Design Document - Group 30

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.

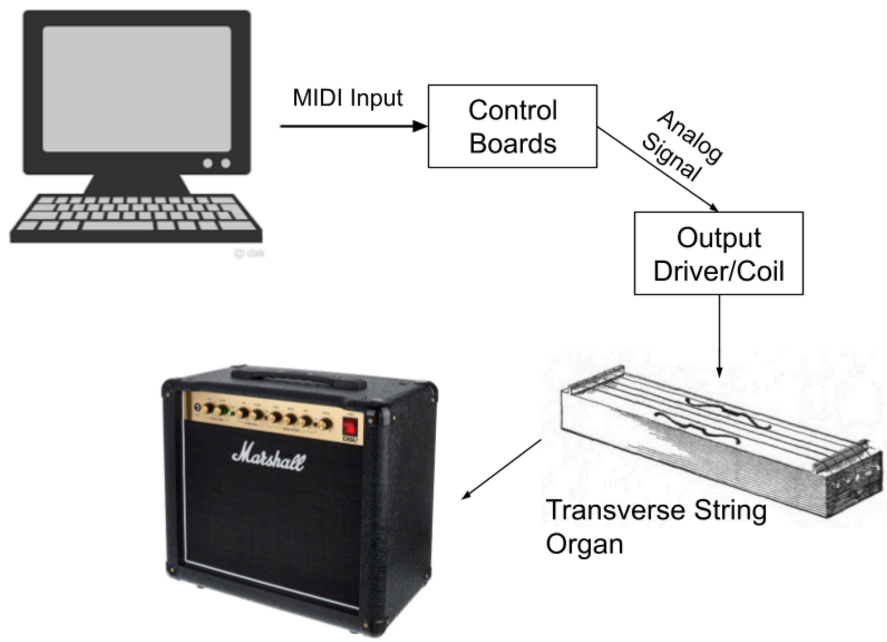
The instrument would comprise two main systems: the Master Board and the individual DSP Feedback Systems.

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

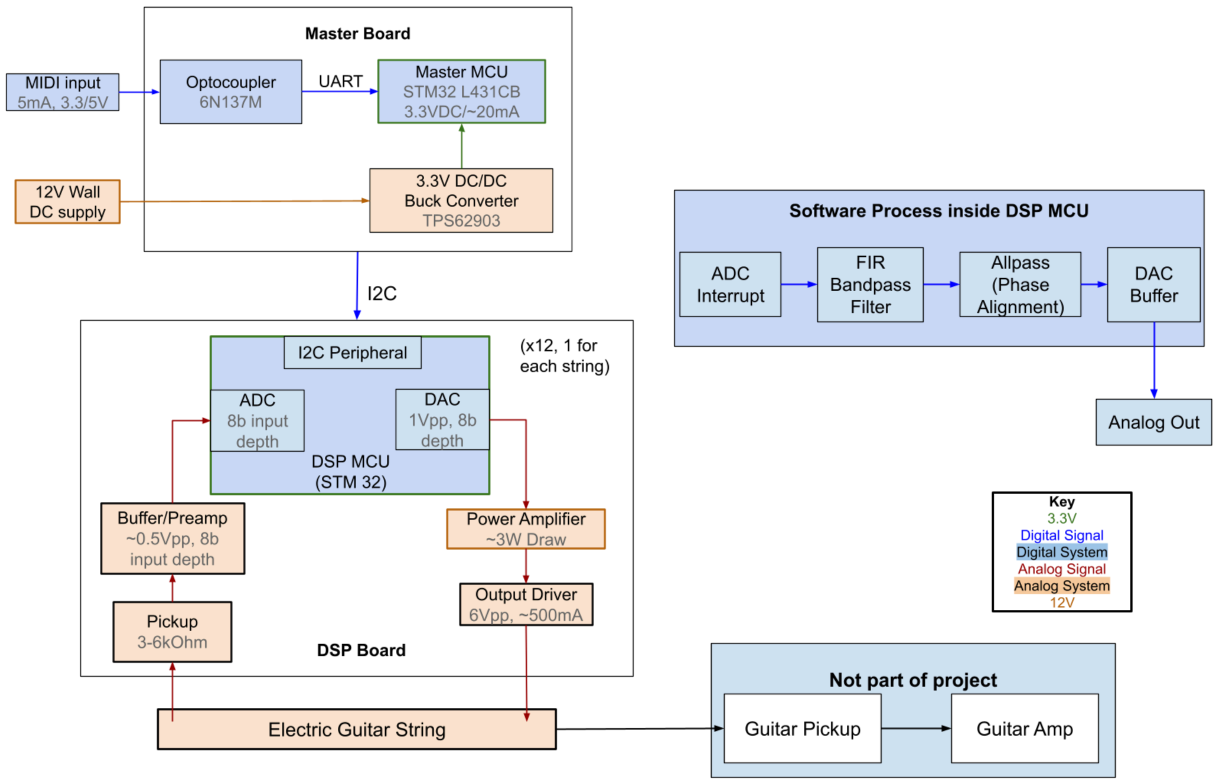
Notes should reach 63.2% max V_{rms} (2.03 dB) within 0.2 seconds during activation, and should reach 36.8% max V_{rms} (-2.66dB) within 0.2 seconds during dampening. This relates to the precision of timing at which we can play this instrument.

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.

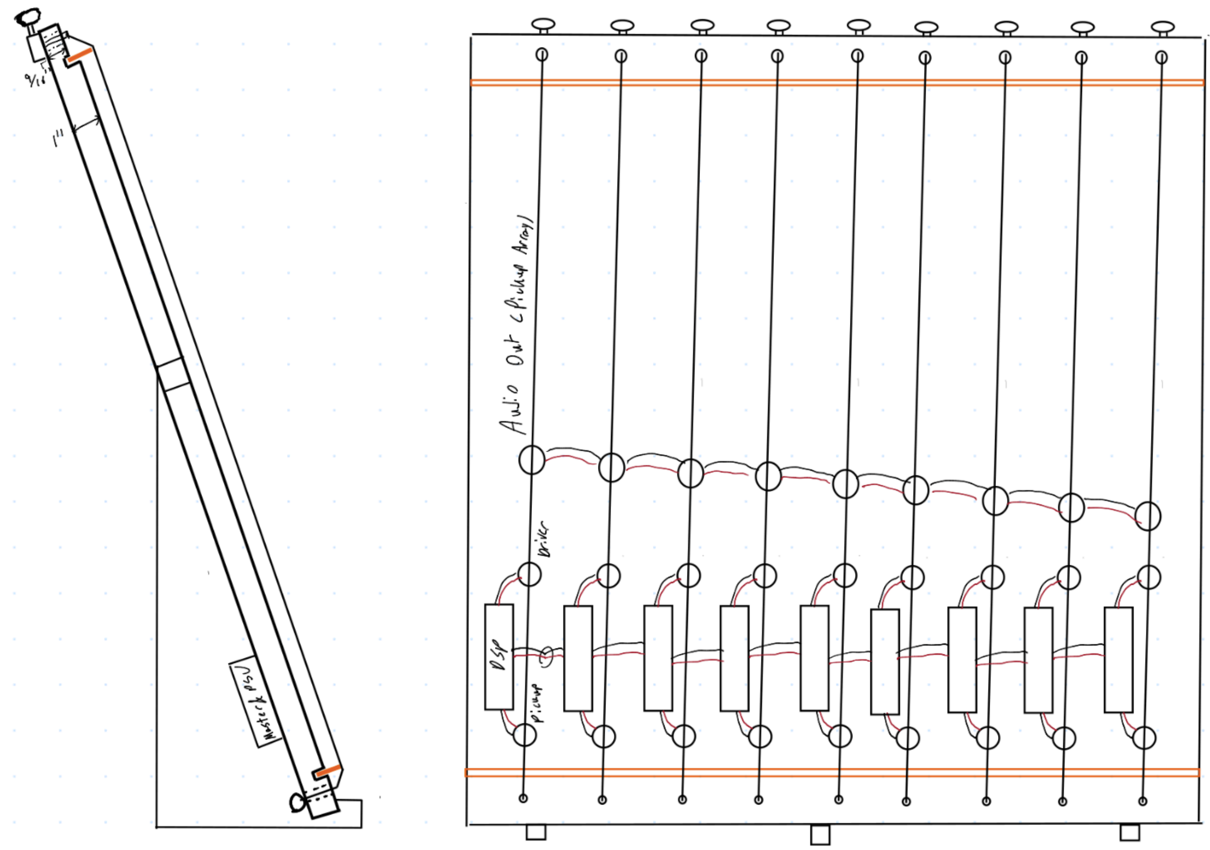
2. Design

Block Diagram:



Above is the overall block diagram for our project. This consists of the following subsystems: Power, Master MCU, DSP MCU, Buffer, Power Amp, Pickup, Driver. These systems will be described in more detail with detailed specifications in following sections.

Physical Depiction (not to scale):

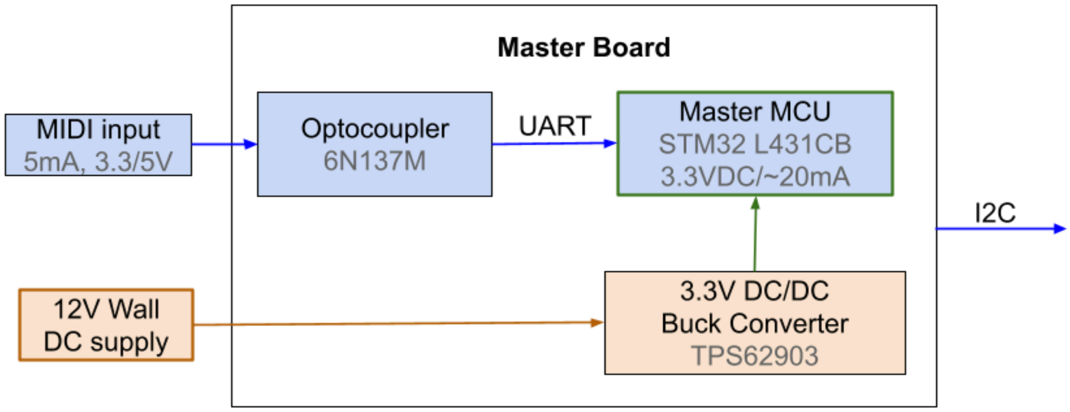


Mechanically, our design consists of a 24"x30" sheet of 1" plywood that has a couple pieces of brass stock attached via dato joints at either end of the board. These stock pieces will act as bridges (resting points) for the ends of the strings. Tuners will be placed at one end, and the ball end of the string will go through the backboard on the other end.

Our design stands upright with the Master Board and power electronics mounted to the back. The DSP boards and coils will fasten to the front of the board through a mix of glue and bolts to securely fasten the pickup and drivers. We are hoping to have twelve DSP units for this project, however this isn't a hard requirement as this number will be primarily limited by cost.

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.

Master Board Subsystem Software Description

The Master Board acts as the only bridge between an external MIDI controller and the Slave Boards connected over I^{2}C. Its primary role is to receive standard MIDI messages using UART, translate them into musical control signals such as “Note On”, “Note Off”, and harmonic mode commands, and determine which slaves should receive what command without lag. The subsystem uses a non-blocking, interrupt-driven design for both the UART and I²C communication, using zero blocking and allowing continuous handling of incoming MIDI data without halting the processor.

Incoming MIDI data is received using the STM32’s USART2 peripheral at a standard baud rate of 31.25 kbps. Incoming inputs are handled in interrupt mode using `HAL_UART_Receive_IT`, which triggers an interrupt each time a byte is received rather than stopping the whole MCU. Within the UART interrupt callback, a very simple state machine reconstructs complete MIDI messages from the incoming byte stream, including support for the MIDI “Running Status” feature [1]. This state machine determines whether each byte is a status or data byte, and once a valid three-byte message is assembled (status, data, data), it passes the information to the parser for translation over the I²C bridge.

The parser translates each MIDI message based on its status byte. Messages beginning with `0x9` are treated as “Note On” events if the velocity value is greater than zero, while messages with `0x8` or a velocity of zero are interpreted as “Note Off” events. Simultaneously, because this is an interrupt based approach, the parsed MIDI data is prepared for transmission over I²C by packing it into a 3-byte container containing the status, note number, and velocity.

After parsing, the Master transmits the MIDI data over I²C using `HAL_I2C_Master_Transmit_IT`, which sends the message asynchronously to the specified Slave’s address. This interrupt-based approach allows the I²C operation to occur in the background without blocking other processes, maintaining responsive non-lagging communication with the MIDI UART interface. Each I²C transmission reports success or errors through the I²C event interrupt callback, providing a reliable way to confirm data delivery. During development and testing, the on-board LED and UART debug messages with the help of a serial terminal offer visual feedback, allowing easy confirmation that the Master Board is decoding and transmitting data as intended.

The system is designed to be scalable to a number of Slaves. Additional MIDI features, such as velocity curves, modulation, or harmonic mode controls, can be added to the parser without modifying the communication architecture making this system easily modifiable. Even without connected Slave Boards, I²C activity indicators can be demonstrated using an oscilloscope, verifying that the Master Board is generating valid communication signals.

Table 1. Master Board R&V

Input Voltage: The STM32 microcontroller and peripheral circuits must operate within **3.3 V ± 5%**.	Measure voltage at the STM32 VDD pin using a digital multimeter (DMM) while the system is powered and transmitting data. Confirm the voltage is within **3.135 V–3.465 V.**

MIDI Baud Rate: The UART interface will receive MIDI messages at **31,250 bps ± 1%(clock tolerance)** without data loss or framing errors.	Use a logic analyzer connected to the UART RX pin while transmitting MIDI messages from a controller. Verify the measured bit rate and confirm that messages are received correctly.
I^2C Clock Rate: The I^2C master interface must operate at **100 kHz ± 5%** (Standard Mode) during message transmission.	Measure the I^2C SCL line using an oscilloscope or logic analyzer. Verify that the clock frequency remains within the range of **95 kHz–105 kHz.**
Power Draw: The master board should consume no more than**140mA** during continuous MIDI activity.	Measure current drawn from the 3.3 V power source using a DMM. Record average current during MIDI message bursts.

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 1.5A of draw when three amplifiers operate at full output, 3A allocated for MCU loads, and 0.5A reserved for additional losses.

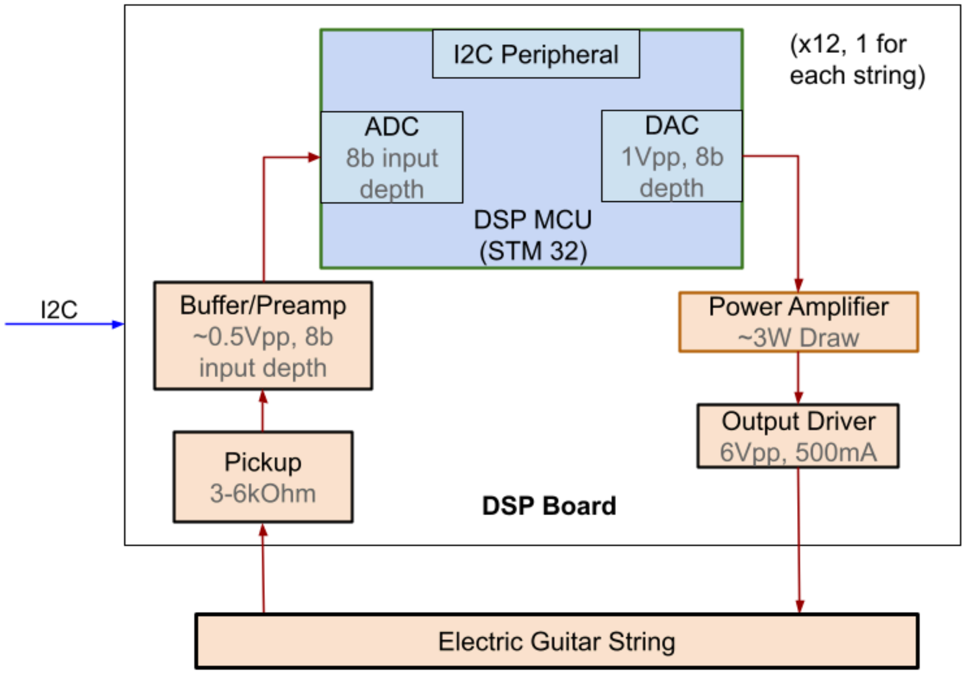
The DC/DC regulator (TI TPS62903) can supply a maximum of 3A, as specified in its datasheet. This requirement exists as each DSP MCU (STM32 L431CB) may draw up to 140 mA, which is specified in the datasheet.

Table 2. Power Subsystem R&V

Output Voltage: The DC/DC regulator should output 3.3V with no more than 150mV of ripple to ensure smooth and predictable MCU operation and output values.	The DC/DC buck converter will be supplied with the 12V AC/DC converter and the output will be measured with an oscilloscope to ensure that the output waveform is consistent at 3.3V with ripple voltage within the acceptable limits.

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.

Table 3. Board R&V

Sampling rate: Sampling rate must be at least two times the highest frequency for each note. For our instrument this will be at least **2kHz** as the highest harmonic that we wish to isolate is slightly under 500Hz. This additional headroom will allow us to use a low Q lowpass filter around 1kHz to prevent aliasing.	Measured by ensuring perfect recreation of signal at half of sample rate. MCU input will be connected to the function generator with appropriate reference and range. The input must be at least 500Hz, and the output signal must show no evidence of aliasing.

Phase alignment (total lag): Phase of offset between input and output signal should be **below** $\frac{\pi}{10}$ **radians** (18°) in order to be considered sufficiently "in phase" to create feedback.	Measured by finding time difference in nodes of pure tone waveform. MCU input will be connected to the function generator with appropriate reference and range. Input and output signals will be measured concurrently through an oscilloscope and their phase difference will be found by measuring the distance between their nodes scaled by frequency.
Sample Reference Voltage: Output signal from MCU internal DAC should be centered around half of Vcc to (**1.65V**) for single ended analog signal output. This bias will be AC coupled before entering the power amp stage.	Can be verified by viewing the signal of the component (connected to the power amp) in an oscilloscope. If nothing is moving near the pickup, the signal should be at 1.65V. If the pickup is actively monitoring a moving string, the waveform should have a DC offset of 1.65V.

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.

Table 4. Pickup R&V

Output Impedance: Pickup must have a DC resistance in the range of **2kΩ - 5kΩ** in order to properly filter through the preamp and maintain the appropriate signal level.	Measured by finding the DC resistance of the driver with a multimeter. LCR characteristics may also be found for this component for additional calibration.
Signal Level: Peak to peak voltage must be in the range of **50mV - 300mV** when the pickup is held near the string. This will allow the pre-amp to amplify the signal such that it saturates the ADC.	Can be verified by viewing the signal of the component (not connected to any other circuit) in an oscilloscope. Pickup will be held approximately 1 cm away from the string, and the string will be plucked loudly. Scope measurement functionality will be used to find Vpp of the waveform immediately after the attack (50ms after pluck).
Reference Point: Reference from ground must be **0V** in order to prevent any DC current flowing through the pickup. This is essential to guarantee that the component will not become hot and pose a safety risk.	Can be verified by viewing the signal of the component (connected to the preamp) in an oscilloscope. If nothing is moving near the pickup, the signal should be at 0V.

	If the pickup is actively monitoring a moving string, the waveform should have a DC offset of 0V.

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.

Table 5. Buffer R&V

Gain: Pre-amp should be able to amplify the signal by a factor of at least **8** in order to better saturate the ADC voltage reference while not going past this value. A passive voltage attenuator is included in this design to allow for fine tuning of this subsystems gain factor.	Measured by viewing input and output signals of the circuit in an oscilloscope. Input of the circuit will be connected to a function generator supplying a pure tone with a frequency of 500 Hz and a peak-to-peak voltage of 200mV. The output of this circuit should be a pure tone at the same frequency with a peak to peak voltage of roughly 1.6V.
Band Limiting: Signal should be filtered such that an attenuation factor of at least **-6dB** is applied to any signal greater than the highest sample frequency. This frequency is **1kHz** for our design, however it may be increased as high as half of the sample frequency (24kHz in the case of professional audio sample rates).	Measured by viewing input and output signals of the circuit in an oscilloscope. Input of the circuit will be connected to a function generator supplying a pure tone with a frequency of 1 kHz and a peak-to-peak voltage of 200mV. The output of this circuit should be a pure tone at the same frequency with a peak to peak voltage of less than 50mV.
Output Reference: The signal output by the DAC integrated in this MCU should have a reference of **1.65V**. This will allow the output range to be as wide as possible without clipping the signal.	Can be verified by viewing the signal of the DAC (connected to the power amp) in an oscilloscope. If the MCU is configured to not put out a signal, the signal should be at 1.65V. If the pickup is actively monitoring a moving string or synthesizing a signal, the waveform should have a DC offset of 1.65V.
Output Range: The signal output by the DAC integrated in this MCU should have an output range of **2Vpp** around the reference voltage. This will allow the microcontroller to utilize most of its bit-depth	Can be verified by viewing the signal of the DAC (connected to the power amp) in an oscilloscope. If the MCU is configured to not put out a signal, peak-to-peak voltage of the signal should be 0V.

DSP without clipping at the ends of the voltage supply.	If the pickup is actively monitoring a moving string or synthesizing a signal, the waveform should vary between 0.65V and 2.65V.

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 2W as this is the upper range of what off the shelf guitar sustainer devices typically output.

Table 6. Output Driver R&V

Input Impedance: of the pickup must be in the range of **6Ω - 10Ω** in order to efficiently convert the power from the discrete amplifier into a magnetic field that can drive the string into oscillation.	Measured by finding the DC resistance of the driver with a multimeter. LCR characteristics may also be measured for additional calibration of this component using a network analyzer to measure the frequency response.
Power Delivery: The coil must be able to continuously withstand **500 mA** of current in order to prevent overheating and short circuits from the output of the power amplifier.	This will be measured by finding the physical width of the wire using a set of micrometers. With these measurements we can verify that the wire is a gauge that will be rated for continuous delivery of our expected amount of current. For 28 AWG wire, we're expecting this diameter to be 0.3211mm.

The final component of this subsystem and our design in general is the **power 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. As op-amps are typically highly efficient, we expect the amplifiers to draw 3W of instantaneous power at the peak output voltage. 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.

Table 7. Power Amplifier R&V

The input impedance should be between **1kΩ - 5kΩ** to ensure that the input current for the op-amp does not exceed 10mA and damage the device.	As the input impedance may affect the integrity of the device, this will first be simulated in LTSpice to ensure that input current for the op-amp is within safe ranges.

Gain Factor: The voltage gain factor of this amplifier should be 10 to ensure that the output voltage is utilizing the full 12V range of the supply voltage, and the maximum possible amount of power is used to drive the coil and oscillate the string.	This can be measured by measuring the input and output waveforms from this amplifier and ensuring that the peak output voltage has 10 times the magnitude of the input waveform.
Current Draw: This amplifier is expected to output approximately 500mA into the coil so that the coil is driven with a strong enough magnetic field to oscillate the string. This gives a maximum power output of 3W.	The output current can be measured using a multimeter. When the input voltage is at 6V, the output current into a 12 ohm load such as the coil should be measured to be 500mA.

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.

Passive attenuators are included in the pre-amplifier for the DAC to ensure that the signal from the string never grows large enough to clip inside the op-amp or DAC. By including these points of manual tuning, we allow ourselves to push our signal and power levels higher and provide tunability for the feedback system.

3. Cost and Schedule

Bill Of Materials

Subsystem	Value	Package	Component Price	Supplier	Supplier PN	Manufacturer	Manufacturer PN	Quantity
Master Board	Optocoupler	THT	$0.83	DigiKey	160-1792-ND	onsemi	6N137M	1
Master Board/DSP - Power	Buck Converter	SMD	$1.76	DigiKey	296-TPS62903QRYTRQ1TR-ND	Texas Instruments	TPS62903QRYTRQ1	1
Master Board/DSP - Power	10uF, 50V	SMD	$0.11	Mouser	187-CL31A106KBHNNNE	Samsung	CL31A106KBHNNNE	12
Master Board/DSP - Power	1uH, 3A	SMD	$0.23	Mouser	652-SRP2510TMA-1R0M	Bourns	SRP2510TMA-1R0M	1
Master Board/DSP - Power	113 kΩ	SMD	$0.10	Mouser	667-ERA-3AED1133V	Panasonic	ERA-3AED1133V	1
Master Board/DSP - Power	24.9 kΩ	SMD	$0.16	Mouser	652-CRT0805DY2492ELF	Bourns	CRT0805-DY-2492ELF	1
Master Board/DSP - Power	Power Supply		$12.99	Amazon	ALITOVE DC 12V 5A Power Supply	Alitove		1
DSP / Master - MCU	MCU	QFP	$2.17	DigiKey	497-STM32L431CBT6TR-ND	STMicroelectronics	STM32L431CBT6TR	13
DSP - Buffer	Op Amp	THT	$0.25	DigiKey	296-TL071HIDRTR-ND	Texas Instruments	TL071HIDR	12
DSP - Buffer	500kΩ Trim Pot	THT	$2.05	DigiKey	3296W-504LF-ND	Bourns Inc.	3296W-1-504LF	12
DSP - Buffer	1MΩ	SMD	$0.10	DigiKey	541-1.00MCTR-ND	Vishay Dale	CRCW08051M00FKEA	24

DSP - Buffer	110kΩ	SMD	$0.10	DigiKey	541-10KATR-ND	Vishay Dale	CRCW080510K0JNEA	12
DSP - Buffer	15kΩ	SMD	$0.10	DigiKey	541-15.0KCTR-ND	Vishay Dale	CRCW080515K0FKEA	12
DSP - Buffer	22uF	SMD	$0.08	DigiKey	1276-CL21A226MAYNNNETR-ND	Samsung Electro-Mechanics	CL21A226MAYNNNE	24
DSP - Buffer	1uF	SMD	$0.08	DigiKey	1276-1029-2-ND	Samsung Electro-Mechanics	CL21B105KBFNNNE	12
DSP - Buffer	4700pf	SMD	$0.10	DigiKey	1276-2525-2-ND	Samsung Electro-Mechanics	CL21B472KDCNNNC	12
DSP - Output driver	Op Amp	20-TSSOP	$1.35	Mouser	835-RT9147ZQW	Richtek	RT9147ZQW	12
DSP - Output Driver	10uF, 50V	SMD	$0.11	Mouser	187-CL31A106KBHNNNE	Samsung	CL31A106KBHNNNE	12
DSP - Output Driver	220uF, 25V	Radial, Electrolytic	$0.22	Mouser	647-UVR1E221MPD	Nichicon	UVR1E221MPD	12

Project Design Schedule

Weeks 3 - 6: Circuit design and initial breadboard testing

- Master board prototyping with UART and I2C Peripherals
- Pre-amp breadboard development
- Component research and specification
  - MCU selection, component selection for master board, pre-amp, and power amp.
- Development of physical specification for project

Weeks 7-8: DSP Design and development

- PCB design for Master board and DSP testing platform (includes DSP MCU & preamp)
- Continued revision of power amp design

Weeks 9-10: Gear up for demo & Begin finalization of software

- Preparation for second breadboard demo
  - Finalization of power amp design
  - Custom coils are made (done at home by eddy)

        - Basic feedback loop should work at this point
- Full PCB design for DSP boards
- Revision for Master board PCB design
- Code for MIDI to I2C should begin to take shape

Weeks 11-15: Finalization of design

- Filter development in DSP controller
    - Tuning may need to be done to optimize performance of harmonic modes
- Finalization of MIDI to I2C controller (master board) - ongoing throughout period
- Board assembly (weeks 11 and 12)
- Physical assembly pending chassis from machine shop
    - Boards attached to chassis (week 13)

4. 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.

References:

[1] Midi Fanatic, "Running Status" [Online]. Available: MIDI Specification: Running Status