Design Document

Sound Effect Devices for Musicians:

Synthesizer

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Frontal materials

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Acknowledgement

The Sound Effect Devices for Musicians team would like to kindly thank both Dr. Chen and Dr. Geiger for advising us as well as supporting us with knowledge through the duration of this project.

Problem statement

In today's world musicians can be put in a difficult spot when trying to find new equipment that will help them excel in the music world while staying under a reasonable budget. Musical devices, such as synthesizers, can cost anywhere from $500 - $10,000, with this most musician don't have the cash to afford upgrading their equipment to compete in the competitive industry of music.

Our goal is to create a synthesizer for musicians is easy to use, modular, and affordable. Our target is to create the synthesizer for under $250. Musicians thrive on having the latest and greatest devices to create music, so we plan on creating our synthesizer with our own unique spin on it by using a capacitive touch keyboard instead of the traditional keyboard layout. This device we are creating will not only look like a state-of-the-art synthesizer but will also sound like one too as we plan to give the musicians ample freedom to make music their way with our device being modular. With this it will also be extremely user friendly as it will also be able to be paired with an iPad to allow users to have an interface to control values.

Operating environment

The operating environment for the synthesizer, in a perfect world, would be in a controlled environment where the users can make sure that the product is always in the best shape. Naturally, that will not be the case always. Operating environments are really determined by the users. If the user would like to use the product in a controlled environment that is fine and if they would like to take it to live shows that is fine as well. As you can imagine, that means the product needs to be able to withstand normal everyday use both inside and outside a controlled environment. It will need to withstand rough conditions, even if it might never be exposed to these conditions. Simply putting it, our product will be built to last and perform at its highest capabilities.

Intended user(s) and intended use(s)

Intended users of the synthesizer include everyone that has an interest in making music or making noise for entertainment. All musicians, from amateurs to professionals, are the target audience for this product. We aim at making this not only a highly sophisticated tool, but also one that is easy and fun to use.

The intended use is up to the users. Our hope is this tool is being used to help professionals create music but also people that are just having fun. This product in the general realm is intended for musicians who want to make high-quality music on a lower end budget.

Assumptions and limitations

Assumptions:

- Users can read and understand English - to understand brief introduction
- Users will have access to a wall outlet

- All components will work inside the product
- Product will be taken care of by owners
- Users will have access to a Windows or Mac computer with *Max* software, and an iPad with the *Mira* application if they want to use the external UI features.

Limitations:

- People without musical knowledge will have a learning curve
- Group members have conflicting schedules so finding an ample amount of time each week to work is difficult

Expected end-product and other deliverables

The final deliverable for this project will aim at handing over a sophisticated synthesizer that is not only easy to use but is also a very powerful tool for musicians to have. The synthesizer will cost under $250 to be made, but that will not affect the functionality of the synthesizer. The tool will be fully functional and include eight modules that give musicians the capability to make a wide range of music.

The final deliverable for this project will also include a brief instruction, specifying modules that are included in the synthesizer and how they work. The instruction manual will target users that already have a musical background, so they can quickly look through the manual and get into making music as quickly as possible.

We estimate completion and delivery of this project to be May of 2019, the end of the Spring 2019 semester.

Specifications and Analysis

Proposed Design

We have decided to create a synthesis device that implements the basic sound generation functionality of conventional synthesizers, as well as some additional effect processing features and external control features. For the basic functionality we need eight separate modules; Voltage-controlled audio frequency oscillators, voltage-controlled low frequency oscillators, a multichannel mixer, voltage-controlled filters, envelope generators, voltage-controlled amplifiers, a keyboard that generates control voltages, and a power delivery module. Figure 1 shows a block diagram showing how the basic modules will be connected inside of the device. With these 8 modules, the user can design new sounds by setting the available parameters and then playing those sounds with the keyboard. All of the modules have their inputs and outputs available as control voltage sources and destinations on the hardware user interface. With these inputs and outputs, the user is able to reconfigure the internal signal paths. The basic modules have controls on the hardware user interface to set all of the sound influencing parameters. The controls all go through the keyboard module's microcontroller which acts as the digital brain for the entire device. By having the controls go through the microcontroller, the user has the ability save and recall presets. The microcontroller also allows for a wired communication path via USB with an external PC machine. The PC machine can run a *Max* patch that communicates with an iPad through WiFi. With the iPad the user can have a wireless user interface that mimics the hardware user interface with additional touch and motion controls. The keyboard module also allows the user to play the synthesizer in 3 voice modes: monophonic, 2-note paraphonic, and 4-note paraphonic. Additionally, the keyboard module has multiple sequencers for notes, gates, and signal paths. The last additional feature the keyboard module incorporates is interchangeable and reconfigurable effect loops. There will be three integrated effects, a short loop sampler, chorus, and delay, and there will be six additional effect loops with dedicated inputs and outputs that the user is able to use for any effect they wish to incorporate into their signal chain. The FX signal chain can be made up of the onboard effects and any of the six effect loops; it can be configured to have the loops in any order; and the device can save and recall 8 preset FX-chains.

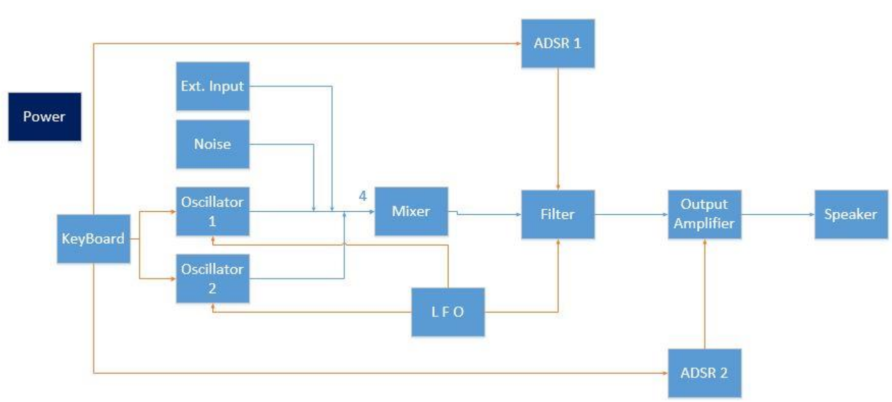

Figure 1: Block Diagram for basic module connectivity

Design Analysis

This segment will be broken down by each module and will be explain the decisions that have been made to reach the final product.

Oscillator

The oscillators are designed around the CEM 3340 voltage-controlled oscillator. This primary reason this chip was selected was because it very easily handles the 1 volt per octave standard. The chip also provides 4 output waveforms, triangle, ramp, pulse, and square. Given a 15-volt supply, the output amplitudes for the triangle, ramp, and pulse/square are 5V, 10V, and 13.5V respectively. The triangle and ramp outputs are feed into amplifier stages to center them around 0 volts. This is done because audio signals are centered around 0 volts and the CEM outputs waveforms that are above 0V. The triangle is also amplified to 10 V_{pp}. The pulse/square is divided to 10V. Amplifying every signal to 10V makes the output uniform and easier to deal with for the mixer team. The CEM also has a temperature compensator circuit and has been designed in accordance with the datasheet.

The oscillator module includes two oscillators. These will be connected using the hard sync pin on the CEM.

Filter

The design for the low pass and high pass filter both Butterworth filters and of fourth order. The reason to do the low pass and high pass filter this way is because this will allow us to reach our specification of 12/24 dB per octave. The reason for a Butterworth filter design is to achieve maximum flatness on the top of the frequency response. This will make the allow the signal going through the filter to be modified easier. I chose to design a high pass and low pass filter because adding these circuits together will therefore create a band pass and notch filter while minimizing components and without having four separate circuits. We want to have the cutoff frequency and resonance to be adjustable between 15 Hz and 8kHz. These values are the frequencies of each octave from the first to the tenth octave. Having the cutoff frequency adjustable allows the user to determine what frequencies they want to let through to create the sound they desire.

The reason a fourth order filter was designed is because the higher the order of the filter, the more flexibility we have in shaping the frequency response. The higher the order also means there is minimum space in the transition band, which is what we desire. The low pass filter was designed by cascading two active second order non-inverting filters together. The function for a low pass filter should have a zero in the numerator that is at infinity, meaning the numerator should be constant. Q in the transfer function equation is the pole quality factor which determines the sharpness of the features in the frequency response. For example, high Q means the cutoff is very sharp and is not smooth and low Q returns smooth cutoff. For our application a Q of ~ 1 or slightly less is ideal. The high pass filter was designed in a similar way. Two second order high pass filters are cascaded to form a fourth order. The function for high pass should have numerator that is equal to the gain*s^2 which is just s^2 because the gain is usually a unity gain. For a fourth order system the normalized denominator in factored form is the following:

$$(1+0.765s+s^2)(1+1.848s+s^2)$$

Equation 1

Noise

To create a noise generator the first approach was to see what noise should be amplified. There are two major types of noise that we chose to investigate thermal, and avalanche. The method that thermal noise can be collected is amplifying it off a resistor. The thermal or Johnson noise comes from the random motion of charge from thermal excitation [1]. This noise source can be described by this equation:

$$V_{th}^2 = 4kTR\Delta f$$

Equation 2

k	Boltzmann Constant
T	Absolute Temperature
4kT	$1.61 \cdot 10^{-20}$V·C at room temperature

Figure 1

We turned against this idea because of the amplification and the number of DC biases need to get the desired white noise.

Instead we selected to use avalanche noise. Avalanche noise originates from the breakdown of the reversed bias junction of a BJT. This noise originates from carriers in the junction gaining energy from the high electric field then colliding with the crystal lattice [1]. This equation for this noise is characterized by the equation below:

$$\text{Sav} = \frac{(2qI)}{(2\pi f\tau)^2}$$

Equation 3

To get the brake down voltage a 2222 npn transistor 12 volts was placed above the emitter to get the brake down according to the datasheet the brake down voltage from the emitter to base is 7 volts. It was found that this breakdown did occur. From here a common emitter amplifier and an operational amplifier along with DC biases provided the output waveform of the noise with an RMS value in between 500mV and 1V.

Output Amplifier

The output amplifier is a simple circuit designed with a TI LM386 Low Voltage Audio Power Amplifier. The reasoning for this decision is that with our project being in the realm of audio, an audio amplifier is most ideal for our case and a power amplifier is needed to be able to power a speaker. The circuit was designed with the following specifications. Going into the circuit is an audio signal with a waveform of roughly 10 V_{pp}. Going into the amplifier that voltage needed to be cut down to roughly 2 V_{pp} as it specified the max input voltage for the LM386 to be .4 V on the data sheet. Then for the LM386 the non-inverting terminal is connected to the wiper/middle terminal of a 100k Ohm potentiometer. This potentiometer will allow us to adjust the amplitude of the audio output. Finally, coming out of the LM386 is a signal that is fed through a voltage divider to put the voltage down to a max of 1.5 V_{pp} as that is a safe range to insure, we don't harm the speakers.

Mixer

The idea of the mixer is to use a summing amplifier to combine the two oscillators, noise, and the external inputs. The design for the oscillators should produce an output of 10Vpp, the noise will need to get amplified in this process to 10Vpp, and the external inputs are presumed to be 10Vpp. The output of the mixer should be a 10Vpp value that will then go through a step down that will either bring it to 5Vpp or 3Vpp.

Keyboard

The basic functionality of a keyboard is to generate different control voltages depending on what key is pressed, similar to a DAC. In order to increase the functionality of the entire device, we decided to make a keyboard that has digital functionality. We have decided to use the MK66FX1M0VMD18 ARM Cortex-M4 processor as our microcontroller. With the microcontroller, we can program all of the buttons to have multiple functions. This allows us to shrink the hardware interface controls and also allows for presets of those controls to be saved and recalled. With a wired USB connection, the user also has the ability to control the device with external controllers and the included software interface. The use of a microcontroller also allows for the implementation of sequencers. With this feature the user can program note patterns to be played in time by the keyboard. The user can also program a sequence for multiple gates to be sent to an external drum machine. Another sequence that can be programmed is a signal path sequence that allows for different effect chains.

Testing and Instrumentation

Interface Specifications

The interface that we will be using to confirm that the device works is by using the lab equipment at Coover Hall at Iowa State University. Below is a list of the equipment that will be used in the lab.

Equipment	**Purpose**
Function Generator	This device will allow us to test the voltage source for a sine, pulse, and ramp. The major benefit of this device is being able to select the amplitude and frequency of the wave.
Oscilloscope	The device measures the waves throughout the circuit. It also provides a good measurement system for voltage peak to peak, amplitude, and RMS of an AC wave.
Multimeter	This device measures the resistance, voltage, and current throughout the circuit.
DC source	This will provide a consistent voltage source that we will use for our Vdd and Vss of the circuit.
Lab View	This will allow us to automate the testing of the device if needed.

Figure 2

Hardware / software:

To model out the schematics of each module the program that we decided to use is National Instruments Multisim. Multisim is a SPICE simulator and circuit design software. This allows the user to be able to run analog, digital, and power electronic simulations and be able to capture the data. This program is useful for our needs because it is a clear way to represent a simulation of our modules since it has the characterization of many parts. It is an easy to use software that allows us to make a circuit diagram for our schematics so that we can present them to each other and our advisors.

One of the biggest benefits of NI Multisim is that is contains the footprints for the components which makes using the PCB designer National Instruments Ultiboard. This is a program that allows us to design the PCB for each module. This program is renowned for being an excellent way to do PCB design and routing which will help us optimize the dimensions of the boards that we are fabricating.

Prior to creating the PCB's, we will solder the device onto a perf board. This will allow us to confirm that the device will work with the components soldered down. The testing of this device will be confirmed using Coover's laboratory equipment.

To have our own customized PCB design we are using OSHpark. This company has a great user interface when submitting a PCB design to be printed on a PCB. They have a cost that we support of 5- 10 dollars per square inch. Having them fabricate our PCB's is something that we need in our final product.

Once the PCB has been fabricated it will be tested using Coover's lab equipment. This will show final proof that the module will work in our final product.

Process:

Oscillator

Once the CEM 3340 chip is obtained, the proposed schematic will be built on a breadboard. The chip will be given inputs from a power supply that follow the 1 volt/octave standard. The output waveforms will be tested using an oscilloscope. Parameters tested will be amplitude and frequency. The actual frequency and desired frequency should differ by less than 5 cents. This specification was selected because this is widely considered the just noticeable difference for pitch. The amplitude will be tested to the values given on the datasheet. Errors in this stage will be corrected through amplification.

Filter

The first step here was drawing out by hand what a fourth order low pass and high pass filter look like. This was to attempt to see what resistor controls the cutoff and what controls the resonance. This is still being determined, but once that is confirmed, the filters can be connected in the same circuit. Then testing can be done to determine function ability.

Noise

The first step was calculating what the current is across the emitter to the base of the BJT. This provided the first resistor of the device. From here amplifiers were calculated to provide a gain between 500mV and 1V. This schematic was first draw by hand and then tested on the breadboard. These tests were conducted in Coover lab.

Output Amplifier

Testing for the output amplifier has been completed on a simulator using NI Multisim. For testing of this circuit, we needed to create the component for the LM386 which was not too difficult as there was plenty of online help for this. Following this we created the rest of the circuit, we used a function generator to imitate the output of the filters which will be the input to this circuit. Then we used NI Multisim’s oscilloscope component to display the output while we varied the potentiometer value. The reasoning for this was to ensure that for all ranges the circuit would be within the threshold.

Mixer

This circuit design will be calculated to get the desired output of 10Vpp to then to the step down. The circuit design is still in progress for this module.

Keyboard

Since the keyboard module interacts with every other module, it will be a bit difficult to test in the initial phases. The components that need to be tested are all the buttons, knobs, and sliders; they should all produce the correct sensor signals when pressed or adjusted. Next, the microcontroller has to interpret those signals and set the correct parameters on the digital controls that are in all of the modules. Once that is confirmed, testing for the preset save function is needed. This can be done by saving a preset, turning the device off then back on, and checking that all of the parameters are correctly readjusted after power on. Finally, tests have to be conducted on all the different sequencers. These will confirm that the correct notes are saved and played back in the desired tempo.

Results

Oscillator

Results have not been obtained for the CEM 3340 yet.

Noise

Below is a circuit schematic for the noise module.

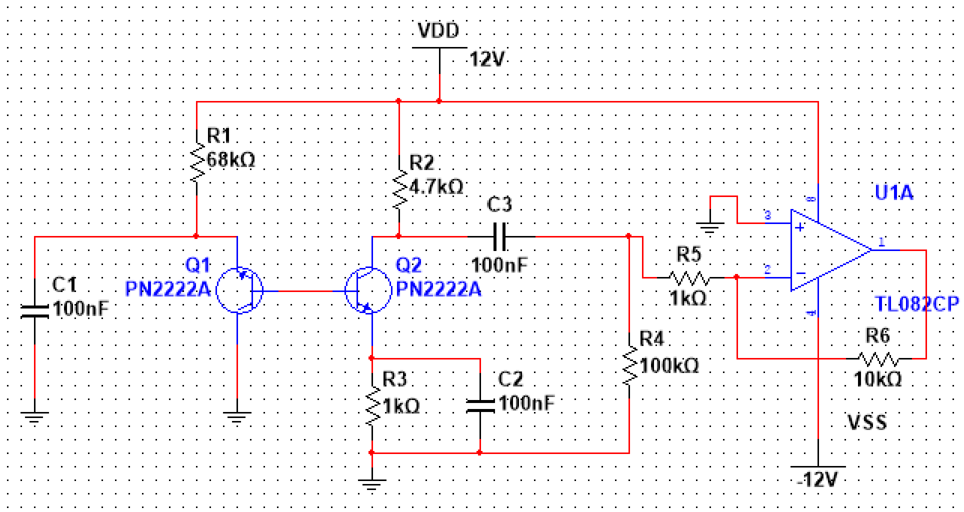

Figure 2

The interesting part about this design was trying to amplify the noise instead of suppressing it which is a typical occurrence in circuit design. The first challenge was solved by increasing the values for the capacitors from nano-farads to micro-farads. Not only did this create a better DC bias it also made the circuit work as expected. The next challenge that was faced was trying to maximizer the gain through the amplifier without distorting the noise. It was found that the gain of 10 will maximize the gain of the white noise without distorting it.

Below is the output waveform from the oscilloscope

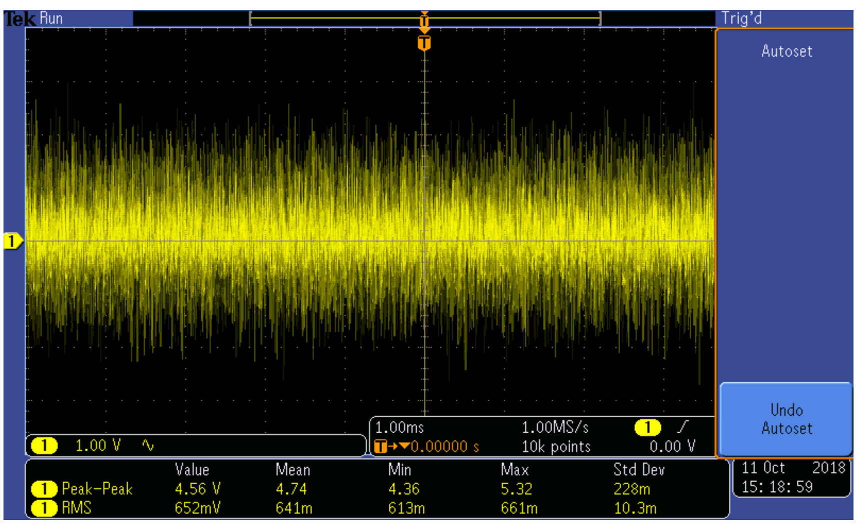

Figure 3

This waveform shows that result that was expected of having an RMS value between 500mV to 1V and providing an output that is white noise.

Output Amplifier

While testing the output amplifier the original circuit did not include a capacitor. When we ran the simulation the circuit, we were getting some DC gain in the output. After some discussion with all the team members we decided to add a large capacitor to cut the DC gain off. After this was done the circuit was simulating correctly. Also, while testing this section another problem that was quickly resolved was instead of having a 12 V supply to the LM386 we were using the input signal, a team member quickly recognized that, and it was an easy fix and resolved. Over the duration of this testing there was a lot to learn. Some of the most important information we learned was when designing the circuit, we need to take our time on it. When we rush a circuit, we are bound to make mistakes that we will just have to fix later when testing. Learning from this is something that every member can benefit from. Attached below the schematic that was found to be functional for our use.

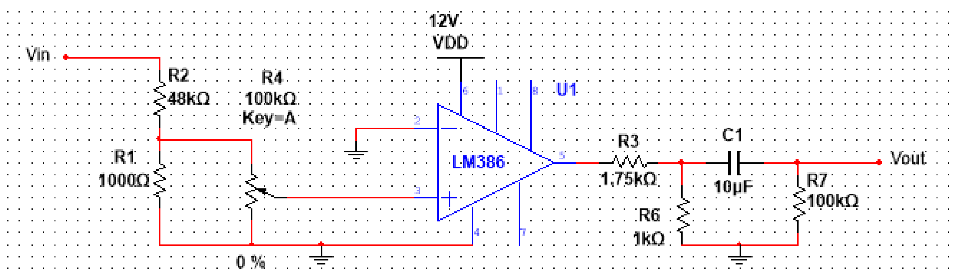

Figure 4

Mixer

Below is previous concept for the mixer.

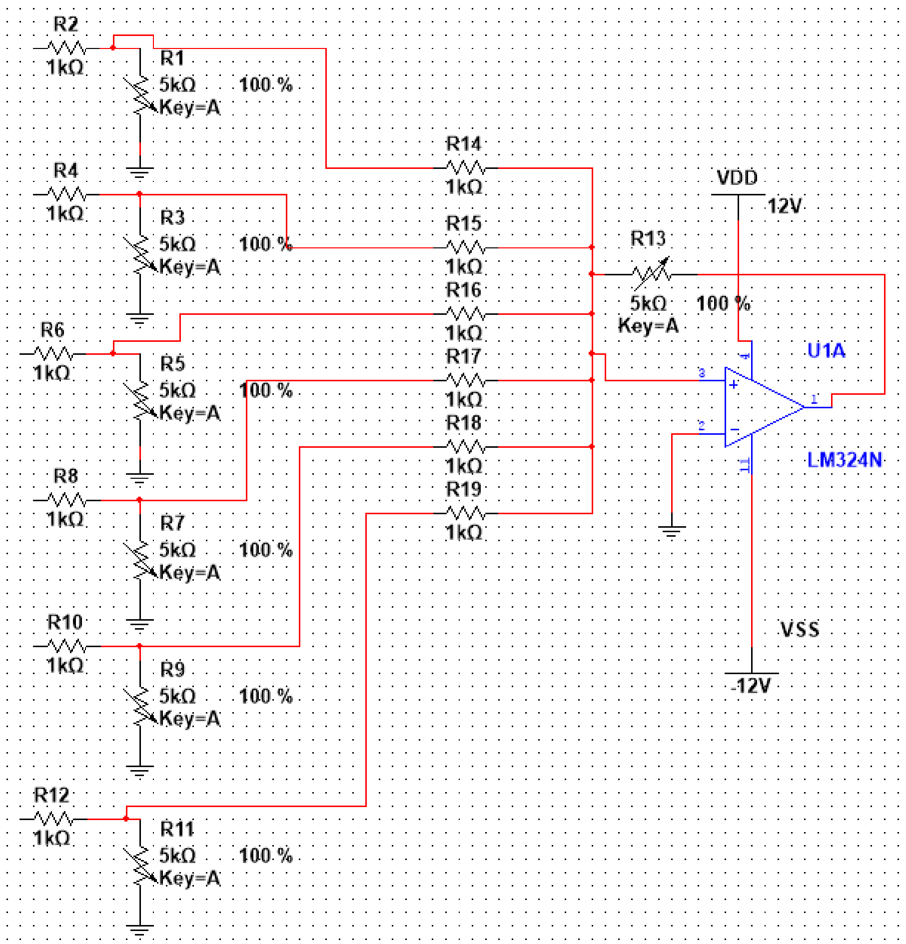

Figure 5

This circuit will be modified to having larger resistors to reduce power loss, and instead of having a customizable voltage swing for the inputs. They will have the option to be either on or off. From here the mixer will not have a gain since it will combine all the inputs to have a 10Vpp output. Then it will go through a step down. To get the output to either be 5Vpp or 3Vpp.

Keyboard

The only results that have been obtained are for the implementation of capacitive sensors. We are able to implement capacitive sensors that function as buttons. In our test, the press of a button turns on an LED. Once the other modules advance in the design phase, the keyboard will be programmed to control those modules vs just turn on LEDs. The figure below shows the current prototype for a 3-button keyboard with an Arduino.

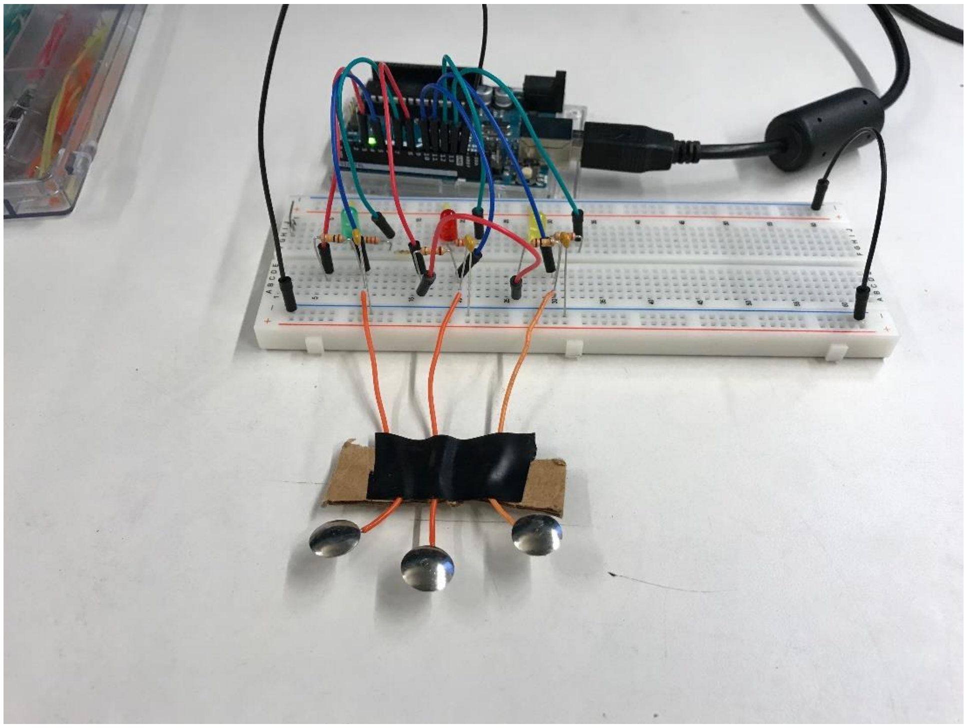

Figure 6: 3-button keyboard prototype using an Arduino UNO

Closure Materials

As stated in the previous sections, these are our designs for each module. We chose to implement the designs in this way because this is the best way to meet practically all our design specifications as well as reach the overall end goal product. Each module has a specific set of design constraints and specifications that need to be met in order to function properly. We made our designs to meet these specifications and the way we did this has been explained in good detail in the previous sections. We have a few tasks left in the schematic design phase, but once we complete these tasks, we can fully start the testing phase. The testing phase is quite comprehensive, so we have allotted enough time to catch minor errors before sending out our schematics to be printed on PCBs. Some modules have already begun the testing phase, so we are moving along with good progress and staying on track of our deadline goals. For many modules, there are several different ways the circuit can be designed to reach the same specifications. We all have designed each module in a way to help reduce cost as much as possible, for example implementing a design that is the simplest yet still meets our criteria. We also have designed each module so that it is very easily transferrable to a PCB while minimizing room for error when OSHpark designs the PCBs. Therefore, we believe our designs are the best match to meet the design criteria we created when we determined the path to reach our final product.

Resources

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