA, 5V charge pump created by Texas Instruments. The 60150 is able to receive input voltages from 2.7 to 5.5V and outputs a controlled 5V. This input range easily meets the output range provided by the lithium-ion battery. The output current reaches its maximum at 140mA, which would be too small for many applications but is sufficient for the very low current draw of the op-amp channels. The charge pump would have been a useful option for the Pocket Amp as it does not utilize an inductor saving both cost and space. Unfortunately, the TPS60150 was disqualified for similar reasons to the TPS61256, output voltage and package type. The 60150 is limited to only a 5V output, which would work for optimum output conditions but might turn off the op-amps' supply if the voltage drops for any reason. The TPS60150 is also limited to the WSON package, which is a proprietary TI technology. The WSON is very compact and is generally inexpensive which would normally be an excellent choice for this project, but it is almost impossible to find breakout boards for WSON. Because of the lack of breakout boards, prototyping and breadboard testing becomes very difficult.

4.2.5.3.3 LM27313

Another boost converter considered for the op-amp rail supply was the LM27313 by Texas Instruments. The LM27313 is listed as a 1.6MHz boost converter with 30V internal FET switch in SOT-23. The 27313 supports input voltages from 2.7 to 14V and can produce a range of output voltages from 4 to 28V. The voltage ranges meet both the output requirements and the input voltages provided by the battery. The high output voltage range permits the design to exceed the minimum voltage requirements of the op-amps. Additionally, the SOT-23 package is able to be put on breakout boards and thus easily tested. From this viewpoint the LM27313 was a very viable choice for the power supply. However, a quick simulation on TIWebench revealed a lower efficiency rating than the selected converter (see Figure 46). While it ultimately was not the chosen converter the LM27313 was purchased a backup in case the chosen converter proved unable to realize for unforeseen circumstances.

4.2.5.3.4 TPS61085

The final boost converter that was considered was the Texas Instruments TPS61085. The TPS61085 is an 18.5V, 650kHz / 1.2MHz, DC/DC converter with forced PWM mode. This converter accepts an input range of 2.3 to 6V and can output voltages from 2.8V to 18.5V. These ranges easily meet both what the battery can provide and what the op-amp rails will require. The output range also allows the design to produce a voltage higher than the bare minimum, which the op-amps require. The high frequencies of 650 kHz or 1.2 MHz allow for the use of smaller inductors and capacitors. These high switching frequencies also provide quicker transient responses. The forced pulse-width modulation (PWM) mode allows for a steady output even as the input declines over the discharge of the battery. The TPS61085 also comes in both VSSOP and TSSOP packages which are not only compact packages but also can be tested easily as breakout boards are readily available for them. Ultimately, the decision was made to utilize the TPS61085 for the Pocket Amp. The TPS61085 is able to provide an able 8V for the op-amps voltage supply, which exceeds the minimum 5V by 60%. The available packaging allows for proper testing of the design. Since the TPS61085 is a TI product, the process of selecting passive components is greatly streamlined by using TIWebench. Utilizing TIWebench the following schematic (Figure 47) was designed and simulated.

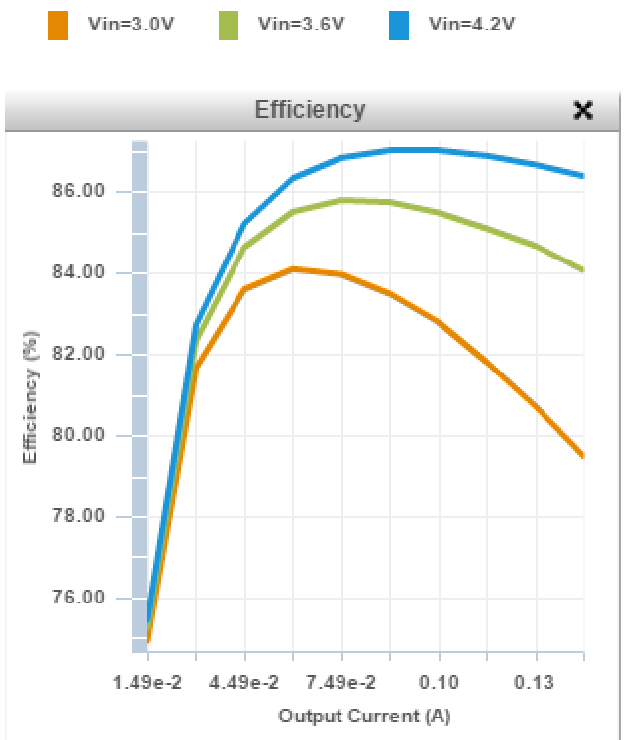


Figure 46. Efficiency for the LM27313 Circuit
Created by Trent Coleman Using TIWebench Power Designer

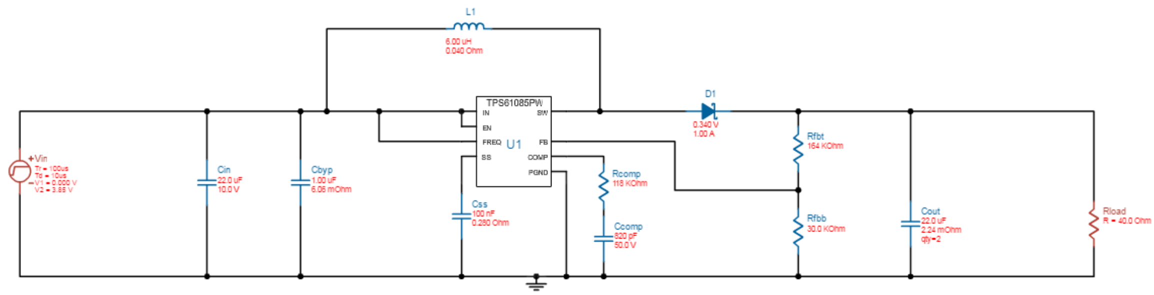


Figure 47. TPS61095 Schematic
Created by Trent Coleman Using TIWebench Power Designer

Because the TPS61085 does utilize fixed outputs it must rely on set resistors to create the desired output voltage level. For an 8V output, two resistors are used to set the voltage a 164kΩ and a 30kΩ. The ratio between these two resistors is used to modify the feedback level and produce 8V. For the rectifying diode D1, a Schottky diode was selected in order to reduce the power consumption across the diode. The input and output capacitors are 22µF capacitors, which provide both current protection and rectification for the output capacitor. A 6.00µH inductor was selected to produce the energy storage requirements for the switching converter. The external compensation pin is attached to a 118kΩ resistor and 820pF capacitor. This RC network improves the circuit's ability to handle transient responses. Using TIWebench's simulation tools, the startup and steady state waveforms were generated.

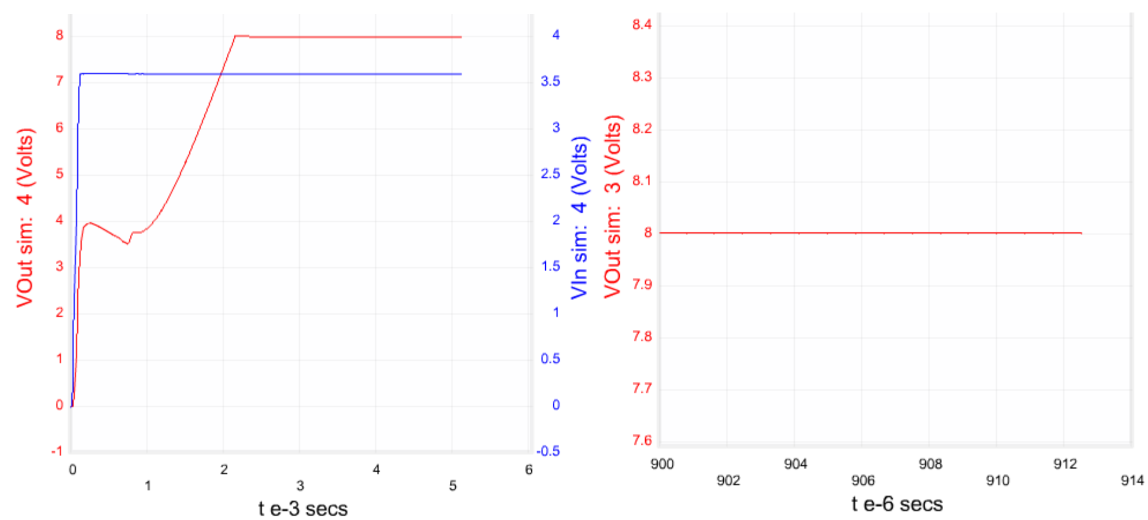


Figure 48. TPS61085 Startup and Steady State Outputs
Created by Trent Coleman Using TIWebench Power Designer

As can be observed from the output waveforms in Figure 48, this circuit design for the TPS61085 provides a very constant voltage at 8V and reaches 8V from start up within 3ms. This kind of response is exactly what is necessary for op-amp initialization, a highly consistent steady state with a quick initial startup. Just as importantly, the efficiency of the device remains high even for the low output currents.

As can be seen from the graph in Figure 49, the efficiency of the circuit decreases as the battery will continue to drain. The efficiency of the circuit does remain high, around 90%, for the nominal voltage of 3.7V. While the individual current draw of a single op-amp is very low, only about 5mA, there will be quite a few op-amps are require a voltage. With all of these supplies in parallel, they will draw a sufficient current in order to reach the high efficiency produced by larger current draws.

Despite this drop in efficiency the TPS61085 proves to be an excellent option for providing the positive rail voltages for the op-amps of this project. The easy 3.7 to 8V boost does not cause a large amount of power consumption and results in a nominal efficiency of roughly 90%. The converter's ability to maintain these high efficiencies while providing low currents is very useful and not always a common feature in boost converters. Finally, the ability to utilize TIWebench as a design tool is extremely important. The use of free, accurate simulation software ensures that the design process is simplified. For these reasons the TPS61085 was an excellent option for the power supply design.

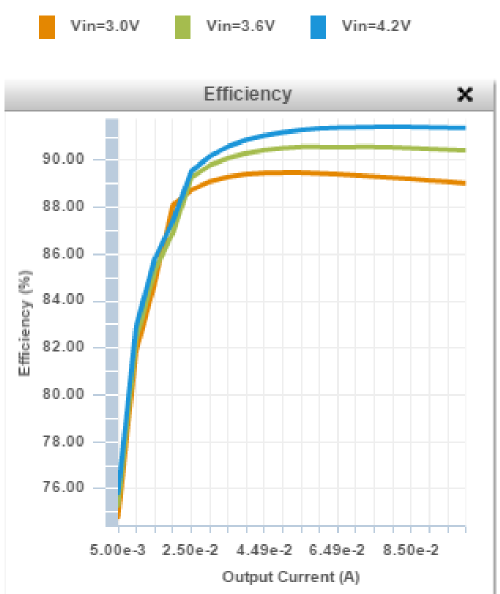


Figure 49. Efficiency of the TPS61085 Circuit
Created by Trent Coleman Using TIWebench Power Designer

4.2.5.4 Inverting Converter

The final IC necessary for the voltage conversion section of the design was the inverting regulator. For the negative rails the op-amps several options of inversion were considered. A normal op-amp powered inverting buffer was the original option. However, this inverting buffer would require its own positive and negative rail supply and furthermore an op-amp cannot operate correctly if its input and rails come from the same sources. An inverting flyback transformer was also considered, but such an option might require a custom built transformer, a difficult and expensive proposition. Due to these limitations its was decided to provide the negative rails their own dedicated regulator which gives out a negative voltage. Several inverting regulators were considered before the final selection was made.

4.2.5.4.1 LT3479

The LT3479 by Linear Technologies is listed as a 3A, full features, DC/DC converter with soft start and inrush current protection. The 3479 is capable of inverted outputs and accepts an input voltage range of 2.5 to 24V. The initial design was to utilize the LT3479 as an inverter only, and power it from the end of the boost converter. However, this design was not preferred as it introduced a large amount of components for less efficiency and possible strain on the boost converter. The LT3479 was purchased in case it could be used for a backup in case the primary inverting regulator proved impractical for unforeseen circumstances.

4.2.5.4.2 LT3483

The LT3483 is an inverting micropower DC/DC converter with an in-built Schottky diode. This converter was designed by Linear Technologies and possess many of the qualities necessary for this project. The converter can receive input voltages from 2.5V to 16V and can output inverted voltages up to -38V depending upon the circuit topology. The LT3483 is also quite flexible in its circuit topology. Where most converters ICs are designed specifically for inductive or charge pump applications, the LT3483 is able to be used for charge pump, inverting flyback or inductive DC/DC conversion. However, an inductive conversion will be used as it is easier to design utilizing the LT3483's datasheet. Unfortunately, the efficiency of the LT3483 is much lower when compared to the boost converter and step-down regulators. However, this lowered efficiency is unfortunately common among inverting regulators. Part of the LT3483's specialization for this project is that it's efficiency is large for particularly small currents. The LT3483's datasheet provides are very excellent circuit design which can directly be used for the Pocket Amp's power supply design. This ideal circuit application can be observed in Figure 50.

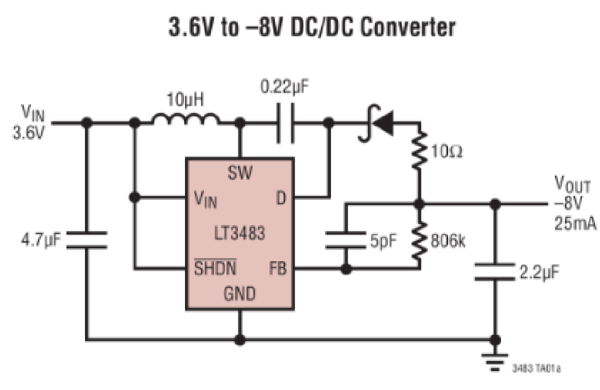


Figure 50. LT3483 Schematic
Permission Pending from Linear Technologies

The design works with minimal power loss as the set resistor which would draw much of the output voltage is only 10Ω. This circuit is almost exactly what is likely to be required from the Pocket-Amp's design. Each op-amp will require -8V and draw 4.5mA from their negative rails. Thus this circuit could easily supply five op-amps and supply even more with greater efficiency. While this provided circuit is very optimal, it is less efficient than the other regulators in the design. This efficiency is shown in Figure 51. However, with the high efficiency of the other regulators and simplicity of this design, some efficiency loss was deemed acceptable.

4.2.5.5 Battery Charging Schemes

With the selection of lithium-ion as the primary supply design, a method of recharging the lithium battery is required for the design. The charging circuit is designed to take the 5V input from a standard USB charger and apply it safely to the lithium-ion battery so that the battery reaches a 100% charge and is not damaged by the charging process.

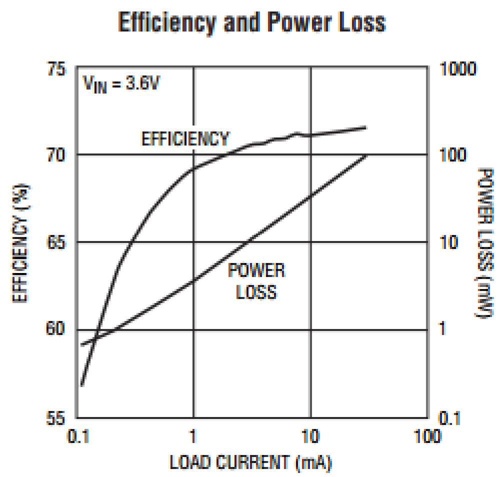


Figure 51. Efficiency of the LT3483 Circuit
Permission Pending from Linear Technologies

For a lithium-ion battery, charging can be done in several different methods. The first method is constant voltage charging. This method works by applying a constant voltage across the battery, driving it to its max charge. The source current can be varied in order to match how quickly the battery is filled. However, a current greater than the battery's hourly capacity should not be applied to lithium-ion batteries. Once this maximum charge is met the source current is reduced greatly to a point where it only provides enough current to counteract the battery's inherent self-discharge rate. For the battery in this project the set voltage is 4.2V which details what voltage should be supplied and the hourly rate is 2500mAh, though the manufacturer of the lithium-ion polymer battery recommends that it not be charged at a rate greater than 1200mA. This kind of circuit can be created through the use of voltage converter to create the 4.2V set voltage, and by setting larger resistor values in parallel with the battery the overall current through the battery can be effectively limited. However, such a design is only suitable for slow charging, where the input current is always safe for the battery to handle. There is no method of switching the current to a lower state once it reaches a certain threshold of charge.

Another regulator method utilizes a regulator and a BJT transistor to create the current for the battery. The design works by allowing the regulator to regulate the base current of the BJT. The BJT determines a fixed current to pass to the battery cell. As the battery reaches its maximum charge voltage then the regulator will stop driving current to the BJT as its desired voltage has been met. Thus, the current flow to the battery will also slow to a trickle. A downside of this design is that is produces a fair amount of energy waste between the regulator and the transistor, especially once the final charge has been met. This method also requires very precise resistors, which can be difficult to find and requires a current limited source so that the BJT does not permit an excessive current.

Another option is to use a specialized charging controller such as the Texas Instruments LM3420. These controllers are regulators at the core of their function but are hooked with more complicated circuits. One of these charging circuits utilizes two MOSFETS in addition to the current driving BJT. This circuit is similar in using the BJT to drive the current into the battery with the regulator monitoring the feedback and adjusting the current flow. While the battery is below its set voltage the two MOSFETs are fully on allowing the full current of the current restrained source to flow to the battery. Once the battery reaches its full set voltage, the BJT lowers its driven current. Once this occurs, the regulator takes the position of driving the MOSFETs in linear mode instead of allowing them to be fully open. These dedicated charging ICs are by far the superior charging method, as they do not require an overly precise resistor network and more accurately regulates the current flowing into the battery. The downside of these circuits is that they can be quite costly and more importantly difficult to design. Additionally, while these networks are already very complicated they do not handle the challenge of charging during simultaneous usage of the battery nor do they handle how the battery is connected to both the load and the charger.

For this project – instead of designing an original charging network – it was decided to purchase a dedicated charging network from the battery supplier. While a simple slow charging network could be made to charge the battery, this is insufficient for the intended needs of the pocket amp, portability and user friendliness. Both portability and user friendliness demand a charging speed faster than trickle charging will permit. Fast charging networks can be incredibly complex however, including numerous transistor and diode components. With this complexity, the probability of design error increases exponentially. While other components can be troubleshot and adjusted with time, failures concerning the battery are more troublesome. If a regulator fails, then another regulator can be purchased for relatively low cost and a new design tested. If a battery charging network fails, then a battery can be irrevocably damaged, leading to a potentially dangerous situation. Replacing batteries is a much costlier item than a regulator and the failure modes of lithium batteries could cause severe damage or injury. As stated earlier in this paper overcharging a lithium-ion paper can have explosive consequences. For safety and project security reasons, a manufacturer designed dedicated charger, which has more time and expertise behind its design, was chosen for this project.

4.2.6 Block Diagrams

In Figure 52 below, the diagram shows the basic block diagram of the flow of analog effects that we will be implementing. The audio signal is first put through a filter before it is split into three to parts to allow for tone control of the low, mid, and high frequency ranges. In musical terms, this controls the bass, midband, and treble ranges. The three parallel signals are multiplied by the desired amount of gain and then subsequently added back in together before they can be put through the overdrive effect.

In Figure 53 below, the block diagram shows the order of effects that the pocket amplifier will implement. If multiple effects are desired to be implemented simultaneously, the order in which they are produced is very important. In music, there is not set order of effects with rules that cannot be broken. This is because different combinations of effects produce drastically different sounds, which even

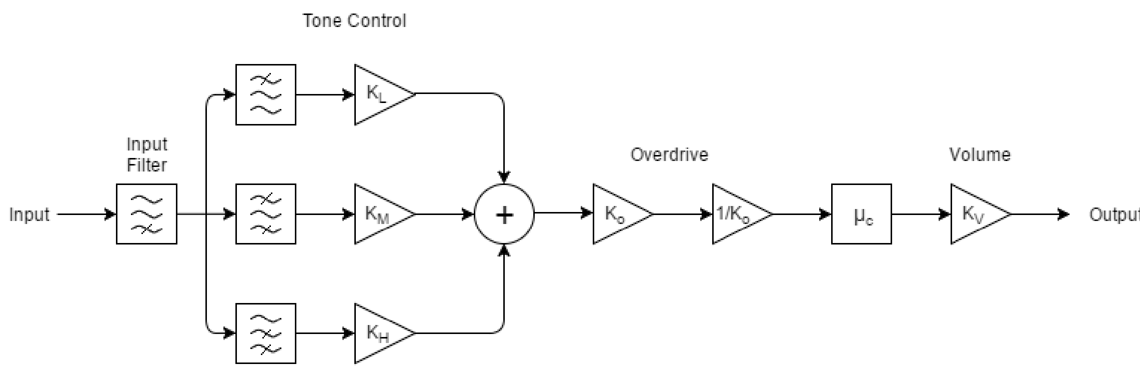


Figure 52. Analog Effects Block Diagram
Created by Marissa Kane

if they are uncommon they may still be desired at times. Our order of effects was chosen to produce the purest sound. Tone control and filters come at the beginning of the effects chain to ensure that the signal that is then sent through the other effects is as clear as possible. Overdrive is the next effect, because distorting the sound is much cleaner before any time-based effects are implemented, as all of the following effects are. Flanging and chorus are modulation effects that follow overdrive. Flanging and chorus are similar effects with variable delay lengths. Modulation effects have the ability to overpower any effects that come before them if they are set to have a high resonance. This makes it not ideal to have them at the end of the effects chain. Having these modulation effects in the middle of the order allows for a clearer tone control and overdrive, while not overwhelming the other time-based effects in the process. The time-based effects delay and echo follow immediately after the modulation effects. In this way the sound is developed from all of the other effects first and then the delay or echo will appear significantly more natural than if those effects had been implemented earlier on in the chain. Reverb is the final effect to be added in before the audio is output. Reverb is a combination of multiple types of delays and is intended to imitate the sound the instrument would make while reverberating in a natural environment. Therefore, to produce the most authentic tone, reverb should be the final effect.

Figure 53. Effects Order Diagram
Created by Marissa Kane

4.2.7 Processor Design

Out of all the processors and architectures researched, the candidate that stands out most is the MSP432 ARM processor. The 32-bit ARM architecture is extremely appealing for the Pocket Amp due to its widespread use – there will already be large amounts of code and many developers to draw knowledge from. The RISC implementation of ARM gives a predictable number of instructions per clock cycle and since the MSP432 ADC will have a fixed sample rate and fixed clock rate for the MSP432 core we can easily figure out the number of instructions we can execute within one sample. Knowing the number of instructions per sample gives us an upper bound on the complexity of the digital effects.

The built-in ADC reduces the number of parts and complexity of the PCB and hardware design. The ADC features 14 bits of depth, which is more than enough for audio quality for our application. Sampling audio at an industry standard of 44,100 Hz will give us a similar quality to audio CDs but with a slightly lower dynamic range. The MSP432 has a maximum clock rate of 48MHz which gives the MCU over 1,000 clock cycles for every sample taken. This clock frequency is enough that most DSP effects are possible, especially time-domain functions.

The MSP432 features a Launchpad system which would make prototyping easy. Breadboarding the components the MSP432 will talk to will be extremely easy as the Launchpad breaks out the pins to breadboard compatible female headers. All power management and programming is taken care of on the Launchpad board which for prototyping leaves just the coding and communication to developed. This removes many hurdles and greatly speeds up the development of the prototype.

The MSP432 code development is all done in TI’s Code Composer Studio which the team has experience with and is widely used and features many code examples built-in and online. Having a feature-rich environment for code development and debugging will greatly accelerate the prototype development. MSP432 physical development has many projects to use as reference material including TI implementations and recommended setups.

5. Prototyping

Prototyping will begin at the end of November and will continue through the start of January. The goal of prototyping is to implement all the components in a temporary development environment in order to verify the design. The hardware prototyping is done in breadboards and breakout boards while the software prototyping is done in debug environments for the respective components.

5.1 Power Supply

The power supply is what will charge the batteries and provide power to all the different parts of the Pocket Amp. Prototyping is done by physically creating the designed circuits and testing their output to verify it is within the required specs. Heavy focus was done during the initial prototyping on the power supply as it will be required to run all the other parts – no integration can be done without a single working power supply.

5.1.1 Digital Loads Regulator

The prototyping process for the digital loads supply will consist primarily of constructing the circuits detailed in Section 4.2.5.2.3 LP2989. These circuits will be constructed on breadboards for easy modification and portability and with materials found in UCF student labs and purchased from various suppliers. The LP2989 regulator chip was placed on a breakout PCB board so it could be easily implemented on a breakout board. While the final passive components of the design will be surface mounted components, for the prototyping stage through-hole components will be utilized. Once the circuit has been constructed and double checked, ideal voltage inputs will be fed to the circuit and the output will be recorded with a digital multimeter. The input voltage will be varied to different voltages within the battery output range to show how the regulator will handle the different voltages the battery will supply. It should be noted that the circuit in Figure 53 will likely undergo several small changes as more ideal passive components are purchased. If an inconsistent or unregulated output is produced then the circuit will be rechecked thoroughly to ensure the circuit is properly constructed. Next, the different components of the prototype will be removed and checked individually for errors, replacing any defective components. Finally, if all components are working properly and the circuit still is outputting an incorrect output, then the overall design of the circuit will be redesigned. Figure 54 shows the first initial prototype of the LP2989, 3.3V output linear regulator circuit.

5.1.2 Analog Loads Converters

For this project the analog loads are supplied by two boost converters. The first boost converter outputs a positive voltage while the second boost converter is an inverter. As both converters are inductive converters they will be built apart from one another to prevent mutual induction. For the final design either shielded inductors will be used or an inductive shield will be created.

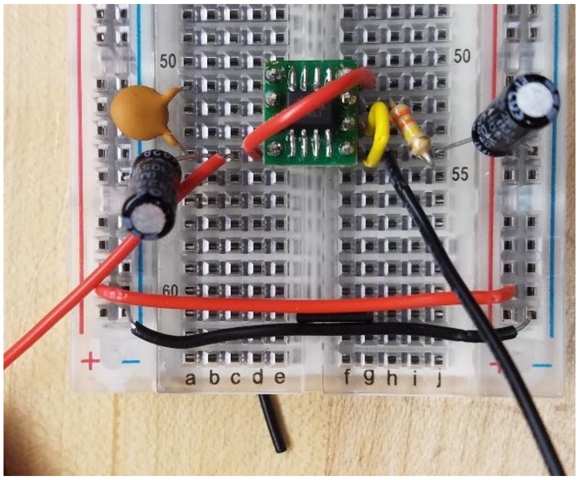


Figure 54. Prototype LP2989 Circuit

The prototyping of the two boost converters will take place by creating the circuit shown in the design sections 4.2.5.3.4. and 4.2.5.4.2. These circuits will be realized on a breadboard for easier modification and troubleshooting. The circuits will be utilizing through-hole passive components and a PCB breakout board for the TPS61085 and LT383 chips. For the final PCB implementation all components will be surface mounted for a more compact design. For the circuit shown in Figure 55 the input is fed in from the right on a red wire, from there the input is connected to input and bypass capacitors as well as the circuit's inductor. This input is also fed into the input, enable, and frequency pins of the chip. The output of the chip exits the top right pin and connects to the inductor and the Schottky diode. The diode then connects to the resistor network which forms the feedback loop. The diode also connects to the output capacitor and the final output which is the black wire exiting to the left. The chip also has a compensator pin on the bottom left which forms an RC network to ground.

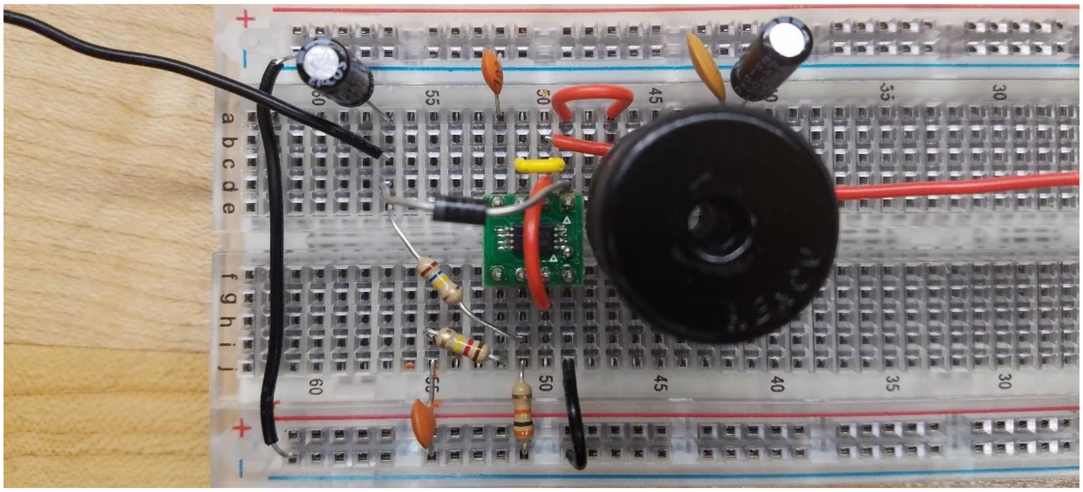


Figure 55. Prototype TPS61085 Circuit

The LT3483 circuit shown in Figure 56 has its input fed from the red wire on the bottom of the circuit. Immediately, the input is connected to ground with the input capacitor to aid with transient responses. After the input capacitor the input is fed to the inductor, and the input and shutdown pins via red wires. The other end of the inductor connects to the switching pin via a red wire. The switching pin connects to the D output pin via a capacitor which is linked to a reversed Schottky diode. On the other side of the diode, the resistive network is connected to the feedback pin and to the output capacitor. The output of this inverting converter is found in the connection between the resistive network and the output capacitor. The ground pin was connected to the ground bus using a black wire. It should be noted that the initial prototypes in Figure 55 and Figure 56 were constructed using materials from UCF's student labs and will be modified with more ideal components as they are acquired. Finally, as the boost circuits are feeding into the same devices, they should be tested to insure that both circuits will operate correctly when tied to the same source. This will be done by constructing the circuits shown above and connecting them to the same input and once again testing their outputs separately.

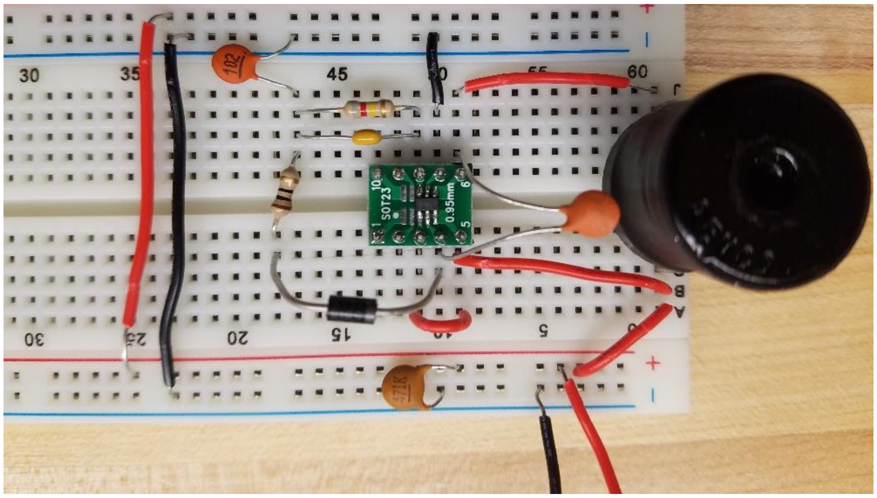


Figure 56. Prototype LT3483 Circuit

5.2 Analog Effects

Prototyping analog effects subsystems will largely consist of building the proposed selected designs on breadboards and using various tests to ensure that they are operating as possible. There may be small changes to component parts such as using mechanical potentiometers in lieu of digital potentiometers to simplify the testing process, in this case so that this test does not rely upon a microcontroller. Passive components which are planned to be surface mount devices in the final build such as resistors, capacitors and inductors where needed may be replaced with through hole versions as they will be much easier to prototype with as they will not require breakout boards to be able to use on a breadboard. This replacement will only be done if doing so will not change the relevant parameters of the device to be replaced. If any of the designs being tested do not perform as expected they will be rejected and new designs produced and tested until a satisfactory performance is achieved. It should be noted that the prototypes in the following sections are shown using the digital potentiometers so that these images may be used as reference material at a later date.

5.2.1 Tone Control Prototyping

The breadboard prototype in Figure 57 utilizes the selected digital potentiometers, TPL0501 on the blue breakout boards, and the selected Operational Amplifier, the OPA1611 on the green breakout board, to realize the selected tone control network. The unused header pins on the digital potentiometers belong to the Chip Select, SPI Input, SPI Clock, and low terminal. Because in this case these are being used as variable resistors, either the high or the low terminal may be left floating. In this case the low terminal is floating and the high terminal is one end of the effective resistance with the wiper being the other. In Figure 57 the yellow wire traces the signal path with the black wire belonging to the ground node, the green wire being the negative rail and the red wire being the positive rail of the Op-Amp and VDD for the potentiometers. Key parameters to test when prototyping this design will be the ability for the tone control network to differentiate a signal into three distinct frequency ranges and the ability for it to amplify or attenuate all three of these ranges.

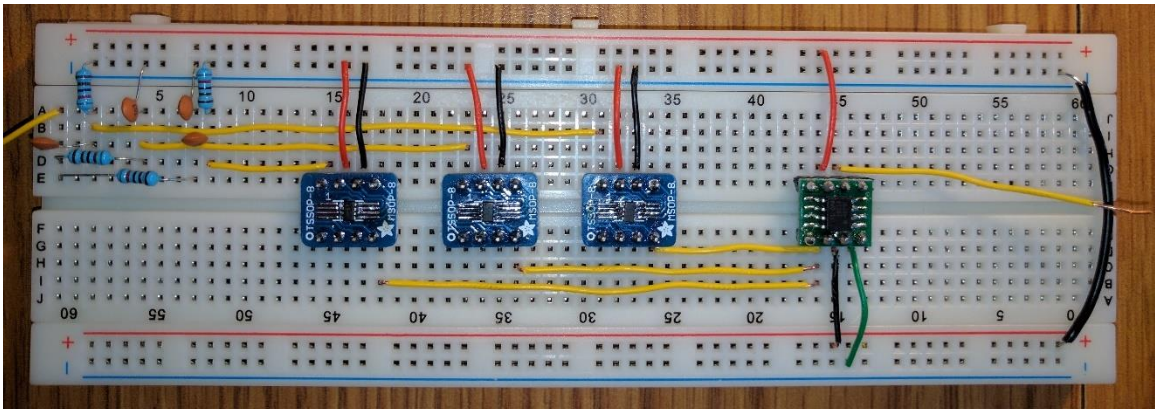


Figure 57. Tone Control Breadboard Prototype

5.2.2 Overdrive Prototyping

The breadboard prototype in Figure 58 shows the use of the selected Operational Amplifiers part number OPA1611 from Texas Instruments and the digital potentiometers part number TPL0501 also from Texas Instruments in order to construct the selected tone control schematic. In this figure the signal enters the network via the green wire on the left hand side and exits using the green wire attached to the second Op-Amp on the right hand side. The black wire denotes ground and occupies the outside rails on the top and bottom with the positive and negative sources for the Operational Amplifiers and potentiometers occupying the inner rails on the top and bottom respectively.

Key attributes that must be seen in the testing of this design are its ability to clip a signal, for the level of clipping to be changed as desired, and to limit the overall output to a safe voltage so that it will not damage later components.

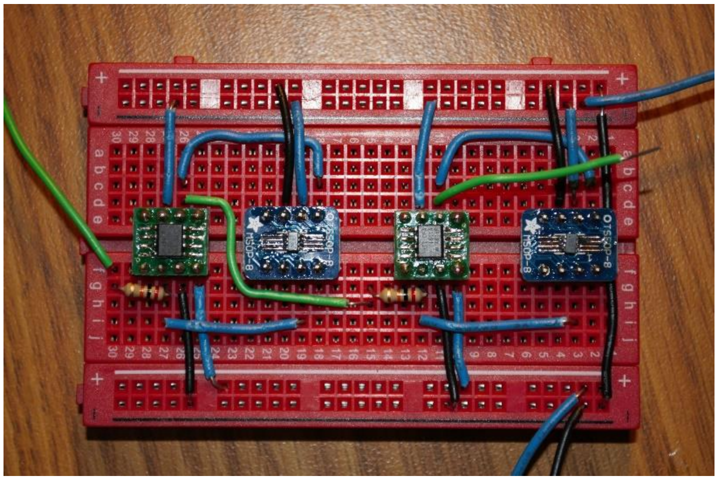


Figure 58. Overdrive Breadboard Prototype

5.3 Analog Filters

Prototyping the analog filters subsystems required by the Pocket Amp to ensure a clean signal will consist of building the proposed designs on breadboards and then sweeping them with a variety of frequencies to produce a magnitude response and then comparing that response to the expected response of the filter as designed. Again, some of the smaller components may be replaced with easier to use versions as long as they hold to the same device parameters and as such are unlikely to affect the overall performance of the filter. If any of the proposed designs do not perform as expected or rely upon tolerances too tight for practical components they will be redesigned and retested as required.

5.3.1 Input Filter Prototype

Figure 59 shows the use of two OPA1611 Operational Amplifiers from Texas Instruments to implement the selected input filtering network, the Fliege low pass notch filter. In this prototype the yellow wire on the left hand side is the overall input and the yellow wire on the right hand side is the overall output. The green node is the negative rail for the Operational Amplifiers and the red node is the positive rail. The black wires denote the ground node. Important figures of merit for this prototype are its ability to filter out unwanted frequencies under and including 60 Hz, the amount that it attenuates these frequencies, and its ability to pass wanted frequencies above 80 Hz and the amount that it attenuates these wanted frequencies.

5.4 Volume

Prototyping the volume subsystem of the Pocket Amp will be done using a breadboard and the selected operational amplifier on a breakout PCB so that it may be used with the breadboard. The resistor required will be replaced with a resistor or corresponding value and tolerances in a through hole package and the digital potentiometer will be replaced with a mechanical version with the same or approximately the same parameters until such

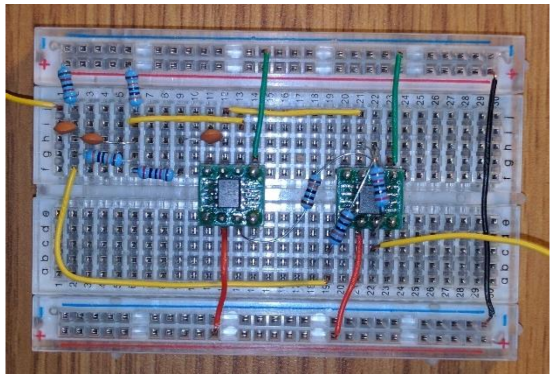


Figure 59. Input Filter Breadboard Prototype

A time as integration testing may occur to bring the microcontroller inputs necessary to control the digital potentiometers. The prototype in this final form is presented below in Figure 60 does include the digital potentiometers so that these images may be used as references at a later date.

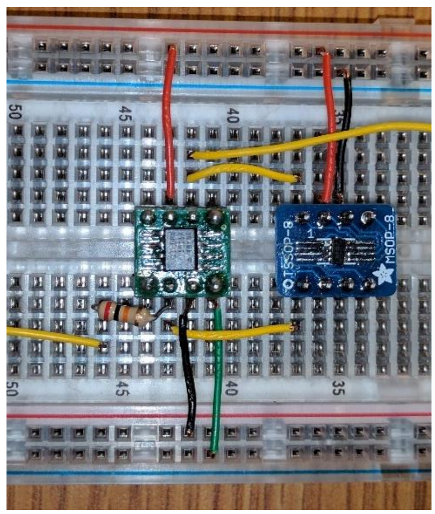


Figure 60. Volume Network Breadboard Prototype

5.5 MSP432

The first step to prototyping the MSP432 system is to make stub functions for all the relevant parts to the code. The stub functions simply simulate the actual functions that will be implemented on the MSP432, and are used to lay out the framework on the code. The first functions that were worked on to flesh out beyond stub functions are the I/O functions, namely I^{2}C for the DAC so we can at least have audio output for testing even if the MSP432 is simply passing it through.

TI has a large set of driver libraries that ships with TI's Code Composer Studio. Initially the engineers only had Code Composer Studio set up to interface with the MSP430; switching from developing for the proprietary 16-bit processor to the 32-bit ARM processor requires changing all dependencies, libraries and imports to the appropriate MSP432 versions. TI has a software suite called MSPWare, which supports both of their MSP processors by delivering libraries and compilers to Code Composer Studio, which reduced the number of independent downloads to get the IDE up to running the MSP432.

The driver library for the MSP432 is called driverlib, and contains device drivers for almost every feature of the MSP432, including ADC functions, I^{2}C, SPI, encryption and floating point operators. Our focus for the DAC will just be on the I^{2}C functionality. The primary I^{2}C feature used from the library is the master config object. It contains all the data used to initialize the protocol, and is show in Figure 61 below.

```
/* I2C Master Configuration Parameter */
const eUSCI_I2C_MasterConfig i2cConfig =
{
        EUSCI_B_I2C_CLOCKSOURCE_SMCLK,          // SMCLK Clock Source
        3000000,                                // SMCLK = 3MHz
        EUSCI_B_I2C_SET_DATA_RATE_100KBPS,      // Desired I2C Clock of 100khz
        0,                                      // No byte counter threshold
        EUSCI_B_I2C_NO_AUTO_STOP                // No Autostop
};
```

Figure 61. I^{2}C Configuration Code Snippet
Created by William McKenna

After creating the data structure for configuration, the program enters the main function that sets pins to the correct I/O settings and sets up the I^{2}C registers to hold the slave address (0xA0 for our DACs) and information then initialize the submaster clock to drive the I^{2}C bus when transmitting. The program then enters the infinite while-loop to transmit the data – here just being a repeated pattern of 0x0FFF then 0x0000 – to the DAC. This is supposed to drive the DAC to the highest voltage level then lowest, and from there the clock will be tuned to output samples at 44100Hz and be able to send notes from the DAC. The functionality of the DAC and I^{2}C has not been thoroughly tested at this time, as access to oscilloscopes is limited to when the team is on-campus.

5.6 App Prototyping

In order to build an app prototype, the official Android Studio tutorial (https://developer.android.com/training/index.html: "Building Your First App") to build an app was followed. From the tutorial's app, functionality was added until it contained all of the functionality that the Pocket Amp App would require. From this initial version, functionality will be incrementally added until the final version is reached which fulfills all the requirements for the app. The projected timeline of app versions is seen below in

Table 7.

The Android Tutorial's first chapter "Getting Started" deals with creating a basic UI, developing for the multitude of Android systems, and interacting with the system through permissions, data storage and messaging. A majority of our app will deal with front-end things and not have to worry about things like content sharing or location services. This makes following the Android Tutorial and excellent starting point and allows us to customize the app we are building while we work through the tutorial.

Version	Features Added	Target Date	Progress
v0.1	Initial app from Android Tutorial	12/1/16	Complete
v0.2	Add buttons and mockup pages	12/2/16	Complete
v0.3	Add setting controls for every effect	12/5/16	In Progress
v0.5	Add app background – store options for effects	12/10/16	In Progress
v0.8	Add basic Bluetooth connection	12/24/16	Not Started
v0.9	Communicate options over Bluetooth	12/31/16	Not Started
V1.0	Final release, improve look/feel and finalize effects and options	1/7/16	Not Started

Table 7. App Version Timeline
Created by William McKenna

Pocket Amp app version 0.1 was created by following the first segment in the first chapter of the tutorial. While the tutorial and tasks were straightforward, using Android Studio was less straightforward. System requirements for running a full-featured IDE while compiling, building and debugging are quite high, so also running an emulated phone with multiple cores and multiple Gb of RAM is out of the question on the developers' laptops. To combat this, the computer engineer used his phone, an LG G3, as a debugging environment. After setting the phone and computer up to connect via USB and allow the computer to build and modify apps on the fly on the phone, debugging become a much easier process. App version 0.1 features a main screen with an editable textbox and a button. When the button is pressed, the text box's contents are sent to another page, which opens and displays the contents. This is a very simple app, but it will serve as the backbone of our Pocket Amp app since a large part of our frontend will be passing data between options on separate pages. This simple app can be seen in the screenshots in Figure 62.

At this time, the next stage of development is to get rid of the button and text box, refactor all the code so it is more relevant to our project, and add in the buttons and some background code for the option pages. Both the pages – called activities in Android – were given more appropriate names, the name of the project and app were changed to reflect the Pocket Amp app project, and the internal functions were all refactored. The standard linear view was changed to allow scrolling and all the sizes, padding, and offsets were made to be relative sizes in dp. “dp” are dimensionless pixels, Android’s way of making everything similar sized on devices with different pixel densities. Enough buttons were added to cover all the effects, and a save/load, and a sync button for when synchronizing with the Pocket Amp is desired. On the LG G3, the menu stays mostly on the screen and scrolling is not needed but on other devices with more normal-sized or smaller screens scrolling will be essential.

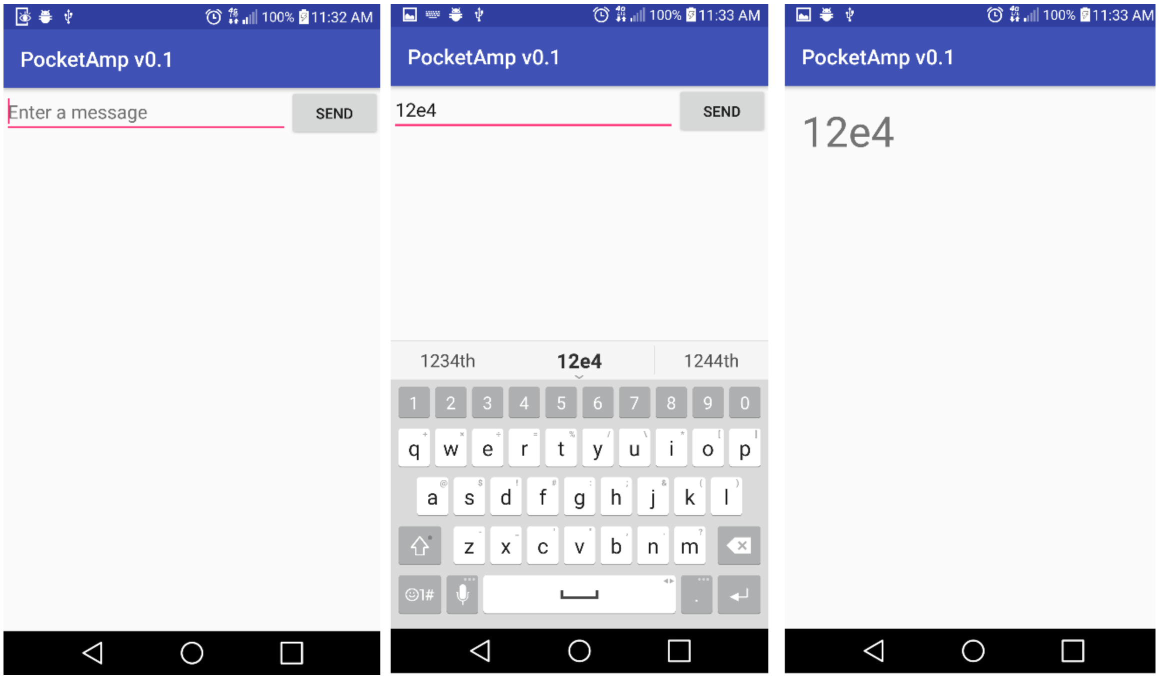


Figure 62. Pocket Amp App v0.1
Created by William McKenna with Android Studio

At this point, research was done into Android platform compatibility. The initial setup had an API level of 15, which covers every phone from Android 4.0.3 designated “Ice Cream Sandwich” through modern version such as “Marshmallow” or “Nougat.” While there are still many devices running Ice Cream Sandwich, a majority of Android devices have moved on to much higher API levels, and with that comes much more functionality for developers. After reviewing what functionality might be required, an API level of 17 was decided upon. Android 4.2 Jelly Bean featured a rework of the user interface system which brought many more functions for designing themes and for interacting with the user, so targeting this platform and above was deemed ideal.

The next step was to add in buttons and have a way for them all to call an option window and let the window know what button was pressed. Instead of having a different window for every single option, it was determined that for such a simple UI it would be best to have one option window be customized based on the option that called it. This is done by every button passing the name of the function it is performing to the option window and letting the option window build itself based on what name it received. The basic handoff of information was accomplished and some of the java-generated option windows were created, which wrapped up version 0.2 of the Pocket Amp app. The main menu and the equalizer option screen can be seen in Figure 63.

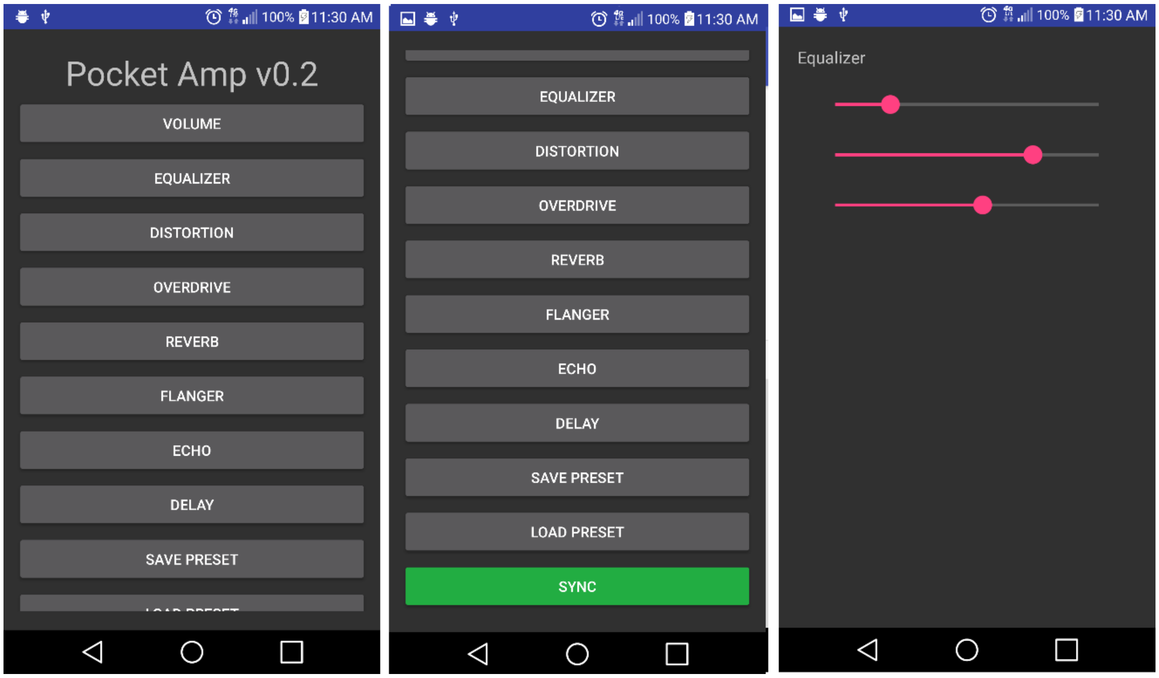


Figure 63. Pocket Amp app v0.2
Created by William McKenna with Android Studio

6. Test Plans

Just as designing was divided into software and hardware, testing will be divided into software and hardware testing. In addition to these divisions, we will also have to test the integration between systems – both hardware and software – and test against the requirements of the Pocket Amp.

6.1 Software Unit Testing

Software unit testing is done to verify the individual software components can function by themselves and are correct individually. Since unit testing is such a common practice in the software development world many of the software suites used have some unit testing functionality built in, or there are libraries which automate unit testing.

6.1.1 UI Testing

For user interface unit testing, each function will be tested for validity in input and output. The testing will be done both in Android Studio debugger for checking output of each function and on multiple Android devices to ensure the validity of the app across multiple setups. The functions of the app will be broken into three groups for testing: app background functions, amplifier control functions, and communication functions.

App background functions cover things such as saving data when the app is closed, loading data when it is open, storing presets, and managing any permissions the app will need. The functions of these will be tested in Android Studio by checking save states and load states for multiple amplifier setups. The permissions will be tested by installing on multiple devices and ensuring that it requests Bluetooth and any other permissions that are required for the software to run.

Amplifier control functions will be tested for only allowing valid inputs and for correctly updating the internal state of the application. Input validation will be made easier by reducing the user input to only allow valid values, such as using check boxes, radio buttons, and sliders. The internal state will be checked by Android Studio debugging tools to ensure that the options are being set and stored internally.

Communication functions will be tested for reading the internal state when the user opts to change the amplifier set up in addition to setting up and maintaining a Bluetooth low energy connection with the Pocket Amp. Communication will be tested mainly through Android Debug and through testing with the MSP432 and Bluetooth device.

6.2.1 Digital Effects Testing

Digital effects testing will be done by simulating the input of a perfect note, such as exactly 440 Hz which is the musical note A in the octave above middle C. This is a very common tuning note for musicians, the tone will be produced through the use of an AC function generator and output of the DAC will be measured with an oscilloscope. The digital effects are all variations on time-delay functions and as such the waveforms will be easy to visually verify that an effect is being applied properly. If this is not satisfactory, debugging tools can be used on the MSP432 Launchpad to step through execution. Since stepping through execution is not done in real-time, the input would have to be simulated since the ADC would not be sampling the function generator at the proper moments. This can be done by writing test functions to generate a perfect wave, which can easily be done by storing the data structures from function generator testing.

6.1.2 MSP432 Output Testing

The MSP432 module must be tested for correctly receiving input instructions, outputting the appropriate signals to components, receiving input analog signals, and outputting audio signals after applying the appropriate transformation functions. MSP432 testing will be done discretely from the Bluetooth module and components. Any analog inputs will be created by a function generator and any analog outputs will be measured by an oscilloscope. Any digital inputs will be created by MSP432 debug tools and digital outputs will be measured by debug tools or visually monitored by LEDs.

6.1.3 Bluetooth Testing

The Bluetooth module testing will be done by connecting and manually sending packets to from Android Studio Debug tools and reading and storing outputs with the MSP432. From the MSP432 we will ensure the packets or attribute changes have arrived and are in the correct format to update MSP432 settings. Due to the fairly unidirectional nature of the communication it is only necessary to ensure that the app communicates with the Pocket Amp – as long as the Pocket Amp is able to effectively set up and terminate Bluetooth connections that is all the communication back that we need from the Pocket Amp.

6.2 Hardware Unit Testing

Hardware unit testing is done to ensure the hardware subsystems themselves are within spec. Integrating faulty hardware together is far less costly than integrating thoroughly tested hardware together and can be far safer.

6.2.1 Power Testing

Like all components in the Pocket Amp, each component of the power supply will need testing to ensure that each component operates sufficiently in order for the Pocket Amp to operate. This testing is conducted component by components using ideal sources before merging components together to form combined testing.

6.2.1.1 Battery Testing

The battery and its charging network is the first and perhaps easiest section to test, as it uses predesigned components. This network requires two forms of testing, testing output and testing charging conditions. For the output testing the battery will be connected to the charging circuit using the available JST connectors. The JST wires from the load side of the charging network will then be connected with a 1kΩ resistor. The voltage and current for the resistor will be measured using a sensitive digital multi-meter. The voltage should be equal to around 4.2 to 3.3V depending on how long the battery has been in storage since its purchase. The current should thus equal to 4.2mA to 3.3mA depending upon the present charge of the battery. Next, the battery will be connected to a resistor for a time in order to drain some of the charge within the battery, a drop in voltage from 4.2V will indicate enough charge has been spent. Once some of the charge has been used, the charging network can be removed from the resistor. Next, a simple 5V USB charger will be connected to the charging circuit and the respective LEDs of the charging network should light to indicate the battery is charging. Once the charging network indicates a full charge, the battery's voltage will be tested, which will equal to 4.2V for a full charge.

6.2.1.2 Step-Down Regulator Testing

The first component to undergo testing will be the step-down regulator the LP2989. The first step for the LP2989 is to solder the IC to a breakout board so that it can be placed onto a breadboard for easier testing. Once the breakout board is completed, each passive component will be placed to form the circuit shown in Figure 34. With this circuit completed, it will be connected to a 4.2V input created by a dedicated voltage supply. The load side of the circuit will be connected with a 1kΩ resistor and the resistor's voltage and current will be measured. The voltage should be equal to 3.3V and the current equal to 3.3mA. The input voltage will then be slowly lowered in increments to ensure that the output voltage continues to be well regulated. If a heat monitor is available, then the heat dissipation of each component will be recorded so that they can be compensated for when on the PCB. Next, still utilizing the dedicated power supply, the output of the regulator will be connected to the input of the digital components to ensure that they can be powered from the regulator and that the regulator can handle the current draw provided by the digital components.

6.2.1.3 Boost Converter

The next important component to test is the boost converter. Like the other IC devices, placing the boost converter on a breakout board for testing is ideal. Once on a breakout board the circuit shown in Figure 38 can be built around the converter on a breadboard. A voltage supplier will be connected to the input at 4.2V and the output will be connected to a 1kΩ resistor. The voltage and current for the resistor will be tested with a digital multimeter, the voltage should be 8V and the current should be equal 8mA. The input voltage will slowly be lowered from there to test if the boost converter consistently boosts the voltage up to 8V.

If a heat monitor is available, it will be used to measure the heat dissipation for the circuit, which will be recorded to use on the PCB. This next step will be done concurrently with the inverting boost converter. The output of the circuit will be connected to the positive rail of one of the op-amps. A test signal will then be fed to the op-amp circuit to ensure that the op-amp operates normally when being supplied by the boost and inverting boost converter.

6.2.1.4 Inverting Converter

The inverting converter will also require proper testing to ensure that it operates correctly. The first step in testing will be to place the inverting converter on a breakout board for breadboard testing. The circuit from figure 41 will be constructed on the breadboard. The input will be provided with a dedicated voltage supplier set at 4.2V and the output will be set with a 1kΩ resistor. A digital multimeter will be used to record the resistor's voltage and current, which should be -8V and -8mA respectively. The input voltage will then be reduced incrementally to ensure that the inverting converter produces a consistent output voltage of -8V. Ideally, the different heat dissipation for each component will be measured if a heat monitor is available. This final step will be done concurrently with the boost converter. The output of the inverting converter will be supplied to a negative rail of one of the op-amp circuits. A sample signal will then be inputted into the op-amp circuit to ensure that the op-amps function properly when powered from the power supply.

6.2.1.5 Combined Testing

The final step of testing the power supply is to ensure that each of the DC/DC converters do not interfere with one another. This will be tested by constructing the circuits from figures 34, 38 and 41each on the same breadboard. The output of each circuit will be connected to a different resistor corresponding to their current draw. For example, the inverting boost and boost converters will be connected to a 320Ω resistor to draw approximately 25mA. The input of each circuit will all be fed from the same dedicated voltage supply at 4.2V. As with the individual testing, the voltage level of the supply will slowly be reduced, measuring the output voltages and currents. Once the testing has been completed with the dedicated voltage supply, it will be swapped with the battery and charging circuit. The output voltage and currents will be measured again across the various resistors. The heat dissipation of the circuit will be measured if a heat monitor is available. Assuming the output voltages are correct, the design will pass testing and be ready for PCB implementation.

6.2.2 Analog Effects Testing

This section will explore the methods that will be used to test the prototype devices to ensure that they perform to expectations and requirements and are stable devices that will perform well over their operating range.

6.2.2.1 Tone Control Testing

The tone control subsystem will be tested by first applying frequencies throughout the three frequency bands that the tone control circuit will control, bass, midfrequencies and treble, with varying amounts of attenuation and amplification selected for the tone control circuit with the amplitude and frequency of the input and output being measured on an oscilloscope as well as using a spectrum analyzer to inspect the amount of energy in each of the frequency bands. If the results of this simple test are positive a more complicated signal such as a sum of sinusoidal waves will be produced to use as a simulated input for the tone control network and the output again observed using an oscilloscope to ensure that the portions of that signal that are to be attenuated or amplified based upon the value of the potentiometers are being modified by an acceptable amount. Finally, if these tests are met with success a similar process will be used to observe the output when a signal from a guitar, or a recording of a guitar at least, is used as an input.

6.2.2.2 Overdrive Testing

To test the overdrive effect a simple sinusoid will be used at first as an input. The resulting output in time domain will be observed using an oscilloscope to ensure that the resulting waveform is clipped by the amount set by the value selected on the potentiometers and the amplitude of the output wave is within reasonable limits without being too high such as would damage the following components or so low that noise will be easily introduced. Assuming that this test is passed the subsystem will once again be subjected to a more rigorous test of summed sinusoids to ensure that the proper output is once again observed. Finally, a natural or recorded guitar input will be used to ensure that the Overdrive subsystem can withstand the frequencies generated by a true instrument and its imperfect output.

6.2.2 Filters Testing

Each filter used in the design of the Pocket Amp will be tested in relatively the same way. Firstly, it will be subjected to a range of frequencies and its amplitude response measured using an oscilloscope and compared to the expected response of that filter obtained through computer simulations. If the expected and experimental responses align it will then take a wave containing multiple frequencies such as an impulse, square or triangular wave and, using a spectrum analyzer, its output spectrum will be observed to ensure that it is capable of rejecting the portions of the overall spectrum that are associated with noise or any other unwanted energies. Finally, it will be exposed to a natural or recorded guitar signal as an input to ensure that it only passes the parts of that signal that it is configured to while passing those parts with relatively little attenuation.

6.2.3 Volume Testing

The volume subsystem will be tested not so much for design function, as it is a simple and well tested design, although it will undergo preliminary functional testing to ensure it was built correctly. More importantly it will be used to test the volume that the amplifier is able to produce for differing output voltages and currents in a variety of headphones to ensure that the Pocket Amp will be able to be used with a wide range of consumer headphones. This test will be partly objective and partly subjective. The volume subsystem will take a sinusoid at audible frequency and relatively low voltage and its output power will be measured at all available amplification levels starting with the lowest. The power that it produces will be measured by either an ammeter and a voltmeter or by a wattmeter if available. This is the objective part of the test. The subjective portion of the test is to allow the headphones to be worn and the volume to be given a value on a purely arbitrary 0 to 100 scale that roughly corresponds to that of one of the designer's laptops with the same headphones. By using these two values a table will be created from which recommended values for headphone impedance can be derived to ensure that they meet the specifications of the documentation of the Pocket Amp.

6.3 Integration Testing

Integration testing is the phase after unit testing where the individual subsystems are tested together to ensure they will interact correctly in a predictable and verifiable way.

6.3.1 App and Bluetooth

Integration testing between the Pocket Amp phone app and the Bluetooth system will be done after the unit testing for both has been accomplished. The integration testing will consist of sending Bluetooth packets from the app by changing the settings for the Pocket Amp and initiating connections. The reception of the packets will be verified by monitoring the output of the nRF51822 to verify the packet has been processed and is being transmitted to the MSP432. The app will also be used to verify that the Pocket Amp Bluetooth can connect to the phone; since there is no communication back from the Pocket Amp through Bluetooth there is no need to force the nRF51822 to send packets back to the app.

6.3.2 Bluetooth and MSP432

Testing to ensure that the MSP432 and Bluetooth can communicate will be done by either simulating output from the nRF51822 by directly coercing it to send information through its SPI connection with the MSP432, or – if app and Bluetooth integration (see Section 6.3.1) has already been completed – then Bluetooth and MSP432 integration can be tested by sending packets through the app to ensure they are arriving at the MSP432. Because of the unidirectional flow of data, it is apparent that integrating and verifying the app and Bluetooth systems together before the Bluetooth and MSP432 is ideal.

6.3.3 MSP432 and Hardware

Integrating and verifying the MSP432 will be done by verifying the outputs of the MSP432 are interacting with the hardware in the correct way and the MSP432 is correctly receiving the analog signals from the hardware. Verification of all communication from the MSP432 is key; all SPI interfaces and any logical I/O or multiplexed hardware selection must be verified with each possible intended effect configuration. To test the input from the hardware to the MSP432, a function generator will be used to create a standard function, such as 440Hz at 100mV, which is the musical note “a” above the “middle c” in music theory played at a standard voltage seen from a guitar. To this function we will monitor the frequency received by the MSP432 and verify the waveform matches the appropriate waveform from each effect applied.

6.3.4 Hardware and I/O

The final integration test is the integration between all hardware and the musical input and output. Namely, the guitar input will be tested with an oscilloscope to ensure that the first hardware elements are receiving the guitar input correctly. The musical output will be tested by creating a program for the MSP432 that generates a standard wave form in the ADC data structure. This function will simulate a single note being input from the MSP432 into the DAC and being output to the headphone jack. The accuracy of the output will be tested by generating a standard wave function, such as using the 440Hz “a above middle c” note discussed in Section 6.3.3. This will be tested with a digital tuner to ensure that the note is accurately being output by the DAC and MSP432.

6.3.5 Full Validation Testing

Full system validation testing will be performed by ensuring that a well-tuned guitar’s input into the system comes out as a well-tuned output from the system and that effects applied sound appropriate. Since musical features are largely subjective, the entire team will be asked to verify that the sounds are appropriate to the effect and that the music is pleasing. Due to the team’s limited musical ability, outside guitarists will be consulted for verification that the system is producing correct results. This verification will be done with the entire system functioning, from app to guitar input to headphone output.

6.4 Requirement Fulfillment Testing

In order to ensure that our product meets the requirements set forth in this document the end prototype will be tested by a method known as requirement driven testing. A set of test cases is created based on the requirements of the design and the prototype will be tested against this set, in addition to unit testing and integration testing to verify individual parts. The requirement driven testing creates at least one test per requirement with each test directly relating to fulfilling a requirement. The requirement is listed below with the requirement number and the description, then the test case is listed. The test case shows how the test will be performed to demonstrate that the Pocket Amp prototype meets all the requirements.

Requirement: E.1 1-hour battery life
Test Case: The Pocket Amp shall be charged to maximum battery capacity then discharged through standard guitar playing at a tolerable but sufficiently loud volume with at least 2 effects applied. The Pocket Amp will last more than 1 hour of playing, for at least 3 test runs.

Requirement: E.2 Fit in a 5x6x3 inch pocket
Test Case: The Pocket Amp without any headphones, charging cords or audio output cords plugged in will be measured by the engineers and will be smaller than 5 inches by 6 inches by 3 inches measured externally.

Requirement: E.2 Fit in a 5x6x3 inch pocket
Test Case: The Pocket Amp without any headphones, charging cords or audio output cords plugged in will be measured by the engineers and will be smaller than 5 inches by 6 inches by 3 inches measured externally.

Requirement: E.3 Weigh less than 5lb
Test Case: The Pocket Amp with all components will be weighed by the engineers and will weigh less than 5lb.

Requirement: E.4 Have a 6.35mm “phone jack” audio input jack
Test Case: The engineers will verify the Pocket Amp design and prototype includes a 6.35mm audio input jack to accept audio from the guitar.

Requirement: E.5 Have a 3.5mm “mini phone jack” audio output jack
Test Case: The engineers will verify the Pocket Amp design and prototype includes a 3.5mm audio output jack to output audio to the headphones.

Requirement: E.6 Will NOT have an onboard speaker
Test Case: The Pocket Amp design and prototype will not at any time include a device to emit sound built in to the Pocket Amp.

Requirement: E.7 Will have an external charging port
Test Case: The engineers will verify the Pocket Amp prototype includes a charging port capable of using a plug to charge the internal battery of the prototype. This will be done by inserting a plug and fully charging the battery of the prototype.

Requirement: I.1 Power a pair of headphones
Test Case: The Pocket Amp will be tested with a high-quality consumer-grade pair of headphones to ensure that volume is high enough that it is audible and without noticeable distortion. The test will be conducted on at least 3 people with average hearing to ensure satisfactory volume levels.

Requirement: I.1 Power a pair of headphones
Test Case: The Pocket Amp will be tested with a standard quality headphones. This test will ensure that the Pocket Amp can power the average user's headphones and will be conducted by testing at least two pairs of common headphones with at least two users to ensure the volume level is acceptable.

Requirement: I.2 Use an onboard MSP432 MCU to handle digital signal processing
Test Case: The engineers will verify the design and prototype include an MSP432 to process the digital audio signals from the hardware.

Requirement: I.3 Use MSP432 built-in ADC
Test Case: The engineers will verify that the design is using the MSP432's ADC for converting the analog signal to the digital signal within the MSP432.

Requirement: I.4 Amplify the standard range of guitar and bass guitar sounds
Test Case: The engineers will test that a range of high frequencies from guitars is able to be processed by the Pocket Amp prototype without unwanted distortion of the signal or lost harmonics. This will be verified by playing the highest harmonics on a guitar and having at least two engineers verify that the sound is acceptable.

Requirement: I.4 Amplify the standard range of guitar and bass guitar sounds
Test Case: The engineers will test that a range of low frequencies from bass guitars is able to be processed by the Pocket Amp prototype without unwanted distortion of the signal. This will be verified by playing the lowest notes on a bass guitar and having at least two engineers verify that the sound is acceptable.

Requirement: I.5 Sample at least 12-bits with a frequency of at least 30 kHz
Test Case: The computer engineer will verify that the MSP432 operating system is set up to use the onboard ADC with at least 12-bits of depth in all functions.

Requirement: I.5 Sample at least 12-bits with a frequency of at least 30 kHz
Test Case: The computer engineer will verify that the MSP432 operating system is sampling the analog signal with a frequency greater than 30 kHz on average for the highest latency setup of digital effects.

Requirement: I.6 Introduce less than 1% total harmonic distortion
Test Case: The electrical engineers will verify that the analog system is not introducing more than 1% total harmonic distortion by removing digital effects and inputting at least 3 frequencies into the audio in and measuring the harmonic distortion on the audio output.

Requirement: F.1 Implement digital delay effects
Test Case: The engineers will set up the Pocket Amp to apply a delay effect and verify that playing music with the delay sounds similar to a standard guitar delay effect and does not introduce any unwanted distortion.

Requirement: F.2 Implement digital reverb effects
Test Case: The engineers will set up the Pocket Amp to apply a reverb effect and verify that playing music with the reverb effect sounds similar to a standard reverb effect and does not introduce any unwanted distortion.

Requirement: F.2 Implement digital reverb effects
Test Case: The engineers will set up the Pocket Amp to apply a reverb effect and verify that the effect gives the impression of sound in a large space by having at least two engineers compare the effect to recordings from a concert hall or other large space.

Requirement: F.3 Implement digital echo effects
Test Case: The engineers will set up the Pocket Amp to apply an echo effect and verify that the echo effect sounds like a stereotypical echo of the input audio.

Requirement: F.4 Implement digital flanger effects
Test Case: The engineers will set up the Pocket Amp to apply the flanger effect and verify that playing music with the flanger effect sounds similar to a standard guitar flanger effect.

Requirement: F.5 Implement digital chorus effects
Test Case: The engineers will set up the Pocket Amp to apply a chorus effect and verify that the playing music with the chorus effect sounds like a standard guitar chorus effect in that it sounds like it imitates more than one instrument playing the same music.

Requirement: F.6 Implement analog distortion effects
Test Case: The engineers will set up the Pocket Amp to apply the distortion effect and verify that it sounds like a standard guitar distortion effect.

Requirement: F.6 Implement analog distortion effects
Test Case: The electrical engineers will set up the Pocket Amp to apply the distortion effect and input a signal from a function generator and observe the waveform on an oscilloscope to verify the signal is properly distorted.

Requirement: F.7 Implement analog overdrive effects
Test Case: The engineers will set up the Pocket Amp to apply the overdrive effect and verify that it sounds like a standard guitar overdrive effect.

Requirement: F.7 Implement analog overdrive effects
Test Case: The electrical engineers will set up the Pocket Amp to apply the overdrive effect and input a signal from a function generator and observe the waveform on an oscilloscope to verify the signal is properly distorted.

Requirement: F.8 Implement analog tone control
Test Case: The engineers will test at least 3 different settings of the analog tone control to verify the tone control is affecting the correct parts of the audio spectrum.

Requirement: F.9 Implement analog volume control
Test Case: The engineers will test at least 2 different volume levels to ensure that the volume control changes the audio output of the Pocket Amp.

Requirement: U.1 Communicate through Bluetooth with smartphone application
Test Case: The engineers will connect at least 3 times to verify the ability to connect the Pocket Amp to its companion app with Bluetooth.

Requirement: U.2 App will be developed for Android devices
Test Case: The computer engineer will verify the app is built for Android devices.

Requirement: U.3 App will control effects
Test Case: The engineers will verify the app can send commands to the Pocket Amp by changing settings on the Pocket Amp at least three times.

Requirement: U.4 App will control volume
Test Case: The engineers will test the volume controls with the app by changing the volume of the audio output at least 2 times with the app.

Requirement: A.1 Unit production hardware cost will be less than $100
Test Case: The engineers will assemble a budget containing estimates for all parts for a production run of 1,000 Pocket Amps and verify that the total is less than $100 dollars per Pocket Amp.

Requirement: A.2 Unit prototype will be finished by the end of Spring 2017
Test Case: The engineers will finish the prototype by the end of the spring semester at UCF in order to present it to the senior design committee.

7. Standards

Every design has a wide array of applicable standards. The main standards affecting the Pocket Amp are Bluetooth standards developed by the Bluetooth Special Interest Group, the Federal Communication Commission's standards and the International Special Committee on Radio Interference's standards on electromagnetic interference and electromagnetic compatibility, and JEDSEC's nominal 3 V/3.3 V supply digital integrated circuits standards.

7.1 Bluetooth

Bluetooth is a wireless communication standard used for personal area networks and very small local area networks. Bluetooth communicates in the commonly used 2.4 GHz range for unlicensed Industrial, Scientific, and Medical band of frequencies. Bluetooth was designed for low-power applications, which was solidified by the branching off of "Bluetooth with Low Energy" from what became "Bluetooth Classic" or officially "Bluetooth Basic Rate/Enhanced Data Rate." Bluetooth Classic is designed for more traditional usage where a continuous connection is formed and maintained during the entire communication. Bluetooth Low Energy relies on short, ephemeral burst connections that can travel much longer ranges than Bluetooth Classic.

Both Bluetooth standards follow the same stack structure not entirely unlike the TCP/IP stack. The Bluetooth stack is broken into 3 main layers with 10 smaller layers within the main layers. The three main layers from highest level to lowest level are the application layer, the host layer, and the controller layer. Between Bluetooth Classic and Bluetooth Low Energy the 10 sub-layers perform similar operations, but have some changes to their implementations to support the different use cases of Bluetooth Classic and Bluetooth Low Energy. In addition to the standard controller layers for Bluetooth Classic and Bluetooth Low Energy, the Bluetooth standard allows for secondary controllers called Alternative MAC/PHY (AMP) controllers which replace the standard Bluetooth controllers with a different physical layer, allowing for a Bluetooth handshake to check for compatible AMP controllers then switching to the appropriate AMP physical layer to continue the communication.

The Bluetooth Classic physical layer consists of 2.4 GHz, frequency modulated, frequency hopping, time slotted, shaped binary wave transmitting 1 million symbols per second. The 2.4GHz band is divided into 79 frequencies of 1 MHz. Bluetooth Classic defines one device as the host and the rest as slaves, where the host dictates the clock signal. The clock signal is used for time slots and as a seed (along with each device's address) for the pseudo-random hopping of the slaves among the 79 frequency slots. The host can also exclude frequency slots from the hopping, which avoids interfering with static users of the 2.4 GHz band. Bluetooth Basic Rate transmits 1 bit per symbol, giving and effective data rate of up to 1 Megabit/second while Bluetooth Enhanced Data Rate allows either 2 or 3 bits per symbol for 2 or 3 Megabit/second.

Bluetooth Low Energy follows a similar scheme as Bluetooth Classic, with implementations for both Frequency Division Multiple Access (FDMA) and Time Division Multiple Access (TDMA) schemes for contention management. FDMA subdivides the frequency domain into 2 MHz segments and uses 3 for advertising and 37 for data transmission channels. TDMA divides each of the FDMA channels into time units called events. All transmissions occur within these events in data packets. There are two transmission types, relating to the two types of frequency channels: advertising and connection. Advertising is essentially a broadcast to every device on the channel and is used both for creating connections and for one-time unidirectional data broadcasting. Another device can then respond on the advertising channel or request for the conversation to move to a dedicated connection channel for true communication. When a device advertises a connectable packet, another device can initiate communication and the advertiser becomes the initiator's slave device. During the connection, packets are swapped between the two then the connection is immediately ended. This scheme of low-power advertising allows devices to lay dormant, waiting for a central device to advertise and essentially wake up the peripheral device.

A sub-layer above the physical layer is the link layer. The Bluetooth link layer – like the HTTP link layer – creates at least one logical link between the two Bluetooth controller layers. This logical layer serves as an end-to-end connection in the piconet allowing asynchronous communication between higher layers, which may or may not have data ready to send or be able to receive at all times. The logical layer also supports the physical layer by creating and reading physical layer instructions, such as packet structuring and discovery/detection or advertising/connecting protocol. Finally, the logical layer is in charge of multiplexing together the potentially multiple logical links created for the higher layers of Bluetooth.

The link layer has a partner layer at the same level that is used for debugging called direct test mode. Direct test mode is accessed either through debugging UART connections or through commands from the layer above it, the Host Controller Interface, and is used to force the Bluetooth controller to send or receive a packet. This is used for component testing of the Bluetooth physical layer radio or of the host and other upper layers. This can also be used for integration testing between two Bluetooth devices to isolate issues to a host or the physical layer.

The Host Controller Interface is not necessary for Bluetooth operation, but is a sub-layer that crosses the layer gap between host and controller. The layer gap between host and controller in some implementations is more than an arbitrary division or software division, in some implementations the controller or controllers can be physically separated on a printed circuit board. The HCI allows for a UART interfaces between physically discrete host and controllers. The HCI also allows for each host to have multiple controllers even if they are physically integrated together. This setup is used in dual-mode implementations where a single Bluetooth host has both a Bluetooth Classic and a Bluetooth Low Energy controller. The HCI also communicates with each AMP, which internally has a Protocol Adaptation Layer (PAL) to translate between Bluetooth HCI instructions into the lower MAC and PHY layers.

The Bluetooth Logical Link Control and Adaptation Protocol (L2CAP) consists of two managers, that communicate with upper layers and the HCI. The two managers consist of a channel manager and a resource manager. The channel manager manages the channels for communication available to the HCI and upper layers. The resource manager performs segmentation of data when data comes from the lower layers, reassembling any fragmented packets into their correct order. It also manages retransmission and flow control to increase reliability at this point by retransmitting any dropped packets and slowing down data flow if there are dropped packets. It also communicates with the channel manager to synchronize the packet scheduling. The L2CAP also contains controls for a streaming mode, where having a flow of data stop or be flow controlled is not acceptable, and instead the two devices will first agree upon a data rate.

The attribute protocol (ATT) provides a client/server style communication for devices that have established a connection. The ATT is not a necessary part of the Bluetooth stack, but does provide valuable services to devices and as such is required by many optional higher sublayers. To the server, ATT provides a structure to store attributes in, with each attribute holding at least a type defined by a Universally Unique IDentifier (UUID), a handle, and a set of permissions that are set by higher levels and not directly accessible by the Attribute Protocol. To the client, the ATT provides a framework for making requests from the server for attributes – such as reading or writing attributes – to which the server must somehow respond. What's special about Bluetooth's ATT is that it allows a device to both be a client and server at the same time; the ATT even allows two devices to connect to each other and both be clients and both be servers to each other. Though a device can have both a client and server running, it shall only have one server running and can provide different services. This allows all Bluetooth servers to maintain the same range of attribute UUIDs without collisions for devices running multiple servers. A server can also notify or indicate to a client which will tell the client of an attribute or attribute change without the client needing to constantly polling attributes for changes.

In addition to the ATT, there is a sublayer at the same level within the Bluetooth stack called the Security Manager which, as the name implies, manages the security of the connection between Bluetooth devices. Historically Bluetooth was inherently insecure. Encryption was either not implemented in some cases, or not enforced in most cases; devices could request a non-encrypted link and the other device would have to comply with it. This lead to malicious attacks such as Bluetooth worms, Bluetooth snooping, Bluetooth attacks on computer input devices, and less-malicious attacks such as Bluejacking (connecting to an insecure connection and showing a message that they've been Bluejacked, then disconnecting with no malicious payload being delivered). With specifications following Bluetooth 2.1 all connections enforce authorization, authentication, and encryption for each physical link.

The Security Manager performs most of the security actions within the Bluetooth stack. Authorization enforces the data integrity of the ATT, and is stored in the permissions that the ATT cannot access. The ATT not being able to directly access permission attributes creates an internal DMZ (a security separation) within the Bluetooth stack. Insufficient authorization will throw an error back to the client stating the issue. The Security Manager also manages authentication of devices. If a device issues a request to the server on a connection that isn't authenticated, the Security Manager will make the server respond with an error and the client must initiate authentication with the server before continuing communication. The Security Manager's errors are the highest priority errors of this level; if the server could return any other errors then information would be leaked about memory usage and data storage on the server.

The Generic Attribute Profile or GATT is a higher sublayer that uses the ATT to provide services to the higher layers. It is not a necessary sublayer; generally only Bluetooth Low Energy uses GATTs and Bluetooth Classic just uses the lower layers and a profile to allow applications to interface the lower layers. The GATT encapsulates the client/server relationship of the ATT and provides interfaces to the higher layers that show the attributes in a style provided by the higher layer, the Generic Access Profile (GAP). The GATT is just an intermediary between the ATT and the GAPs; the GATT takes care of all the client/server tasks for the profile.

The GAP is used almost exclusively in Bluetooth Low Energy and outlines the functions directly to applications in the highest layer – the Application Layer. The GAP provides functions for the advertise and connect protocols used in Bluetooth Low Energy. Is less of a built-in layer and more of a protocol for the application to conform to. A software Bluetooth stack implementing GAP is guaranteed to be able to access certain functions of Bluetooth.

The application layer is the highest layer and sublayer in the stack. The application layer is part of the actual phone, computer or peripheral's software using Bluetooth. Generally, an app or firmware developer will only interact with the highest layer of the stack and develop the application layer. Bluetooth stacks are well-implemented and are usually packaged into driver form for application developers to use.

7.2 Electromagnetic Compatibility and Interference

Electromagnetic compatibility (EMC) is the state of a device being electromagnetically compatible with its environment. This includes the characteristic that the device will not cause any electromagnetic interference in any surrounding electronic circuits. Electromagnetic compatibility standards follow two overarching issues to keep the balance in the environment and correct improper operation of any system that may cause a disturbance. The first issue that electromagnetic compatibility is concerned with is emissions. Emissions of electromagnetic energy can be deliberate or inadvertent, however regardless of its intention emissions can cause electromagnetic interference. The second issue that electromagnetic compatibility is concerned with is susceptibility. The susceptibility of a system is the tendency or likelihood that it can be influenced by the presence of unwanted electromagnetic energy. The susceptibility of a system correlates with the system's ability to not respond, or have an insignificant response, to electromagnetic energy. EMC's intention is to establish a safe electromagnetic environment for the operation of various electronic devices that can be used simultaneously.

Electromagnetic interference (EMI) is a disturbance that is created by an external source that affects an electrical circuit. EMI can be either man-made or naturally occurring. The disruption can have a varying level of degradation on the affected circuit, the circuit could be partially interrupted or even completely stopped from functioning with enough interference. EMI is caused by electromagnetic induction, electrostatic coupling, or conduction. Electromagnetic induction causes radiated EMI. The presence of electromagnetic interference is at best mildly annoying if it causes a circuit to slightly malfunction, and at worst it can be very dangerous if it adversely affects any medical equipment.

There are a variety of standards and regulations for electromagnetic interference that need to be followed in order to meet the legal limits that have been imposed by the Federal Communications Commission (FCC) and the International Special Committee on Radio Interference (CISPR). The FCC developed their legal limits and methods of testing from publications from CISPR. The original standards from CISPR are not necessarily requirements that must be complied with to meet certain legalities, but instead they are used as benchmarks for suppliers, or specifications for manufacturers to include as a product feature. These standards are often the foundation for legislation regarding electromagnetic compatibility and interference.

Some common countermeasures that can be put in place to reduce the risk of electromagnetic interference are grounding and shielding. These simple measures are able to help quiet the source of the interference or inhibit the reception of the interference by either reducing the amount of the EMI or by diverting the emissions to an alternative inconsequential space.

CISPR 16 is one of the most relevant standards which details the equipment and the methods that are used for measuring electromagnetic compatibility and interference. This standard focuses on frequencies greater than 9 kHz and less than 1 GHz. This standard is actually composed of fourteen publications that describe various aspect of EMC and EMI. This standard goes in depth regarding any realistically imaginable scenario including the measuring apparatuses and ancillary equipment that are related to conducted or radiated electromagnetic disturbances. CISPR 16 defines the processes and methods for the measurement of electromagnetic disturbances. CISPR 16 reports any uncertainties that exist regarding the electromagnetic compatibility tests and measurements and this standard also discusses the statistics and models that are considered for EMC compliance of a product.

The most significant regulation from the FCC regarding electromagnetic interference and compatibility standards is known as FCC Part 15. This set of rules and regulations is actually derived from the Code of Federal Regulations, Title 47, Part 15, 47 CFR 15. FCC Part 15 of Title 47 which sets the guidelines for all unlicensed transmissions is the part of the standard that is actually relevant to EMC. FCC Part 15 has some expected variation from the CISPR standards, however, the regulation align on the more important topics.

FCC Part 15 goes in depth regarding the actions that may be taken if a low-power transmitter is found to be non-compliant with its rules and regulations. Penalties range from forfeiture of any equipment that is not in compliance with legislation up to administrative fines or even criminal penalties.

Most emission limits specified by FCC Part 15 are based on the strength of the electric field produced by the transmitter rather than the amount of power that is generated because the electric field is what causes the electromagnetic interference. Emission limits are generally not based on power generated by a transmitter, because there is not a strong correlation between the strength of the electric field produced and the amount of power generated. It is difficult to know if a product that is being made will have any issues with electromagnetic interference or electromagnetic compatibility until after it is designed and constructed. Large problems will appear earlier on in production, however some inconsistencies will not arise until final testing is being undertaken. For large scale production the best route to ensuring that all EMC and EMI regulations are meet is to find a national recognized testing laboratory that is certified by the Occupational Safety and Health Administration. In order to market and sell our product many regulations and requirements must be met.

Products are generally classified into two categories for electromagnetic compatibility and electromagnetic interference requirements: Class A and Class B. Class A products are marketed for use in a commercial or industrial environment. Class B products are marketed for use in a residential environment.

Radiated emissions testing is the testing that must be undertaken to determine the strength of the electric field that is produced by a product. There will be an electric field in every circuit as long as there are switching voltages or a current present. This field must be tested to determine if its size could be detrimental to any surrounding circuits. In radiated emissions testing a procedure called maximization is used to determine the maximum amplitude of the electromagnetic emissions that are generated from the product being tested. During maximization, the product is manipulated to find the several highest amplitudes that the product can produce to determine if they are within the allowed limits.

There are two main types of radiated emissions test sites: open area test sites and semi-anechoic chambers. Open area test sites are the most common type of test site, however semi-anechoic chambers are arguably more accurate. Semi-anechoic chambers are set up in a similar manner to open area test sites, except instead of being out in the open they are housed in a metal room. The semi-anechoic chamber allows background signals to be attenuated and the signals generated by the product to be prominent in testing. In order to test the complete frequency range, radiated emissions test labs must often use multiple antennas of a variety of sizes to cover the entire spectrum of interest.

Conducted emissions testing is the testing that must be undertaken to determine the amount of electromagnetic energy that will be conducted back onto the power supply. The purpose of conducted emissions testing is to eliminate or decrease the amount of electromagnetic interference that can be generated from the device back onto the power supply, because interference could produce adverse effects in nearby devices. Conducted emissions testing focuses mostly on devices that are connected to an AC power supply, however there are some standards that require this form of testing on circuits that are connected to a DC power supply. Conducted emissions limits are generally a little stricter for Class B devices. In residential environments there are often many devices connected to the power supply. Having more devices connected increases the risk of interference which demonstrates the need for stricter conducted emissions limits on Class B devices.

Electromagnetic compatibility immunity testing tests products regarding electromagnetic energy from the other direction. EMC immunity testing subjects a device to electromagnetic energy to examine its response, or preferably its lack of a response, to the input. There are a large variety of tests that examine the electromagnetic compatibility of a product including, but not limited to: radiated, electrical fast transient, surge, and magnetic field immunity testing. Immunity testing involves continuous waves that are applied over a length of time and transient tests that are applied in short impulses. In radiated immunity testing a product is subjected to a uniform electric field over a range of frequencies. Some common issues that products have during radiated immunity testing are with cables acting as receiving antennas at low frequencies and with induced voltages on digital and analog signals at high frequencies. Electrical Fast Transient is another form of immunity testing in which the switching of inductive loads that could affect a device is simulated. A surge immunity test simulates what would happen if a product was subjected to a low frequency power surge. In magnetic field immunity testing the tests are implemented to ensure that the device under examination is still capable of working properly under the duress of an imposed magnetic field. These are some main tests that a product may have to endure to be labeled as EMC compliant, however, depending on the product and the intended use, not all of the tests may be necessary in order to electromagnetically compatible with the environment.

7.3 Nominal 3 V/3.3 V Supply Digital Integrated Circuits

When logic circuits were first implemented by semiconductor devices they utilized bipolar junction transistors. To determine the logical on and off, an input voltage greater than 0.7V would have to be applied. Because Darlington pair transistors are very common the standard became that any voltages above 1.4V-1.6V would be logical 1 and anything below would be considered logical 0. In order to bias these logic circuits utilizing 1.4V it was best to at least double that voltage and they 3 to 3.3V became common for digital logic circuits.

These voltages have long become engineering standards universally, but technological improvements have necessitated that the tolerable voltage range used for biasing logic circuits change over time. Primarily, logic circuits are becoming smaller and denser. Logic chips are also increasing in the number of functions and thus operations they perform. These increases in logic circuit technology increase a number of undesirable effects such as producing EMF interference, noise from the circuit and greater power consumption. These unfavorable effects must be scaled down by also scaling the power supply and achieving greater unity from both users and designers of power supplies.

The JEDEC standard JESD8C.01 seeks to scale biasing voltages by establishing a standard voltage range for V_{bias}. For devices operating with a narrow range bias, then V_{bias} should not exceed a range of 3.15 to 3.45V while its nominal voltage is 3.3V. A power supply operating in a normal voltage range should provide a range of 3.0 to 3.6V, again with a nominal voltage of 3.3V. The extended range is the final range that is accepted by certain logic devices. The extended range varies from 2.7 to 3.6V and has a nominal voltage at 3.0V.

One benefit of obtaining greater standards from power supplies is to ensure the power is compatible with transistor threshold levels (TTL), while maintaining an acceptable noise margin. These standards of logical 1 and 0 were intended for all components pertaining to logic functions, whether memory storage, microprocessors, gate arrays or other logic circuits. The JEDEC standard JESD8C.01 specifies input and output voltages for a low voltage transistor threshold levels (LVTTL) system with a normal bias range. For input voltages the input high voltage should reach its maximum at $0.3V+V_{bias}$, The high input voltage should dip below 2V. For logical low inputs, the V_{IL} should range from -0.3 to 0.8V. The V_{OH} should be a minimum of 2.4V and the V_{OL} should be at maximum 0.4V. A test condition of applying the minimum power supply voltage of 3.0V and ensuring the output current is -2mA for high and 2mA for low.

However while transistor are still used for some applications, most logic circuits today do not utilize transistors, preferring CMOS gates for logic. For voltage inputs, the range for LVCMOS logic circuits is the same as it was for LCTTL circuits. Assuming a V_{bias} range of 3.6 to 2.7V the LVCMOS output voltage levels do vary from LVTTL outputs. V_{OH} should reach its minimum at V_{bias}-0.2V and V_{OL} should be maxed at 0.2V.

If a component meets or exceeds the input voltage requirements of the three ranges then it is said to meet the requirements of the standard JESD8C.01. If the logical inputs and outputs meet the ranges specified above for LVTTL systems then the device is said to be LVTTL-compatible. The same applies for the ranges given for LVCMOS systems, if those ranges are met then the device is said to be LVCMOS compatible.

This standard also has standards for analog inputs that are fed to a Schmitt trigger input in order to provide discrete outputs for the logic. For the normal supply voltage range, the trigger voltages of the Schmitt triggers have their own respective range. The positive trigger V_p should have a range of 0.9 to 2.1V. The negative trigger V_n should range from 0.7 to 1.9V. These trigger voltages should then produce a hysteresis ranging from 0.2 to 1.4V. If these triggers are correct then the Schmitt trigger should produce a V_{OH} with a minimum of V_{bias}-0.2V for a low current draw (~I_{OH}=-0.1mA) or 2.4V for a high current draw(~I_{OH}=-2mA). The low output range, V_{OL}, should have a maximum of 0.2V for low output current draw (~I_{OH}=-0.1mA) or 0.4V for a high current draw (~I_{OH}=-2mA).

If the Schmitt trigger is biased by a supply running the extended range then the high output current is generally very small (~I_{OH}=-0.1mA). All of the voltage ranges are the same as if they were in the normal range, with two exceptions. The output high voltage, V_{OH}, should only have a minimum of V_{bias}-0.2V. The output low voltage, V_{OL}, is maxed at 0.2V.

This list of standards was created to help bring uniformity to some of the various voltage levels of logic circuits. For biasing voltages three voltage ranges were provided in order for users to choose from a narrow, normal or extended range. These ranges were provided for users and manufacturers some flexibility in their power supply selection or design respectively. For the voltage inputs and outputs of the logic circuits, different ranges were provided depending upon the type of logic circuit, TTL or CMOS. With these ranges come further certifications for LVTTL or LVCMOS. Finally, a range of voltages was provided for the different voltages pertaining to Schmitt triggered inputs. Ranges were provided for the positive and negative firing voltages, the input and output voltages, and the biasing voltages. With industry confirmation to these standard voltage ranges, users and designers alike can provide power supplies that enable the logic circuits to be more efficient and ideal.

8. Administrative

Administration determines what products are viable versus what products will not make it out of research or development stages. Administrative topics cover all logistical aspects of the project including budgeting, suppliers, and timekeeping.

8.1 Budgets

Budgeting is an important part of any project. Budgets are required to keep projects such as within the realm of economic feasibility. Our budgeting is only limited by the production cost of an individual unit; we assume that our overall development budget and test unit budget are fairly unlimited given the scope of the assignment.

8.1.1 Overall Budget

The overall project budget is a cost for parts for the entire project of designing, developing, and producing a single Pocket Amp prototype. The project development cost does not have a set hard limit because we want to have the freedom to explore all of our options in building this project to create the best individual prototype.

System	**Part**	**Quantity**	**Unit Cost**	**Line Cost**
Power	Battery	4	$12	$48
Power	Power Components	2	$15	$30
Amp	PCB	4	$10	$40
Amp	Amp Components	2	$5	$10
Microcontroller	Microcontroller	3	$10	$30
Microcontroller	Microcontroller Parts	2	$15	$30
All	Packaging/Case	2	$20	$40
All	Misc Components	1	$20	$20
TOTAL COST:				$248

Table 8. Estimated Overall Budget

8.1.2 Test Unit Budget

A secondary goal of the group is to limit the cost of each individual prototype, which is shown in Table 9. The prototype cost is in essence the cost to build one prototype without counting man-hours for assembly or the cost of shipping each part. This cost incurred is a dependent variable of both the overall project cost in Table 8 and the production unit cost in Table 10.

0System	Part	Quantity	Unit Cost
Power	Battery	1	$12
Power	Power Components	1	$15
Amp	PCB	1	$10
Amp	Amp Components	1	$5
Microcontroller	Microcontroller	1	$10
Microcontroller	Microcontroller Parts	1	$5
All	Packaging/Case	1	$20
All	Misc. Components	1	$5
TOTAL COST:			$82

Table 9. Estimated Individual Prototype Budget

8.1.3 Production Cost Per Unit

The production cost per unit shown in Table 10 is the estimated cost of parts to build a single Pocket Amp unit. These costs are assuming a production run of 1,000 to 2,000 units and is neglecting costs for assembly and shipping. We did not include assembly as we are unsure of the exact cost of assembly. Assembly costs vary for small-run units but due to the advances in robot component picking robots leading to decreasing assembly costs for small production runs this cost is negligible. We used an online tool (smallbatchassembly.com) to get a quote for our design and for a run of 750 (the max for this quote tool) we received a quote of 7 dollars each. Assuming this can be mitigated by the engineering team assembly some parts and for using a larger batch size we estimate assembly costs to be around 5 dollars. Shipping costs vary even far more and as such are left for the end-user to absorb.

System	Part	Quantity	Unit Cost
Power	Battery	1	$8
Power	Power Components	1	$10
Amp	PCB	1	$8
Amp	Amp Components	1	$10
Microcontroller	Microcontroller	1	$4
Microcontroller	Microcontroller Parts	1	$2
All	Packaging/Case	1	$10
All	Misc. Components	1	$5
TOTAL COST:			$57

Table 10. Estimated Production Cost Per Unit

8.2 Suppliers

Suppliers were chosen for a number of reasons. The main reasons for choosing specific suppliers is cost, product availability, and availability of information and other resources. Minor reasons contributing to supplier selection include shipping speed, website usability among other things. Our main suppliers for parts are Texas Instruments, Digi-Key, Linear Technology, Sparkfun, Adafruit, and various Amazon.com suppliers. Texas Instruments was used due to their sampling policy, fast shipping, and wide array of launchpads and other kits. Digi-Key and Linear Technology were chosen due to their large number of integrated circuits and number of different suppliers available through their site. Sparkfun and Adafruit were both used due to their supply of compatible launchpad-style kits and support for all their products. Amazon.com was used due to the vast number of basic components available, free two-day shipping through Amazon Prime, and excellent site search engine.

8.3 Timeline and Milestones

The following milestones are taken from the course syllabus and generated based on time estimates to complete tasks. The milestones are a rough outline of overall progress to make sure the project is on track. These milestones shown in Table 11. Timeline and Milestones.

Task Description	Deadline	Status	Leader
Senior Design 1 8/22/16 - 12/6/16			
Generate Ideas	8/26/16	Completed	Full Group
Pick Idea	8/31/16	Completed	Full Group
Divide and Conquer	9/9/16	Completed	Full Group
Research Similar Projects	9/16/16	Completed	Full Group
Meeting with Dr. Lei Wei	9/23/16	Completed	Full Group
Research Power Systems	10/7/16	Completed	Trent
Research App Building	10/7/16	Completed	William
Research Guitar Effects/Amps	10/7/16	Completed	Joshua
Research DSP	10/7/16	Completed	Marissa
Research Headphone Output	10/7/16	Completed	Marissa
Research Bluetooth	10/14/16	Completed	William
Research Microcontrollers	10/14/16	Completed	Full Group
Research PCB Design	10/14/16	In Progress	Full Group
Table of Contents	11/4/16	Completed	Full Group
Current Draft	11/11/16	Completed	Full Group
App Prototype	12/6/16	In Progress	William
Final Document	12/6/16	Complete	Full Group
Senior Design 2 1/9/17 - 4/26/17			
Build Prototype	TBD	Not Started	Full Group
Testing	TBD	Not Started	Full Group
Finalize Prototype	TBD	Not Started	Full Group
Peer Presentation	TBD	Not Started	Full Group
Final Report	TBD	Not Started	Full Group
Final Presentation	4/26/17	Not Started	Full Group

Table 11. Timeline and Milestones.

8.4 Team Organization

Team organization was done with Google Drive, GroupMe, Microsoft Word, email and text message. A majority of our group communication was done through GroupMe, which is an instant messaging app and website combination built for group communication. It allows basic file sharing, large message formatting, and genera instant messaging. Google Drive was used for most document storage and sharing due to low cost of large storage, ease of sharing files, and live downloading items to local computers with the Google Drive app. Microsoft Word was used for this document editing due to the ease of using styles to automatically format the entire document. Word allows multiple users to edit the same document through OneDrive and merge the edits together. Traditional email and text messages were used for some communication.

9 Conclusion

Senior Design I produced a number of challenges and triumphs which offered tremendous insight and knowledge. Senior Design is a final year course and as such is among the most difficult that an undergraduate engineering student will take during their time at UCF. With this difficulty comes a remarkable opportunity to learn and grow as an engineer. Our group faced challenges and opportunities with several aspects of this project.

A significant triumph for this group was the equal dispersion of workload. While the exact difficulty of each section of the project is difficult to determine no member of the group was not given an essential task to perform. In addition to the work being pretty equally spread out among the group, no member of the group slacked on their designated portion. All members performed their respective portions to the best of their ability.

Another strong characteristic of our group was our interconnection. While each member had a task separate from the rest of the group, each member was very conscious of how their portion affected the other tasks. Not only was each member aware of the others' work but was involved with that work. Design challenges for the project were solved by the team as a whole rather than placing the burden solely on one member. This ability to rely on one another enabled the group to succeed where the efforts of a single member might have been insufficient.

One of the primary learning opportunities for this project was time management. The group was very effective in ensuring consistent time devoted to the project. Before the second week of class was over, the group had setup a weekly time to meet, in addition to the normal class time, to focus on the project. From this stand point of meeting early and often the group excelled. However, the group has had challenges in time management in what aspects of the project to focus on. Due to some misunderstanding from the lecture time, the group delayed writing of the paper to focus on other tasks – such as carefully selecting parts – which proved to be less time sensitive. This led to an accelerated writing pace at the end of the semester. Fortunately, this accelerated writing proved sufficient to meeting the writing goals.

This project and the final paper in particular presented many challenges and opportunities for our group. Our group was able to demonstrate hard work and fairness by sharing the workload equally among the group. Additionally, the group learned to work together to solve difficulties in the design of the Pocket Amp. Finally, the group was faced with some difficult time allocation problems which hindered progress. Despite this setback the group was able to persevere and produce this paper on time. The lessons and successes of this class have been invaluable and will certainly be useful in future engineering projects.

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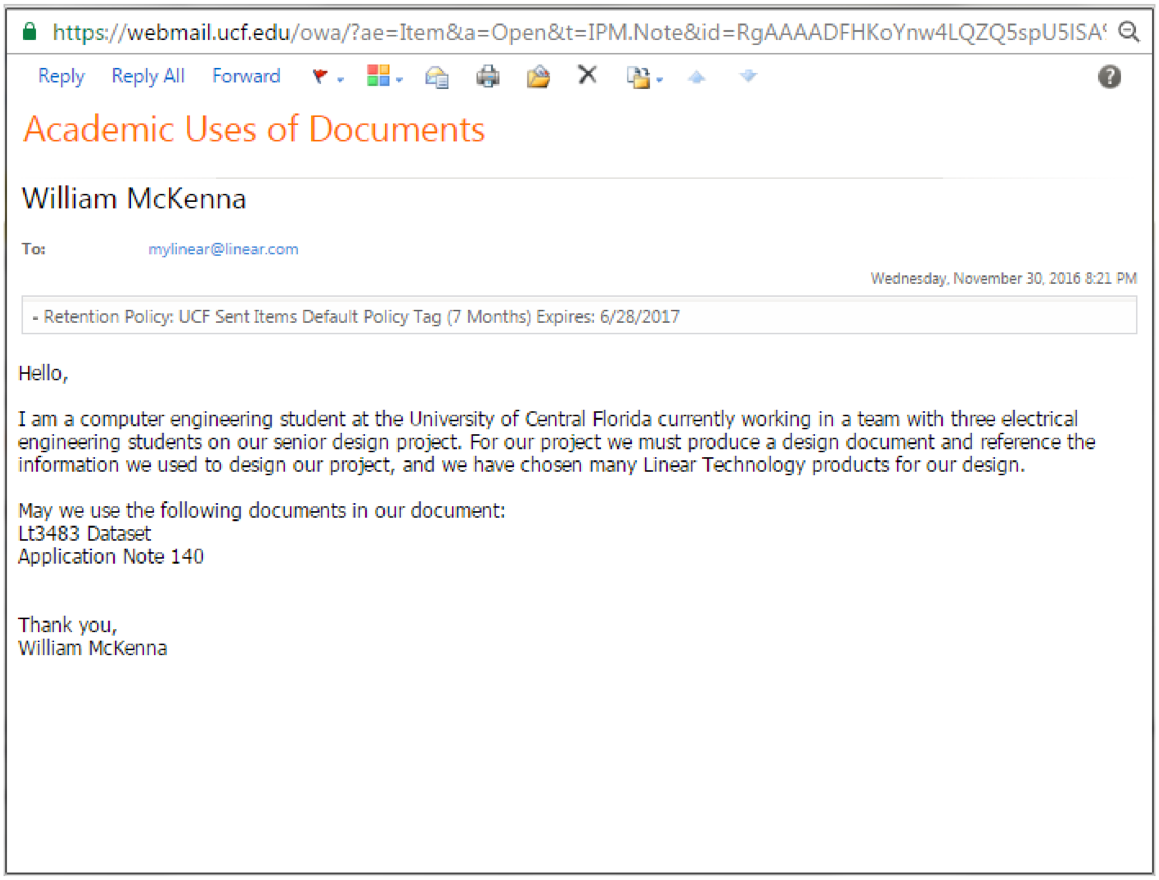
11. Appendices

11.1 Copyright Permissions



https://webmail.ucf.edu/owa/?ae=Item&a=Open&t=IPM.Note&id=RgAAAADFHKoYnw4LQZQ5spU5ISA'

Reply Reply All Forward

Academic Uses of Documents

William McKenna

To: mylinear@linear.com

Wednesday, November 30, 2016 8:21 PM

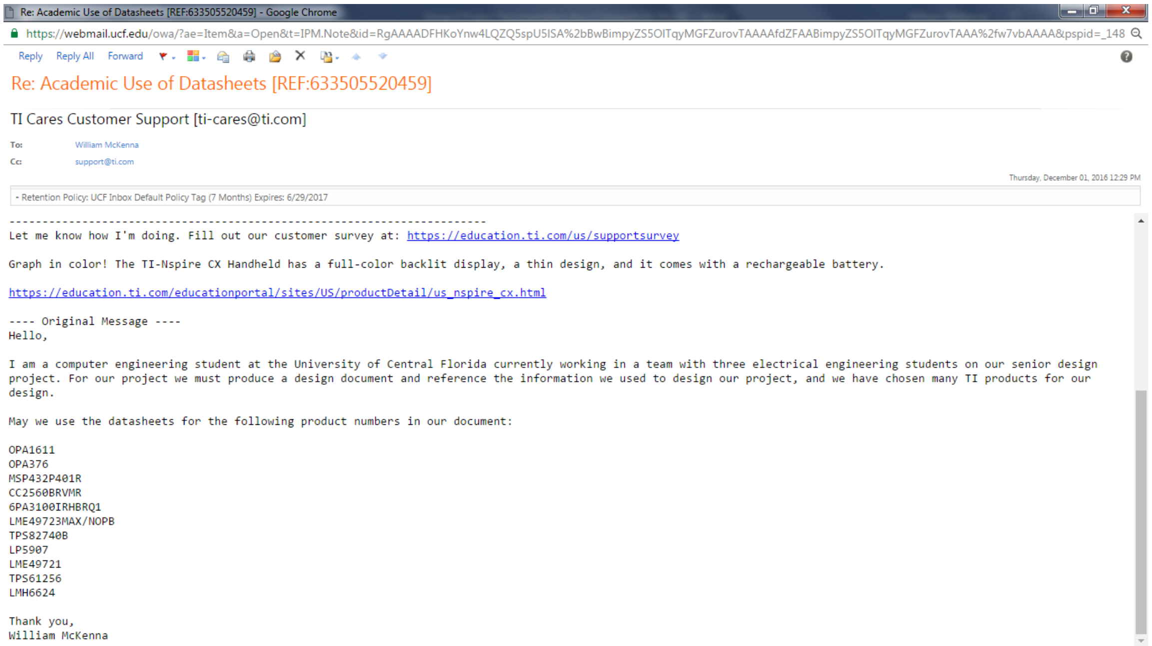
- Retention Policy: UCF Sent Items Default Policy Tag (7 Months) Expires: 6/28/2017

Hello,

I am a computer engineering student at the University of Central Florida currently working in a team with three electrical engineering students on our senior design project. For our project we must produce a design document and reference the information we used to design our project, and we have chosen many Linear Technology products for our design.

May we use the following documents in our document:
Lt3483 Dataset
Application Note 140

Thank you,
William McKenna



Re: Academic Use of Datasheets [REF:633505520459] - Google Chrome

https://webmail.ucf.edu/owa/?ae=Item&a=Open&t=IPM.Note&id=RgAAAADFHKoYnw4LQZQ5spU5ISA%2bBwBimpyZS5OITqyMGFZurovTAAAAfdZFAABimpyZS5OITqyMGFZurovTAAA%2fw7vbAAAA&pspid=_148

Reply Reply All Forward

Re: Academic Use of Datasheets [REF:633505520459]

TI Cares Customer Support [ti-cares@ti.com]

To: William McKenna
Cc: support@ti.com

Thursday, December 01, 2016 12:29 PM

Retention Policy: UCF Inbox Default Policy Tag (7 Months) Expires: 6/29/2017

Let me know how I'm doing. Fill out our customer survey at: https://education.ti.com/us/supportsurvey

Graph in color! The TI-Nspire CX Handheld has a full-color backlit display, a thin design, and it comes with a rechargeable battery.

https://education.ti.com/educationportal/sites/US/productDetail/us_nspire_cx.html

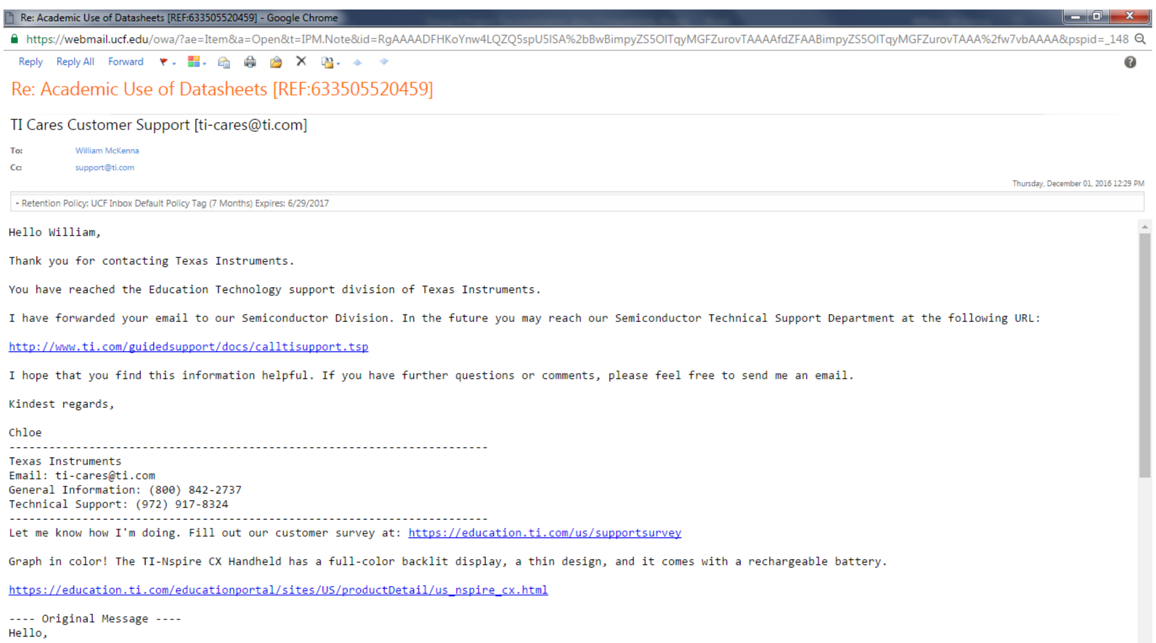
---- Original Message ----
Hello,

I am a computer engineering student at the University of Central Florida currently working in a team with three electrical engineering students on our senior design project. For our project we must produce a design document and reference the information we used to design our project, and we have chosen many TI products for our design.

May we use the datasheets for the following product numbers in our document:

OPA1611
OPA376
MSP432P401R
CC2560BRVMR
6PA3100IRHBRQ1
LME49723MAX/NOPB
TPS82740B
LP5907
LME49721
TPS61256
LMH6624

Thank you,
William McKenna



Re: Academic Use of Datasheets [REF:633505520459] - Google Chrome

https://webmail.ucf.edu/owa/?ae=Item&a=Open&t=IPM.Note&id=RgAAAADFHKoYnw4LQZQ5spU5ISA%2bBwBimpyZS5OITqyMGFZurovTAAAAfdZFAABimpyZS5OITqyMGFZurovTAAA%2fw7vbAAAA&pspid=_148

Reply Reply All Forward

Re: Academic Use of Datasheets [REF:633505520459]

TI Cares Customer Support [ti-cares@ti.com]

To: William McKenna
Cc: support@ti.com

Thursday, December 01, 2016 12:29 PM

Retention Policy: UCF Inbox Default Policy Tag (7 Months) Expires: 6/29/2017

Hello William,

Thank you for contacting Texas Instruments.

You have reached the Education Technology support division of Texas Instruments.

I have forwarded your email to our Semiconductor Division. In the future you may reach our Semiconductor Technical Support Department at the following URL:

http://www.ti.com/guidedsupport/docs/calltisupport.tsp

I hope that you find this information helpful. If you have further questions or comments, please feel free to send me an email.

Kindest regards,

Chloe

Texas Instruments
Email: ti-cares@ti.com
General Information: (800) 842-2737
Technical Support: (972) 917-8324

Let me know how I'm doing. Fill out our customer survey at: https://education.ti.com/us/supportsurvey

Graph in color! The TI-Nspire CX Handheld has a full-color backlit display, a thin design, and it comes with a rechargeable battery.

https://education.ti.com/educationportal/sites/US/productDetail/us_nspire_cx.html

---- Original Message ----
Hello,

11.2 Data Sheets

11.2.1 Op Amps

OPA1611: http://www.ti.com/lit/ds/symlink/opa1612.pdf
OPA376: http://www.ti.com/lit/ds/sbos406g/sbos406g.pdf
LME49721: http://www.ti.com/lit/ds/symlink/lme49721.pdf
LME49723: http://www.ti.com/lit/ds/symlink/lme49723.pdf
LMH6624: http://www.ti.com/lit/ds/symlink/lmh6626.pdf

11.2.2 Regulators and Converters

TPS82740B: http://www.ti.com/lit/ds/symlink/tps61254.pdf
LP5907: http://www.ti.com/lit/ds/symlink/lp5907.pdf

11.2.3 Miscellaneous

MSP432P401R: http://www.ti.com/lit/ds/symlink/msp432p401r.pdf
CC2560BRVMR: http://www.ti.com/lit/ds/swrs121e/swrs121e.pdf
TPS61256: http://www.ti.com/lit/ds/symlink/tlv320dac3100-q1.pdf