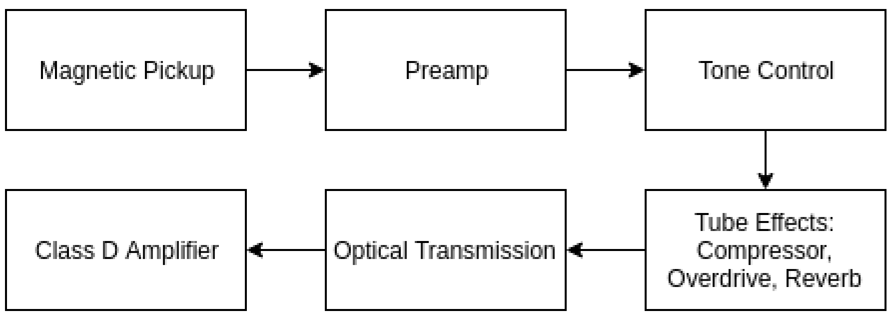
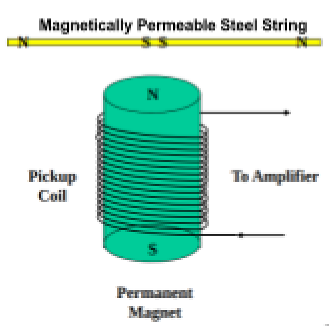
6.101 Project Proposal

The integrated system is consisted of a magnetic pickup, miniature battery powered preamp, ADC stage, long optical transmission line, DAC stage, effects mixer-compressor, and power amplifier. The goal of the system is to allow for multiple preamp devices and long transmission lines for large concert hall performances. The system should be compacted as to not impede the performer, and the transmission line should be robust enough to not introduce any noise along lengthy runs. These transmission lines will converge to a centralized, multi input mixer and effects stage to be processed before being sent to the speaker outputs.Stretch goals of this project would be having a PCB preamp/ADC stage, a tube based compressor/overdrive system using staved cathode configuration, and a powerful enough output stage to drive a small stage monitor speaker.



The magnetic pickup is composed of a N52 neodymium magnet and a tightly wound coil. It's closely placed underneath a steel string. When the steel string moves, it creates a change in the magnetic field and therefore causes a voltage within the coil. The more windings the pickup has the more voltage it can induce. Furthermore, 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.



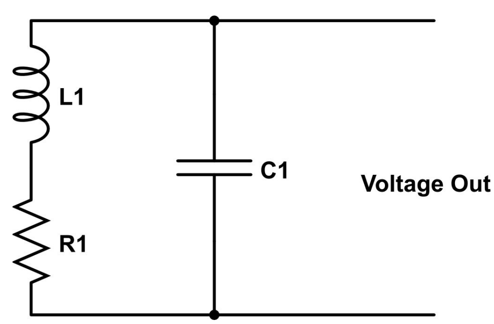
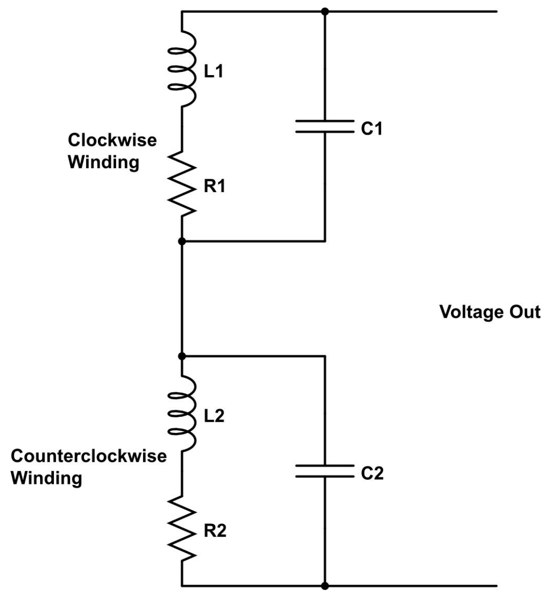


Image Source: [AES Talk, UIUC Nov. 29, 2005]

Since the pickup can be modeled as an RLC circuit, it has self resonance. 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 the first 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.

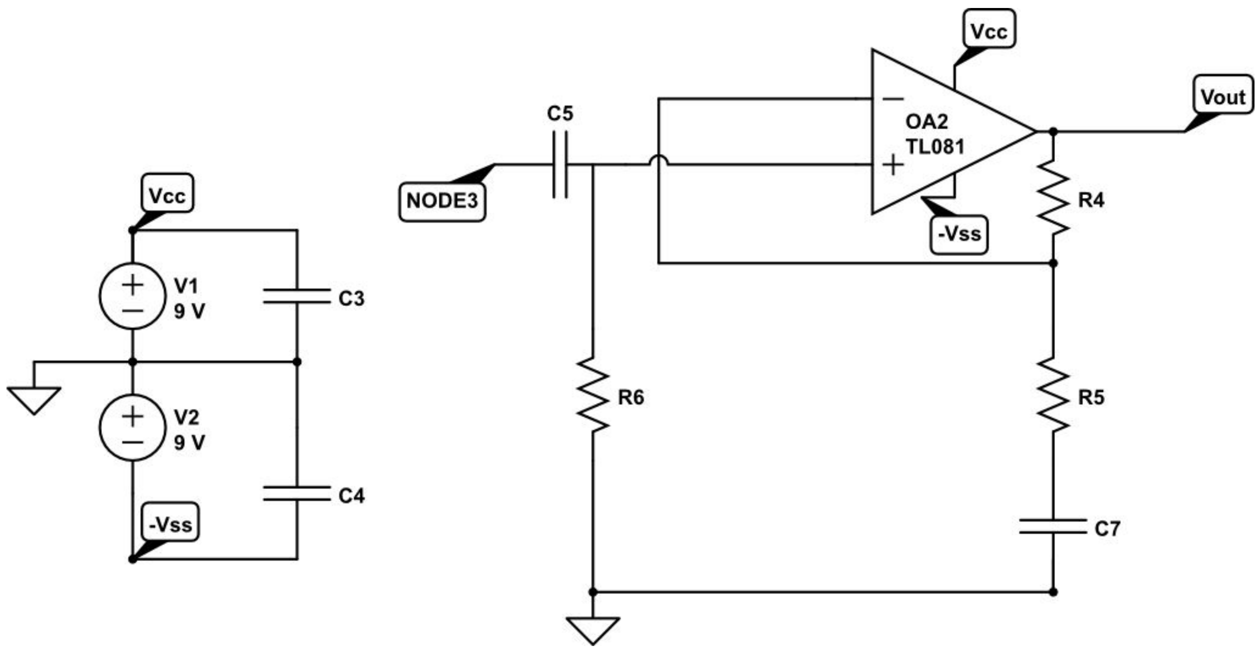
Risk: Another common issue with a magnetic pickup is that it will also pick up the 60 Hz noise from the power supply. This will generate a constant hum on top of the signal from the violin. A solution would be to implement a humbucker design for the pickup. It is pickup with two coils, each wind in opposite directions. This will eliminate any common mode noise.



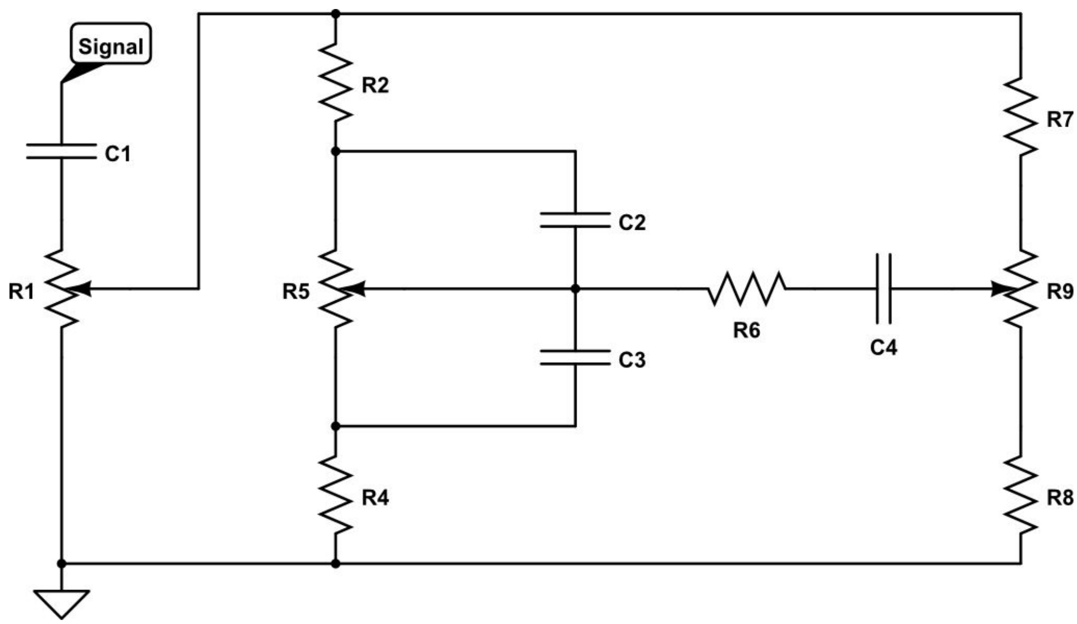
Risk: The magnetic pickup is highly depended on its proximity to the steel strings. For instruments like a guitar, where the strings are all perpendicular and equal distance to a surface, the magnetic pickup works very well in equally sensing all the strings movement. However, on a violin, the four strings are spaced out unevenly and at different heights.

Due to this odd geometry, the magnetic pickup might not work well at evenly sensing the different strings. One solution to this is to shape the magnetic pickup in the same pattern as the violin strings' height. As a backup plan, a piezoelectric pickup can be used instead. The device can be attached anywhere on the violin and still sense the output of the violin accurately.

Since the pickup is inherently a high output impedance system, the output is connected to a high impedance input amplifier. Thus the small output of the pickup (~100 mV) can be boosted up to line level (~1 V RMS). Additionally, the high impedance input will act as a voltage divider with the output impedance of the pickup. This will attenuate the signal and will smooth out any resonance effects of the pickup if there is any.



The preamplifier is an op amp in a non-inverting amplifier configuration. The gain of it is set by R4 and R5. These two values are calculated to give an output of around ~1V RMS line signal. C5 and R6 couple the signal to ground. R6 is set to an extremely high value of around 3-10 Mohm to get a high input impedance to the amplifier. Since C5 and R6 act as a high pass filter, C5 is selected to set the cutoff frequency to be below the lowest note of the violin.

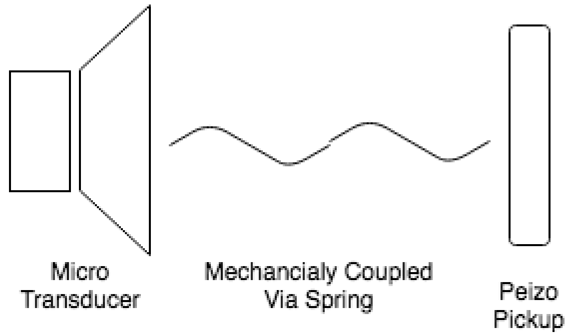


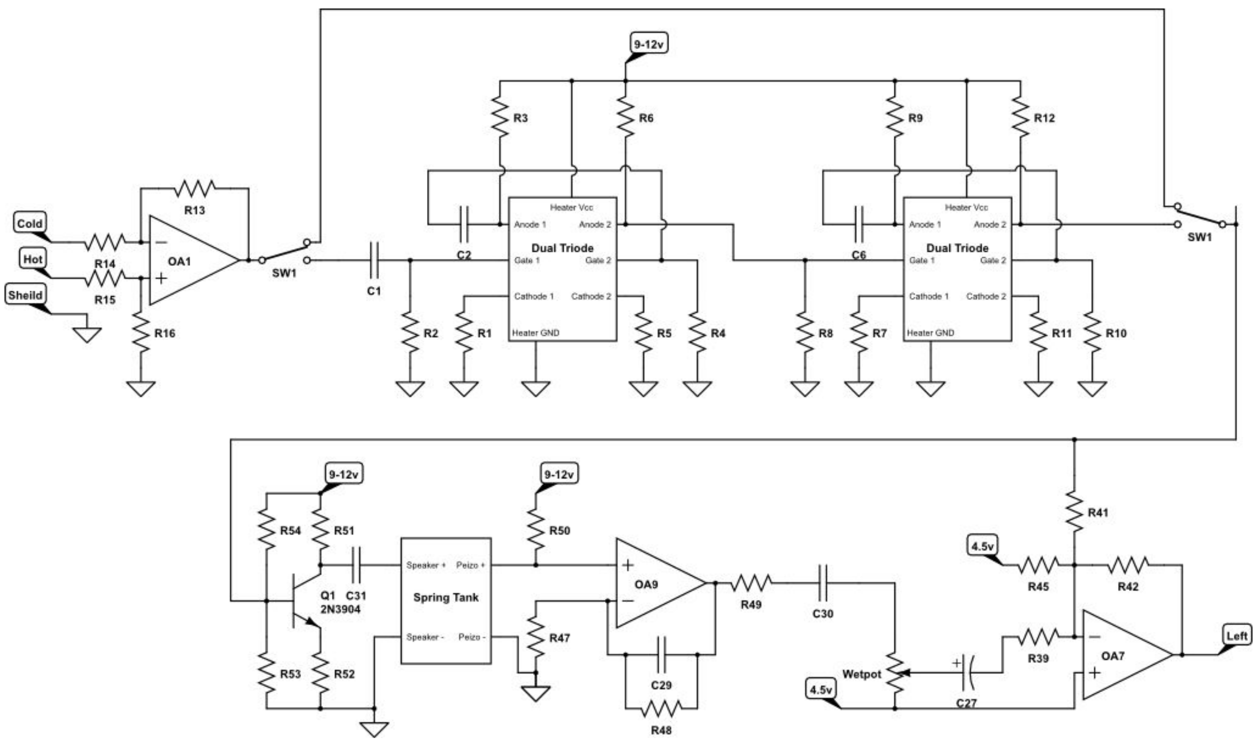
The tone control part of the circuit is first put into the circuit via C1 and R1. This acts as the gain control. The signal is then passed to a Baxendall tone control network. R5 is used to control the bass, and R9 is used to control treble. R1 should be a logarithmic type potentiometer. R5 and R9 should be linear type potentiometers. This will help the performer to easily control the tone control.

Risk: The potential problem with this circuit is that it's a passive filter system which introduces loss. It's possible to feed this into another non-inverting amplifier to boost the signal up again if the loss is too much.

This part of the circuit (magnetic pickup, preamplifer, and tone control) will be done by Thanh Nguyen.

The effects component of this project will be based around two main stages, both of which can controlled by the performer. The balanced line level signal from the preamp is received and converted to single ended before being fed to the tube compressor and overdrive stage, which sends the signal through multiple triode stages in order to produce nonlinear distortion. After this, the signal is sent to the reverb stage, which is essentially a transducer coupled to a pickup via a spring. This stage uses a simple class A amplifier and a simple op-amp piezo pickup design. The recovered signal is then mixed in with the original signal in a proportion governed by a user controlled wetpot.





The tube stage of this effects box uses the first dual triode to create clipping distortion by using the first triode of the first dual triode to boost the signal with a low R3 value on the anode (or plate) resistor and the second triode of the first dual triode to clip the signal by limiting maximum current flow with a higher R6 anode resistor value. The second stage is a two part gain stage running in starved cathode configuration to induce more non-linear tube distortion, white also adding gain control with a potentiometer on the R7 resistor (not shown).

Risk: Most preamp tubes that are designed to work with high plate voltages can be relatively unstable at the much lower voltages used in a starved cathode configuration. 12v operation is rarely documented in tube specification sheets and can vary dramatically between individual tubes. Triode stages may have to be tuned to provide adequate overdrive at line voltages with a variety of trim pots, even after ideal values are calculated.

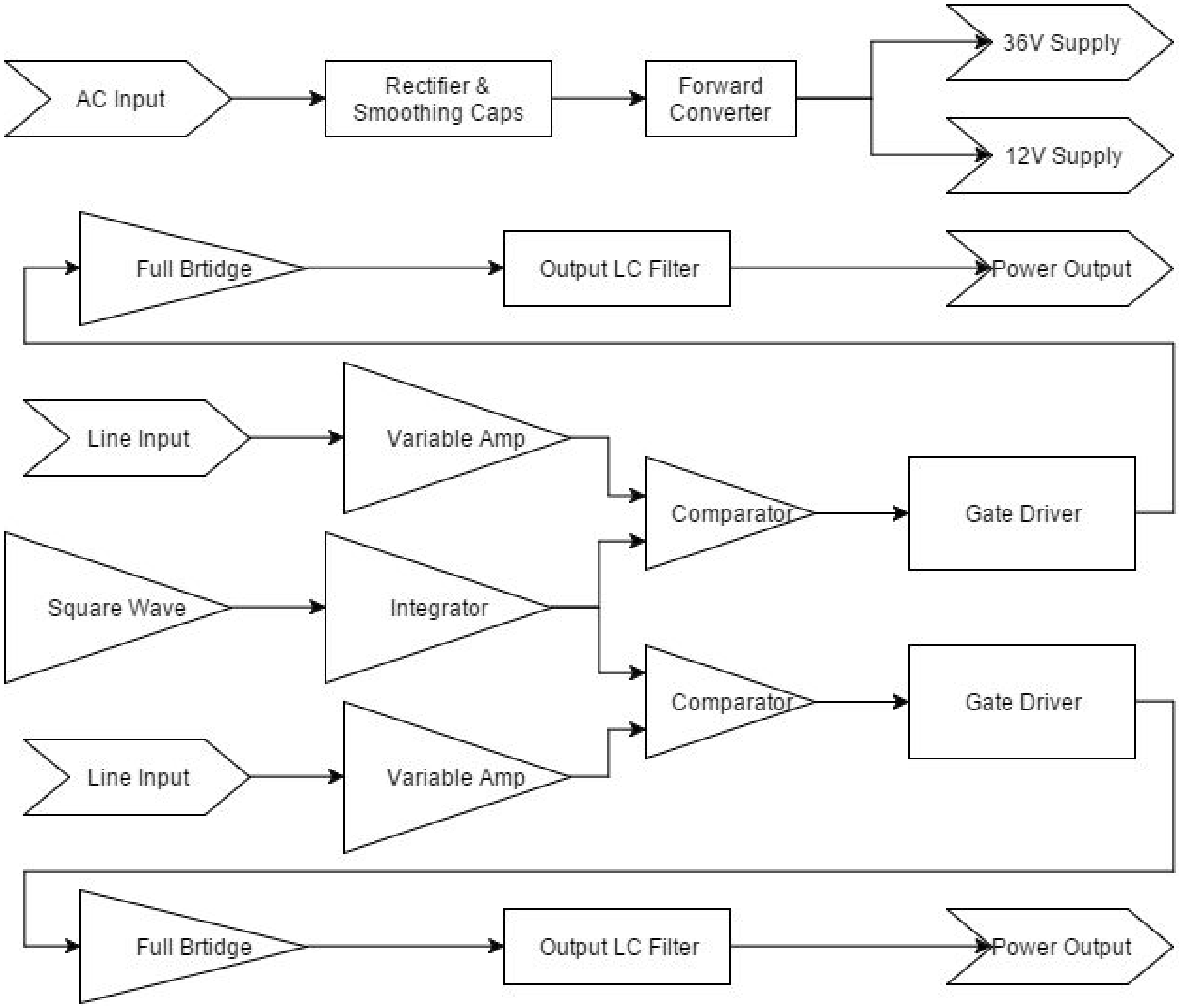
One of the most technically interesting and natural sounding ways to create reverb for audio is via a spring tank or spring coupling system, typically composed of a transducer coupled to a receiver via one or more long, weak springs. The weaker the spring, the longer the reverb delay. For this design, the transducer is a high efficiency, low output full range speaker and the pickup system is a piezoelectric microphone. The spring will be directly mounted to the cone of the speaker on one end and the face of the pickup on the other.

The circuit, therefore, requires a small power amplifier for the speaker and a high sensitivity high impedance op-amp stage with a built in low pass filter to eliminate high frequency after-ringing produced by the spring. The reverb tank itself does not need to be particularly high fidelity, and the amplifier does not need to provide much power to adequately couple the output, so a simple class A amplifier design sufficient.

Risk: With such a high gain pickup system, external noise becomes a huge factor. The system will have to be well isolated mechanically and electrically may require a second order high pass filter to reduce bump noise, potentially via an integrator. This could introduce phase issues on the summing output if not implemented properly.

Risk: It’s quite possible that the mechanical transmission system will prove too sensitive, too unreliable, or too poor quality to be reasonable as a delay line. In this case the only way to produce reverb would be via DSP or with a large, premade spring-transducer assembly.

The effects stages will be designed and implemented by Peter Sudermann.



The amplifier stage of this project will be implemented as a class D amplifier. Our goal will be to make it capable of outputting at least 100w of power with minimal distortion between the frequencies created by a violin. Other target features will include AC input, a flat frequency response, line-level input, and volume control. The system will be implemented as follows:

A fused AC input will be converted into a 36VDC supply for the power stage and a 12VDC supply for powering signal-level components. This will be accomplished through the use of a forward converter with a 36V output and a 12V output; the former will be regulated through feedback and the latter will be further regulated in a linear fashion to ensure a noise-free IC supply. This may change into a store-bought switching supply if needed.

The signal will first be amplified through a variable amplifier controlled by a log-pot. This will both allow volume control and bias the signal to a workable voltage. The signal will then be compared to a 500kHz triangle wave generated by a multivibrator-integrator combination in order to generate a suitable PWM signal. The output will finally be fed into a gate driver that will power a full bridge power stage.The output will be appropriately filtered to remove the modulating signal and then fed to output terminals. There are two channels to be outputted, so this circuit will be doubled (sans the triangle wave generator, which can be shared).

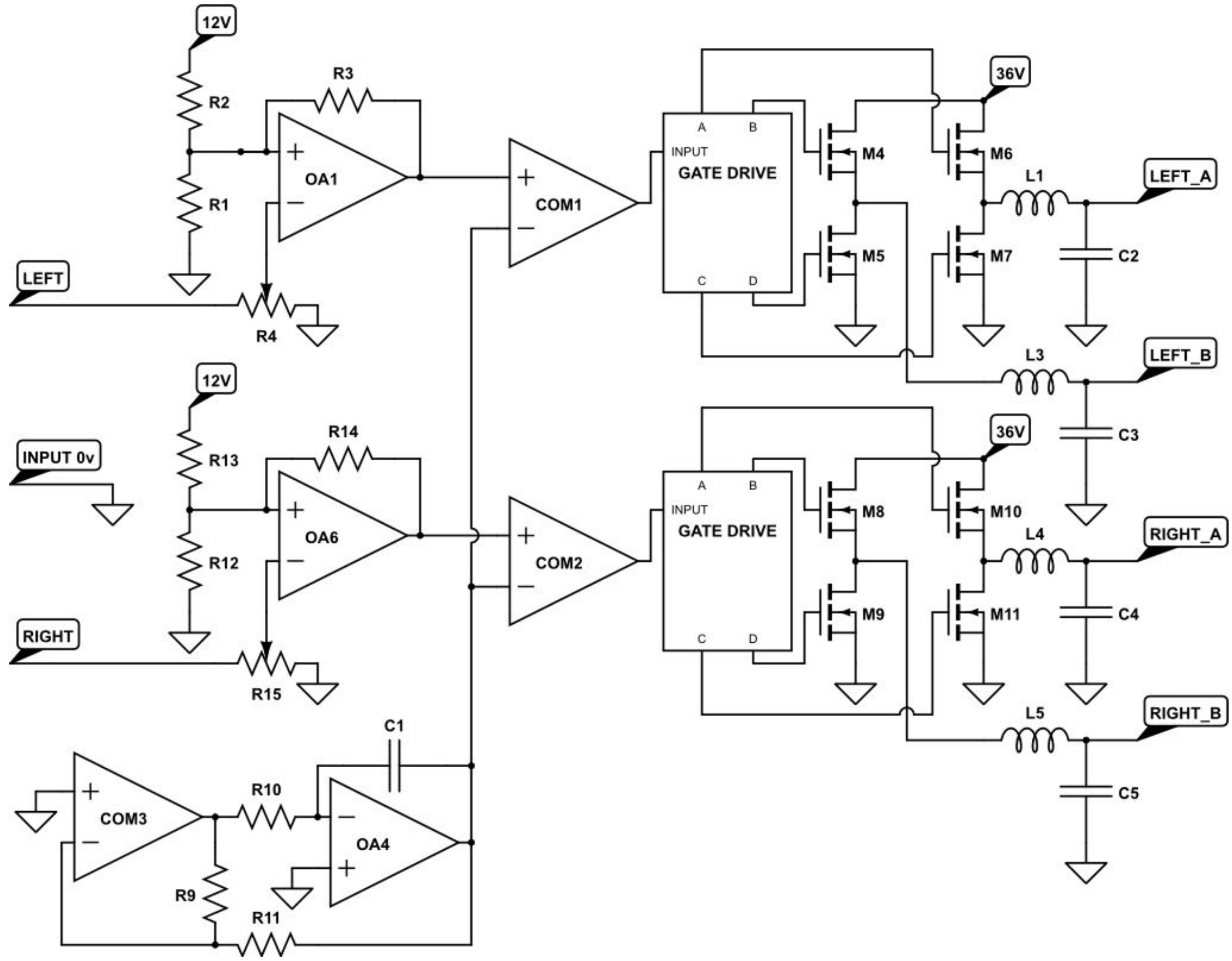
The bare minimum functionality from this amplifier would be a functional PWM modulator capable of driving a small speaker using a premade power supply. This would require creating a triangle wave generator and comparing it to our input, giving us a comparator output that could poorly drive a speaker. Adding variable gain and a low power half-bridge driver would be better, and a full-bridge driver with LC filtering would be considered adequate. A stretch goal would be to build a forward converter capable of powering this circuit.

The riskiest element of this project is the power supply, as creating a noise-free power supply may prove to be both hard and dangerous. This risk can be mitigated by making the power supply inputs modular such that a commercial supply can be swapped in if needed. The next riskiest element will be the full-bridge power stage, as full-bridge drivers must be carefully designed to avoid shoot-through. Careful experimentation and calculations involving gate capacitance and resistance can be made to ensure that this does not occur. The next riskiest element will likely be the triangle wave generator, as any nonlinearities present in this wave will show up as distortion in the final amplifier. This can be dealt with by using fast low-noise audio-grade op amps.

Careful considerations will have to be paid to noise throughout this design. The high switching frequencies of this circuit will likely cause problems for any circuit elements that are not properly bypassed, and a PCB will have to be used to avoid EMI issues. The use of a ground plane and short traces should help greatly.

High quality components will be needed to avoid issues with noise and voltage offset, so audio-grade op-amps in the SOIC form factor will be ordered as samples from TI. IRF540 MOSFETs will suffice for the switching elements, and passives will be taken from the lab space. If a forward converter is used, inductors will be hand-wound on cores taken from PC power supplies and a switching regulator will be ordered as a sample from TI. A log-pot will be purchased from Digikey to allow proper output level control along with a gate driver. Prototyping will begin on a breadboard using lab op-amps and will move to PCBs manufactured with the Othermill in EDS. The final PCB will hopefully use only SMD versions of components purchased from Digikey.

A cursory schematic of the circuit as planned is presented below:



This portion of the circuit will be worked on by Chetan Sharma.

The milestones for magnetic pickup, preamplifier, and the tone control will as follow:

- April 21 - Finish creating the preamplifier and the tone control on breadboard. CAD the circuits and send out for PCBs.
- April 30 - Finish creating the magnetic pick.
- May 2 - Soldering parts and assembling onto a violin.

The milestones for the effects stage are as follows:

- April 20 - Demonstrate functional tube overdrive and gain stage. Design piezo preamp stage based on component specs
- April 23 - Acquire and assemble components for spring reverb system. Test basic functionality and determine feasibility.
- April 25 - Finish tuning reverb and tube stages. Design PCB
- April 25 **[Stretch]** - If tuning is unnecessary, design basic optical transmitter and receiver system with ADAT Lightpipe
- April 26 - Order PCB
- April 29 - Design and print casing for spring reverb system. Order last minute switches, pots, and other interface components.

- May 6 - Have PCB assembled and debugged

The milestones for the amplifier stage are as follows:

- April 18th - Create a functional triangle wave generator.
- April 20 - Create a preamp stage and compare it to the triangle wave, giving a PWM output.
- April 22 - Figure out how to power the amplifier, whether it be through a homemade power supply or a premade power supply with filtering.
- April 23 - Drive a full-bridge power stage with the comparator output; filter appropriately to remove excessive noise.
- April 25 - Design a prototype PCB and test for bugs.
- April 26 - Order a real PCB.
- May 6 - Have final PCB soldered and ready for presentation.