1. Introduction

Problem

Electric guitar feedback is traditionally produced by amplifying the signal from the instrument loud enough that the energy stored as sound can induce a sustained feedback loop in the guitar string. Products such as the EBow take this concept and remove the inefficiency of energy transmission through sound by instead sending the amplified signal through magnetic driver coils (think of speaker drivers) directly into the string. Products such as this implement harmonic controls through analog filters in the signal chain, causing the string to resonate in higher octaves.

Techniques such as this create a unique timbre from this instrument which can be finely controlled by the player and the electronics of the instrument. This unique timbre is restricted to a small number of notes (1-6 strings) at any given time and can only be utilized by musicians who are trained on guitar.

Solution

Our team would like to bypass these restrictions by making a harp or organ-like instrument with one feedback system per string. This instrument would ideally consist of twelve strings representing the chromatic scale in the third musical octave. Our instrument would be controlled over a MIDI interface, allowing it to generalize to a broad range of musical controllers for those with backgrounds in various instruments.

The instrument would comprise two main systems: the Master Board and the individual DSP Feedback Systems.

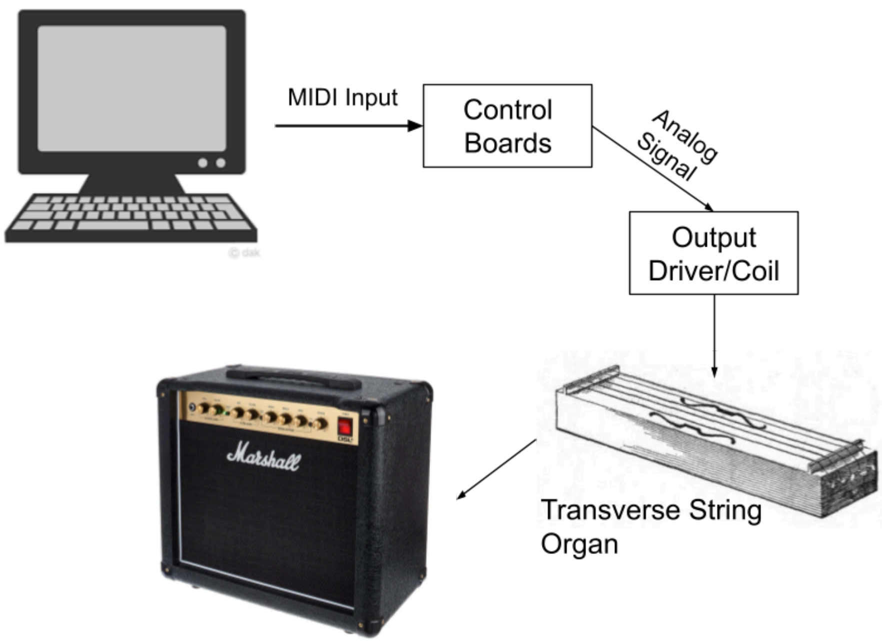
The Master Board will act as the host of the system; it will listen to a MIDI signal through the UART peripheral of an STM microcontroller, and translate specific MIDI commands to an I2C bus, where this system would act as the master. This board will also include a 3.3V DC/DC regulator to power the MCUs of the other boards of the system.

On the slave side of the I2C bus will be several (1 per string) low-power, DSP microcontrollers. These microcontrollers will implement the filtering that traditional sustain systems typically do using DSP rather than analog filters. This will allow us to perform extended functionality such as the automatic muting of notes, and more controlled harmonic filtering.

These DSPs will be paired with an electromagnetic pickup (similar to that of an electric guitar) to sample a signal from the string as it vibrates, and an electromagnetic driver which will receive an amplified & filtered version of the original signal in order to induce feedback into the string. Each electromagnetic driver will be powered by a discrete class AB amplifier.

We would like to use an off the shelf 12V DC power supply to power the entire system.

Visual Aid



High-level requirements list

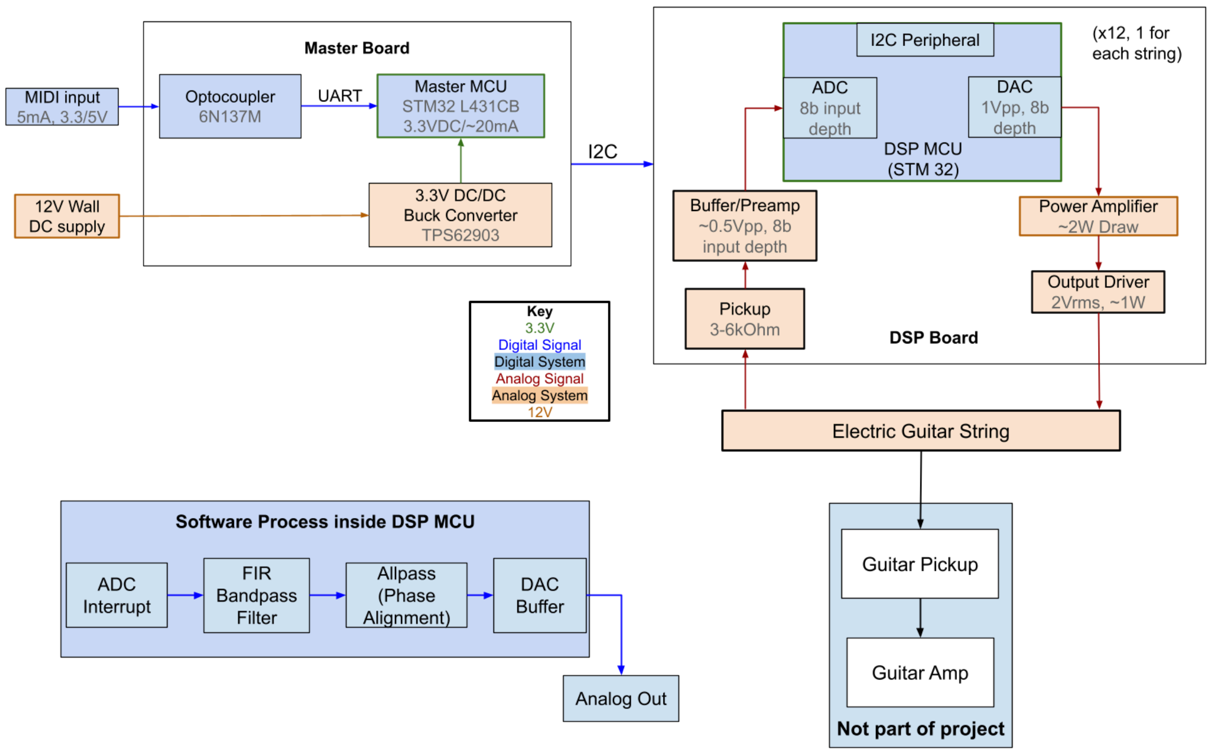
Notes should reach 63.2% max Vrms (2.03 dB) within 0.2 seconds during activation, and should reach 36.8% max Vrms (-2.66dB) within 0.2 seconds during dampening. This relates to the precision of timing at which we can play this instrument.

Small chords can be made: minimum 3 strings can ring out concurrently. These strings must be able to ring out for at least 20 seconds to be considered sustained indefinitely.

Harmonic control of each string is possible: The instrument can isolate strings at their fundamental frequency, and the 2nd & 4th harmonics (octave & 2 octaves). This will be measured using a waterfall diagram generated by an oscilloscope, ensuring that the active harmonic has a larger amplitude in fourier analysis than any other.

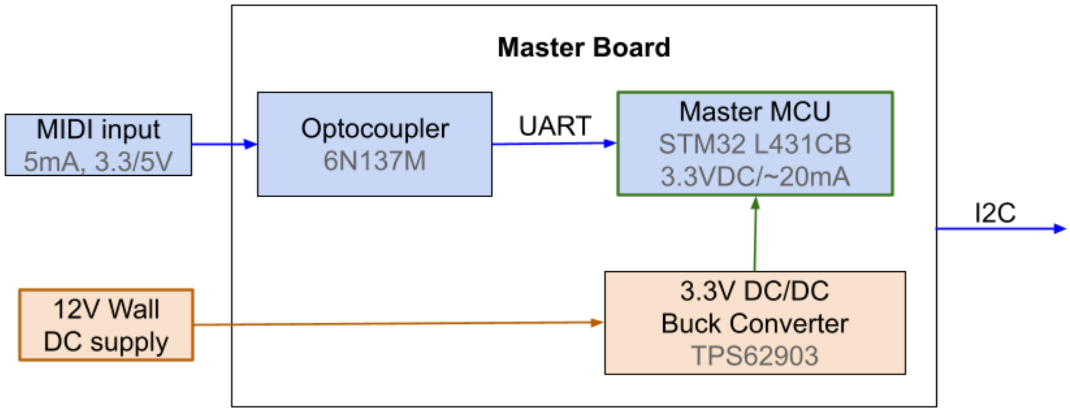
2. Design

Block Diagram:



Subsystem Overview/Requirements:

Subsystem 1: Master Board



The Master Board will be composed of three parts: an optocoupler, the master MCU, and an integrated 3.3V DC/DC converter.

The optocoupler will serve the purpose of reference isolation for the MIDI controller port. This is standard circuit design for a MIDI receiver and will require minor peripheral circuitry to perform tasks such as ESD protection and signal biasing.

The master MCU will be a low power MCU, capable of basic communications such as an STM L0 / L4. Since the bandwidth requirement of this MCU is actually less than that of the DSP boards, we will likely use the same MCU as the DSP boards for cost optimization. In particular, we were considering the STM32 L431CB, the reasoning for which will be explained in the DSP section.

The DC/DC regulator will likely be a TI TPS62903. We've decided to use an integrated regulator as the functional design of this type of circuit is not core to the working concept of this project. This IC has an input range of up to 17 Volts which aligns with the off the shelf 12 Volt power supply that we are hoping to use. This specific regulator has a sustained maximum current output of 3A, which significantly greater than the maximum current draw (140mA) for 13 of the MCUs above, although it is also important to note that these MCUs are not expected to draw nearly that much current as they will not be powering any peripherals. This IC is a QFN package which will require an SMD stencil and reflow soldering, however members of our team have experience with BGA design and reflow soldering from prior classes.

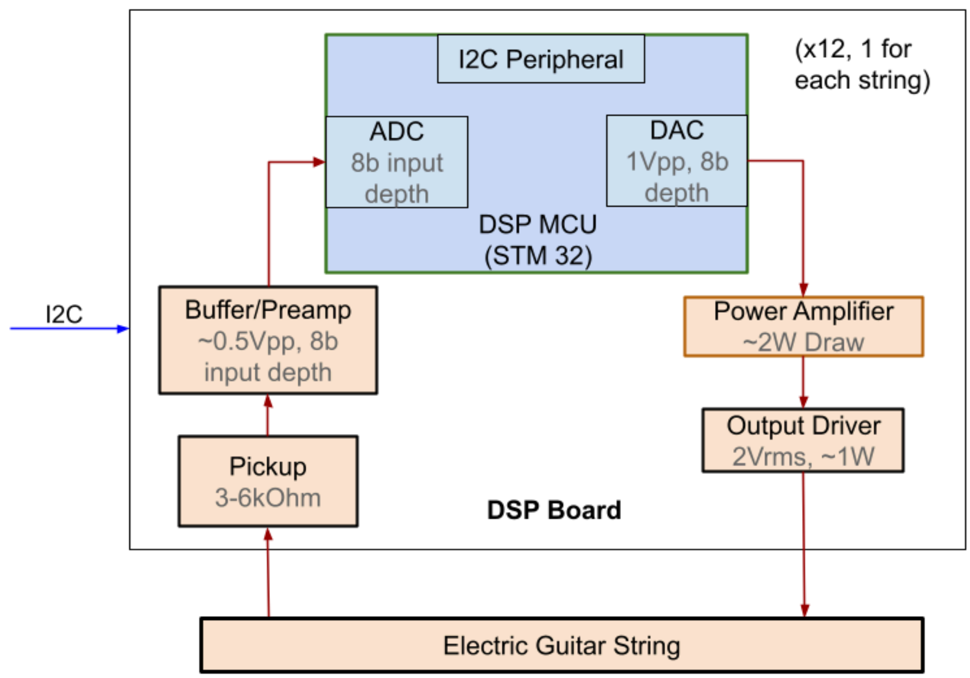
Power Subsystem (bottom of Master Board block):

The off-the-shelf 12V DC power supply listed as an input for the master board must supply 12 V ± 0.5 V and at least 5A of current, however as the system is powering an integrated DC/DC regulator with a broad input range and amplifiers which have forgiving supply voltage requirements, the voltage tolerance is not critical. This current budget assumes up to 1A of draw when three amplifiers operate at full output, 3A allocated for MCU loads, and 1A reserved for additional losses.

The DC/DC regulator (TI TPS62903) must be able to supply a minimum of 3A, as specified in its datasheet. This requirement exists as each DSP MCU (STM32 L431CB) may draw up to 140 mA, which is specified in the datasheet.

Subsystem 2: DSP Boards

The DSP boards are on the slave side of the I2C bus of the Master Boards . These boards will be paired with an electromagnetic pickup to sample a signal from a vibrating string, and this will create feedback through an amplifier into an electromagnetic driver to continue oscillating the string.



The DSP boards will be comprised of five components:

The MCU that we are hoping to use for both these boards and the Master Board is the STM32 L431CB. This MCU is a part of STM's low-power series of microcontrollers, and is likewise cheap and accessible which is important for scalability in a project that uses several of them. This MCU comes in a QFP package and therefore will be hand-solderable. Additionally, this specific MCU has an internal factory trimmed 16MHz oscillator, which is key to reducing the overhead circuitry needed for DSP. In the same vein, this MCU has a built in DAC which will allow us to directly drive amplifier circuitry rather than using PCM output and smoothing circuitry. While it is likely possible to process multiple concurrent channels of audio using this MCU, we would like to use 1 DSP per string to avoid any potential bandwidth restrictions or architectural complication when executing this design. Our choices regarding cost and scalability reflect this decision.

The DSP boards will make use of two electromagnetic coils. The signal for each string will start at a **pickup** (similar to that of an electric guitar) localized to each specific string. These pickups will have an output impedance of roughly 3 kOhms which will be buffered through an op-amp before driving the ADC pins of the DSP MCU. The output of these pickups can be attenuated passively using potentiometers for level matching.

Due to the (relatively) low output impedance or our per-string pickups, an **op-amp buffer** with a slight voltage gain (approx 6dB) will be required before feeding the signal into the ADC integrated in the MCU. Before buffering, the signal from the pickup will be filtered to combat anti-aliasing and passed through a passive attenuator to fine-tune the peak-to-peak voltage during device calibration. Because the highest frequency that this instrument aims to output is slightly less than 1kHz (B5), these antialiasing filters will have a cutoff frequency of 2kHz to satisfy Nyquist.

The second electromagnetic coil will be the **output driver** for the string. This coil functions identically to the voice coil of a speaker: a power-amplified signal is passed through a low impedance coil (~8 Ohms) to move a magnet. A magnet will be positioned next to the string, magnetizing the string and allowing it to capture power from the driver coil. The same concept applies in reverse for the pickup mentioned above. These output drivers should draw approximately 1W at ~2Vrms as this is the upper range of what off the shelf guitar sustainer devices typically output.

The final component of this subsystem and our design in general is the **discrete class AB amplifier** that will be attached to each DSP board and used to drive the output drivers. Each of these amplifiers will be connected to the 12 Volt power supply that powers the entire system, allowing for greater power output than could be supplied by an MCU or battery. Taking into account the inefficiencies of a class AB amplifier, this means the amplifiers will draw ~2W at most. Although we would like to use an off the shelf power supply that can power all twelve sustain devices at max power concurrently, we will implement digital controls such as restriction of how many notes may be turned on concurrently to ensure that we stay below power limits.

Tolerance Analysis:

The biggest tolerance problem in the power system is ensuring it can handle both the amplifier and MCU demands at the same time. Class AB amplifiers will consume about 2W of power per channel to provide 1W output, based on expected losses. With three amplifiers running at full power, this provides an estimated 1A draw from the 12V power supply. This output should suffice to drive strings into feedback as it is significantly higher than other commercially available systems which run off of battery power.

Additionally we are to provide 3A to the MCUs, this is based on the TI TPS62903 datasheet while the expected load will be lower. Another 1A will be provided to cover potential losses. These demands factored together determine the need of an off the shelf 12V ± 0.5V supply rated at least 5A. This gives us an acceptable margin for all three amplifiers, all MCUs and conversion losses all in a max load scenario.

The inclusion of DSP in this project will create a fair level of robustness in our implementation. Although DSP buffer and analog filter delay may create a phase mismatch between the input and output signals of the DSP boards, we will be able to combat this using digital delays for phase realignment. Additionally, the overall gain of the output amplifier can be attenuated digitally before D to A conversion to prevent mechanical or thermal failures.

3. Ethics and Safety

Our project will have some important ethical and safety considerations. Our system deals with electromagnetic pickups, driver coils, and amplifiers, which makes electrical safety of foremost importance. We will be working with a 12V DC supply and class AB amplifiers, which although low voltage, are able to supply high currents. To ensure safety during development and use, we will design the power system with basic protective measures to prevent overheating, limit excess current, and reduce the chance of accidental damage during operation. Wiring and enclosures will meet best practices for avoiding accidental contact between conductive surface and live components.

From the perspective of both the IEEE and ACM Code of Ethics, we will make the health and safety of users our greatest concern and avoid risks. Our design will utilize safe levels of current, and we will also comply with proper standards for consumer devices as per IEC 62368 for audio devices which includes musical instruments like our project. In alignment with responsibility and transparency principles, we will make both the safe operational conditions and limitations of our device available to users as well as maintain a detailed lab log during development. We also acknowledge the importance of accessibility and inclusiveness with our product. Taking this into account, our system employs MIDI as its control interface meaning any musician regardless of experience can play it, making our final product more accessible. This design philosophy aligns with the ACM Code of Ethics provision to benefit society and human life.

Finally, during development, we will be practicing standard laboratory safety practices, like wearing safety gear when soldering, being careful when handling power supplies, and following campus laboratory safety guidelines as outlined in our training. Any software used in the project will comply with intellectual property rights as well as campus policy, and will be open-source or licensed software.