Pocket-Amp

Joshua DeBaere, Marissa Kane, Trent Coleman, and William McKenna

Dept. of Electrical Engineering and Computer Science, University of Central Florida, Orlando, Florida, 32816-2450

***Abstract* — This paper details the design process to create the Pocket-Amp. The Pocket-Amp is designed to provide guitar amplification and effects to a headphone output. The Pocket-Amp's small size and ease of use is designed to make guitar practice a more portable event. The ultimate goal of the Pocket-Amp is to combine the best of analog and digital designs to produce excellent and quick effects that mimic the use of traditional amplifiers.**

***Index Terms* — Music, op-amps, portable, digital effects, amplifiers.**

I. Introduction

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 smartphones, 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.

II. Overview of Requirements

When the members of the Pocket-Amp design team set out to create the Pocket-Amp they analyzed the most important features and set detailed specifications for those features. The Pocket-Amp would be limited to being under 5 pounds and be able to fit within a 5x6x3 inch pocket. These requirements were set to ensure that the Pocket-Amp is extremely portable within a normal cargo pants pocket. A detailed App would be required for the Pocket-Amp to interface with so that different effects features could be altered and tested. The Pocket-Amp was required to implement a variety of guitar effects including delay, chorus, distortion, volume and other effects. These requirements ensure that the Pocket-Amp is useful not for musicians and testing different effects.

III. High Level Block Diagram

Before the overall design process was to begin the members of the Pocket-Amp team decided to partition the separate sections of the project. This was best shown by creating a high-level block diagram. The high-level block diagram in **Error! Reference source not found.** 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 for the final project.

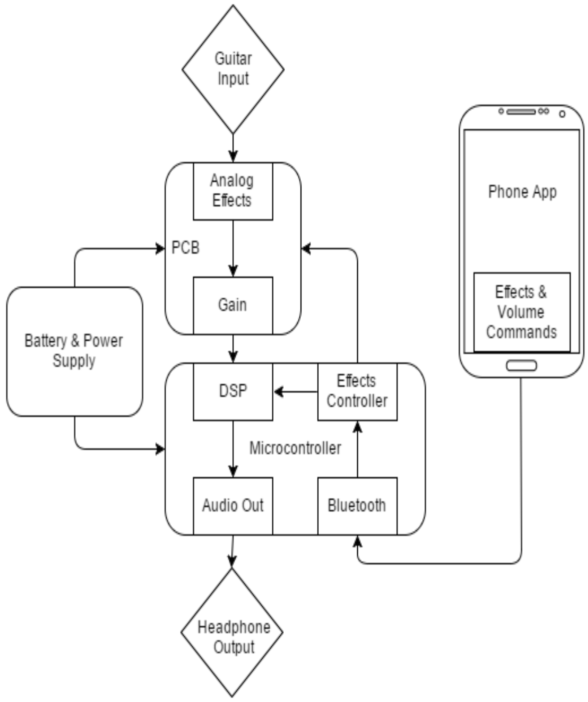

Fig. 1. Pocket-Amp High-level Block Diagram.

IV. Major Components

With the high-level block diagram established the group members could begin distributing work between the major components of the Pocket-Amp each with their own major subsections.

A. Battery and Power Supply

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. The power system would be designed in order to provide a ±5.4V supply for analog loads, a 3.3V supply for digital loads and a ±1.5V reference for the analog to digital convertor. The power supply was constructed of several components to supply these voltages.

(1) The battery- For the Pocket-Amp a compact size and good voltage output were paramount. For the battery, a Lithium-ion Polymer battery was selected. 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.

(2) ±5.4V Supply – Several design iterations were considered for this supply as both positive and negative voltages are required. The initial design called for a single boost converter to create the proper voltage magnitude. After this the output would be split to an invertor which would then form the negative supply. This quickly proved to be a difficult design as creating a small inverter without utilizing an op-amp can be difficult. Another solution was attempted where two separate convertors would be utilized one inverting and the other a conventional boosting convertor. This design was quite feasible but unfortunately utilized two inductors which can introduce noise into the system and tend to be a large component on the board. The third and final approach was to utilize a single inductor split rail convertor. These chips are especially designed to create equal magnitude positive and negative supplies. This design allows for only one inductor in the supply and reduces the number of ICs on the power PCB. For the supply the Texas Instruments TPS65132 was selected as the primary chip. The TPS65132 is a programmable output split rail convertor which outputs a default ±5.4V which was determined to be an excellent supply voltage for the audio Op-Amps.

(3) The 3.3V Digital Supply – The secondary voltage supply was to provide 3.3V for the digital loads, primarily the MSP432. This supply would be required to supply the MSP432, the digital to audio convertor and the for the I2C communication bus. Ideally, this chip was produce little noise and maintain a stable output for the digital loads.

Originally, some type of buffering was considered between the power supply and the digital loads to reduce noise and potential hazards. However, buffering both proved to be difficult and ultimately unnecessary as the battery supply is not prone to noise. The next option 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. Ultimately, the desired chip was the LP2989. The LP2989 sports both a very low noise output and a very low dropout voltage. The LP2989 also came in various packages which made it easy to solder and to test. However, after rigorous testing with the LP2989 the chip proved difficult to utilize so a backup regulator was implemented. The TPS70633 was obtained in order to provide the 3.3V in light of the LP2989's difficulties. TPS70633 presents many of the same qualities of the LP2989, low noise and low dropout voltage, but less optimal in both qualities. However, the TPS70633 does spout a much better price as it is less than $1/3^{rd}$ the cost of the LP2989.

(4) ±1.5V Reference Voltage- in order for the analog to digital filter to operate correctly it must have a positive and negative rail. These rails are utilized as a reference with which to alter the signal from. For the project is was decided that the Texas Instruments TPS78915 was selected to create these rails. The TPS78915 is an ultralow power, low- noise LDO regulator which outputs 1.5V as a fixed value. The negative rail was created by utilizing an op-amp inverter to invert the output of the TPS78915 and create the -1.5V rail.

(5) Battery Charging Circuit - 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. 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.

B. Analog Amplification and Effects

The goal for the Pocket-Amp was to provide many of the same effects that are produced from larger, bulky analog amps but in a much smaller package. This process would be accomplished largely by utilizing different Op-Amp circuits, to realize many analog effects, and for other effects to realize them digitally through the microcontroller. The analog portions of the Pocket-Amp are separated largely into two sections: filtering and analog effects

(1) Amplifier Selection - For the audio effects to be created from the guitar's input several amplifiers would be needed to create the effects' circuits. 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. Though many different Op-Amps were considered each with varying benefits and drawbacks the eventual choices were the 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 2 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 3 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.

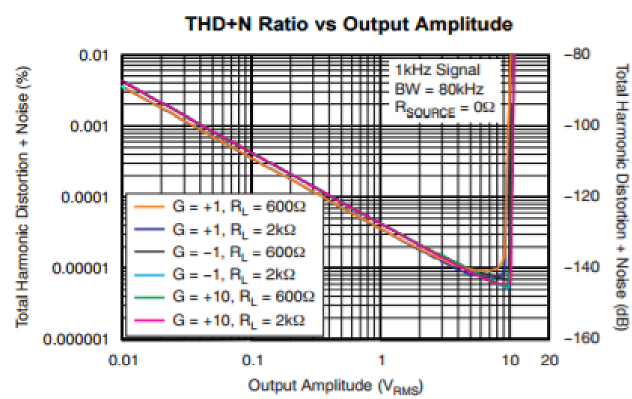

Fig. 2. OPA 1611 THD vs. Output Voltage

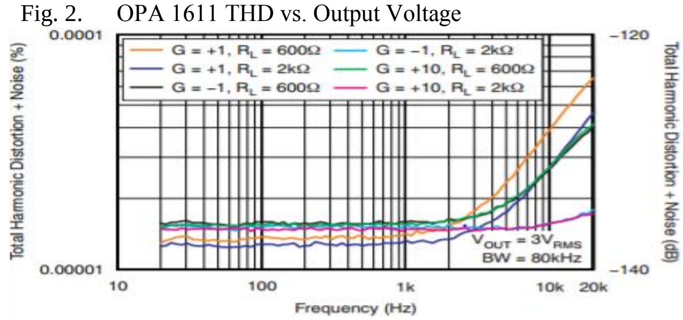

Fig. 3. OPA 1611 THD vs. Frequency

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.

V. ANALOG FILTERING AND EFFECTS

A. 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.

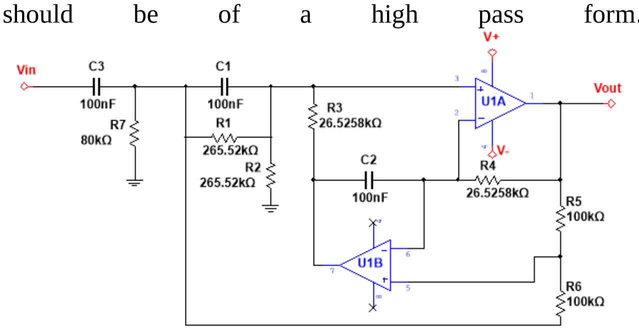

Fig. 4. Fliege High Pass Notch Filter

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 4 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 and results in the magnitude response in **Error! Reference source not found.**. 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.

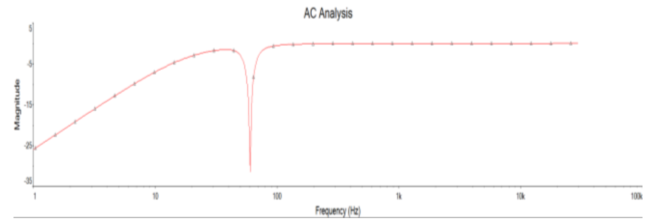

Fig. 5. Fliege High Pass Notch Filter Response

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.

B. Analog Effects

The first step in creating these analog effects was to section them into a block diagram so that each component could be placed in the proper order. With the block diagram in place each effect portion could be built individually before assembling all effects together. This process allowed for an easier and ultimately more efficient design process. With the block diagram in place each effect portion could be built individually before assembling all effects together.

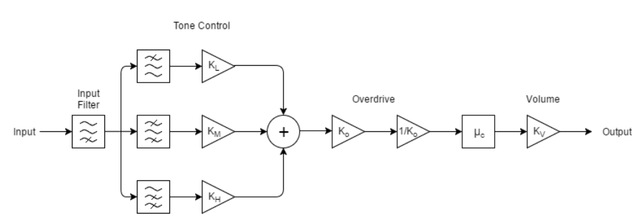

Fig. 6. Pocket-Amp Analog Effects Block Diagram

(1) The first effect to be handled was the 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. Two primary methods of Tone Control implementation were considered for the Pocket Amp: Three Channel Tone Control Network and Baxandall Tone Control Circuit. Ultimately, the Three Channel Tone Control Network was the selected effect method for the Pocket-Amp.

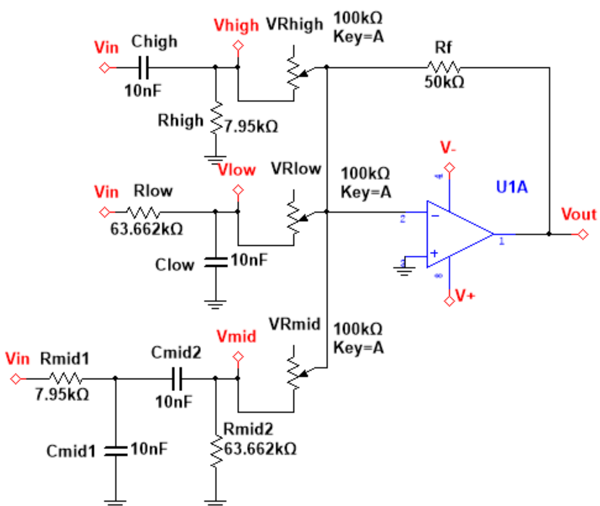

Fig. 7. Three Channel Tone Control Network With Boost and Cut Capability

The Three Channel Tone Control Network presented in **Error! Reference source not found.** has a magnitude response that can be seen in Figure 8 where section (a) shows the range of attenuation and amplification due to changing the potentiometer VRhigh in **Error! Reference source not found.**, section (b) shows the range of attenuation and amplification due to changing the potentiometer VRmid in **Error! Reference source not found.**, and section (c) shows the range of attenuation and amplification due to changing the potentiometer VRlow in **Error! Reference source not found.**. 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 **Error! Reference source not found.** 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.

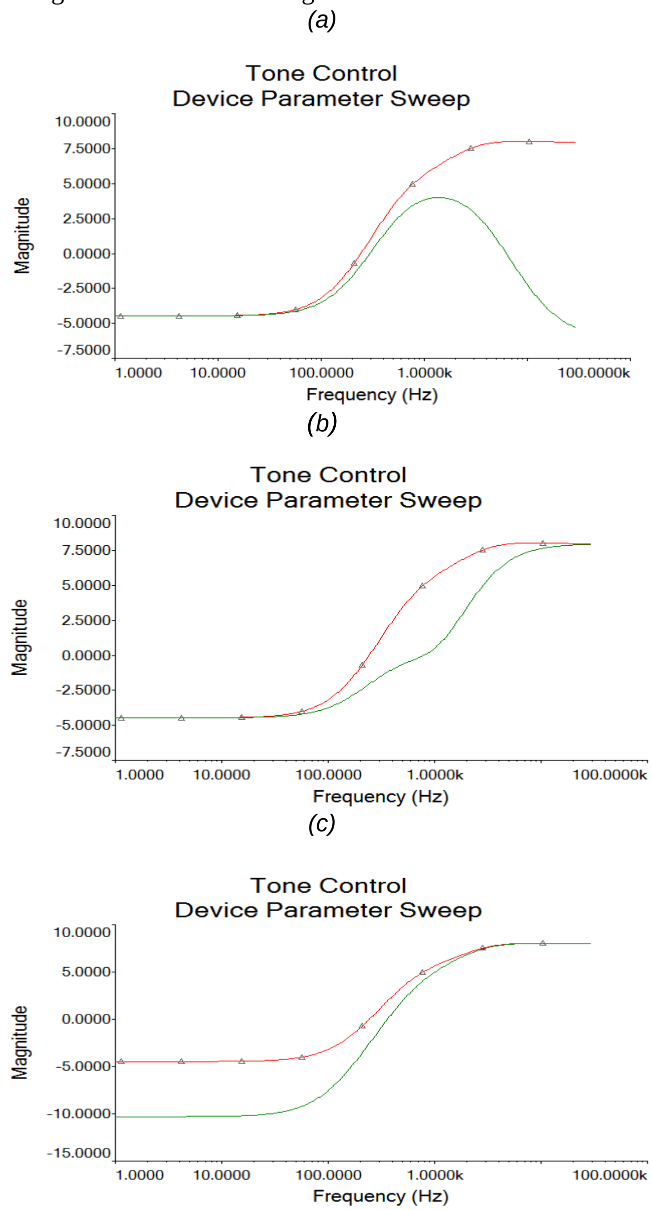

Fig. 8. Magnitude Response of Three Channel Tone Control Circuit

While the Baxandall Tone Control Network is 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.

(2) The second effect to be implemented was the 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 a reliable and variable clipping distortion can be produced. However, this design process would be difficult to implement a variable DC source so another method was chosen.

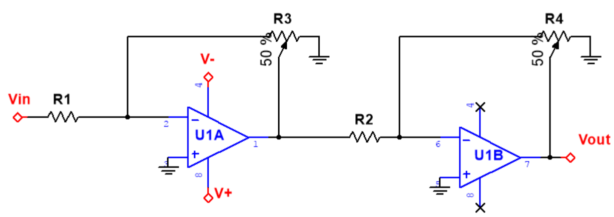

Fig. 9 Overdrive with Cascaded Operational Amplifiers

The chosen method utilizes 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 **Error! Reference source not found.**. 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.

$$K = \frac{R_3}{R_1} = \frac{1}{\frac{R_4}{R_2}} = \frac{R_2}{R_4}. \quad (1)$$

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.

(3) The final effect to be enacted by analog circuits was the 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. 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, 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.

VI. Software

The software involved in developing the Pocket Amp project includes communication protocols between the Pocket Amp and a phone, communications between each chip, namely the MSP432 microcontroller, the Atmel ATmega32U4 microcontroller, and the Nordic nRF8001 Bluetooth low energy module., and digital signal processing. Due to the low-power nature of this system, the processor architecture is be designed with energy efficiency in mind. The TI MSP432P401R processor is utilized because it is a low-power ARM processor with built-in support for many basic analog and communication functions. The MSP432 communicates with several other microchips through a couple different inter-chip communication systems.

A. Inter-chip Communications

Serial Peripheral Interface (SPI) is a master-slave protocol in which in our case the MSP432 is the master controlling several digital potentiometers. This standard allows for full-duplex communication in which the master and slave can both send and receive information at the same time, however we are using SPI as a half-duplex where the master is always sending and the potentiometers are always receiving. These potentiometers allow the digital effects to be controlled and turned on or off.

Inter-Integrated Circuit (I^2C) is another master-slave protocol, however I^2C can have multiple masters. I^2C is physically implemented with two bidirectional wires: the serial clock line (SCL) and the serial data line (SDA). These wires are in an open-drain configuration where the data bus is kept at high voltage until a user pulls it low.

B. Phone Application

For our Pocket Amp we developed a phone application using Google's official Android Integrated Development Environment (IDE) called Android Studio because it cross-platform compatible on Windows, OS X, and many Linux operating systems. The Pocket Amp App will take all of the users inputs and settings for the Pocket Amp and translate them to Bluetooth packets. These packets will then be sent to the MSP432 which turns the packets into appropriate settings for the hardware components and applies any desired digital effects to the audio signal from the instrument. The App Backend mainly consists of an implementation of a modified version of JavaScript Object Notation (JSON). Initially, settings are transferred from the app to the Pocket Amp in a well-formed object transferred by a packet. Using JSON, the object is parsed into text to be sent at which point it is parsed from text back to object again. The App User Interface is a simple Graphical User Interface (GUI) that allows users to select digital and analog effects that will be applied and to control any options for the effects. The GUI was developed using components that restrict the user so he or she is only able to choose valid values and it was designed so that it is intuitive for musicians, but also simple enough for non-musicians to figure out how it works. Bluetooth low energy, specifically the Nordic nRF8001 single-chip BLE module blended with the Atmel ATmega32U4 on Red Bear Lab's Blend Micro, is used to communicate between the Android phone application and the Pocket Amp. Bluetooth low energy is used to significantly lower the power consumption relative to what a standard Bluetooth module would have required, because the BLE module not require a constant connection as does Bluetooth Classic.

C. MSP432 Code

The MSP432 code is an interrupt-driven basic operating system that is used to support a core of digital signal processing (DSP) functions. The DSP functions take up a majority of the code size and clock cycles in runtime. Besides DSP functions, the microcontroller manages communication, power, and hardware. The non-DSP functions are interrupt-driven in order to reduce the need for polling or any other wasteful overhead that could have potentially taken away from audio effects. Any incoming communication from the Bluetooth module will raise an interrupt and the Pocket Amp will cease sending an output while the interrupt is handled. This avoids creating potentially damaging or unpleasant audio output signals by having samples piling up or overwriting current samples. The MSP432 receives the Bluetooth input, then sends the appropriate signals, changes the appropriate internal states, clears the sample cache, then reinitializes 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.

VII. Conclusion

Senior Design II was an important time of growth and learning for every member of the group. In addition to the many technical lessons that were gained over the process, many other lessons in group management were gained. These lessons were gained over a large amount of meetings and work sessions over the two semesters. Each member of the group, while each assigned different tasks, worked closely with each member. Every aspect greatly increased the members of practical engineering knowledge, from component selections, to PCB implementation.

Acknowledgements

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