

Integrated Violin Pickup, Effects, and Amplifier

By Thanh N. Nguyen, Peter Sudermann, and Chetan Sharma

6.101, Spring 2017

Table of Contents

[Abstract](#)

[Introduction](#)

[Project Overview](#)

[Block Diagram](#)

[Magnetic Pickup - Thanh Nguyen](#)

[Objectives](#)

[High Level Implementation](#)

[Low Level Implementation](#)

[Results](#)

[Preamplification and Tone Control - Thanh Nguyen](#)

[Objectives](#)

[Preamplification Implementation](#)

[Tone Control Implementation](#)

[Result](#)

[Effects Stage - Peter Sudermann](#)

[Objectives](#)

[High Level Implementation](#)

[Low Level Implementation](#)

[Results](#)

[Amplification Stage - Chetan Sharma](#)

[Objectives](#)

[High Level Implementation](#)

[Low Level Implementation](#)

[Physical Implementation](#)

[Results](#)

[Conclusion](#)

[Final Project Results](#)

[Final Thoughts and Insights](#)

[Citations](#)

Abstract

This project aimed to create an integrated violin pickup, effects stage, and amplifier. The violin pickup was implemented as a laser-cut humbucker magnetic pickup with a non-inverting preamplifier and a Baxandall tone control network. The effects stage included a tube compressor designed to introduce soft clipping and a spring reverb tank. The final output amplifier took the form of a PCB mounted Class D amplifier. The resulting system worked as expected with minimal distortion and decent reliability. Modular design and testing, as we learned, were invaluable elements of the design process.

Introduction

Most musical instruments can have their notes converted into electrical signals; the violin is no exception. Since the 1920s, violins with electric pickups have been used by performers worldwide. Their popularity could be attributed to their versatility; an electric audio signal can be transformed and amplified in fashions that permit unique and novel sounds to be created. This gives the performer a greater variety of tools to create music and perform it in a large concert stage for thousands of people.

The goal of this project is to create a complete system capable of picking up notes from a standard violin, applying effects to them, and amplifying them to a level suitable for performance. Notes will be captured using a magnetic pickup. Piezoelectric pickups are traditionally used for violins, but a magnetic pickup has the advantage of depending less on the particular acoustics of the instrument. A preamplifier immediately boosts this incoming signal to operational levels.

The captured signal will then be sent to the sound effects stage. The sound effects stage serves to add complexity to the relatively dry, acoustically uninteresting signal of the pickup. This stage makes use of antique analog audio equipment by using a tube compressor and hardware spring reverb.

The wet signal will then be sent to the output stage. The output stage of this project takes the form of a Class D amplifier. The Class D architecture allows one to amplify sound accurately without compromising efficiency; a well-made Class D amplifier can achieve efficiencies above 95%.

Project Overview

Block Diagram

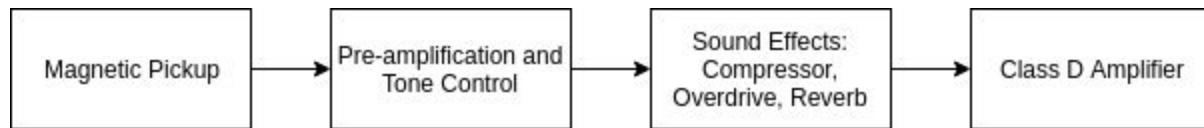


Figure 0.1: Block Diagram

Magnetic Pickup - Thanh Nguyen

Objectives

The first block of our project is the magnetic pickup. It creates a signal that copies the sound of the violin. The captured signal is output as a voltage. Additionally, the pickup should only capture the sound from the violin and not any other EM waves from any other sources. The functional requirement of this block is to create at least a 50 mV peak-to-peak signal with noise at least 1/10 the strength of the signal. The challenges of this block are ensuring that the captured signal is strong enough, the common mode noise from stray EM waves is small, and the sound from each string is equally captured.

High Level Implementation

The magnetic pickup is composed of a magnet and a coil. The pickup is placed underneath the strings of a violin. It's important to make sure that the strings are magnetically permeable i.e. steel and not gut core. As the string is vibrated to make sound, a small current is induced in the string due to it moving in the magnetic field of the magnetic. The small current induced in the string generates its own magnetic field. Since the string is constantly vibrating to make sound, the generated magnetic field from the moving string is also constantly changing. Since a change in a magnetic field within a coil generate a voltage, the coil produces a voltage that represents the constantly changing magnetic field and thus the sound of the moving steel string (Seering).

Schematic diagram of a simple pickup.

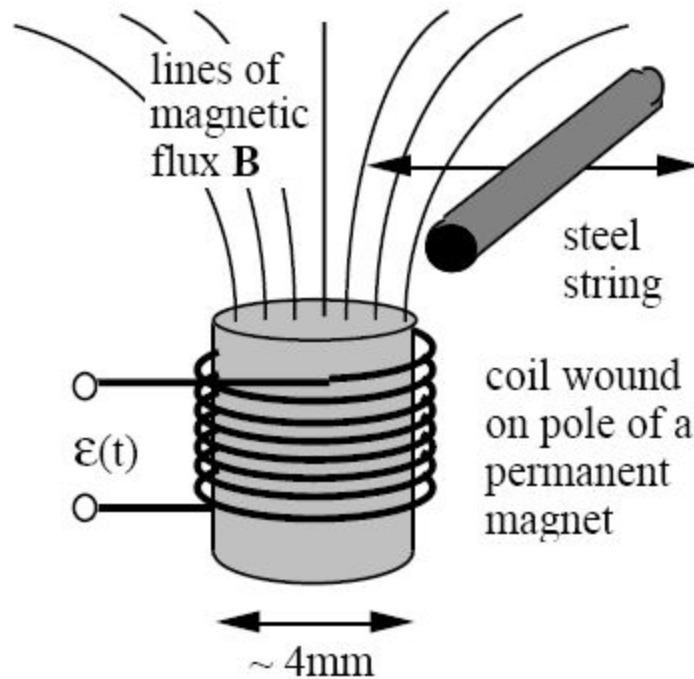


Figure 1.1: Schematic of a magnetic pickup
Photo Credit: audio-pro-central.com

Low Level Implementation

In the design of the magnetic pickup for the violin, there are three major issues. The first issue is that the magnetic pickup will capture any stray electromagnetic wave especially the 60 Hz AC hum from any outlet. This will induce a constant hum in the signal. To cancel out this common mode noise, we employed a humbucker configuration of coil. Essentially, instead of one big coil capturing the signal, we used two small coils -- one wound clockwise and one wound counterclockwise. The common mode noise appears over both coils and is attenuated. However, each individual string produces a local change in magnetic field and is not equally picked up by both coils. Our implementation of the humbucker configuration is shown below in figure 3.2. The violin has four strings. Under string G and D there is one coil wound clockwise. For string A and E, there is another coil underneath wound counterclockwise.

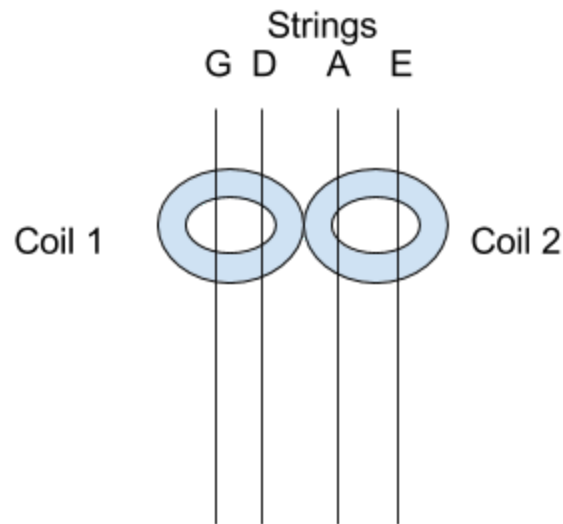


Figure 1.2: Top view of the magnetic pickup.

The next issue is more of a mechanical issue but is worth noting due to its importance to maintaining sound balance. For the sound to be captured equally on each of the violin string, we need to ensure the magnet can place an equal magnetic field at each of the violin string. However, on a violin, each string has an unequal height to the body of the violin, so a simple bar magnet placed underneath the strings will not work.



Figure 1.3: Note the distance of each string to the body of the violin

To solve this issue, four threaded standoffs is placed on top of a bar magnet. Each standoff is spaced out similarly to how the strings are spaced out. A bolt is inserted into each of the standoff. It's important that the standoffs and the bolts are magnetically permeable. The inserted bolt can adjust its own height by screwing down and up the standoff. With this setup, we can transmit the magnetic field from the bar magnet up through the standoffs and bolt. Since the bolt is height adjustable, we can ensure that an equal magnetic field is produced at each string.

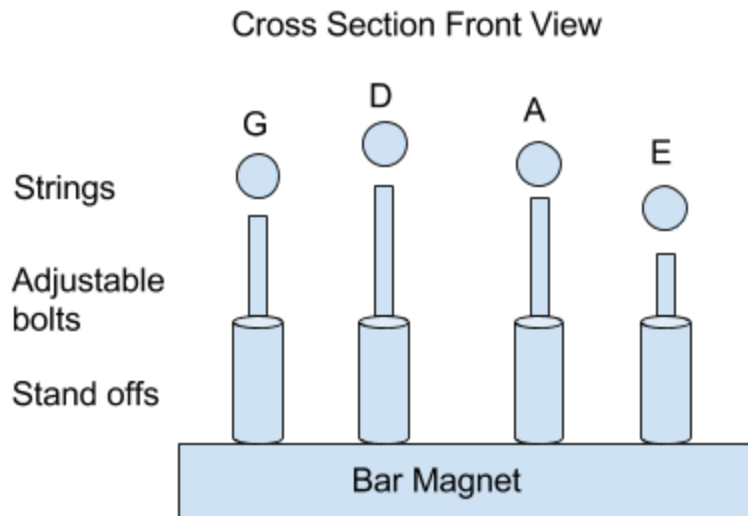


Figure 1.4: Cross section view of the magnetic pickup

The last issue in building the magnetic pickup is self-resonance. The pickup can be modeled as an LRC circuit. The coil is a loop of wires, so it will have inductance. The long length of wire needed to create the coil has resistance. Lastly, the close packing of the coil creates a capacitance. If the resonance is within the hearing frequency (20 - 20kHz), the voltage out would not be balanced across the full range of the violin. There would be frequencies where the sound would be louder and other softer. This brings forth a design constraint with the pickup. The pickup cannot just have a lot of windings to have a big output, but it also has to have a small enough amount of windings to create a self resonance outside the hearing frequency. In order to arrive at the correct amount of windings, an iterative process must be taken, but due to a time constraint, the amount of winding is decided by mimicking the design of another magnetic pickup (Jenn). The final design uses a N52 grade neodymium bar magnet and two coils of 4000 turns.

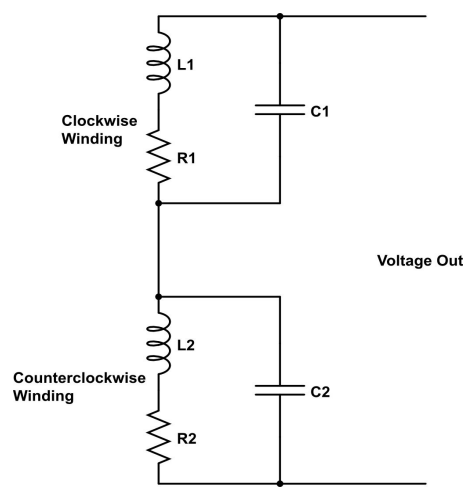


Figure 1.5: Circuit model of the magnetic pickup

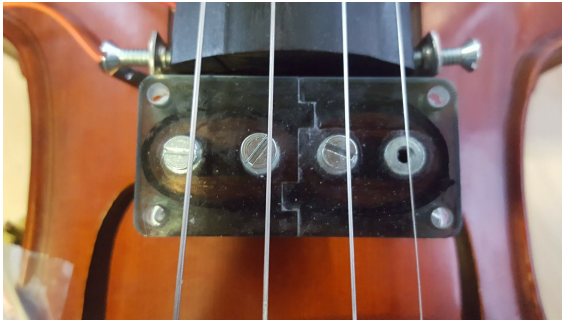


Figure 1.6: Top view of the magnetic pickup.

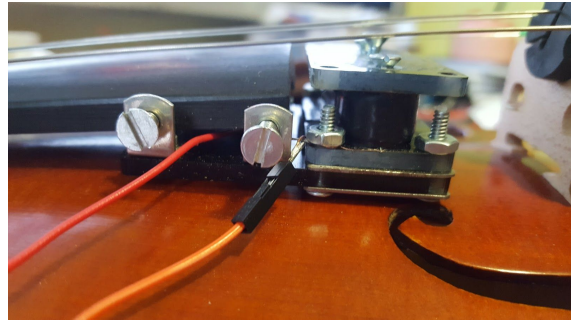


Figure 1.7: Side view of the magnetic pickup.

Results

From a rough hearing test, the higher frequencies, playing on the E string of the violin, have a slight roll off in volume. It's noticeable softer than sounds played on the G, D, and A strings (lower frequency). The output signal at a medium playing level is 200 mV peak-to-peak. Although untuned correctly, the next stage, the preamplifier, will help with equalizing the signal. During the winding process, we also accidentally missed counted, making one coil about 500 turns higher than the other one. Thus, we did not perfectly attenuated the 60 Hz AC hum. The noise signal is measured at 7 mV peak-to-peak in a normal room environment. However, the noise signal can get as high as 50 mV peak-to-peak when the magnetic pickup is near a transformer.

Preamplification and Tone Control - Thanh Nguyen

Objectives

The next step in our project is to amplify and filter the signal produced by the magnetic pickup. The output of the magnetic pickup is roughly 200 mVpp. We want to boost this signal up to line level which is 1 Vrms as the first functional requirement for this block. We want to filter the signal to only capture the signal from the hearing frequencies. Lastly, we want to give the performer more control over their instrument, so we will also have a tone control as the second functional requirement. It will be used to attenuate or boost the signal of the bass frequencies (10 Hz - 1 kHz) and the treble frequencies (1 kHz - 20 kHz).

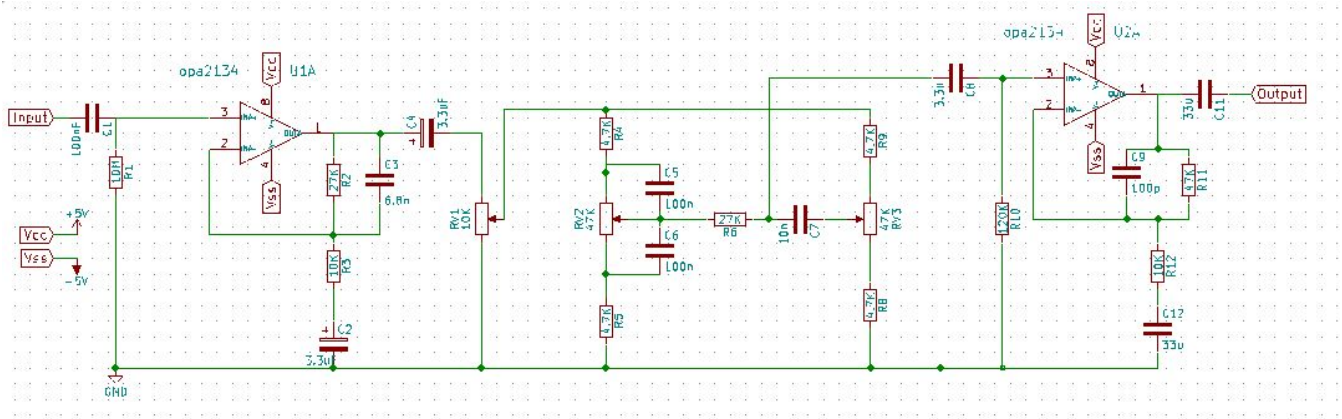


Figure 2.1: Overall schematic of this block

Preamplification Implementation

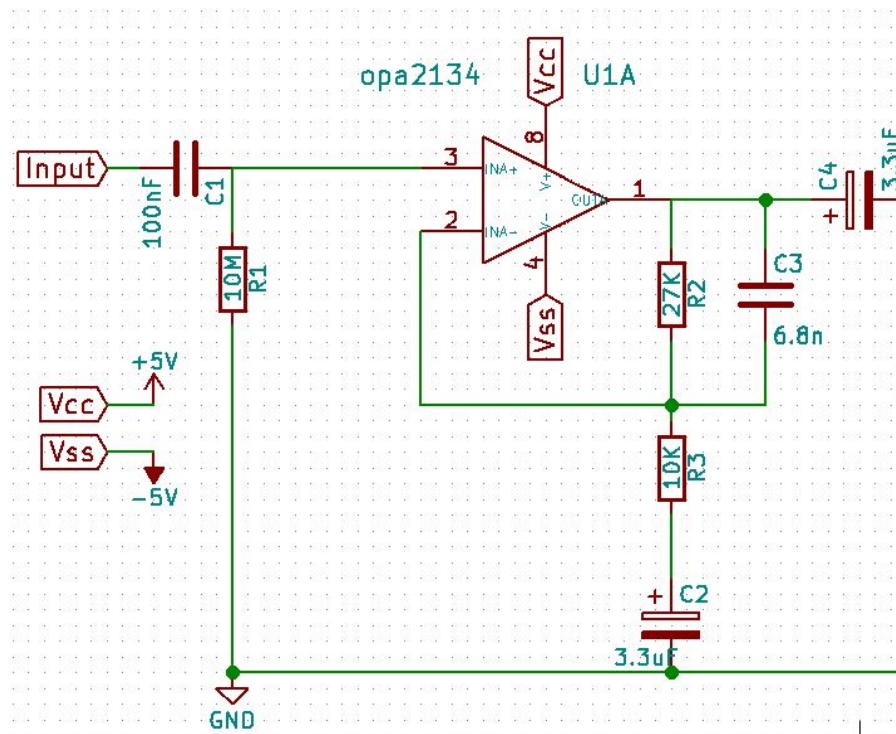


Figure 2.2: First Preamplifier

The preamplification circuit is very simple. It's just an op-amp in a non-inverting amplifier configuration with some capacitors to act as filters. One important aspect of this amplifier is its high input impedance, $R1$, which is at 10 MOhms. This is important because the output impedance of the magnetic pickup is in the 10s of kOhms, so to preserve the voltage signal, we have $R1$ at 10 MOhms -- two order of magnitudes larger than the output impedance of the pickup as is typically done for a preamplifier. With a high input impedance, the input current into the amplifier is low, so the signal will not be distorted from resistive lost.

Another advantage of a high input impedance is that it will act as a voltage divider along with the output impedance of the magnetic pickup. In effect, this high input impedance also equilibrate the output signal of the pickup even if the resonance is in the hearing frequency. The over gain of the amplifier is determined by R3 and R2. As for filtering the signal, C1 controls the high pass filter and C3 controls the low pass filter.

Tone Control Implementation

The tone control is replica of an Baxandall tone control network (Patron). First, the signal is fed into the network using a potentiometer RV1. This acts as an overall volume control, so it has to be a logarithmic potentiometer. In this network, the bass is controlled by RV2, and the treble is controlled by RV3.

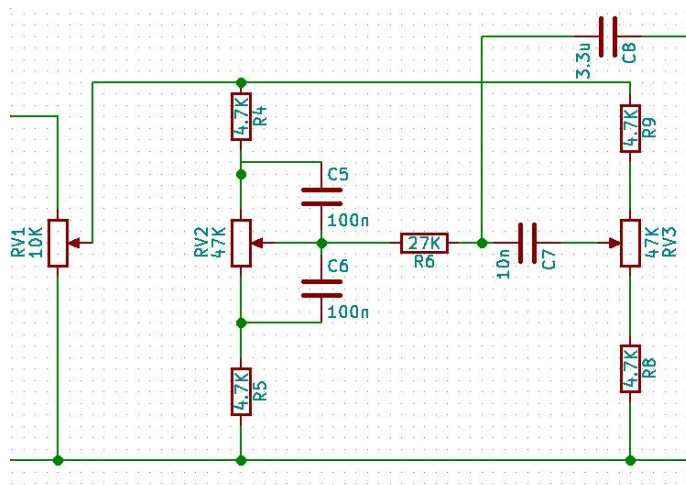


Figure 2.3: Tone Control Network

The operation of the circuit can be splitted into two parts. The left half of the circuit controls the bass response while the right half of the circuit controls the treble response. The circuit works by varying the cutoff frequency of the RC passive filter. The cutoff frequency is changed by varying the resistance of the potentiometer. See figure 4.6 and 4.7 to see how this concept apply to the bass control circuit.

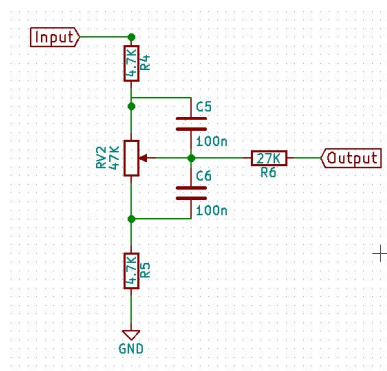


Figure 2.4: Bass circuit of the Baxandall network

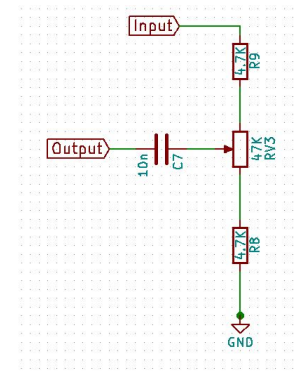


Figure 2.5: Treble control part of the Baxandall network

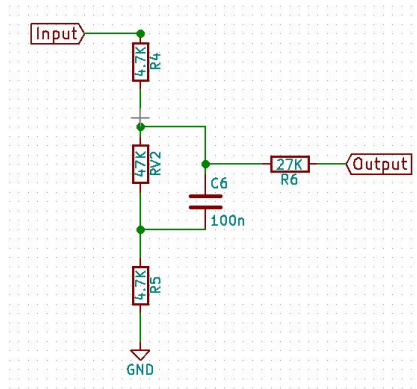


Figure 2.6: Bass control at full boost.

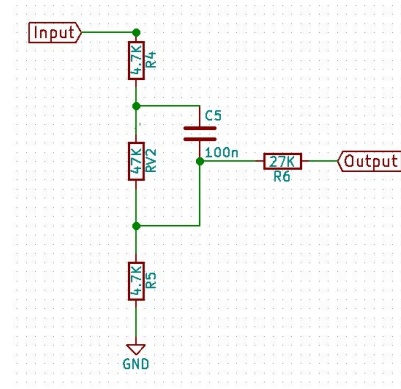


Figure 2.7: Bass control at full cut

The tone control circuit is designed to set the midpoint to around 1 KHz with the two cut off frequencies to be at 10 Hz and 100 KHz, each two decades away from the midpoint, when the potentiometer is half way turned.

Lastly, because the tone control network is a passive network, there is loss in the circuit. Thus the signal is boosted again through a non-inverting amplifier configuration. This works in the exact same way as the previous amplifier. The gains of the two amplifiers were chosen so that the output this stage would be line level or 1 Vrms.

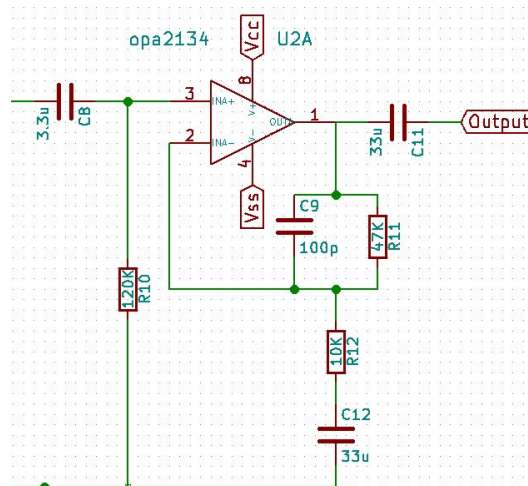


Figure 2.8: The second non-inverting amplifier

Result

As we were designing the tone control circuit, we forgot that the violin does not actually play any note in the bass frequency. The lowest note that the violin can play is only 200 Hz and not as low as 10 Hz. Thus, the bass tone controller is effectively useless since its cutoff point

was placed too low. However, the treble tone controller works just as expected. Using a function generator, frequencies input into this block at 1 KHz and above were attenuated or boosted by rotating RV3. However, changes in RV2 do not affect the signal at these frequencies.

Effects Stage - Peter Sudermann

Objectives

The effects stage of this project was designed to take the dry, harmonically barren signal from the pickup and add complexity in the form of tube compression and reverb. This stage would need to be very quiet, to avoid inducing extra noise into the audio chain. It would also have to limit the output to maintain a line level audio signal and prevent unwanted clipping in the amplification stage.

High Level Implementation

The compressor stage of the audio effects was designed from the start to use tubes, rather than transistors, for the musical quality they add to the sound. Tubes, even in their linear region, produce light second and third harmonic distortion. This replaces the harmonics normally produced by the environment and the acoustics of the wooden body of the instrument.

A compressor works by taking any input signal above a certain threshold and decreasing its gain. This gives the input signal less of a chance of clipping, and also increases the audibility of quieter, more subtle parts of the audio waveform, since the volume discrepancy between louder and softer components is less prominent. Tube compressors are highly popular in the studio environment, as their transition between gain profiles at the threshold (known as the “knee”) is smooth and non-linear, rather than the sharp, linear transition produced by many solid state compressors.

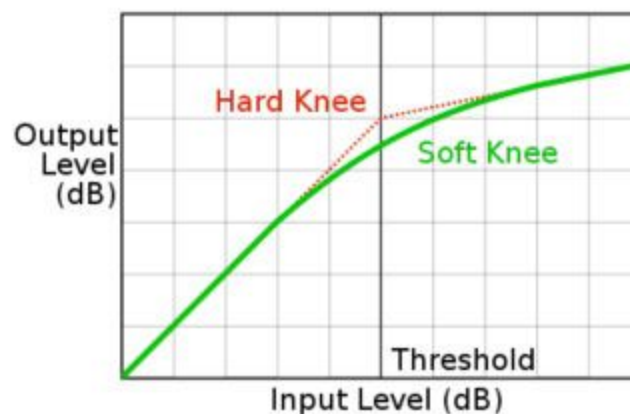


Figure 3.1: Compressor Basics

For this specific implementation, a dual stage triode compressor was placed at the input of the circuit. Triodes were chosen due to their relative ease of implementation and the harmonic spectrum they produce. This stage was placed at the input to insure that the input signal would not be hard clipped by an op-amp stage, since tubes produce less harsh distortion when driven beyond rail voltages. This distortion is known as “soft-clipping,” and produces a spectrum with mostly second and third order harmonics, and less higher frequency and odd order harmonics, resulting in a smooshed looking waveform that sounds more musical than the sharp clip of an op-amp.

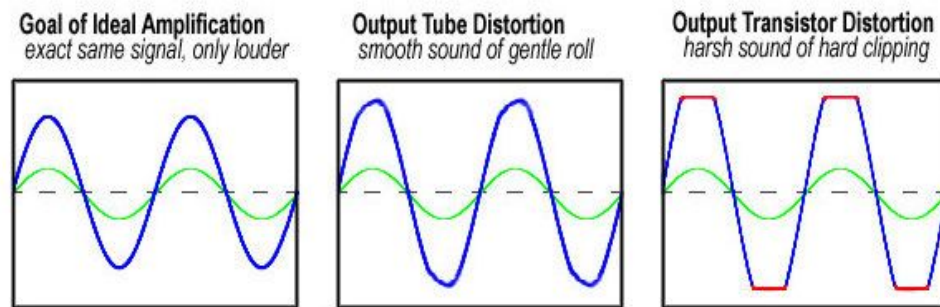


Figure 3.2: *Distortion Tubes Transistors 2*

Reverb effects are very difficult to produce in the analog domain, and nowadays are almost exclusively done in software. However, two popular forms of analog reverb are still in use today; plate reverb and spring reverb. Both of these designs make use of physical systems to transfer sound waves through a medium. In the case of spring reverb, audio is played through a long stretched spring via a transducer and picked up on the other end, creating a delayed signal and echoing back a mirror of that signal as the wave bounces back and forth through the spring.

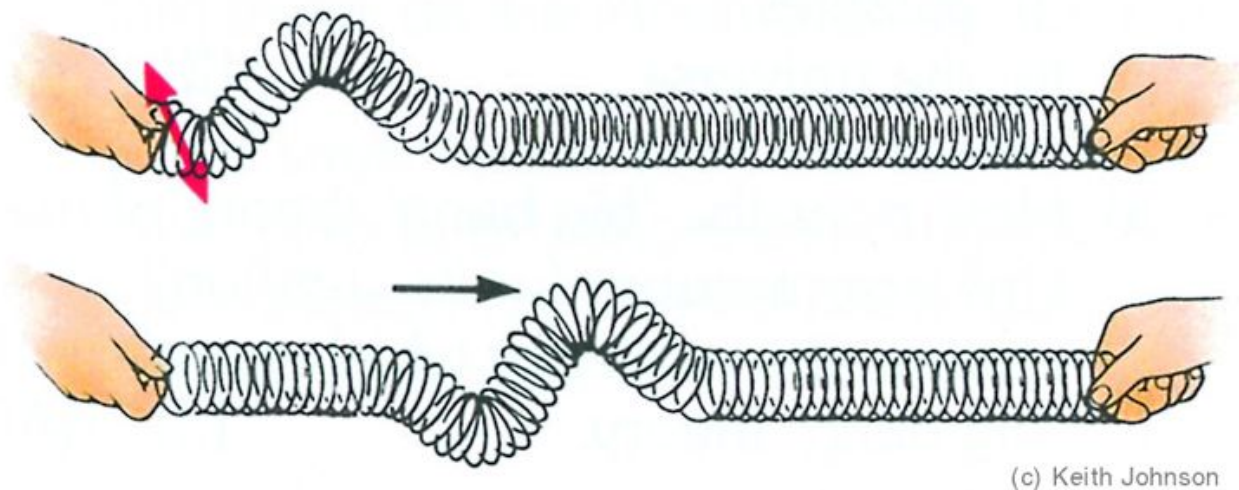


Figure 3.3: *PFY Waves Transversewave*

Not shown on the circuit diagram are the LM7805 and LM7905 linear regulators used to supply the rail voltages. These were chosen because of their exceptionally low noise floor and reasonable max current output of just over 1 amp.

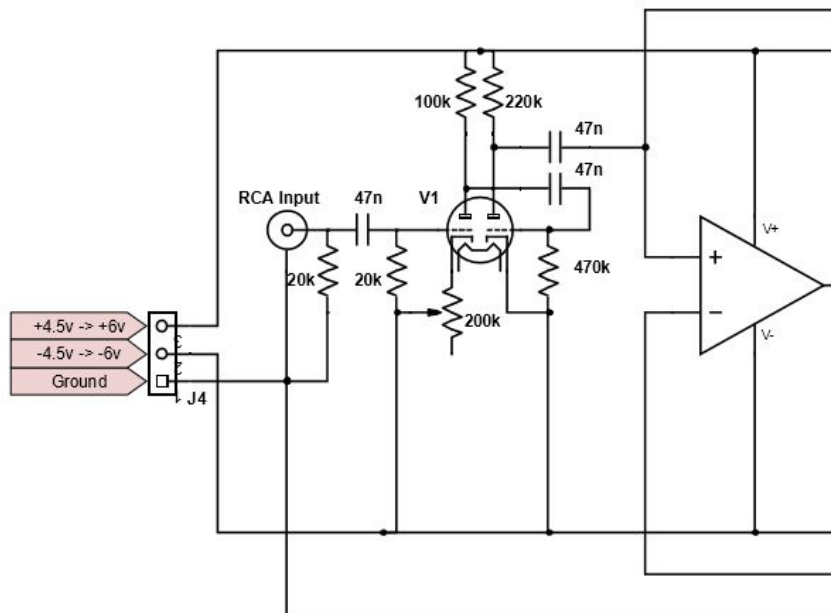


Figure 3.5: Dual Triode Input Stage

The audio signal enters through the RCA input at a max voltage of 1Vrms or 2.8Vp-p, tied to ground via a 20k load resistor, as per line audio specifications. This signal then enters a decoupling capacitor to AC couple the input to the bias on the grid. The grid is also tied to 20k, to provide a matching input impedance into the tube.

The first stage of the tube is the variable gain half of the compressor, which provides a gain equal to $\frac{\mu \times 100k || 470k}{100k || 470k + r_a}$ where r_a is the plate resistance and μ is the gain factor. In a ECC82, the g_m is approximately 20, and the plate resistance is equivalent to the internal plate resistance, 64.5k plus $(\mu + 1) \times r_k$. r_k is the cathode feedback resistor, which is a variable resistor from 10k to 200k. These values were chosen to create a gain range centered roughly around 0dB. As r_k varies, the gain of the stage varies as follows:

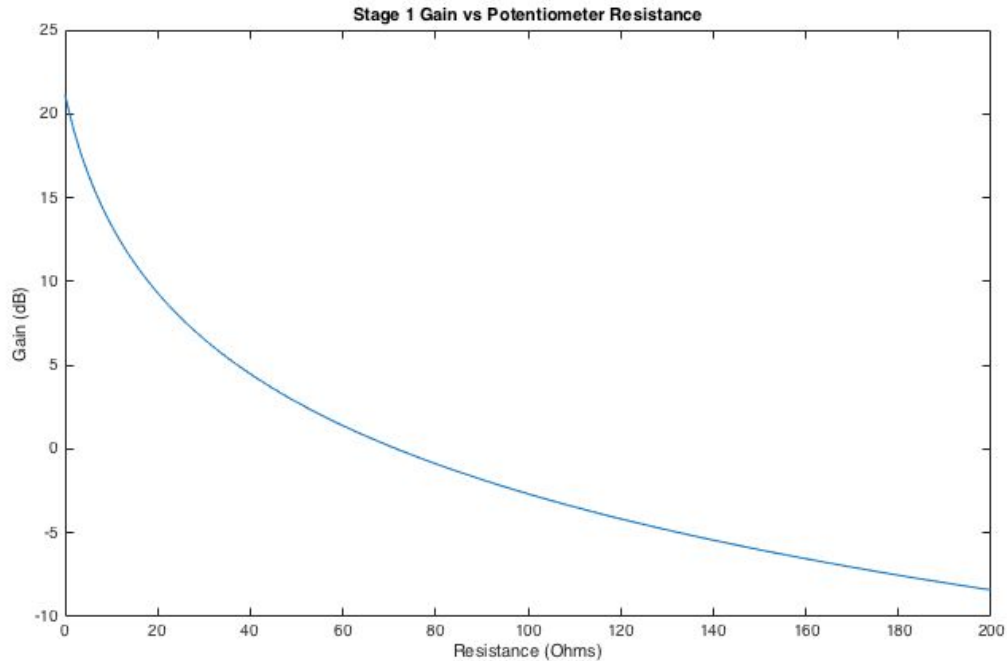


Figure 3.6: Compressor Gain Curve

To induce more tube distortion and center the threshold around -10dB, a second triode gain stage was added. Both tube stages had a voltage headroom of the rail to rail voltage of 10Vp-p, meaning the input signal would need a gain of around 11 to 12dB in order to fully clip. However, a EC882 triode begins to distort at around 60-80% of its rail voltage, meaning the actual signal gain would only need to be around 6dB. Adding in the -10dB requirement, the overall gain of the stage would need to be around 16dB.

The second triode stage omits a feedback plate resistor, which provides a greater gain equal to $\frac{\mu \times 220k}{220k + r_a}$, where r_a is purely the internal plate resistance of 62.5k. This simplifies out to approximately 15-16dB of gain, satisfying the requirements and creating an output signal of approximately 6Vp-p.

Both of these stages induce 2nd and 3rd harmonic distortion even before the compressor threshold. These harmonics are applied unequally to the positive and negative halves of the waveform due to the biased and nonlinear nature of a triode gate. This results in waveforms similar to those below (original signal in blue, triode output in yellow):



Figure 3.7, 3.8: Scope View of Added Harmonics

For the reverb stage, a current amplification circuit was needed in order to properly drive the spring tank. A spring tank driver consists of a coil wrapped around ferrous material bent over as seen below. This acts like a solenoid, and magnetises each end of the ferrous material in opposite polarities, pulling the magnetic bead one direction or the other to vibrate the spring.

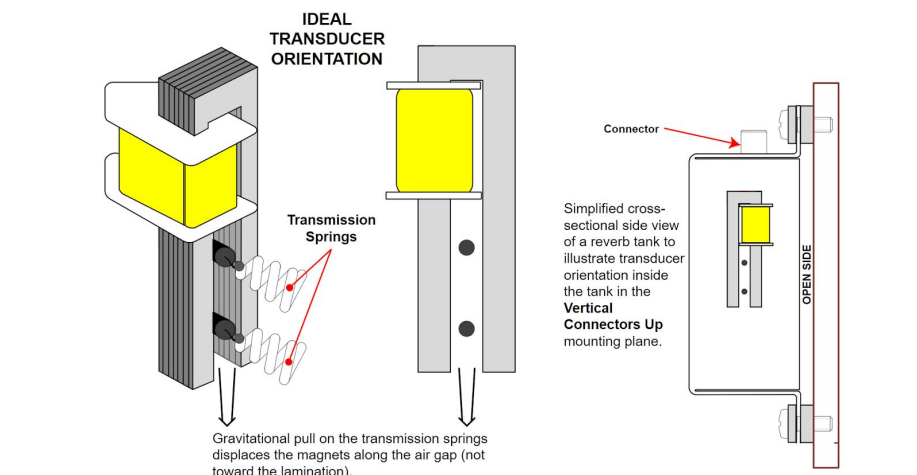


Figure 3.9: Ideal Transducer

In order to determine the scale of current amplification needed, we had to determine the impedance of the coil at all relevant frequencies. The coil chosen was spec'd at 150 ohms at 1kHz, and we needed coverage from 160-200 Hz all the way up to at least 8kHz in order to reasonably cover the root, 2nd, and 3rd harmonic range of the violin.

Unfortunately, a specifications sheet was not available for this specific reverb tank, so an impedance curve for a 1.5k tank was used to approximate:

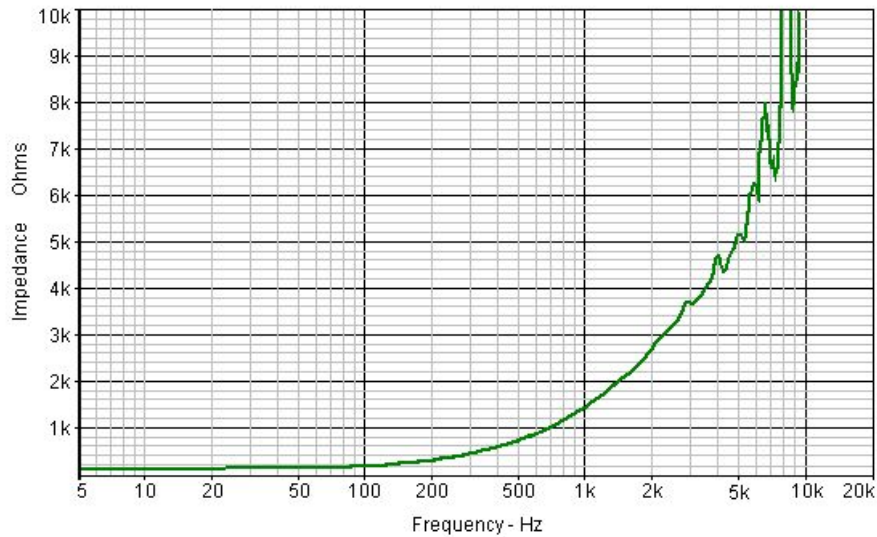


Figure 3.10: Accutronics Reverb Tank Drive Coil Impedance

Down at 160-200Hz, the impedance is roughly 150 ohms, or a tenth of the 1kHz impedance. Extrapolating, we can assume the 150 ohm tank has an impedance of roughly 15 ohms at this frequency. Since recommended voltage used to drive this tank is around 5-6 volts over the whole spectrum, the amplifier required voltage control feedback and a max current output of greater than 400mA. However, since the reverb naturally blurs sound, distortion was not a huge concern in choosing an amplifier.

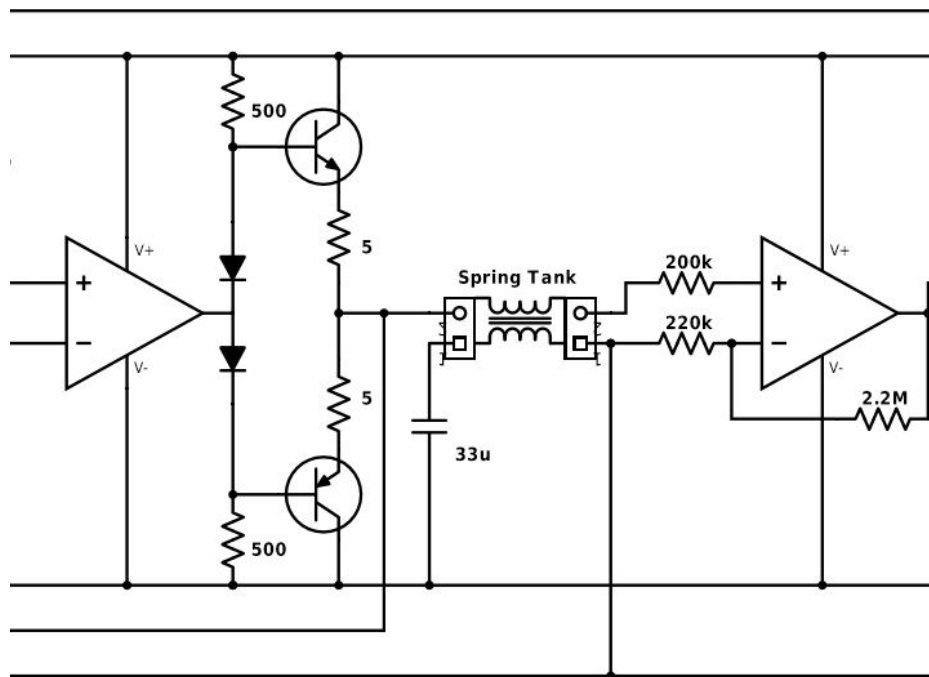


Figure 3.11: Reverb Tank Driving and Recovery Circuit

For the amplifier stage, a simple AB circuit was selected for its relative efficiency and reasonable power output. The transistors chosen were 2N2905 and 2N2219 for their high current output of approximately 600mA, high beta value of approximately 100, and relative abundance in EDS. To provide short circuit protection, 5 ohm, 10 watt resistors were used on the output to dissipate the 5W of heat produced when +/-5V output shorts to ground. These resistors create a voltage divider at of around .75, meaning the op amp need to provide at most 8Vp-p, which is within operating range of an OPA134 op amp. With a beta value of 100, the current to base required is around 4mA. Experimentally, it was determined that the output stage bias current required to prevent clipping was around 10mA, necessitating two 500 ohm resistors. Each resistor dissipates at most 50mW at idle, meaning regulat 1/8 watt resistors could be used.

On the recovery stage, the tank output was around 500mV at absolute maximum, and 20mV minimum. To prevent clipping at high output, a preamplifier with a 10x gain was determined to be the highest safe value. Since spec'd output impedance of the spring reverb tank was listed as 2.25k at 1kHz, so a 100 to 1 input impedance ratio necessitated a 220k input resistor, and thus a 2.2M feedback resistor. A 200k resistor was added to the non-inverting input to reduce bias caused by impedance mismatch.

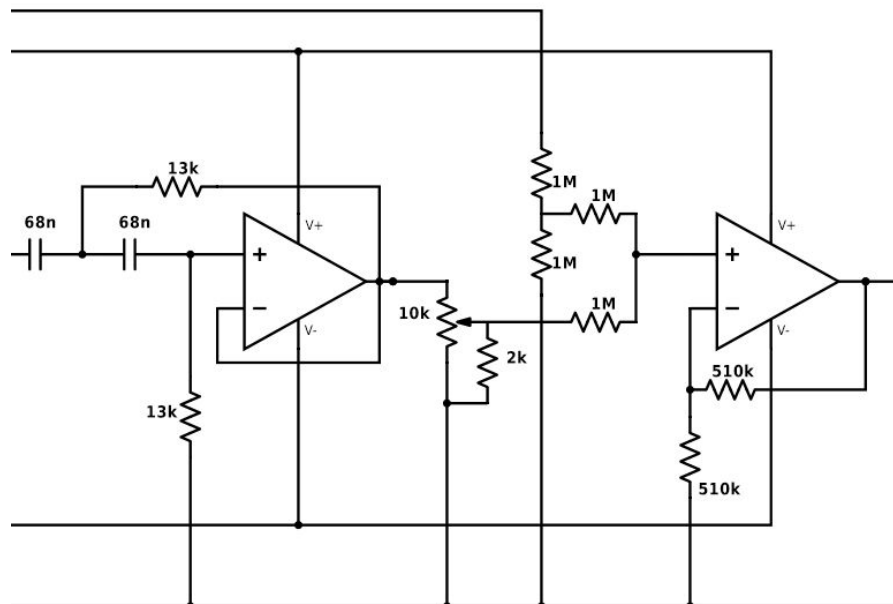


Figure 3.12: Sallen Key Filter and Mixer

To reduce the noise caused by bumping and jostling the spring tank, it was necessary to include a Sallen Key high pass filter. This filter was designed with a crossover at 180Hz, providing reasonable response around 160-200 Hz, the lowest note producible by the violin. As this was a second order filter, 80Hz handling noise was rendered essentially non-existent.

Following the filter, the mixing stage was needed to mix the raw, “dry” signal from before the reverb stage and the muddy, “wet” sound produced by the reverb tank. A 10k potentiometer in a semi-log configuration provided reasonable adjustment of the wet output across the full range of 0-100% signal level. The dry stage fed through a fixed voltage divider, reducing output level to 50% max (note this voltage divider does not connect to the lower resistor of the wet input). This was done to prevent clipping on the summing output, but also to bring the dry signal down to the nominal value of the wet signal, as the wet signal only peaks at 5Vp-p at certain frequencies, but is nominally around 1-3Vp-p. Since the adder acts as an averaging circuit in the gain of 1 configuration, a gain of 2 was selected to maintain signal strength. The resistor chosen for the voltage division and input stages were all chosen to be 1M, as this is a factor of 100 greater than the output impedance of the 10k potentiometer.

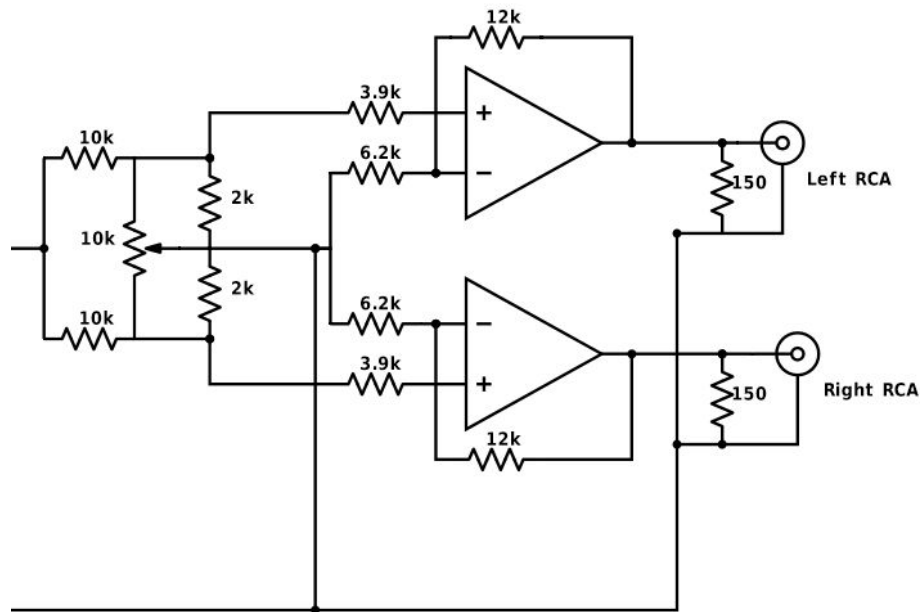


Figure 3.13: Active Pan Pot Output Stage

For the panpot stage, the goal was to create a low loss resistor network with an S-taper output, also known as a double log taper, as well as normalize the signal to 1Vrms line level.

An S-taper output was chosen as it best matches the gain of sounds from ear to ear in real life, and is the preferred configuration used in almost all audio applications. To normalise the signal, a simple 10k resistor divider was added to the inputs of the 10k pot to create a max gain of 0.5x, reducing the 5-6Vp-p signal to 2.5-3Vp-p, which is close to the 2.8Vp-p (1Vrms) of line level. To achieve the S-taper configuration, 2k resistors were added to bridge the center of the pot with each signal line, and then the center of the pot was tied to ground. This creates a variable voltage divider network. That allows the panpot to control output to each side of a stereo output. However, this network also adds an additional voltage divider in the form of the 2k resistor and the non-inverting input resistor, reducing gain again. The non-inverting op-amps are designed to recover that gain, and were manually tuned to ensure the output to each RCA

never exceeded 2.8Vp-p. The outputs of these op amps were tied to ground with 150 ohm resistor, to match the output impedance specified by line level standards.

Results

Every component of the effects stage functioned as intended upon initial construction, successfully passing an audio signal and processing effects as intended. However, it initially suffered several problems with noise and frequency response. These issues were corrected over the course of a week of repeated measurement and refinement.

The stage was initially powered off of a 10v supply with a resistor divider creating the virtual ground. After the tube stage was completed, the measured noise on the output was considerable, as tiny fluctuations in the anode and cathode voltages of the tube caused large fluctuations in output noise. In addition, the AB amplifier stage caused the virtual ground to swing wildly, producing distortion, reducing low to mid frequency response, and causing noise spikes in the tube stage.

To correct for this, a linear power supply was constructed using LM7805 and LM7905 regulators and the output of the AB amplifier was capacitor coupled to the negative rail, rather than the virtual ground, dramatically reducing noise floor and practically eliminating the distortion caused by rail swinging. This capacitor coupling of the input was only possible as the spring reverb tank has a floating ground input, which is not tied to the metal chassis.

With the noise floor reduced to around -35 to -38dB, another problem was revealed. The tube stage was picking up rf interference, as well as emf interference from power supply transformers. This was solved by adding a thick ground plane across the bottom of the breadboard, as well as changing the input from the voltage supply from 14 to 15.5 volts, which changed the switching frequency of the supply transformer to avoid resonance with the tube. These additions decreased the overall noise floor to around -48 to -51dB, hitting our stretch goal.

All other components of the circuit functioned as expected and as designed.

Amplification Stage - Chetan Sharma

Objectives

The amplifier stage of this project was designed to take in a standard line-level signal from any stereo audio source and amplify it to levels that would permit one to drive large speakers at a high volume. The stage would have to introduce minimal distortion and have an

even frequency response; in addition, it would have to provide volume control and allow for custom speakers to be used.

High Level Implementation

The architecture of choice for this project was a Class D design. A Class D amplifier, unlike a Class A/B/AB amplifier, does not use linear modulation to amplify signals; rather, it functions by creating a high-frequency PWM signal with a duty cycle modulated by the input. When this PWM signal is filtered, the output closely follows the input signal.

This style of amplifier has numerous advantages. It does not depend on most of its components operating within a linear region, so it exhibits an even frequency response. Its switching nature also allows it to operate at extremely high efficiency levels; since the switching elements are always completely off or completely on, they dissipate little power in their operation.

Our implementation of the Class D architecture works by taking two input signals and comparing them to a triangle wave with a frequency far above the audible range. When the signal level is zero, this comparison yields a PWM signal with a duty cycle of 50%. As the input voltages rises and falls, however, the duty cycle changes with it in a linear fashion.

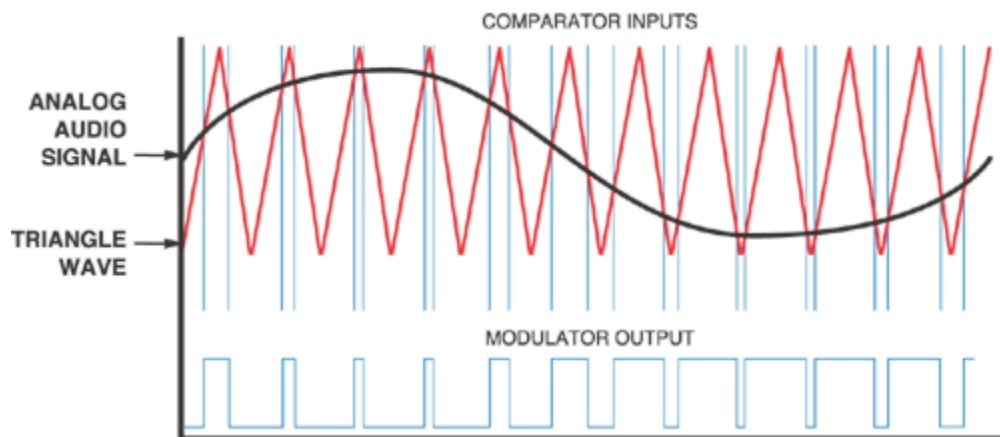


Figure 4.1 - Photo Credit: Steve Somers of Extron Electronics

These PWM signals can then be fed into two H-Bridge configurations of power MOSFETs. This produces two extremely low impedance differential signals that, when filtered, will faithfully reproduce the inputs.

Low Level Implementation

The design was implemented as follows:

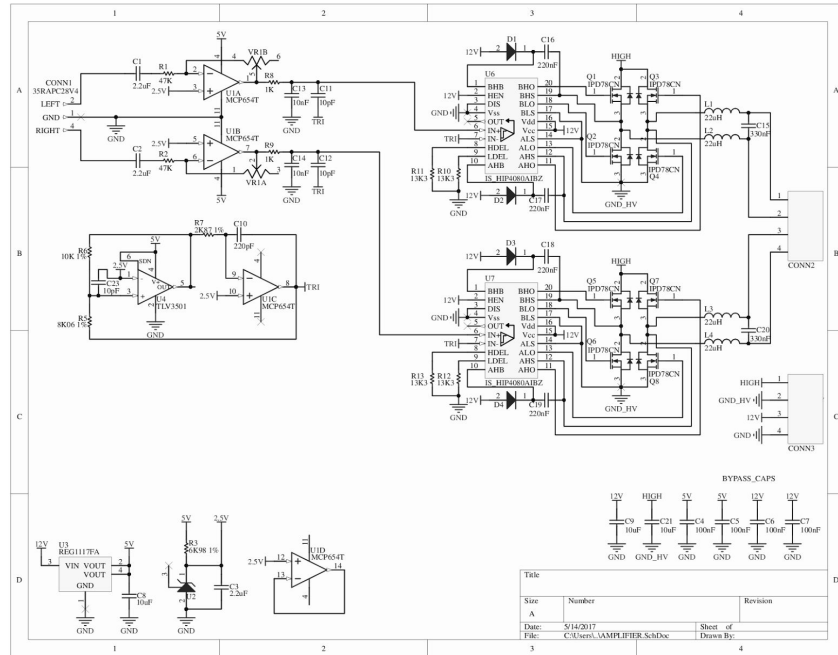


Figure 4.2: Overall schematic

Power for the design is taken in through CONN3. A 12V rail and a HV rail are needed; the former powers the circuit's logic whilst the latter is fed into the H-Bridge to run the gate drive.

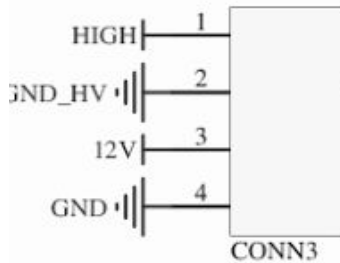


Figure 4.3: Power inputs

A 5V rail is created through the use of a linear regulator, and an additional 2.5V reference rail is created using a precision zener reference IC.

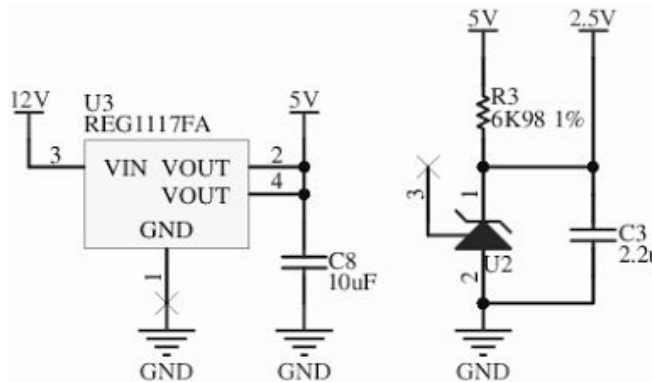


Figure 4.4: Voltage Regulators and References

The signals enter the amplifier through nodes LEFT, RIGHT, and GND. They are immediately passed through bypass capacitors C1 and C2. When in series with resistors R1

and R2, they allow the amplifier to exhibit an input impedance of 47k Ω with a low frequency cutoff of 3.18Hz. The signal is then fed into a high speed quad-package op-amp.

Both op-amps are in the inverting configuration with a false ground at 2.5V (generated elsewhere). In this configuration, their output is centered around 2.5V (allowing for inputs below ground). The amplification is set by VR1, a dual-gang logarithmic potentiometer. The dual-gang design allows both channels to be set to the same volume, and the logarithmic nature of the potentiometer causes the perceived volume changes to be linear (as human hearing perceives volume on a logarithmic scale). The log-pot's maximum resistance of 100k Ω and the input resistor's resistance of 47k Ω allows our inverting amplifier to vary between 0x gain and ~2x gain. An op-amp with a low noise value ($<10\text{nV}/\sqrt{\text{Hz}}$) and a low voltage offset ($<100\mu\text{V}$) was chosen to minimize audio distortion.

After being buffered, the signal is fed through a first-order filter designed to remove any noise above 15.9kHz. This is simply a precaution to avoid any input noise that may have been introduced into the signal.

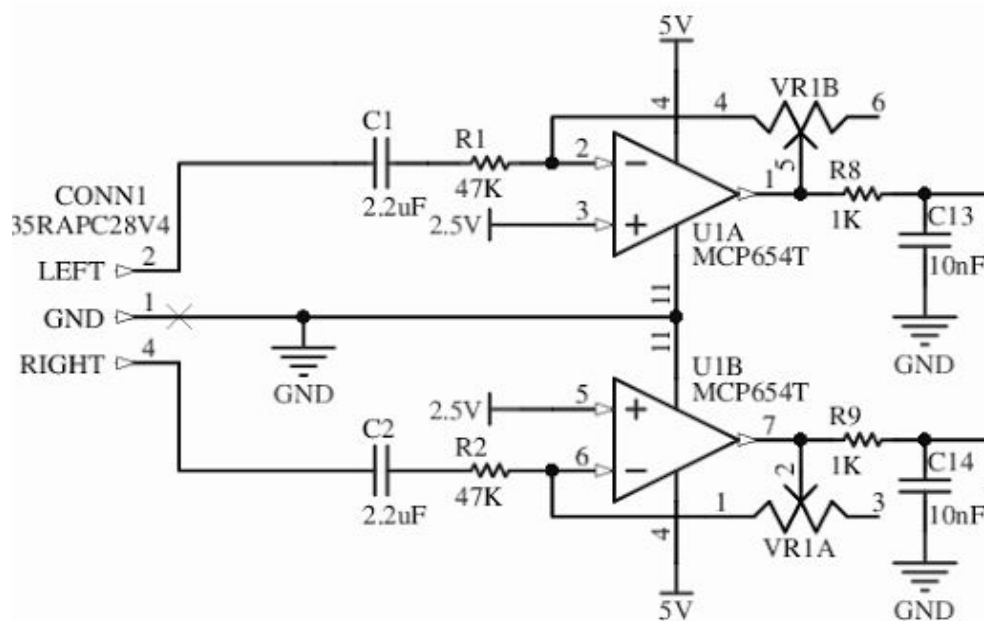


Figure 4.5: Input buffering, preamplification, and biasing.

An accurate triangle wave with minimal distortion is needed to generate a PWM signal. Its frequency must be sufficiently high to permit the usage of a first order output filter; for this reason, a frequency of 500kHz was selected.

The triangle wave generator consists of a hysteretic comparator producing a square wave and feeding it into an op-amp integrator. The integrated square wave is centered about 2.5V and has an amplitude of 2V.

Both the op-amp and comparator operate at high frequencies, so the comparator was selected to have extremely low switch times and the op-amp was selected to have an extremely high gain bandwidth product ($>10\text{MHz}$, as the fourier decomposition of a triangle wave shows that the highest significant harmonics are around 20 times the base frequency).

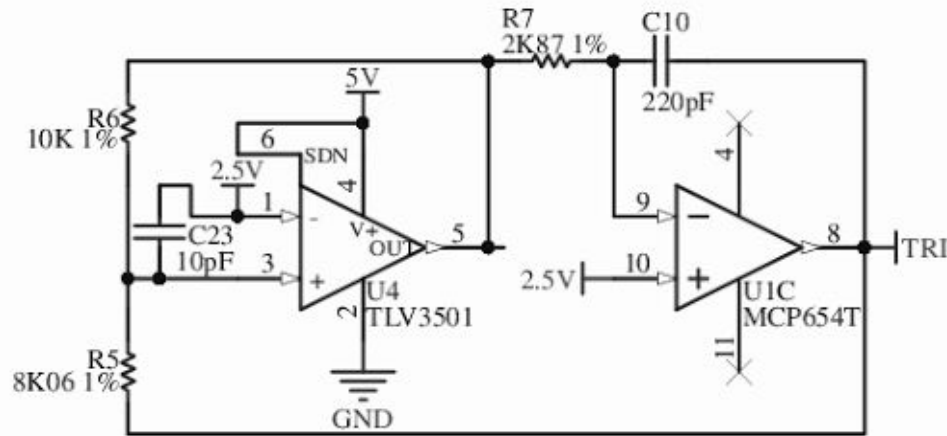


Figure 4.6: Triangle Wave Generator

Both the triangle wave and buffered signal are fed into a high speed comparator to create a varying duty cycle (this comparator is conveniently built into the gate driver). Both comparators have a 10pF capacitor across their inputs to introduce hysteresis. The gate driver drives an H-Bridge configuration of N-Channel MOSFETs; this allows both terminals of the speaker to swing from rail to rail and allows us to forgo an output filtering capacitor.

The gate driver requires an additional bootstrap capacitor/diode to power the high side MOSFET gates; with the addition of these, the HIP4080A can hold the gates at 12V above the high voltage input. A 220nF bootstrap capacitor and a Schottky diode were used for this portion of the circuit.

The gate driver also has a programmable “dead time” set by R10/11/12/13. Proper resistor values are critical here, as setting the dead time too low could result in MOSFET shoot-through and the possible catastrophic failure of the device. 47k Ω resistors were selected to provide a dead-time of around 18ns (equivalent to the maximum turn-off delay of the MOSFETs).

The MOSFETs dissipate power during continuous conduction and during their switching transitions. Assuming 1M switches at $\sim 22\text{ns}$ per switch, the MOSFETs will dissipate less than a watt of average power at full power; as a result, heatsinking was deemed unnecessary. Regardless, thermal dissipation was aided by the MOSFET’s drain connection to the board..

After this, an LC filter is used to filter out the PWM carrier with minimal power loss. The values used will allow the circuit to have a cutoff at 40kHz with a Q of 0.707 with a 8 Ω load on the output; this results in a smooth frequency rolloff with no spikes near 40kHz.

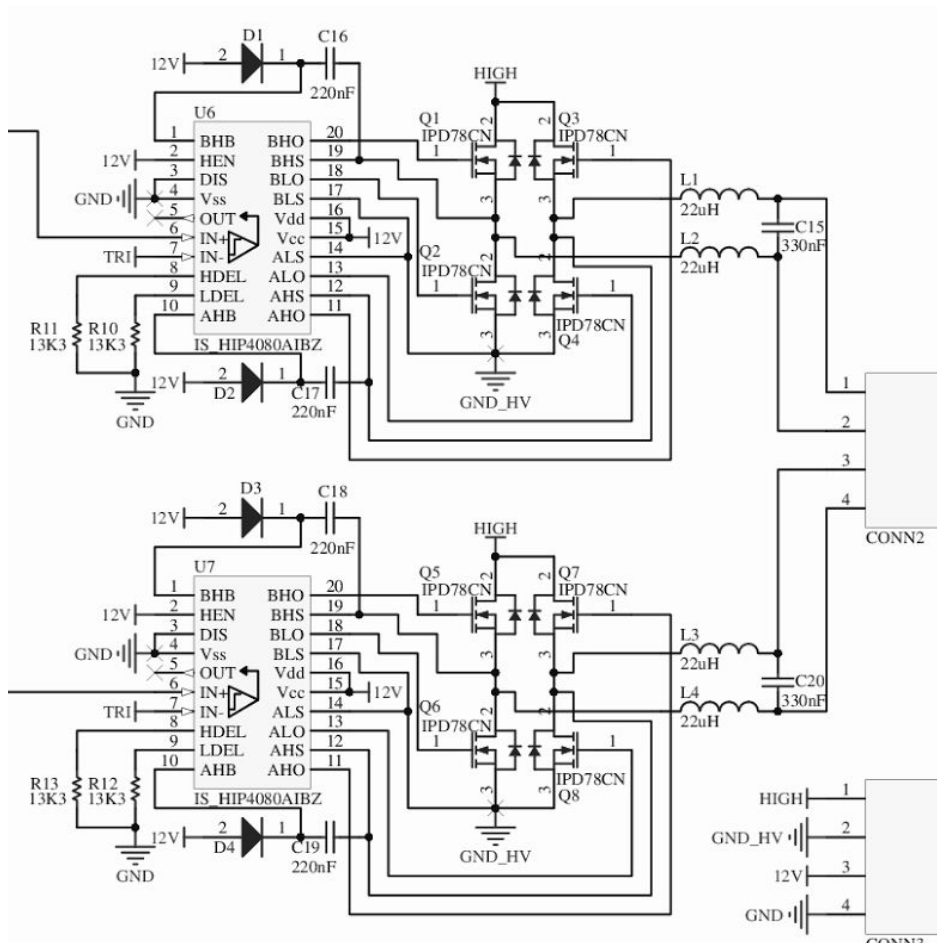


Figure 4.7: Gate Driver, Full Bridge & Output Filtering

0.1uF decoupling capacitors were placed across all IC components, and 10uF bulk capacitances were placed on both power rails.

Physical Implementation

The high-frequency switching nature lends itself to producing electrical noise; if long wires were used in the design, electrical ringing may have resulted and compromised the quality of the system. For this reason, a PCB was utilized. With a PCB, components in the SMD form factor could be used and connected with short trace lengths that minimize stray inductances. A ground plane was also used to reduce ground loops, reduce coupling, and shield the board from EMI. Great care was taken to route as many connections as possible on the front of the board to maintain the continuity of the ground plane. Components interoperating at high speeds were placed in close proximity to each other, and large traces were used to route high-amperage connections.

The PCB was designed in Altium Designer.

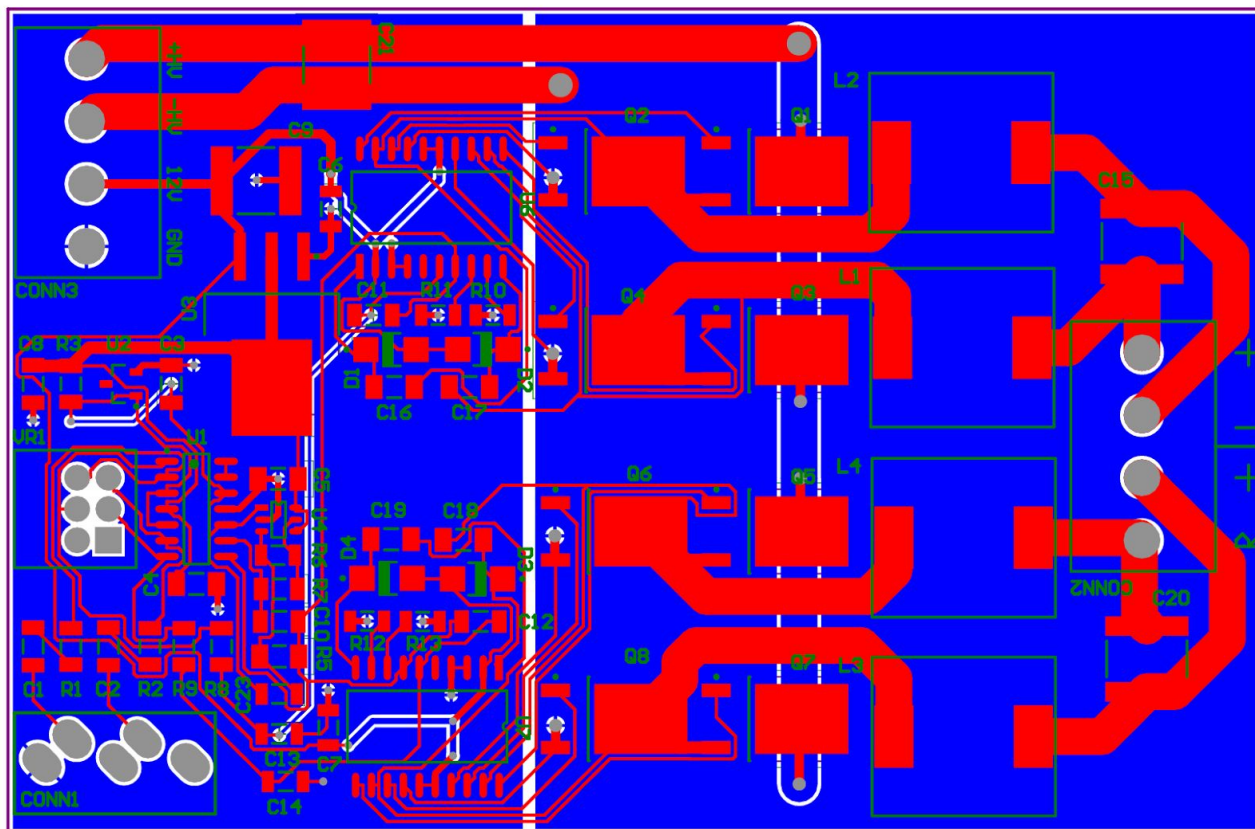


Figure 4.8: Full PCB Design.

In order to test the amplifier's subsystems, small boards containing a subset of the final schematic were manufactured through isolation routing at EDS and tested for functionality. Two such modules were produced. The first board contained the input buffering stage and the triangle wave generator whilst the latter stage contained the gate drive and output filtering. These boards revealed that a few corrections needed to be made to the board: a SHDN pin on the comparator had to be rewired, the grounds of the HV input and the 12V input had to be connected, and gate resistors had to be added to the MOSFETs.

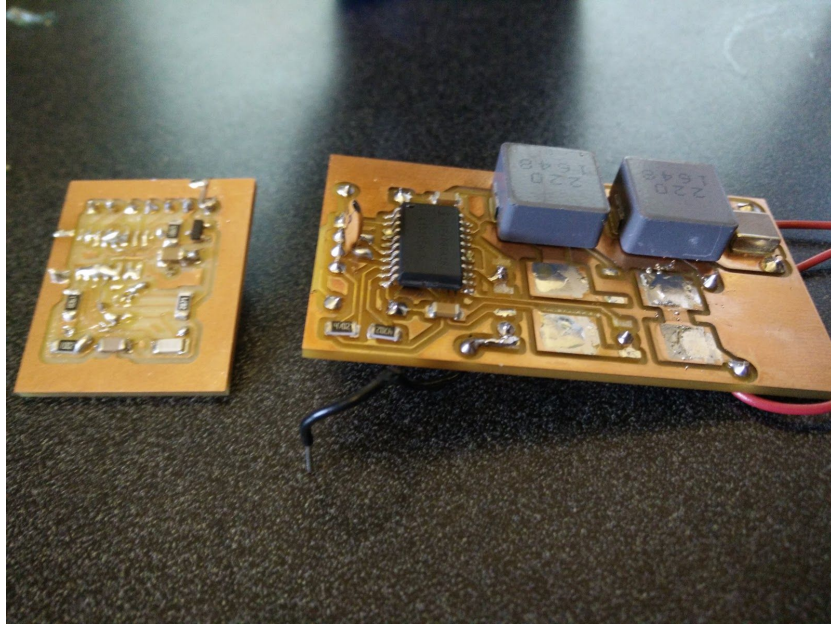


Figure 4.9: Modular Test Boards

The final board was soldered with a reflow oven to ensure that every component would be correctly soldered without shorting; through hole components and botch wiring were added afterwards.

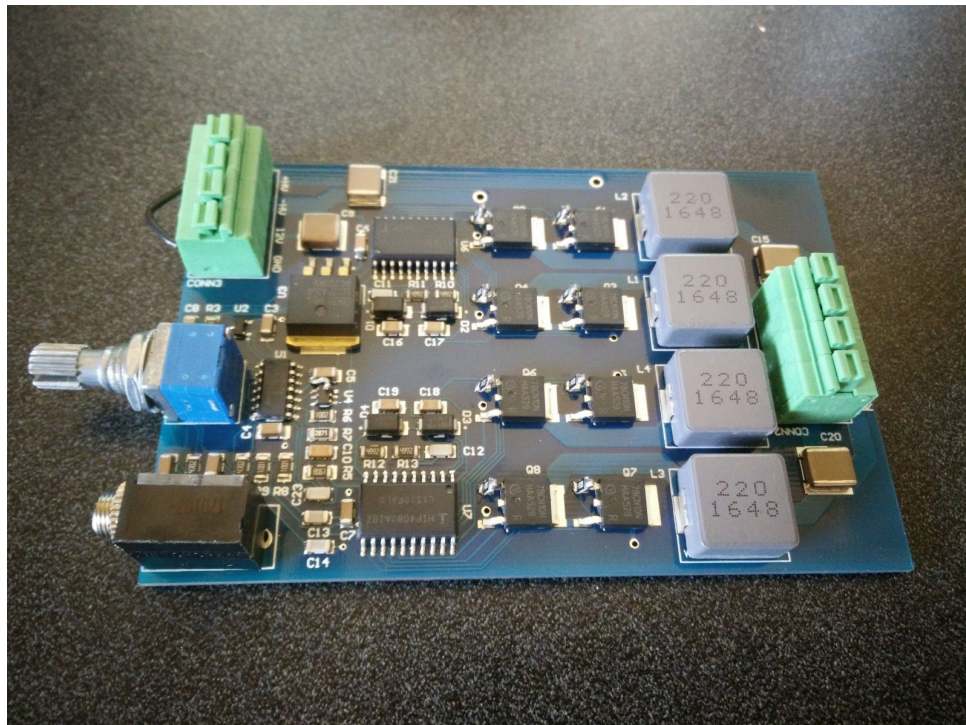


Figure 4.10: Completed board.

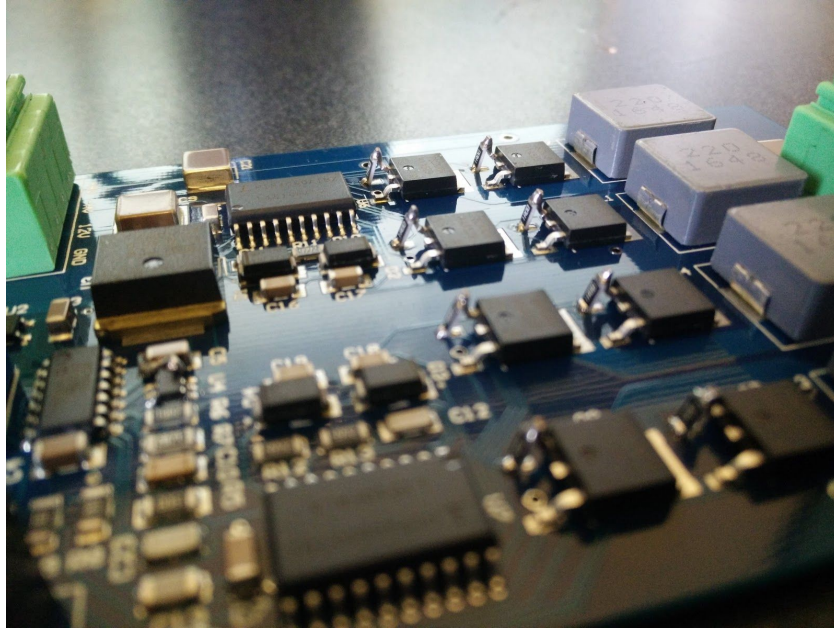


Figure 4.11: Additional picture.

Results

The board functioned correctly upon first being tested. The frequency response was audibly satisfying and an oscilloscope measurement proved it to be flat within the desired region. Erroneous noise was minimal and could only be heard when one pressed their ear to the speaker. A miscalculation regarding the RMS voltage across the speakers led to the amplifier delivering less power than expected; in addition, the usage of Nha's test speakers (which we discovered to be naturally quieter than normal speakers) led to a lower volume output. We believe that a higher input voltage could solve the prior two problems.

During later testing, it was discovered that the dead time used in the gate drive was too low to prevent shoot-through; as a result, the amplifier suffered a breakdown when input voltages were increased due to thermal runaway. The board was rebuilt with larger dead time setting resistors (to increase the time between switching) and smaller gate resistors (to decrease actual switching speeds) and worked reliably afterwards.

Conclusion

Final Project Results

The final project functioned mostly as expected with few problems. The magnetic pickup and preamplifier successfully picked up notes played from the violin with a good deal of accuracy and with minimal hissing. The tone control and effects stage manipulated audio as expected and the amplification stage made the audio audible. Some components were slightly

unreliable and required some degree of fine-tuning to function, but the system could consistently be used to capture, process, and amplify the sounds coming from a standard violin. By our standards, this was a success!

Final Thoughts and Insights

Modular design and an iterative approach proved to be key to this project; by breaking the problem into multiple stages, work was better divided and integrated with little wasted effort. The final project proved to be only a subset of the goals described in the project abstract, but this was deemed to be reasonable given tight time constraints. A long period of testing proved to be vital for many of the modules in this project; modules that were left untested proved to be the most unreliable. Future iterations of this project will likely see modules more tightly integrated and built in a more rugged fashion.

Citations

Compressor Basics. Digital image. *Audio Compression Basics*. Universal Audio, n.d. Web. 18 May 2017.

Distortion Tubes Transistors 2. Digital image. Milbert, n.d. Web. 18 May 2017.

Ideal Transducer. Digital image. *Big Surf Radio*. Oberlin, n.d. Web. 18 May 2017.

Jenn. "Crafting Pickups Is an Art." *Electric-Style Pickups*. N.p., n.d. Web. 18 May 2017.

Johnson, Keith. *PFY Waves Transversewave*. Digital image. *Anrophysics*. Patana, n.d. Web. 18 May 2017.

Jonathan, A. Honda, J. (2005). Class D Audio Amplifier Basics. (Application Note AN-1071). Retrieved from Infineon website: www.infineon.com

Patron, Ramon Vargas. "The James-Baxandall Passive Tone-Control Network." *Dave's Homemade Radio*. N.p., 23 July 2004. Web. 18 May 2017.

Seering, Adam. "Building the Pickup." *MIT OpenCourseWare*. MIT, n.d. Web. 18 May 2017.

Elliott, Rod. *Accutronics Reverb Tank Drive Coil Impedance*. Digital image.

Care and Feeding of Spring Reverb Tanks. Elliott Sound Products, Oct. 2014. Web. 18 May 2017.

Somers, S. *Class D Amplifiers - Not 'Just Audio'*. Extron Electronics. Digital image.